CE - Communication Engg.

Module-II - Analog Transmission & Reception

Analog Signal transmission & Reception

Introduction :-

Modulation may be defined as a process by which some characteristics of a signal known as carrier is varied according to the instantaneous value of another signal known as modulating signal.

→ These modulating signals contains information & are also called baseband signals.

→ The carrier frequency is greater than the modulating frequencies. The signal which results from the process of modulation is known as modulated signal.

Band pass signal :-

If the modulated signal is transmitted over the channel, it is known as band pass / pass band ~~signal~~ signal.

e.g:- If $f_m = 1\,KHZ$,

$f_c = 100\,KHZ$.

During AM, 2 side bands will be generated. Lower side band will be at $(100-1)$ i.e 99 KHZ and upper side band will be at $(100+1)$ i.e 101 KHZ. Thus the AM signal will have frequencies from 99 KHZ to 101 KHZ. Thus frequencies are bandpass type.

Types of modulation :-

Modulation are basically of two types. (For Analog data / modulating signal)

(i) Continuous wave modulation :-

When any carrier wave is continuous in nature, the modulation process is known as continuous wave (CW) modulation or analog modulation. The continuous wave modulations are of 2 types

→ Amplitude modulation
→ Angle modulation.

AM :-
When the amplitude of carrier is varied in accordance with message signal it is known as amplitude modulation.

Angle modⁿ :-
When the angle of carrier is varied according to the instantaneous value of the modulating signal it is called angle modulation. They are of 2 types

→ Frequency Modulation
→ Phase Modulation

(ii) **Pulse Modulation :-**
When the carrier wave is a pulse type wave form the modulation process is known as pulse modulation.

Pulse modⁿ are of 2 types
→ Analog (PAM, PWM, PPM)
→ Digital (PCM, DM, ADM, DPCM)

Analog modⁿ	Digital modⁿ
1) Transmitted modulated signal is analog in nature.	1) Transmitted signal is digital in nature.
2) Coding is not possible	2) Coding technique can be used to detect & correct the errors.
3) Bandwidth required is lower than that for digital modulation methods.	3) Due to higher bit rates, higher channel bandwidth is required.
4) Analog modulation system are AM, FM, PM, PAM, PWM	4) Digital modulation system are PCM, DM, ADM, DPCM etc.

(4) ②

Concept of multiplexing:-

Multiplexing is a technique in which several message signals are combined into a composite signal for transmission over a common channel.

In order to transmit a number of such signals over the same channel must be kept apart so that they don't interfere with each other at the receiver side. and they can be separated easily

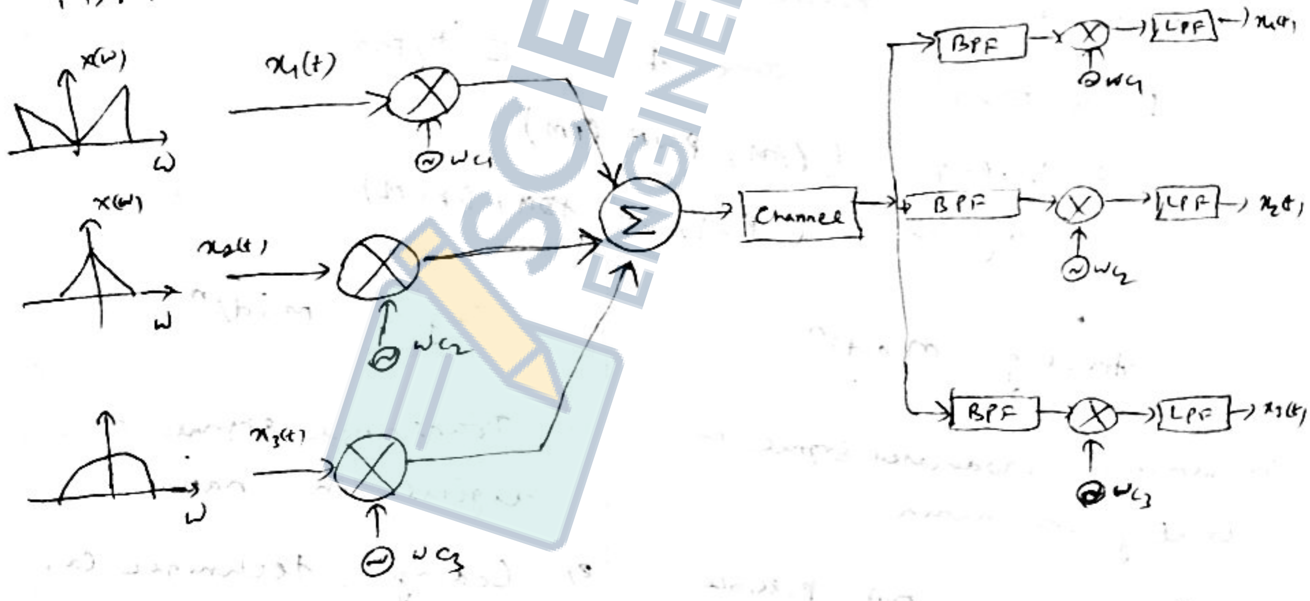
Multiplexing are of 2 types:-

(i) Frequency division multiplexing (FDM)

(ii) Time division multiplexing (TDM)

(i) FDM:-

With the transmission of 3 baseband signals the FDM scheme is shown in figure.



(fig: Freq division multiplexing)

At the receiving end of the channel, the three modulated signals are separated by band pass filter (BPFs) and then demodulated.

It is used in telephone system, telemetry, commercial broadcast etc.

(ii) TDM :-

In this scheme, the complete channel bandwidth is allotted to one user for a fixed time slot. e.g. if there are 7 users, then every ~~7 users ca~~ user can be given the time slot of 1 second.
This method is suitable for digital modⁿ.

Need for (modⁿ / or freq. translation).

(i) Frequency Multiplexing:-

Suppose we want to transmit several different signals having same spectral range in a single communication channel and at the receiving end the signals can be separately recoverable and distinguishable from each other, then it is achieved by translating each one of the original signal to different frequency range. If the frequency ranges don't overlap then the signals may be separated at the receiving end.

(ii) Practicability of antenna:-

If the channel is free space then antenna radiates & receives the signal. The antenna length depends on the wavelength of the signal. Say $(\frac{\lambda}{2})$. To design practical antenna, the freq ~~should~~ translation or modulation should be done

(iii) Narrow banding:-

Suppose the audio range extends from 50 to 10^4 Hz. The ratio of the highest audio freq to lowest is 200. Therefore an antenna used for this purpose for one end range would be entirely too short or too long for the otherband.

Thus the processes of frequency translation is used to change a wideband signal into a narrowband signal.

(iv) Common processing :-

If we want to process a number of similar signals then it is necessary to adjust the frequency range of our processing apparatus to correspond to the freq range of the signal to be processed. Then it is wiser to leave the processing apparatus to operate on some fixed freq range and instead to translate the freq range of each signal.

A method of frequency translation :-

A signal is translated to a new spectral range signal by multiplying the original signal with a sinusoidal signal.

Consider one sinusoidal signal,

$$V_m(t) = A_m \cos\omega_m t \quad \text{--- (1)}$$

$$= A_m \cos 2\pi f_m t$$

$$= A_m \left(\frac{e^{j2\pi f_m t} + e^{-j2\pi f_m t}}{2} \right) \quad \text{--- (2)}$$

Here A_m = Const. amplitude

$$f_m = \frac{\omega_m}{2\pi} = \text{frequency.}$$

Consider another sinusoidal signal,

$$V_c(t) = A_c \cos\omega_c t = A_c \cos 2\pi f_c t$$

$$= \frac{A_c}{2} \left[e^{j2\pi f_c t} + e^{-j2\pi f_c t} \right] \quad \text{--- (3)}$$

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

⊙

(7)

$$V_m(t)\, V_c(t) = \frac{A_m A_c}{2} \cdot 2 \cos\omega_m t \cdot \cos\omega_c t$$

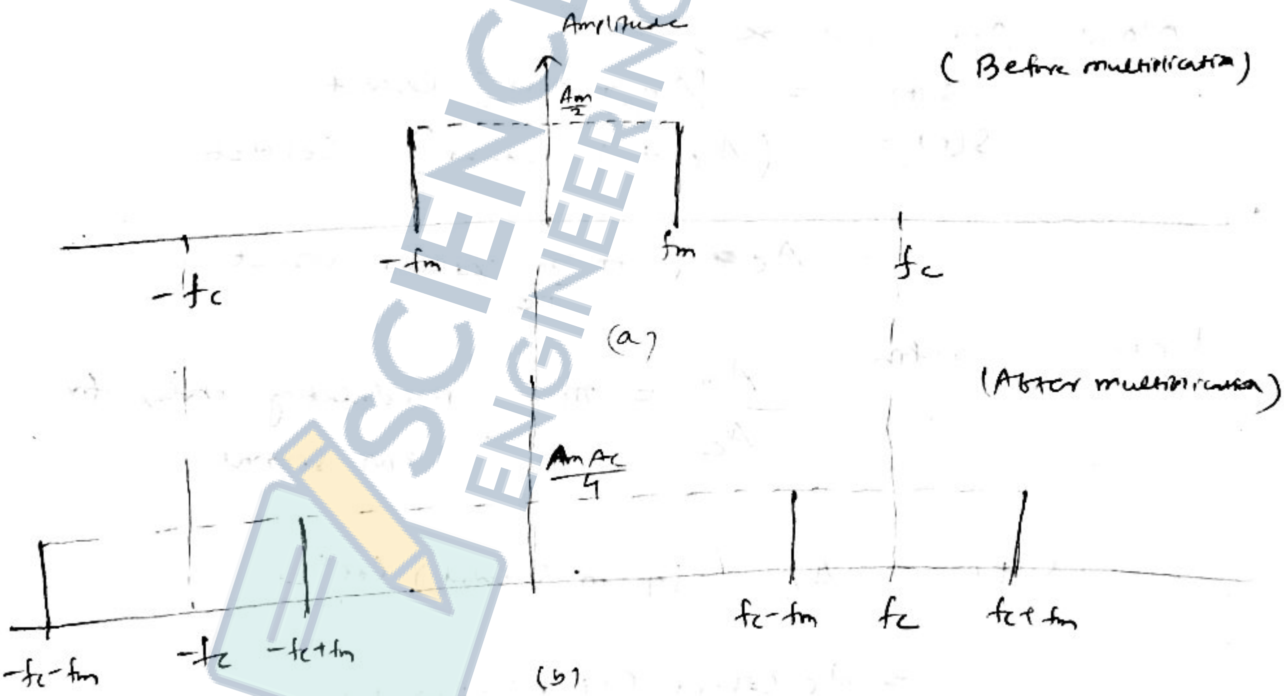
we know

$$2 \cos\alpha \cdot \cos\beta = \cos(\alpha+\beta) + \cos(\alpha-\beta)$$

$$V_m(t) \cdot V_c(t) = \frac{A_m A_c}{2} \cdot \left[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t \right]$$

$$= \frac{A_m A_c}{2} \left[\frac{e^{j(\omega_c+\omega_m)t} + e^{-j(\omega_c+\omega_m)t}}{2} + \frac{e^{j(\omega_c-\omega_m)t} + e^{-j(\omega_c-\omega_m)t}}{2} \right]$$

$$= \frac{A_m A_c}{4} \left[e^{j(\omega_c+\omega_m)t} + e^{-j(\omega_c+\omega_m)t} + e^{j(\omega_c-\omega_m)t} + e^{-j(\omega_c-\omega_m)t} \right]$$



From the new spectral amplitude pattern, shown on fig (b), we see that the original spectral lines have been translated both on +ve frequency direction by an amount f_c and on the -ve frequency direction by an amount $-f_c$. That means there are 4 spectral components like f_c+f_m, f_c-f_m, $-f_c+f_m$ and $-f_c-f_m$.

Single tone amplitude mod^n :-

In AM, the amplitude of carrier signal is varied with instantaneous value of the modulating signal.

Lets consider AM in which the modulating signal consist of only one frequency. That means modulating is done by single frequency or tone modulating signal,

Let a single tone modulating signal,

$$x(t) = A_m \cos w_m t$$

Let the carrier signal be

$$c(t) = A_c \cos w_c t.$$

Now AM will be,

$$s(t) = [A_c + x(t)] \cos w_c t$$

$$s(t) = (A_c + A_m \cos w_m t) \cdot \cos w_c t$$

$$= A_c \left(1 + \frac{A_m}{A_c} \cos w_m t\right) \cos w_c t$$

Lets define, $\frac{A_m}{A_c} = m$ = modulating index for AM signal.

$$s(t) = A_c (1 + m \cos w_m t) \cos w_c t$$

$$= A_c \cos w_c t (1 + m \cos w_m t)$$

$$= A_c \cos w_c t + m A_c \cos w_c t \cos w_m t$$

$$= A_c \cos w_c t + \frac{m A_c}{2} 2 \cos w_c t \cos w_m t$$

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

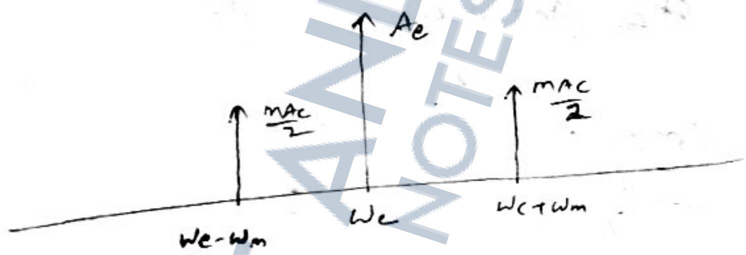
$$\therefore \boxed{s(t) = A_c \cos w_c t + \frac{m A_c}{2} \cos(w_c + w_m)t + \frac{m A_c}{2} \cos(w_c - w_m)t}$$

(9) There are 3 frequency components in AM signal.

(i) Carrier frequency $\underline{\omega_c}$ having amplitude $\underline{A_c}$

(ii) upper sideband freq. $\underline{\omega_c+\omega_m}$ having amplitude $\frac{mA_c}{2}$

(iii) Lower sideband frequency $\underline{\omega_c-\omega_m}$ having amplitude $\frac{mA_c}{2}$.



<u>Power content in AM wave :-</u>

~~The Carrier Po~~

Let's consider a signal,

$$x(t) = A \cos\theta$$

Power content of the signal is, <u>mean square</u> value of $x(t)$

$$\left[\because P = \frac{1}{T}\int_{-T/2}^{T/2} |x(t)|^2 dt \right]$$

$$P = \frac{1}{2\pi}\int_0^{2\pi} A^2 \cos^2\theta \, d\theta.$$

$$= \frac{1}{2\pi} \cdot A^2 \cdot \int_0^{2\pi} \left(\frac{1+\cos 2\theta}{2}\right) d\theta$$

$$= \frac{A^2}{2\pi} \times \frac{1}{2}\left[\theta + \frac{\sin 2\theta}{2}\right]_0^{2\pi}$$

$$= \frac{A^2}{4\pi}\left[2\pi + 0 - 0 - 0\right]$$

$$\boxed{P = \frac{A^2}{2}}.$$

$\therefore$ Power content of a signal having amplitude A is $\frac{A^2}{2}$ i.e $\frac{(\text{Amplitude})^2}{2}$.

⑧ ⑩ Content in AM wave

$\therefore$ Total power

= Power of Carrier + Power of upper side band + Power of Lower side band

$= P_c + P_u + P_L$

$$P_c = \frac{(\text{Amplitude})^2}{2} = \frac{A_c^2}{2}$$

$$P_u = \frac{\left(\frac{mA_c}{2}\right)^2}{2} = \frac{m^2A_c^2}{8}$$

$$P_L = \frac{\left(\frac{mA_c}{2}\right)^2}{2} = \frac{m^2A_c^2}{8}$$

$$P_{total} = P_c + P_u + P_L$$

$$= \frac{A_c^2}{2} + \frac{m^2A_c^2}{8} + \frac{m^2A_c^2}{8}$$

$$= \frac{A_c^2}{2} + \frac{m^2A_c^2}{4}$$

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

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

ex:- An AM Broadcast radio ~~transmitter~~ station radiates 10KW of power. If modulation percentage is 60. Calculate How much is the carrier power.

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

$$\Rightarrow 10 \times 10^3 = P_c\left(1 + \frac{0.6^2}{2}\right)$$

$$\Rightarrow P_c = \frac{10\ kW}{(1 + 0.18)} = 8.47\ kW.$$

(Ans)

Transmission efficiency of AM Signal

Transmission Efficiency $(\eta) = \frac{\text{Side band power}}{\text{Total power}} \times 100.$

Because out of total power P_t, the useful message or baseband power is carried by the sidebands (P_s). The large carrier power (P_c) is a waste from the transmission point of view because it does not carry any information or message.

$$\eta = \frac{P_s}{P_t} \times 100$$

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

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

$$= P_c + P_s$$

$$\therefore P_s = \frac{P_c m^2}{2}$$

$$\eta = \frac{P_s}{P_t}$$

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

$$\boxed{\eta = \frac{m^2}{2 + m^2}}$$

we have

For η_{max}, $m = 1$ $\left(m = \frac{A_m}{A_c}\right)$

$$\boxed{\eta_{max} = \frac{1}{2+1} = \frac{1}{3} = 33.33\%}$$

This implies that only $\frac{1}{3}$ of total power is carried by side bands and rest $2/3$ is wasted.

Current Calculation for AM :-

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

$$I_t^2 \cdot \cancel{R} = I_c^2 \cancel{R}\left(1+\frac{m^2}{2}\right)$$

$$\Rightarrow I_t^2 = I_c^2\left(1+\frac{m^2}{2}\right)$$

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

Note :-

1) The baseband or modulating signal will be preserved in the envelope of AM signal only if

$$A_m < A_c$$

or

$$m < 1.$$

2) If $m > 1$, % of modulation is greater than 100%. In that case the baseband signal ~~power~~ recovered from the envelope will be distorted. This type of distortion is called envelope distortion and AM signal is called overmodulated signal.

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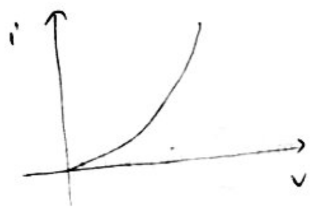
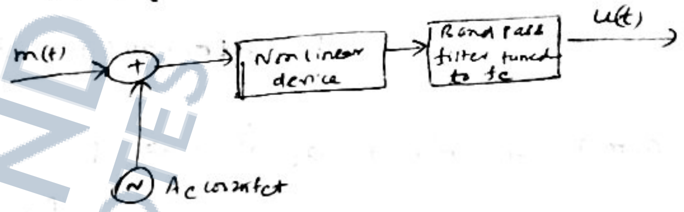
Generation of DSB AM Signal :-

Power-law Modulation :-

Lets consider the use of non linear device such as p-n junction diode which has voltage-current characteristics shown in fig.



(V-I characteristics of a p-n diode)



(Block-diagram of power-law AM Modulator)

Suppose that the voltage input to such a device is sum of message signal m(t) and carrier $A_c \cos 2\pi f_c t$. The non linearity will generate a product of message m(t) with the carrier, plus additional terms. The desired modulated signal can be filtered out by passing the o/p of the non linear device through a bandpass filter.

Let o/p of non-linear device, given by

$$V_0(t) = a_1 V_i(t) + a_2 V_i^2(t)$$

where V_i is the i/p signal, $V_0(t)$ is the o/p signal, a_1, a_2 are the constants.

The i/p to the non linear device is,

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

Its o/p will be

$$V_0(t) = a_1 V_i(t) + a_2 V_i^2(t),$$

$$\Rightarrow V_0(t) = a_1\left[m(t) + A_c\cos 2\pi f_c t\right] + a_2\left[m(t) + A_c\cos 2\pi f_c t\right]^2$$

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

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

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

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

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

When this signal is passed through a bandpass filter the o/p is

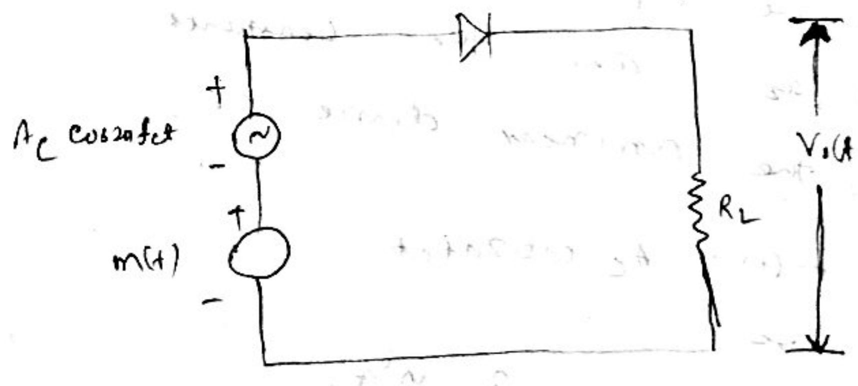
$$u(t) = A_c a_1\left[1 + \frac{2a_2}{a_1} m(t)\right]\cos 2\pi f_c t$$

where $\frac{2a_2|m(t)|}{a_1} < 1$ by design.

Thus, the signal generated by this method is Conventional DSB AM signal.

Switching Modulator:-

Another ~~modulator~~ method for generating AM modulated signal is by means of a switching modulator. Such a modulator can be implemented by the system in the figure given below.



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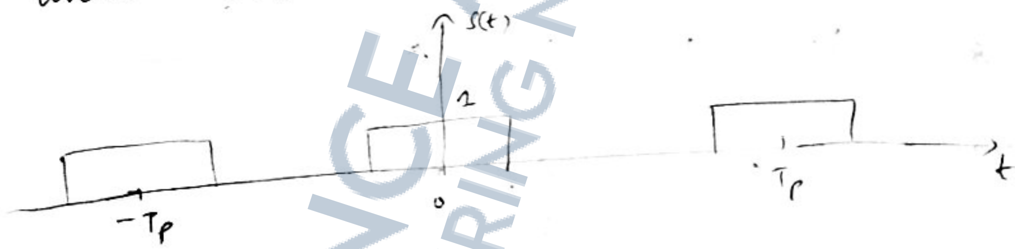
The sum of the message signal and the carrier, i.e $v_i(t) = m(t) + A_c \cos 2\pi f_c t$, are applied to the diode.

The switching operation of a diode may be viewed mathematically as a multiplication of input $v_i(t)$ with switching function $s(t)$ i.e

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

where $s(t)$ is given below,



Hence

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

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

Since $s(t)$ is a periodic function, it is represented in the Fourier series as,

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

So $V_o(t)$ can be reduced to,

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

The desired AM modulated signal is obtained by passing $V_o(t)$ through a bandpass filter with center frequency $f = f_c$ and bandwidth $2W$.

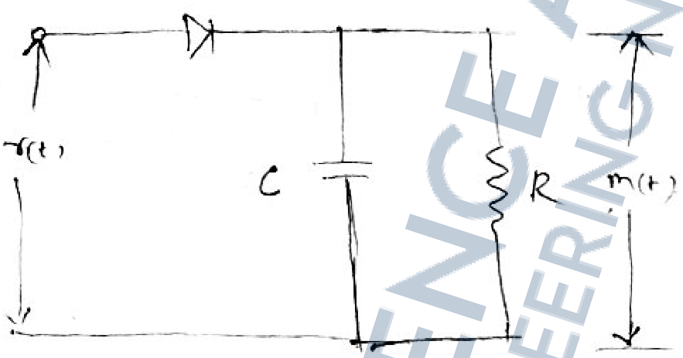
At ckts o/p, we have the desired DSB AM signal.

$$\boxed{u(t) = \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c} m(t)\right] \cos 2\pi f_c t}$$

Demodulation of DSB-AM

Envelope Detector:-

The ckt diagram for an envelope detector is shown below.



→ The Conventional DSB AM signals are easily demodulated by means of an envelope detector. It consist of a diode and an RC ckt, which is basically a simple low pass filter.

→ During the +ve half cycle of the i/p signal, the diode is conducting and the capacitor charges up to the peak value of the i/p signal.

→ When the i/p falls below the voltage on the capacitor, the diode becomes reverse-biased and the i/p becomes disconnected from the o/p.

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→ During this period, the capacitor discharge slowly through the load resistor R. On the next cycle of the carrier, the diode conducts again when the i/p signal exceeds the voltage across the capacitor.

→ The capacitor charges up again to the peak value of the i/p signal and the process is repeated again.

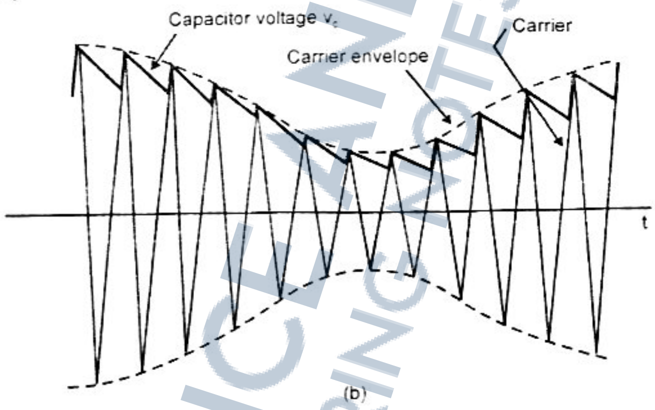

Fig.3.9 : *(a) A demodulator for an AM signal. (b) Input waveform and output voltage v_c across capacitor.*

The time constant (RC) must be selected so as to follow the variation in the envelope of the carrier-modulated signal. So the cond^n is

$$\frac{1}{f_c} << RC << \frac{1}{f_m}$$

In such cases, the capacitor discharges slowly through the resistor and thus, the o/p of the envelope detector closely follow the message signal.

DSB-SC (Double Sideband - Suppressed Carrier) System:-

The general form of single tone modulation is

$$s(t) = \underline{A_c \cos\omega_c t} + \frac{mA_c}{2}\cos(\omega_c+\omega_m)t + \frac{mA_c}{2}\cos(\omega_c-\omega_m)t$$

We see the carrier component in AM wave remains constant in amplitude and frequency.

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

P_t = Total power content of AM wave

P_c = Carrier power.

$$\frac{P_t}{P_c} = \left(1+\frac{m^2}{2}\right), \text{ for } \underline{m=1}, \frac{P_t}{P_c} = 1+\frac{1}{2} = \frac{3}{2}$$

$$\Rightarrow \frac{P_c}{P_t} = \frac{2}{3} = 67\%.$$

∴ So for 100% modulation, about 67% of total power is required for transmitting the carrier which does not contain any information. Hence the carrier is suppressed only the sideband remains. This type of suppression of carrier doesn't affect the baseband signal. The resulting signal obtained by suppressing the carrier from the modulated wave is called Double Sideband Suppressed Carrier signal or DSBSC signal.

Carrier) System :-

modulation is

$\cos(\omega_c - \omega_m)t$

AM wave remains

AM wave

$=1, \quad \frac{P_t}{P_c} = 1 + \frac{1}{2} = \frac{3}{2}$

$\Rightarrow \frac{P_c}{P_t} = \frac{2}{3} = 67\%$

about 67%. If

transmitting ~~the~~ the

information

only the

suppression of

baseband signal. The

suppressing the carrier

called Double

or DSBSC signal.

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~~Approx~~ Power Content of DSB-SC Signal.

For DSB-SC signal, carrier is eliminated, The modulated signal $s(t)$

$$s(t) = \frac{mA_c}{2}\cos(\omega_c+\omega_m)t + \frac{mA_c}{2}\cos(\omega_c-\omega_m)t.$$

We have, $(P_t)_{DSB} = P_c\left(1+\frac{m^2}{2}\right)$, for DSB signal.

Now $(P_t)_{DSB-SC} = (P_t)_{DSB} - P_c$

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

$$= \cancel{P_c} + \frac{P_c m^2}{2} - \cancel{P_c}$$

$$\boxed{(P_t)_{DSB-SC} = P_c \cdot \frac{m^2}{2}}$$

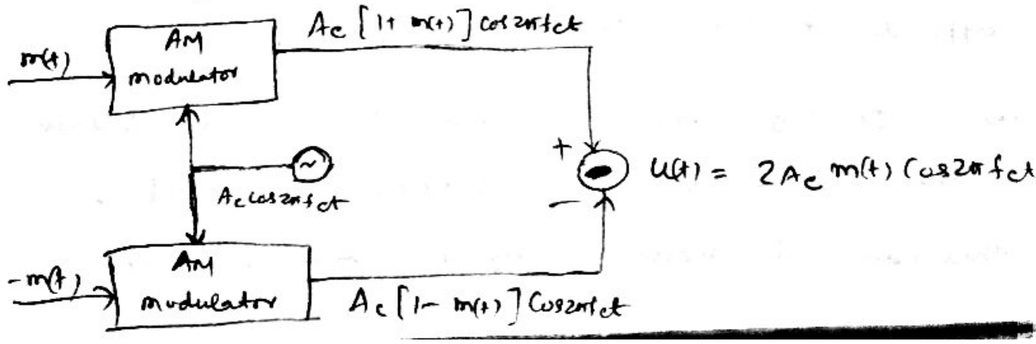
$$P_t = \frac{A_c^2}{2} \cdot \frac{m^2}{2}$$

$$\boxed{(P_t)_{DSB-SC} = \frac{m^2 A_c^2}{4}}$$

Generation of DSB-SC Signal :-

Balanced modulator :-

A relatively simple method to generate DSB-SC signal is to use two conventional AM modulator arr in the configuration shown in fig.



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

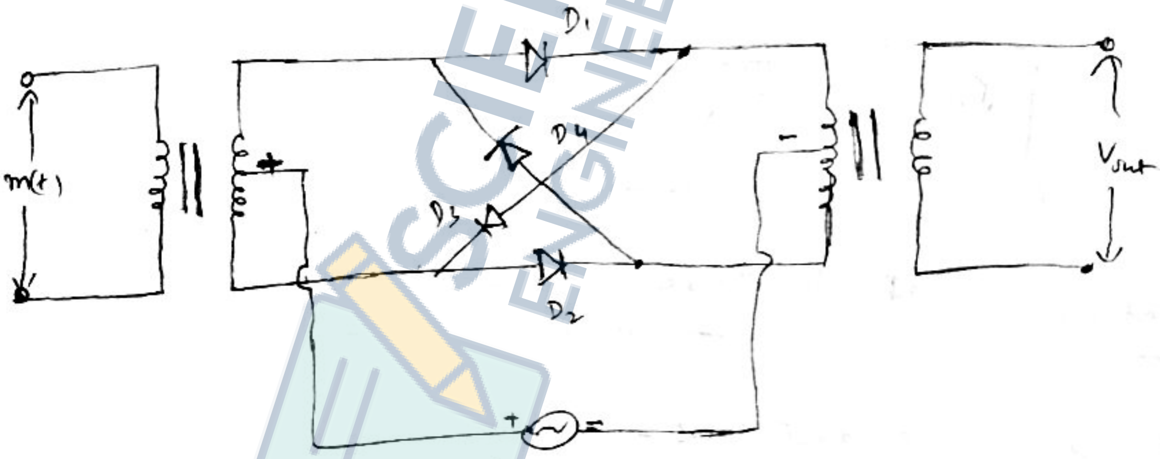
$$= \cancel{A_c \cos 2\pi f_c t} + A_c \cos 2\pi f_c t\, m(t) - \cancel{A_c \cos 2\pi f_c t} + A_c \cos 2\pi f_c t\, m(t)$$

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

We may use 2 square law AM Modulator here. Care must be taken to select modulators with approximately identical characteristics so that the carrier component cancels out at the summing junction.

Ring Modulator:-

Another type of modulator for generating a DSB-SC AM signal is ring modulator.



Square-wave carrier at $f = f_c$.

→ The switching of the diodes is controlled by a square-wave of frequency (f_c), denoted by $c(t)$, which is applied to the centertaps of two transformers.

→ When $c(t) > 0$, the top and bottom diode conduct, while the 2 diodes in cross arms are off.

→ In this case, the message signal $m(t)$ is multiplied by +1.

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→ When $c(t) < 0$, the diodes in the crossarm of the ring conduct, while the other two are switched off. In this case, the message $m(t)$ is multiplied by -1.

→ Consequently, the operation of ring modulator may be described mathematically as a multiplier of $m(t)$ by the square wave carrier $c(t)$ i.e

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

The ring modulator is an ideal form of product modulator and hence it produces the required DSB-SC o/p.

~~c(t)~~

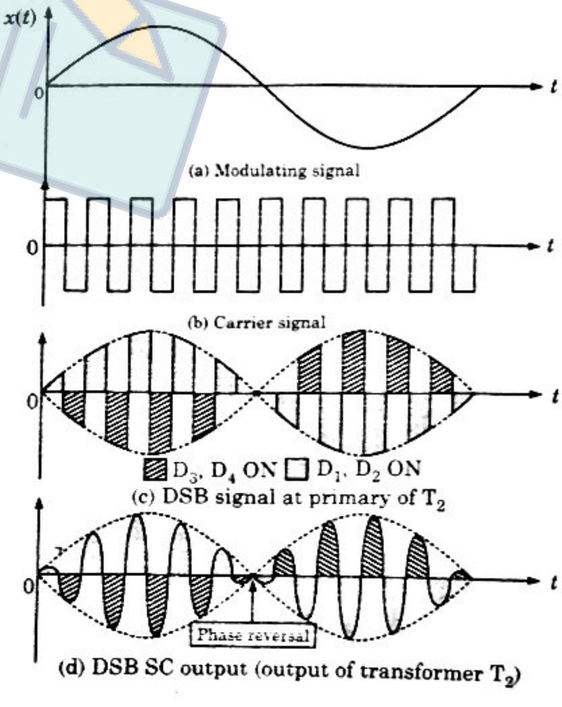
The square-wave carrier signal can be represented by the Fourier series as

$$c(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos\left[2\pi f_c t\,(2n-1)\right]$$

$$\text{So}\quad v_o(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos\left[2\pi f_c t\,(2n-1)\right] \cdot m(t)$$

The desired DSB-SC AM signal $u(t)$ is obtained by passing $v_o(t)$ through a bandpass filter.

Waveform

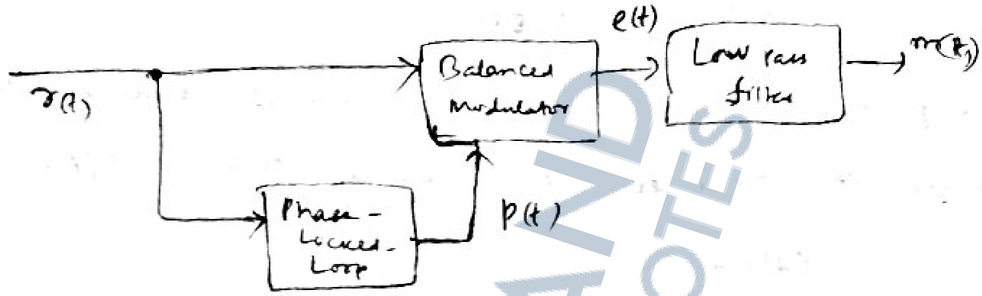


(Page 149 :- Sanjay Sharma)

Fig. 3.28 *Waveforms of the lattice type balanced modulator*

Demodulation of DSB-SC AM signal :-

The demodulation of DSB-SC AM signal requires a synchronous demodulator. That is, the demodulator must use a coherent phase references, which is usually generated by means of a Phase-locked loop (PLL) to demodulate the received signal.



Here the frequency and Phase of r(t) must be identical to the o/p of the Phase locked loop, otherwise signal would get distorted.

A PLL is used to generate a Phase-coherent Carrier signal that is mixed with the received signal in a balanced modulator.

Mathematically, Let $r(t) = m(t).\cos\omega_c t$.

$p(t) = \cos\omega_c t$.

$$e(t) = r(t).\ p(t)$$

$$= m(t)\cos\omega_c t.\ \cos\omega_c t$$

$$= m(t).\ \cos^2\omega_c t$$

$$= m(t)\cdot\left(\frac{1+\cos 2\omega_c t}{2}\right)$$

$$= m(t)\cdot\left(\frac{1}{2}+\frac{\cos 2\omega_c t}{2}\right)$$

$$e(t) = \frac{m(t)}{2} + \frac{m(t)}{2}.\cos 2\omega_c t$$

When e(t) is passed through a low-pass-filter (LPF)

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the term $\frac{1}{2} m(t) \cos 2\omega_c t$, centered at $\pm 2\omega_c$ is suppressed by LPF and thus the o/p of LPF, the original modulating signal $\underline{\frac{1}{2} m(t)}$ is obtained.

SSB-SC (Single sideband Suppressed Carrier) AM

A DSB-SC AM signal requires a channel BW $= 2W$ Hz for transmission, where W is the bandwidth of the baseband signal. However, the two sidebands are redundant. So, the transmission of either sideband is sufficient to reconstruct the message signal m(t) at the receiver. Thus, we ~~reduce~~ reduce the BW of transmitter to that of the baseband signal.

The general form of single-tone modulation is

$$s(t) = A_c \cos\omega_c t + \frac{mA_c}{2}\cos(\omega_c+\omega_m)t + \frac{mA_c}{2}\cos(\omega_c-\omega_m)t$$

$$P_{SSB-SC} = \frac{\left(\frac{mA_c}{2}\right)^2}{2} \qquad \left\{\because \text{Power} = \frac{(\text{Amplitude})^2}{2}\right.$$

$$\boxed{P_{SSB-SC} = \frac{m^2A_c^2}{8}}$$

$$\%\ \text{power saving} = \frac{\frac{A_c^2}{2} + \frac{m^2A_c^2}{8}}{\frac{A_c^2}{2} + \frac{m^2A_c^2}{8} + \frac{m^2A_c^2}{8}} = \frac{\cancel{\frac{A_c^2}{2}}\left[1+\frac{m^2}{4}\right]}{\cancel{\frac{A_c^2}{2}}\left[1+\frac{m^2}{4}+\frac{m^2}{4}\right]}$$

$$= \frac{1+\frac{m^2}{4}}{1+\frac{m^2}{2}} = \frac{4+m^2}{\cancel{4}\,2} \times \frac{\cancel{2}}{(2+m^2)}$$

$$= \frac{4+m^2}{4+2m^2} \times 100\%$$

For m = 1,

$$\% \text{ of power saving} = \frac{4+1}{4+2} \times 100$$

$$= \frac{5}{6} \times 100$$

$$\boxed{\% \text{ Power saving} = 83.33\%}$$

Hilbert Transform:-

It is defined as,

$$x_h(t) = \frac{1}{\pi} x(t) * \frac{1}{t} = \frac{1}{\pi} \int_{-\infty}^{\infty} x(\tau) \cdot \frac{1}{(t-\tau)} d\tau$$

Also inverse Hilbert transform is defined as

$$x(t) = -\frac{1}{\pi} \int_{-\infty}^{\infty} \frac{x_h(\tau)}{t-\tau} d\tau$$

Properties of Hilbert transform:-

1) A signal $x(t)$ and its Hilbert transform $x_h(t)$ have same energy density spectrum.

2) A signal $x(t)$ and its Hilbert transform $x_h(t)$ have same auto-correlation function.

3) A signal $x(t)$ and its Hilbert transform $x_h(t)$ are mutually orthogonal.

$$\int_{-\infty}^{\infty} x(t) \cdot x_h(t)\, dt = 0$$

4) If $x_h(t)$ is a Hilbert transform of $x(t)$, then the Hilbert transform of $x_h(t)$ is $-x(t)$ i.e

If $H[x(t)] = x_h(t)$, then

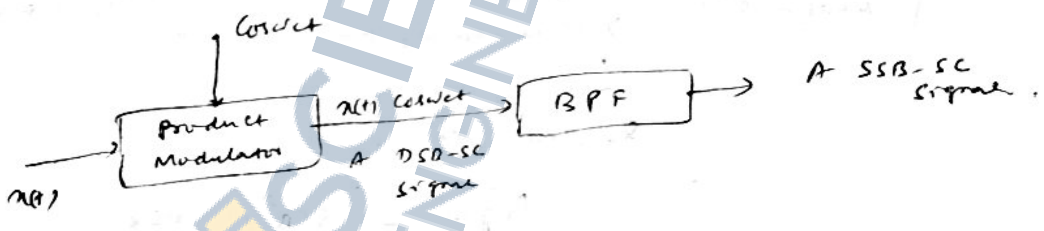
$H[x_h(t)] = -x(t)$, $H \rightarrow$ Hilbert transform.

Generation of SSB-SC signal

1) Frequency discrimination method or filter method.
2) Phase discrimination method or phase-shift method.

1) Frequency discrimination method or filter method :-

In this method, firstly a DSB-SC signal is generated by a product modulator or balanced modulator. After this, from DSB-SC signal, one of the two sidebands is filtered by a suitable bandpass filter (BPF).



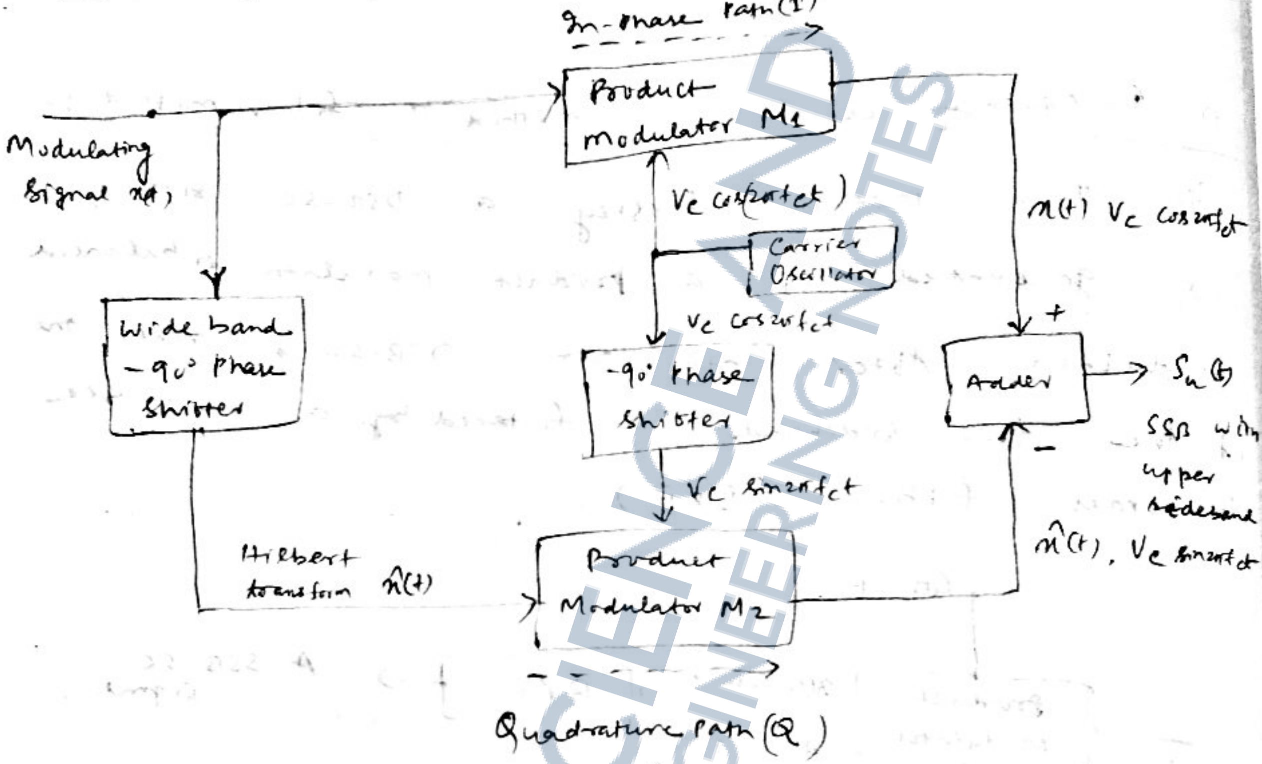
Limitation :-

1) The freq-discrimination method is useful only if the baseband signal is restricted at its lower edge due to which the upper & lower sidebands are non overlapping.

2) The baseband signal must be appropriately related to carrier frequency. Infact, the design of BPF is difficult if the carrier freq. is quite higher than BW of baseband signal.

2) Phase Shift method for SSB-SC generation.

Figure shown below, shows the method for generating SSB-SC signal. This system is used for supression of sideband. It uses 2 balanced modulators M_1 & M_2 and and two 90° phase shifting n/ws shown below.



fig(1):- SSB-SC with USB

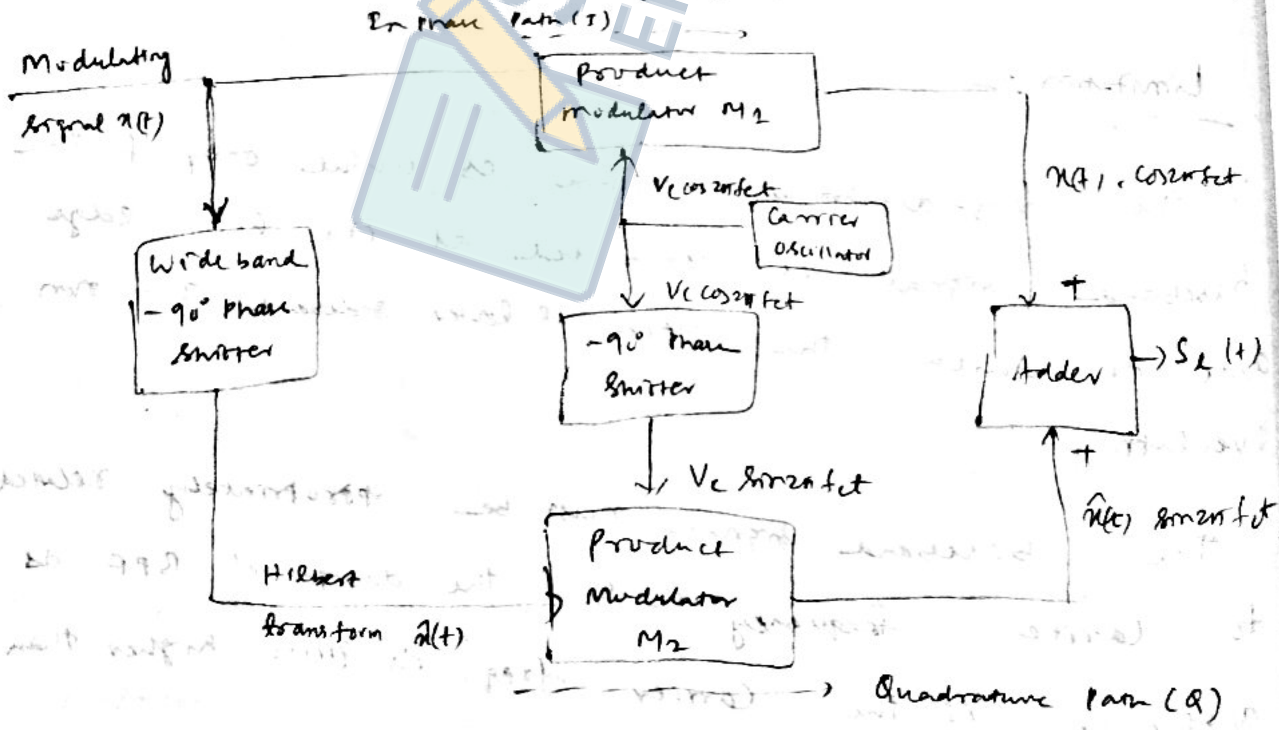


fig 2:- SSB-SC with LSB

Working operation :-

The message signal $x(t)$ is applied directly to the product modulator M_1 & through a $-90°$ phase shifter to the product modulator M_2. Hence we get the Hilbert transform $\hat{x}(t)$ at the o/p of the wideband $-90°$ phase shifter. The o/p of carrier oscillator is applied as it is to modulator M_1 whereas it is passed through a $-90°$ phase shifter and applied to the modulator M_2.

$$\text{o/p of } M_1 = x(t) \cdot V_c \cos 2\pi f_c t .$$

$$\text{o/p of } M_2 = \hat{x}(t) \cdot V_c \sin 2\pi f_c t .$$

The o/p of M_1 & M_2 are applied to an adder.

Note :-

For SSB-SC with only USB (upper sideband) i.e it rejects LSB.

we have

$$\text{adder o/p} = x(t)\, V_c \cos 2\pi f_c t - \hat{x}(t)\, V_c \sin 2\pi f_c t \quad \text{—— (1)}$$

For SSB-SC with only LSB (Lower side band)

i.e it rejects USB

we have

$$\text{adder o/p} = x(t)\, V_c \cos 2\pi f_c t + \hat{x}(t)\, V_c \sin 2\pi f_c t . \quad \text{—— (2)}$$

→ For getting USB, $-ve$ sign on quadrature path.

→ For getting LSB, adder polarities for in phase & quadrature path are both $+ve$.

Demodulation of SSB-SC wave:-

Coherent SSB Demodulator:-

The product modulator is a type of coherent SSB demodulator. To recover the modulating signal from SSB-SC signal, we require a phase coherent or synchronous demodulator.

The received signal is first multiplied with locally generated carrier signal. The locally generated carrier should have exactly the same frequency as that of suppressed carrier. The product modulator multiplies the 2 signals at its i/p and the product signal is passed through a low-pass filter with a bandwidth of f_m. At the o/p of the filter, we get the modulating signal back.

Mathematically,

Let SSB wave at the i/p is given by,

$$S(t) = \frac{1}{2} V_c \left[x(t) \cos 2\pi f_c t \pm \hat{x}(t) \sin 2\pi f_c t \right]$$

The locally generated carrier is $\cos 2\pi f_c t$

The o/p of product modulator is given by,

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

(24) $v_c \quad v(t) = \frac{1}{2} V_c \left[x(t) \cos 2\pi f_c t \pm \hat{x}(t) \sin 2\pi f_c t \right] \cdot \cos 2\pi f_c t$

$$= \frac{1}{2} V_c \left[x(t) \cos^2 2\pi f_c t \pm \hat{x}(t) \cos 2\pi f_c t \sin 2\pi f_c t \right]$$

$$= \frac{1}{2} V_c \left[x(t) \left(\frac{1 + \cos 4\pi f_c t}{2} \right) \pm \frac{\hat{x}(t)}{2} \cdot \left[\sin(4\pi f_c t) + \sin 0 \right] \right]$$

$$\left(\because 2 \cos\alpha \cdot \sin\beta = \sin(\alpha+\beta) + \sin(\alpha-\beta) \right)$$

$$= \underbrace{\frac{1}{4} V_c \, x(t)}_{\text{scaled message signal}} + \underbrace{\frac{1}{4} V_c \left[x(t) \cos 4\pi f_c t \pm \hat{x}(t) \sin 4\pi f_c t \right]}_{\text{unwanted terms.}}$$

When $v(t)$ is passed through the filter, it will allow only the first term to pass through & will reject all other unwanted terms.

$$\therefore \quad V_0(t) = \frac{1}{4} V_c \, x(t)$$

(15PU7-2N2-03)

Phase error & frequency error in coherent detection :-

In coherent detection, we have assumed locally generated carrier is in perfect synchronization. But in practice, a phase error ϕ may arise in the locally generated carrier wave. The detector O/P will get modified due to phase error as follows

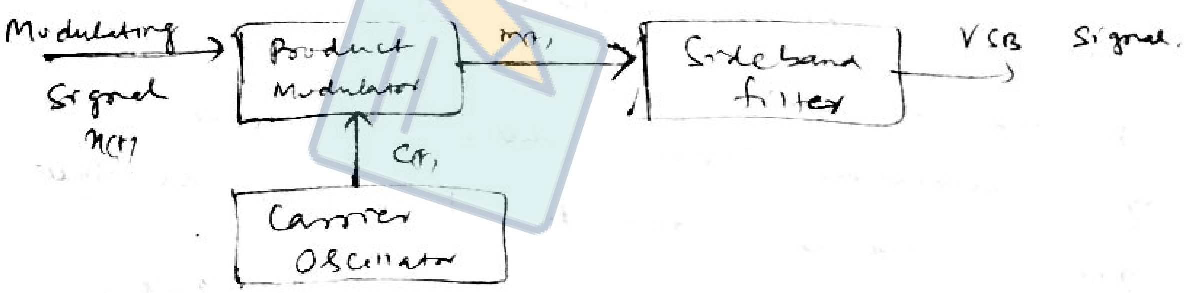
$$V_0(t) = \frac{1}{4} V_c \, x(t) \cos\phi \pm \frac{1}{4} V_c \, \hat{x}(t) \sin\phi .$$

Due to presence of Hilbert transform $\hat{x}$ in the O/P, the detector O/P will suffer from phase distortion. Such distortion does not have serious effect on voice communication but in transmission of music and video, it will have untolerable effect.

Vestigial Sideband transmission (VSB)

The stringent frequency-response requirements on the sideband filter in a SSB AM system can be relaxed by allowing a part, called a vestige, of the unwanted sideband to appear at the o/p of the modulator. Thus, we simplify the design of the sideband filter at the cost of a modest increase in the channel BW required to transmit the signal. The resulting signal is called the Vestigial-sideband (VSB) AM.

Generation of VSB Modulated wave:-



The modulating signal $x(t)$ is applied to a product modulator. The o/p of the carrier oscillator is also applied to the other i/p of the product modulator.

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

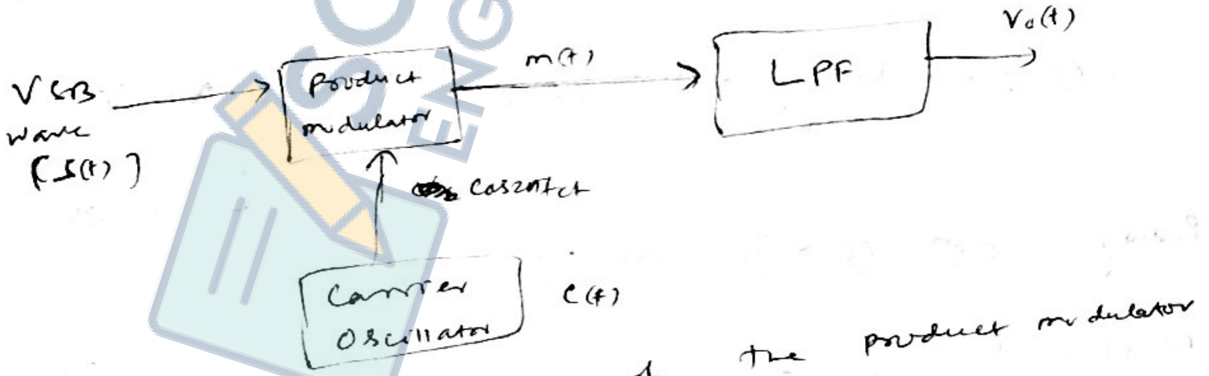
This represent DSB-SC modulated wave. This DSB-SC signal is then applied to a sideband shaping filter. The design of this filter depends on the desired spectrum of the VSB modulated signal. This filter will pass the wanted sideband and the vestige (part) of the unwanted ~~sidepass~~ sideband.

~~Let~~ Let the transfer function of filter be $H(f)$. Hence, the spectrum of the VSB modulated signals given by,

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

Demodulation of VSB wave:-

The synchronous detector for the detection of VSB modulated wave is shown in figure



Mathematically, the O/P of the product modulator is given by

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

Taking Fourier transform both the sides,

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

(32)

$$\therefore M(f) = \frac{1}{2} S(f+f_c) + \frac{1}{2} S(f-f_c) \quad \text{--- ①}$$

by [convolution property]

But $S(f) = \frac{V_c}{2} \left[X(f-f_c) + X(f+f_c) \right] H(f)$

--- ②

$\therefore$ ~~$M(f) = \frac{1}{2} \left[\frac{V_c}{2} \right.$~~

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

$$S(f+f_c) = \frac{V_c}{2} \left[X(f) + X(f+2f_c) \right] \cdot H(f+f_c)$$

--- ③

Similarly

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

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

--- ④

Putting eqn ③ & ④ in eqn ①

$$M(f) = \frac{1}{2} \left[\frac{V_c}{2} \left[X(f) + X(f+2f_c) \right] H(f+f_c) \right]$$

$$+ \frac{1}{2} \left[\frac{V_c}{2} \left[X(f-2f_c) + X(f) \right] H(f-f_c) \right]$$

$$= \frac{V_c}{4} X(f) \left[H(f-f_c) + H(f+f_c) \right]$$

$$+ \frac{V_c}{4} \left[X(f+2f_c) H(f+f_c) + X(f-2f_c) H(f-f_c) \right]$$

The first term represents the spectrum of demodulated VSB o/p.

The second term will be eliminated by the filter to produce o/p $V_o(t)$.

Appl^n :-

VSB modulation has become standard for transmission of T.V signals.

Advantage :-

1) The main advantage of VSB modulation is the reduction of BW. It is almost as efficient as the SSB.

2) Due to allowance of transmitting a part of lower sideband, the constraint on the filters have been relaxed. So practically, easy to design filters can be used.

3) It possesses good phase characteristics and makes the transmission of low frequency components possible.

Frequency Division multiplexing :-

The process of combining a number of separate message signal into a composite signal for transmission over a common channel is called multiplexing.

Two methods commonly used for signal multiplexing

1) Time-division-multiplexing (TDM)
2) Frequency-division-multiplexing (FDM)

(1) (34)

TDM is usually used in the ~~+~~ transmission of digital ~~modulation~~ information.

FDM can be used both in analog & digital communication.

FDM :-

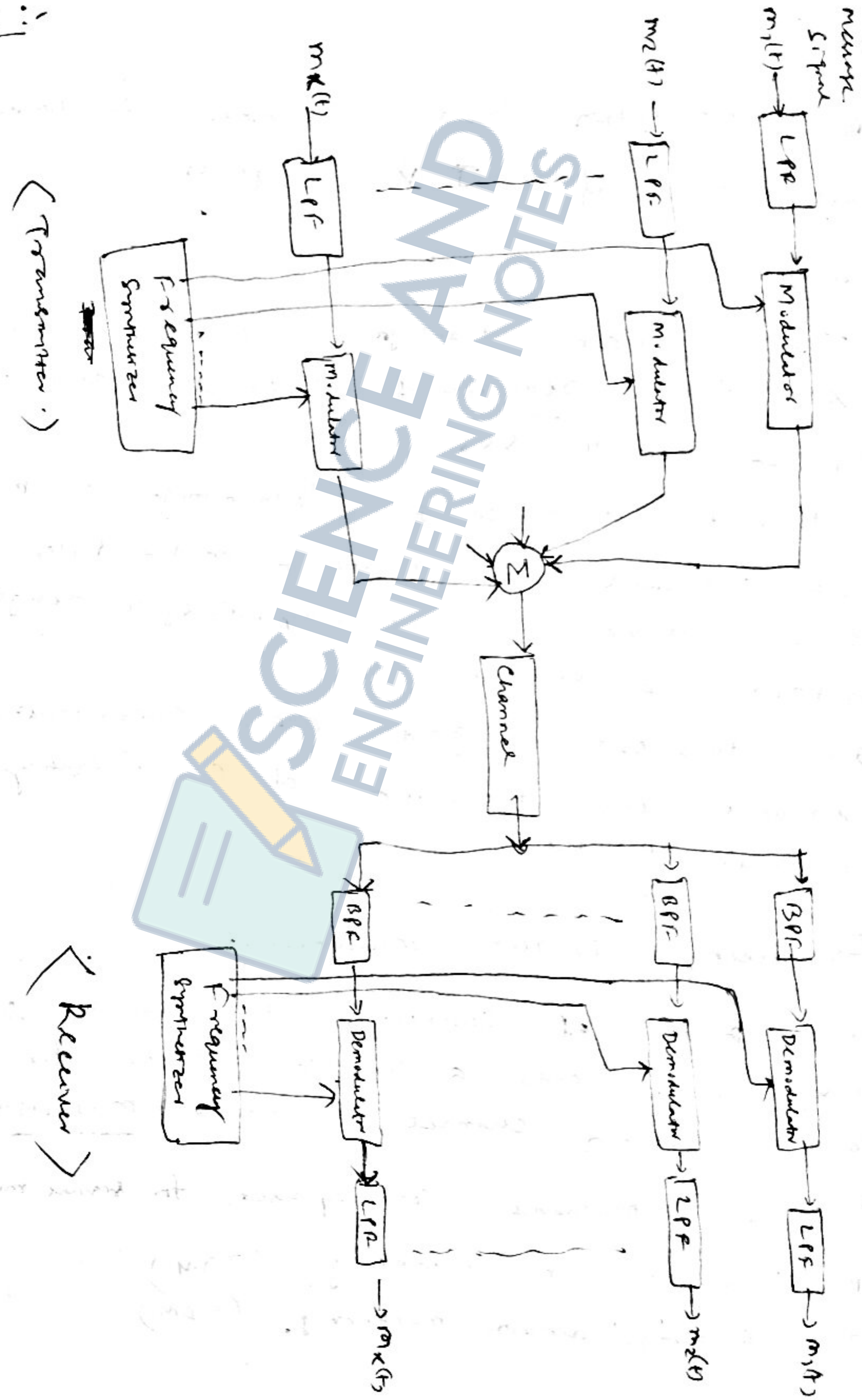


fig:- Frequency - division multiplexing of multiple signals

(35) In FDM, the message signals are separated on frequency. The figure shows FDM of K message signal at the transmitter and their demodulation at the receiver.

The LPF at the transmitter are used to ensure that the BW of message signal is limited to 'W' HZ. Each signal modulates a separate carrier; hence K modulators are required. Then, the signals from the K modulators are summed and transmitted over the channel.

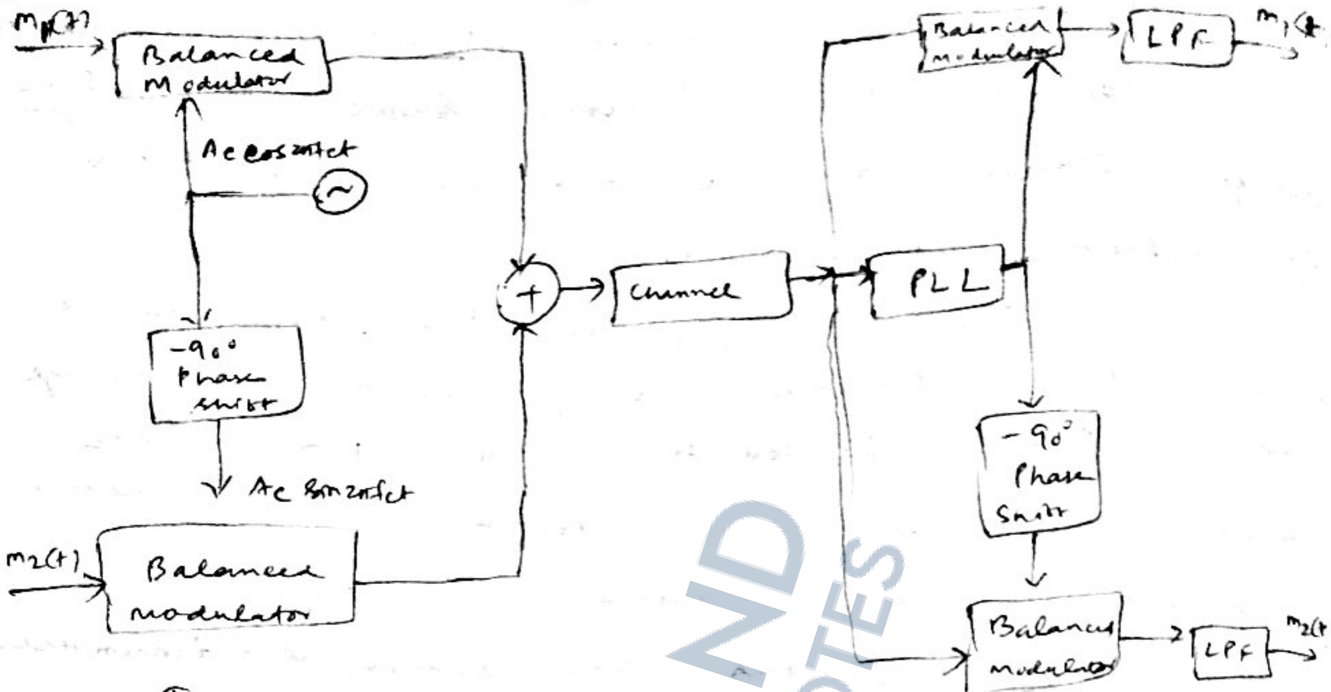
At the receiver of FDM system, the signals are usually separated by passing through a parallel band pass filters, where each filter is tuned to one of the carrier frequencies and has bandwidth sufficiently wide to pass the desired signal. The O/P of each BPF is demodulated and each demodulated signal is fed to a low pass filter that passes the baseband message signal.

FDM is widely used in Radio and telephone communication.

Quadrature-Carrier Multiplexing:-

This type of multiplexing allows us to transmit two message signals on the same carrier frequency, using 2 quadrature carriers $A_c \cos 2\pi f_c t$ and $A_c \sin 2\pi f_c t$.

Suppose $m_1(t)$ and $m_2(t)$ are two separate message signals to be transmitted over the channel.



Transmitter

fig:- Quadrature-carrier multiplexing.

The signal $m_1(t)$ amplitudes the carrier $A_c \cos 2\pi f_c t$ and the signal $m_2(t)$ amplitude modulates the quadrature carrier $A_c \sin 2\pi f_c t$. The two signals are added and transmitted over the channel. Hence the transmitted signal is

$$u(t) = A_c m_1(t) \cos 2\pi f_c t + A_c m_2(t) \sin 2\pi f_c t$$

Therefore, each message signal is transmitted by DSB-SC AM. This type of multiplexing is called <u>quadrature-carrier multiplexing</u>.

As shown, a synchronous demodulator is required at the receiver to separate and recover the quadrature-carrier modulated signals.

Quadrature-carrier multiplexing results in a band-width efficient communication system that is comparable in bandwidth efficiency to SSB AM.

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Angle Modulation:-

Angle modulation may be defined as the process in which the total phase angle of a carrier wave is varied in accordance with the instantaneous value of the modulating signal while keeping the amplitude of the carrier constant.

Lets consider than an unmodulated carrier signal is expressed as

$$c(t) = A\cos(\omega_c t + \theta_0)$$

where A = Amplitude of the carrier.

ω_c = Carrier frequency.

θ_0 = some phase angle.

Substituting $\omega_c t + \theta_0 = \phi$, we get

$$c(t) = A\cos\phi$$

where $\phi = \omega_c t + \theta_0 =$ total phase angle of the carrier signal.

Now we can vary this phase angle (ϕ) in two ways and there are two types of angle modulation.

Angle Modulation

- Frequency Modulation (FM)
- Phase Modulation (PM)

Frequency Modulation:-

In frequency modulation, frequency of carrier is varied in accordance to the instantaneous value of message signal.

Let w_c = Carrier frequency.

$x(t)$ = message signal.

w_i = instantaneous angular frequency value of modulated signal.

then,

$$\boxed{w_i = w_c + K_f\, x(t)} \quad \text{--- (1)}$$

where K_f is proportionality constant and is known as frequency sensitivity of the modulator. This expressed in (Hz/volt).

Now, Let the unmodulated carrier is given by

~~$c(t) = A_c \cos\phi$~~

$$c(t) = A \cos\phi$$

where $\phi = w_c t + \theta_0$

Let ϕ_i be the instantaneous phase angle of the modulated signal.

So the expression for frequency modulated wave will be

$$s(t) = A \cos\phi_i \quad \text{--- (2)}$$

We know

$$\phi = w_c t + \theta_0$$

$$\Rightarrow \frac{d\phi}{dt} = w_c$$

$$\Rightarrow \phi = \int w_c\, dt$$

$$\therefore \phi_i = \int w_i\, dt$$

$$= \int \left(w_c + K_f\, x(t) \right) dt$$

$$\left(\because w_i = w_c + K_f\, x(t),\ \text{eq}^n\ (1) \right)$$

$$\Rightarrow \quad \phi_i = \omega_c t + \int K_f\, x(t)\, dt \quad \text{--- (3)}$$

Putting eq^n (3) in eq^n (2),

$$S(t) = A \cos\left[\omega_c t + \int K_f\, x(t)\, dt\right]$$

Now, if phase angle of unmodulated carrier be taken at $t=0$, then the limit of integration will be 0 to t.

In this case the expression for FM wave will be

$$\boxed{S(t) = A \cos\left[\omega_c t + K_f \int_0^t x(t)\, dt\right]}$$

which is the required general expression for FM wave.

Phase Modulation (PM)

In PM, the phase angle of the carrier is varied according to the message signal.

Mathematically,

Let $c(t) = A \cos(\omega_c t + \theta_0)$

$c(t) = A \cos \phi$

where $\phi = \omega_c t + \theta_0$

Neglecting θ_0, we get total phase angle of unmodulated carrier as

$\phi = \omega_c t$.

Now, according to phase modulation, this phase angle ϕ is varied linearly with baseband or modulating signal $x(t)$.

Let the instantaneous value of phase angle be denoted by ϕ_i

Therefore

$$\boxed{\phi_i = \omega_c t + K_p\, x(t)} \text{ ——— } (1)$$

where K_p is the proportionality constant and is known as phase sensitivity of the modulator. This is expressed in radians/volts.

The expression for phase modulated wave will be

$$s(t) = A \cos \phi_i \text{ ——— } (2)$$

Putting the value of ϕ_i from eqn (1) in eqn (2), we have

$$\boxed{s(t) = A \cos [\omega_c t + K_p\, x(t)]}$$

which is the required mathematical expression for phase modulate wave.

Frequency deviation :-

The max^m change in instantaneous frequency from the average frequency (ω_c) is called frequency deviation.

We know,

$$\omega_i = \omega_c + K_f\, x(t)$$

The max^m change in (ω_i), from the avg of carrier frequency ω_c depends on the magnitude and sign of $K_f\, x(t)$.

$K_f\, x(t)$ is max^m i.e. $|K_f\, x(t)|_{max}$

(41) If $x(t) = A_m \cos \omega_m t$.

Then frequency deviation $(\Delta\omega)$

$$\Delta\omega = |K_f \; x(t)|_{max}$$

$$\boxed{\Delta\omega = K_f A_m}$$

BPUT-2004-05

Relation between (phase modulation) & (Freq. modulation)

We know that an angle modulated wave is given as,

$$S(t) = A \cos \phi_i$$

where A = Amplitude

ϕ_i = Instantaneous total phase angle of modulated wave.

The angle modulated wave is given by

For FM, the modulated wave is given by

$$S(t) = A \cos\left[\omega_c t + K_f \int_0^t x(t)\, dt\right]$$

For PM,

$$S(t) = A \cos\left[\omega_c t + K_p\, x(t)\right]$$

It is observed that PM & FM are closely related to each other because in both the cases there is a variation in the total phase angle.

In PM, the phase angle varies linearly with with baseband signal $x(t)$ whereas in case of frequency modulation, the phase angle varies linearly with integral of baseband signal $x(t)$. This

(42) means FM may be obtained using PM and vice versa.

FM using PM :-

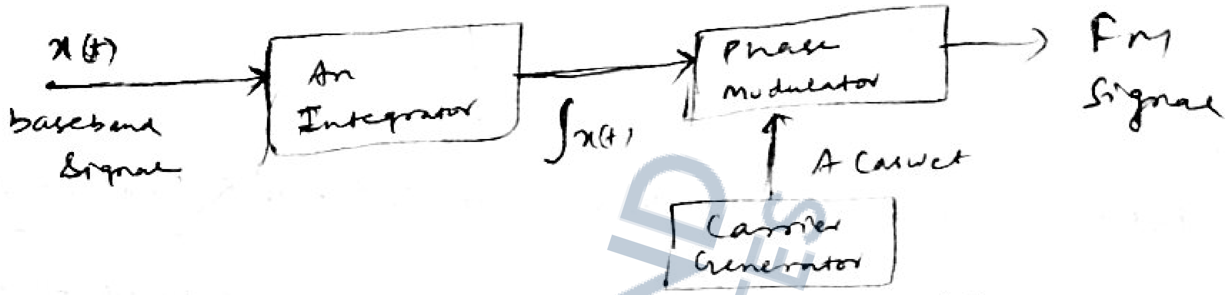


fig:- Generation of FM using Phase modulator

To get FM by using PM, we first integrate the baseband signal and then apply to the phase-modulator.

PM using FM :-

Similarly, PM wave may be generated by using frequency modulator by first differentiating modulating or base band signal x(t) and then applying to the frequency modulator.

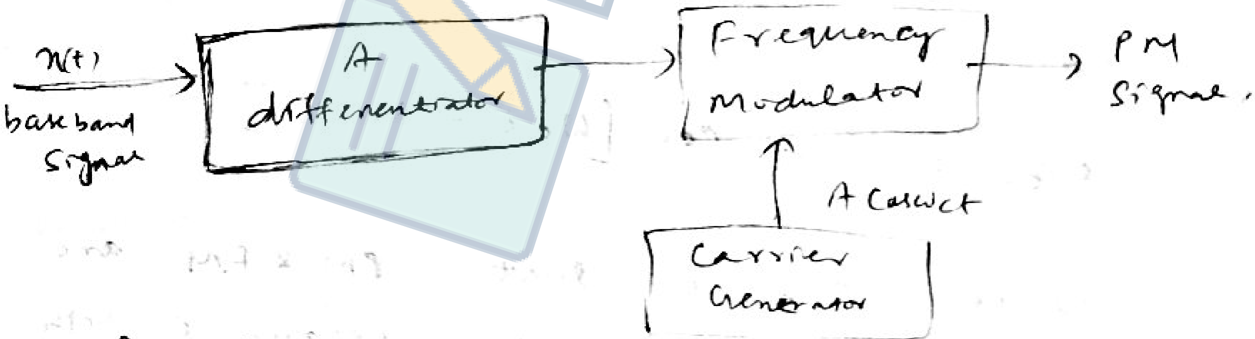


fig:- Generation of PM using frequency modulator.

Single-tone frequency modulation:-

↳ Assuming base band signal contain single frequency.

Let us consider a carrier signal as

$$c(t) = A_c \cos \omega_c t \qquad (1)$$

Let modulating signal be

$$x(t) = V_m \cos \omega_m t \quad \text{—— (2)}$$

Where V_m = Maximum amplitude of modulating signal.
ω_m = frequency of modulating signal.

Let the expression for FM wave be

$$S(t) = A_c \cos \phi_i \quad \text{—— (3)}$$

where ϕ_i is the instantaneous Phase angle of the modulated wave.

We know for FM,

$$\omega_i = \omega_c + K_f\, x(t) \quad \text{—— (4)}$$

Putting the value of $x(t)$, we get

$$\omega_i = \omega_c + K_f V_m \cos \omega_m t \quad \text{—— (5)}$$

frequency deviation is given by

$$\Delta \omega = |K_f\, x(t)|_{max} = K_f\, |x(t)|_{max} = K_f . V_m$$

$\therefore$ frequency deviation = $\boxed{\Delta \omega = K_f V_m}$, Putting the value of $K_f V_m$ in eqn (5), we get

Therefore,
$$\omega_i = \omega_c + \Delta \omega \cos \omega_m t \quad \text{—— (6)}$$

Total phase angle $\phi_i = \int \omega_i \, dt$

$$\therefore \phi_i = \int (\omega_c + \Delta \omega \cos \omega_m t)\, dt$$

$$= \omega_c t + \Delta \omega . \frac{\sin \omega_m t}{\omega_m}$$

$$\phi_i = \omega_c t + \frac{\Delta \omega}{\omega_m} \sin \omega_m t$$

$\cancel{\omega_c t + \Delta}$

Let $\frac{\Delta \omega}{\omega_m} = m_f$ = Modulation index

$\therefore$ Modulation index $(m_f) = \dfrac{\text{Frequency deviation}}{\text{modulating frequency}}$.

$\therefore \phi_i = \omega_c t + m_f \sin \omega_m t$ —— (7)

Substituting eq^n (7) in eq^n (3), we have

$$\boxed{S(t) = A \cos[\omega_c t + m_f \sin \omega_m t]} \quad \text{—— (8)}$$

Which is the required mathematical expression for single tone FM wave.

<u>Figure:-</u>

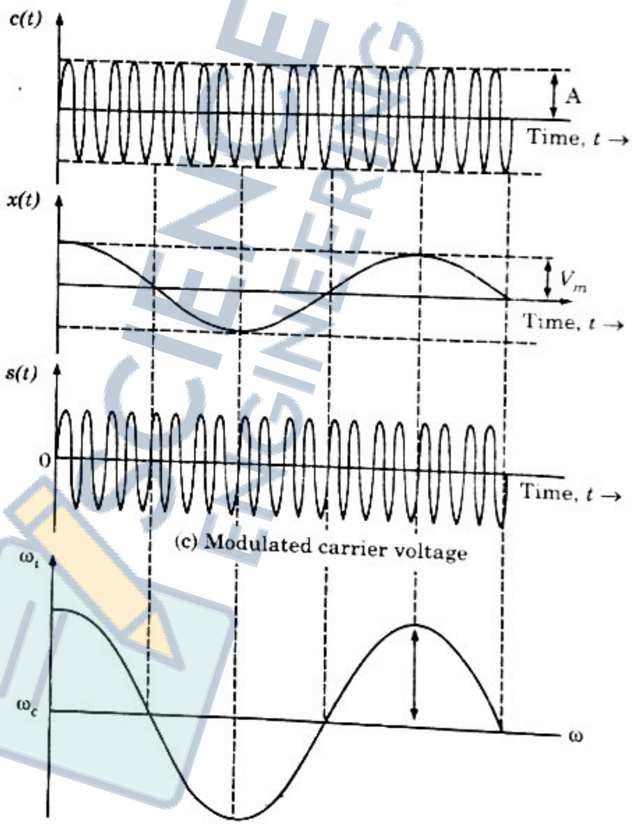


Fig. 4.5 *Illustration of FM (a) Unmodulated carrier (b) Modulating signal (c) FM signal (d) Instantaneous carrier frequency.*

45) **Prob. 1)** A single tone FM is represented by a voltage eqn as

$$v(t) = 12 \cos(6\times10^8 t + 5 \sin 1250 t)$$

Determine the following

(a) Carrier frequency

(b) modulating frequency

(c) the modulation index

(d) maximum deviation

(e) What power will this FM wave dissipate in 10Ω resistors.

Ans ⇒ $V(t) = 12 \cos[6\times10^8 t + 5 \sin 1250 t]$

The standard eqn for FM,

$$V(t) = A \cos[w_c t + m_f \sin w_m t]$$

Comparing, $A = 12$, $w_c = 6\times10^8$, $m_f = 5$

$w_m = 1250$

(a) Carrier frequency $w_c = 6\times10^8 \ \frac{rad}{sec}$

$\Rightarrow f_c = \frac{6\times10^8}{2\pi} = 95.5 \ MHz$

(b) modulating freq, $w_m = 1250 \ \frac{rad}{sec}$

$\Rightarrow 2\pi f_m = 1250$

$\Rightarrow f_m = \frac{1250}{2\pi} = 199 \ Hz.$

(c) modln index $m_f = 5$

(d) maxm deviation $= \Delta w$

we know $m_f = \frac{\Delta w}{w_m} = \frac{\Delta f}{f_m}$

$\Rightarrow \Delta f = m_f \times f_m = 5 \times 199 = 995 \ Hz$

(i)

(4b) (e) The power dissipated is,

$$P = \frac{V_{rms}^2}{R} = \frac{\left(\frac{12}{\sqrt{2}}\right)^2}{10\Omega} = \frac{12^2}{2\times 10} = \frac{144}{20} = 7.2 \text{ watt}$$

Types of Frequency Modulation (FM):-

(1) Narrow band FM:- In this case, K_f is small. Hence the bandwidth of FM is narrow.

2) Wideband FM:- In this case, K_f is large hence the bandwidth of FM is large (wide band)

1) Narrowband FM:-

$$X_{FM}(t) = A\cos(\underbrace{w_c t}_{\alpha} + \underbrace{m_f \sin w_m t}_{\beta})$$

This is in a format $A\cos(\alpha+\beta)$

$$= A[\cos\alpha . \cos\beta - \sin\alpha . \sin\beta]$$

$$X_{FM}(t) = A[\cos w_c t . \cos(m_f \sin w_m t) - \sin(w_c t) . \sin(m_f \sin w_m t)]$$

For narrowband FM, $K_f \to 0$, $m_f = \frac{\Delta w}{w_m} = \frac{K_f A_m}{w_m}$

$\therefore m_f \to 0$

when $\theta \to 0$, $\sin\theta \to \theta$

$$X_{FM}(t) = A[\cos w_c t . \cos 0 - \sin w_c t\,(m_f \sin w_m t)]$$

$$= A[\cos w_c t - m_f \sin w_c t . \sin w_m t]$$

$$X_{FM}(t) = A\cos w_c t - \frac{A m_f}{2} . 2\sin w_c t . \sin w_m t$$

(47) $X_{FM}(t) = A\cos w_c t - \frac{m_f A}{2}\left[\cos(w_c - w_m)t - \cos(w_c + w_m)t\right]$

$$\boxed{X_{NBFM} = A\cos w_c t + \frac{m_f A}{2}\cos(w_c + w_m)t - \frac{m_f A}{2}\cos(w_c - w_m)t}$$

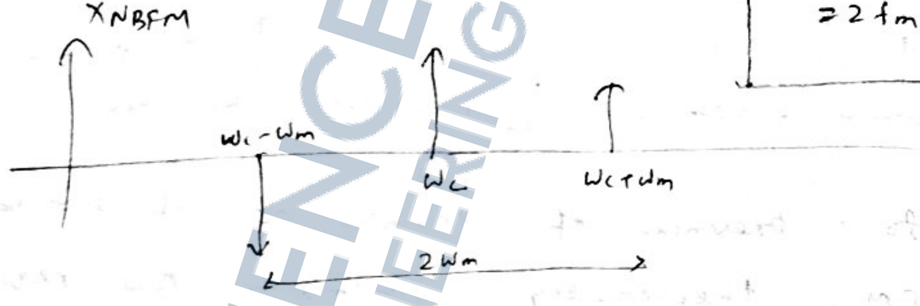
NBFM → Narrowband FM.

But

$$\boxed{X_{AM}(t) = A\cos w_c t + \frac{m A}{2}\cos(w_c + w_m)t + \frac{m A}{2}\cos(w_c - w_m)t}$$

So in narrow band FM, the lower sideband is 180° out of phase with the carrier and upper sideband.

For NBFM,
$BW = 2w_m \ \frac{rad}{sec}$
$= 2f_m \ Hz$



2) Wideband FM:-

When the value of the modulation index (m_f) is quite large, then in FM, a large no. of sidebands are produced and hence the bandwidth of FM is sufficiently large. This type of FM system is known as wideband FM.

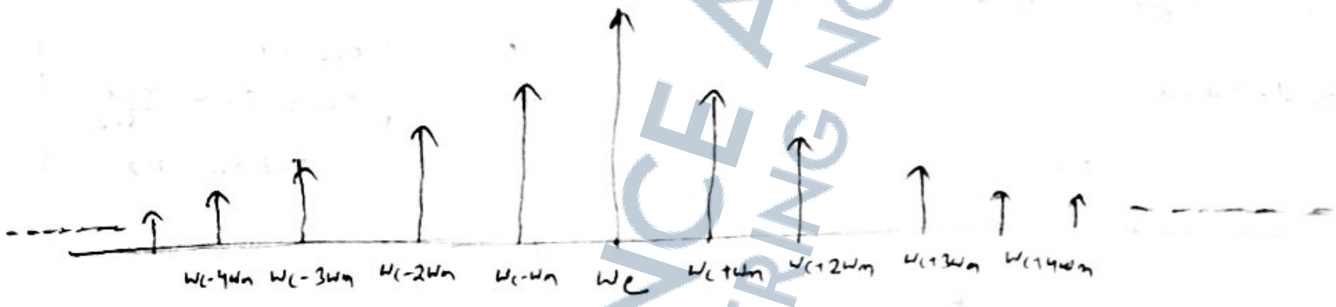
$X_{FM}(t) = A\cos\left[w_c t + m_f \sin w_m t\right]$

$= A\left[\cos w_c t \cdot \cos(m_f \sin w_m t) - \sin w_c t \cdot \sin(m_f \sin w_m t)\right]$

This expression can be expressed in terms of Bessel's function $J_n(m_f)$.

$$X_{WBFM} =$$

$$A\Big[J_0(m_f)\cos\omega_c t - J_1(m_f)\big[\cos(\omega_c-\omega_m)t - \cos(\omega_c+\omega_m)t\big]$$
$$+ J_2(m_f)\big[\cos(\omega_c-2\omega_m)t + \cos(\omega_c+2\omega_m)t\big]$$
$$- J_3(m_f)\big[\cos(\omega_c-3\omega_m)t - \cos(\omega_c+3\omega_m)t\big]$$
$$+ J_4(m_f)\big[\cos(\omega_c-4\omega_m)t + \cos(\omega_c+4\omega_m)t\big]$$
$$+ \text{-----------} \Big]$$



Due to presence of infinite no of sidebands in WBFM, theoretically transmission BW reqd. is infinite. But higher sidebands have amplitude very small such that if these side bands are not transmitted it does not affect the quality of transmission. Practical transmission BW in WBFM is given by Carson's Rule.

$$\boxed{\begin{aligned} BW &= 2(\Delta\omega + \omega_m) \ \ \frac{rad}{sec} \\ BW &= 2(\Delta f + f_m) \ \ Hz \end{aligned}}$$

But $m_f = \frac{\Delta\omega}{\omega_m}$, ~~$BW = 2\Delta\omega($~~

$$BW = 2.\omega_m\left(1 + \frac{\Delta\omega}{\omega_m}\right) = 2\omega_m(1 + m_f)$$

(49)

$\therefore$ ~~BW = 2 mf~~

$\therefore$ $\boxed{BW = 2(1+m_f)\, w_m}$

If $m_f << 1$, ie Narrow band

$\boxed{BW = 2 W_m \text{ or } 2 f_m}$

If $m_f >> 1$ ie wide band

$BW = 2(1+m_f)\, W_m \ \frac{rad}{sec}$

$BW = 2(1+m_f)\, f_m \ HZ$

Prob:- The max^m deviation allowed in an FM broadcast system is 75 KHZ. If the modulating signal is single-tone sinusoid of 8 KHZ, determine the BW of FM signal. What will be the BW if modulating signal amplitude is doubled.

Ans :- Given $\Delta f = 75$ KHZ.

$f_m = 8$ KHZ.

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

$= 2(75+8)$ KHZ

$\boxed{BW = 166 \text{ KHZ}}$

$\therefore$ $(\Delta W = K_f A_m)$

If Amplitude is doubled,

Δf is doubled.

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

$= 2[150+8]$ KHZ $= 2 \times 158 = 316$ KHZ.

(Ans)

Phase modulation :-

The distinct feature of phase-modulation is that the deviation in the carrier frequency(ω_c) is linearly proportional to the baseband frequency (ω_m).

However in FM, the deviation is independent of baseband frequency.

The total phase angle of PM wave is expressed as

$$\phi_i = \omega_c t + K_p x(t) \quad \text{--- (1)}$$

Let single tone modulating signal be

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

$$\therefore \phi_i = \omega_c t + K_p V_m \cos \omega_m t$$

$$\boxed{\text{Phase deviation}(\theta_d) = K_p V_m}$$

$$(S(t))_{PM} = A \cos[\omega_c t + \theta_d \cos \omega_m t].$$

$$= A \cos \phi_i$$

Instantaneous frequency related to ϕ_i is expressed as,

$$\omega_i = \frac{d\phi_i}{dt} = \frac{d}{dt}[\omega_c t + \theta_d \cos \omega_m t]$$

$$= \omega_c + \theta_d . (-\omega_m) \sin \omega_m t$$

$$\omega_i = \omega_c - K_p V_m \omega_m \sin \omega_m t$$

$\therefore$ Thus, the max^m departure in the frequency from ω_c is $K_p V_m \omega_m$.

$$\therefore \boxed{\Delta \omega_{PM} = K_p V_m \omega_m},$$

$\therefore$ frequency deviation of PM depends on the modulating freq.

(5)

$$\boxed{\begin{aligned}(S(t))_{PM} &= A\cos(\omega_c t + K_p V_m \cos\omega_m t)\\ \cancel{\Delta\omega_{PM}} &\cancel{= K_p V_m \omega_m}\\ \Delta\omega_{PM} &= K_p V_m \omega_m\end{aligned}}$$

Transmission BW of PM

$$(BW)_{PM} = 2(\Delta f + f_m) \quad Hz$$

$$= 2(\Delta\omega + \omega_m) \ \frac{rad}{sec}.$$

mod^n index

$$m_p = \frac{\Delta f}{f_m} = \frac{K_p V_m f_m}{f_m}$$

$$\boxed{m_p = K_p V_m}$$

where $\boxed{\begin{aligned}\Delta\omega &= K_p V_m \omega_m\\ \text{or} \quad \Delta f &= K_p V_m f_m.\end{aligned}}$

mod^n index $\boxed{m_p = K_p V_m}$

Prob: 3 A baseband signal $x(t) = 5\cos 2\pi 15\times10^3 t$ angle modulates a carrier signal $A\cos\omega_c t$.

(i) Determine the modulation index & BW for

(a) FM system (b) PM system.

(ii) Find the change in BW & modulation index for both FM & PM if f_m reduced to 5 KHZ.

Assume $K_f = K_p = 15 \frac{KHZ}{volt}$.

Ans: $x(t) = 5\cos 2\pi\times15\times10^3\times t$

$f_m = 15\times10^3 = 15KHZ.$

$V_m = 5$ volt.

(i)

(a) for FM

$$m_f = \frac{\Delta f}{f_m} = \frac{K_f V_m}{f_m} = \frac{15\times5\times10^3}{15\times10^3} = 5$$

$$BW = 2(\Delta f + f_m) = 2(15\times5\times10^3 + 15\times10^3)$$

$$= 2(5+1)\times15\times10^3$$

$$\boxed{BW = 180\ KHZ}$$

$K_f V_m$

$= 15\times10^3\times5$

(b) For PM

$$\Delta f = K_p V_m f_m$$
$$= 15\times10^3 \times 5 \times 15\times10^3$$
$$= 1125 \text{ MHz}.$$

$$BW = 2(\Delta f + f_m)$$
$$= 2(1125\times10^6 + 15\times10^3)$$
$$= 2(225\times15\times10^3\times10^3 + 15\times10^3)$$
$$= 2\times15\times10^3(225000 + 1)$$
$$\simeq 2\times15\times10^3 \times 225\times10^3$$
$$\boxed{BW \simeq 2250 \text{ MHz}}$$

modi^n index $m_p = K_p V_m = 15\times10^3\times5 = 75 \text{ KHz}$

(ii) If $f_m = 5 \text{ KHz}$.

FM:
$$m_f = \frac{\Delta f}{f_m} = \frac{K_f V_m}{f_m} = \frac{15\times10^3\times5}{5\times10^3} = 15.$$
$$BW = 2(\Delta f + f_m) = 2(15\times10^3\times5 + 5\times10^3)$$
$$= 160 \text{ KHz}.$$

PM:
$$\Delta f = K_p V_m f_m = 15\times10^3\times5\times5\times10^3$$
$$= 375 \text{ MHz}.$$
$$BW = 2(\Delta f + f_m) = 2(375 \text{ MHz} + 5 \text{ KHz})$$
$$\simeq 2\times375 \text{ MHz}$$
$$= 750 \text{ MHz}$$

$m_p = K_p V_m = 15\times10^3\times5 = 75{,}000 = 75 \text{ KHz}$

(Same as that previous one)

(53) Bput 2006-07

FM	AM
1) Amplitude of FM wave is const.	1) Amplitude of AM wave will change with modulating voltage.
2) Hence transmitted power remains const.	2) Transmitted power dependent on modulation index.
3) All transmitted power is useful.	3) Carrier power & sideband power are useless
4) FM receiver are immune to noise,	4) AM receiver are not immune to noise.
5) $BW = 2(\Delta f + f_m)$	5) $BW = 2 f_m$.
6) BW is large. wide channel required.	6) BW much less than FM
7) FM transmission & reception equipments are more complex	7) AM equipments are less complex
8) The number of sidebands having significant amplitude depends on modulation index (m_f)	8) Number of sidebands in AM will be const. and equal to 2.
9) The information is contained in the freq. variation of the carrier.	9. The information is contained in the amplitude variation of the carrier.

FM	PM
1) If baseband signal is $V_m \cos w_m t$ and Carrier is $A \cos w_c t$ than FM signal is given by $S(t) = A \cos[w_c t + m_f \sin w_m t]$	1) If base band signal is $V_m \cos w_m t$ & Carrier is $A \cos w_c t$ the PM signal is given by $S(t) = A \cos[w_c t + m_p \cos w_m t]$
2) ~~$\Delta f = K_f A$~~ $\Delta f = K_f V_m$ frequency deviation independent of modulating freq	2) ~~$\Delta f = K_p V_m f_m$~~ $\Delta f = K_p V_m f_m$. freq. deviation depends on modulating freq.
3) modulation index $m_f = \frac{\Delta f}{f_m} = \frac{K_f V_m}{f_m}$	3) Modulation index $m_p = \frac{\Delta f}{f_m} = K_p V_m$
4) It is possible to receive FM on a PM receiver.	4) It is possible to receive PM on a FM receiver
5) Noise immunity is better than AM & PM	5) Noise immunity better than AM but less than FM
6) Amplitude of FM is Const.	6) Amplitude of PM is also Constant.
7) Signal to Noise ratio (SNR) is better than PM.	7) SNR is ~~less~~ inferior to FM.
8) FM is widely used.	8) PM is used in some mobile system

55 FM Generation:-

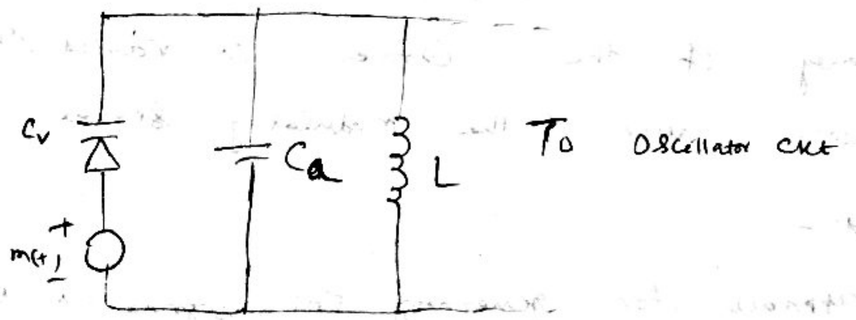
Angle modulators are in general, time-varying and nonlinear systems. One method for generating a FM signal directly is to design an oscillator whose frequency changes with i/p voltage. When i/p voltage is zero, the oscillator generates a sinusoid with frequency fc and when the i/p voltage changes, frequency changes accordingly.

There are 2 approaches to design such an oscillator, usually called a VCO (Voltage Controlled Oscillator).

1st method:- (To use Varactor diode) ⟨Direct method⟩

A varactor diode is a capacitor whose capacitance changes with applied voltage. Therefore if this capacitor is used in a tuned ckt of the oscillator & the message signal is applied to it, the frequency of the tuned ckt will change in accordance with message signal.

Methods of FM generation
- Direct method
  - Varactor diode modulator
  - Reactance modulator
- Indirect method
  - Armstrong method.



(fig:- Varactor diode implementation of an angle modulator)

(56) Lets assume that the inductance of the inductor in the tuned ckt is L_0 and the capacitance of the varactor diode is given by

$$C(t) = C_0 + K_0 m(t)$$

When $m(t)=0$, the frequency of the tuned ckt is given by $f_c = \frac{1}{2\pi\sqrt{L_0 C_0}}$. In general for nonzero $m(t)$, we have

$$f_i(t) = \frac{1}{2\pi\sqrt{L_0(C_0+K_0 m(t))}}$$

$$= \frac{1}{2\pi\sqrt{L_0 C_0\left(1+\frac{K_0 m(t)}{C_0}\right)}}$$

$$= \frac{1}{2\pi\sqrt{L_0C_0}} \cdot \frac{1}{\sqrt{1+\frac{K_0 m(t)}{C_0}}}$$

Assuming, $\frac{K_0 m(t)}{C_0} << 1$,

$$= \frac{1}{2\pi\sqrt{L_0C_0}} \left(1+\frac{K_0 m(t)}{C_0}\right)^{-\frac{1}{2}}$$

$$f_i(t) = \frac{1}{2\pi\sqrt{L_0C_0}}\left(1-\frac{K_0 m(t)}{2C_0}\right) \quad \left((1+x)^n \simeq 1+nx \text{ when } x<<1\right)$$

$$f_i(t) = f_c\left(1-\frac{K_0 m(t)}{2C_0}\right)$$

$\therefore$ frequency of the carrier is varied with instantaneous value of the modulating signal.

2nd Method:-

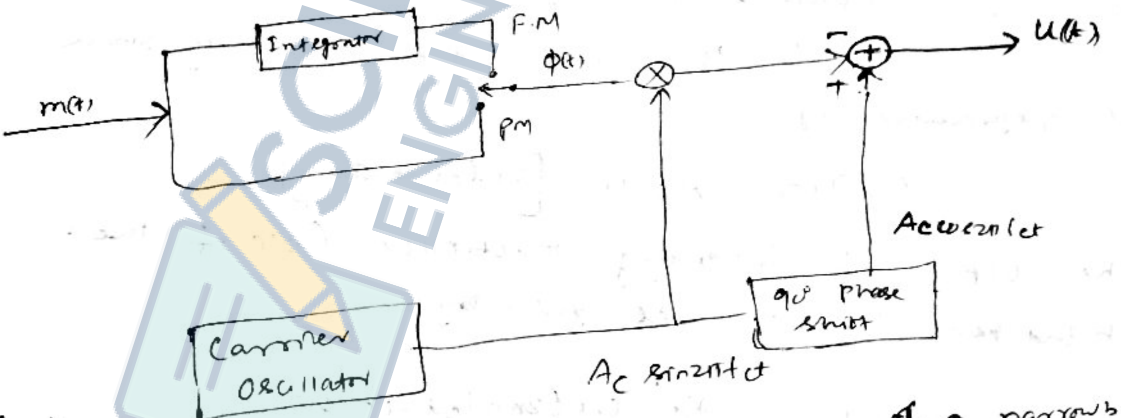
A second approach for generating FM signal is by use of a reactance tube. In a reactance tube

57 implementation, an inductor whose inductance varies with the applied voltage is employed and the analysis is very similar to the analysis presented for the varactor diode.

Since PM & FM are closely related & PM can be formed using FM modulators, the same method is used for PM ~~generator~~ generation.

Indirect method (Armstrong Method)

In this method, first narrowband angle-modulated signal is generated and then it is changed to wideband signal. Due to similarity of conventional AM signals, generation of narrowband angle-modulated signal is straight forward. In fact any modulator for conventional AM generation can be easily modified to generate a narrow-band angle-modulated signal.



fig(1)

Fig(1) shows the block diagram of a narrowband angle modulator.

The next step is to use the narrowband angle-modulated signal to generate wideband angle-modulated signal. Fig(2) shows the block diagram of a system that generates wideband

1 (58) angle modulated signals form narrow band angle-modulated signal.

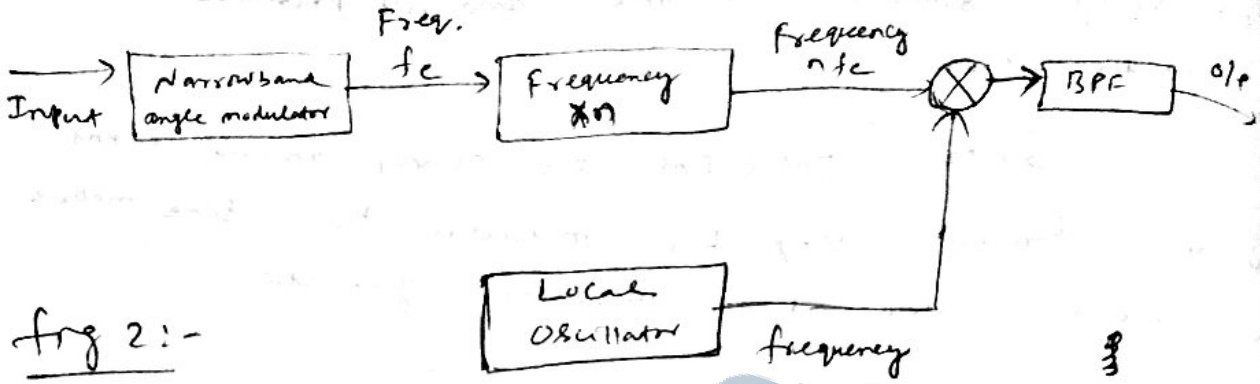


fig 2:-

The first stage of such system is, of course a narrowband ~~angle modulated~~ angle-modulator shown in fig (1). The narrowband angle modulated signal enters a frequency multiplier that multiplies ~~that~~ the instantaneous frequency of the i/p by some constant n. This is usually done by applying the i/p signal to a nonlinear element and then passing its o/p through a bandpass filter tuned to the desired central frequency.

If the narrowband modulating signal is represented by

$$u_n(t) = A_c \cos\left[2\pi f_c t + \phi(t)\right]$$

the o/p of frequency multiplier (o/p of the band pass filter) is given by

$$y(t) = A_c \cos\left[2\pi n f_c t + n\,\phi(t)\right].$$

In general, this is a wideband angle modulated signal. However, there is no guarantee that the carrier frequency of signal, nfc, will be the desired carrier frequency. The last stage of modulator performs up or down conversion to shift the modulated signal to the desired center frequency

This stage consist of a mixer and a bandpass filter. If the frequency of the local oscillator of the mixer is f_{LO} & we are using a down converter, final wideband angle modulated signal is given by

$$u(t) = A_c \cos\left(2\pi(nf_c - f_{LO})t + n\phi(t)\right)$$

FM demodulation -

FM demodulators are implemented by generating an AM signal whose amplitude is proportional to the instantaneous frequency of the FM signal and then using an AM demodulator to recover the message signal.

To implement the first step i.e transforming FM signal into AM signal, it is enough to pass FM signal through an LTI system whose frequency response is approximately a straight line in the frequency band of FM signal. If the frequency response of such system is given by

$$|H(f)| = V_0 + K(f - f_c) \quad , \text{for } |f - f_c| < \frac{B_c}{2}$$

and if the i/p to the system is

$$u(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\tau)\, d\tau\right)$$

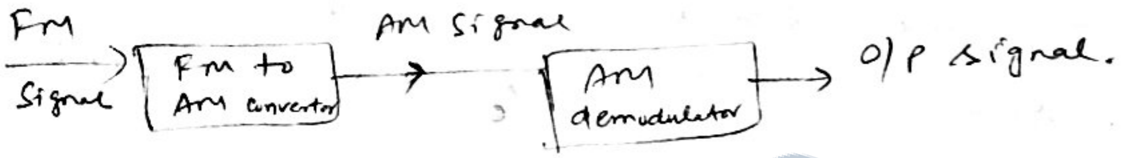
then o/p will be the signal

$$v_0(t) = \underbrace{A_c\left(V_0 + K k_f m(t)\right)}\ \underbrace{\cos\left(2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\tau)\, d\tau\right)}$$

(multiplied by T.F)

(6)

The next step is to demodulate this signal to obtain $A_c(V_0 + K K_f\, m(t))$, from which the message $m(t)$ can be recovered.



There exist many ckt that can be used to implement the first stage of FM demodulator i.e FM to AM converter. One such method is a simple ~~differentiation~~ differentiation [Hints: In FM, we have integration $\int m(t)$ but in AM no integration. If we differentiate FM, get AM)

Another method is to used, i.e two ~~tuned~~ ckts tuned at two frequencies f_1 and f_2 are connected in a configuration which is known as balanced discriminator.

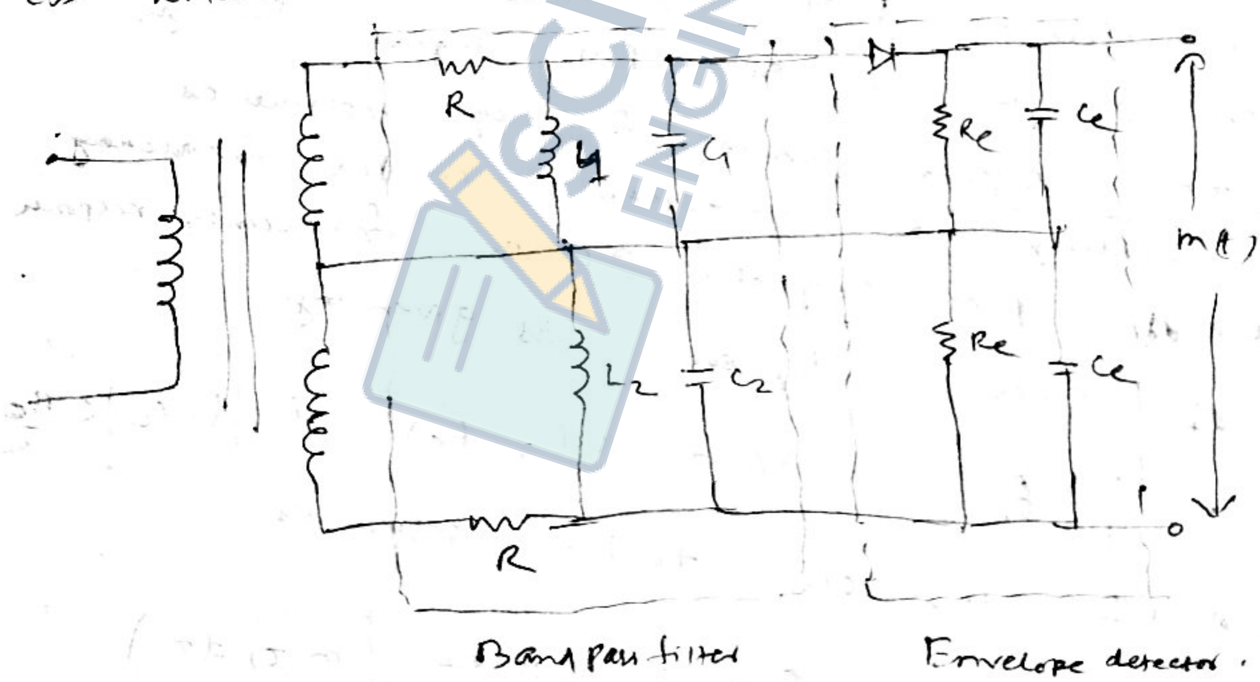
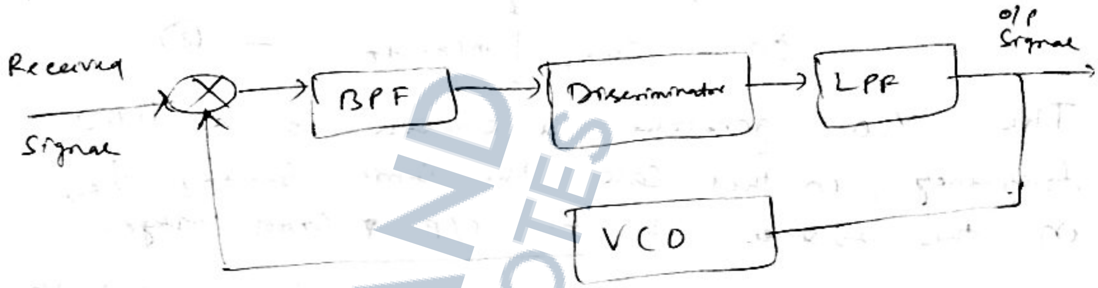


fig:- A balanced discriminator and corresponding frequency response.

(61)

2nd method:- (FMFB)

The second approach is to use feedback on the FM demodulator to narrow the band width of FM detector.



This method is used to reduce the noise power at the o/p of demodulator.

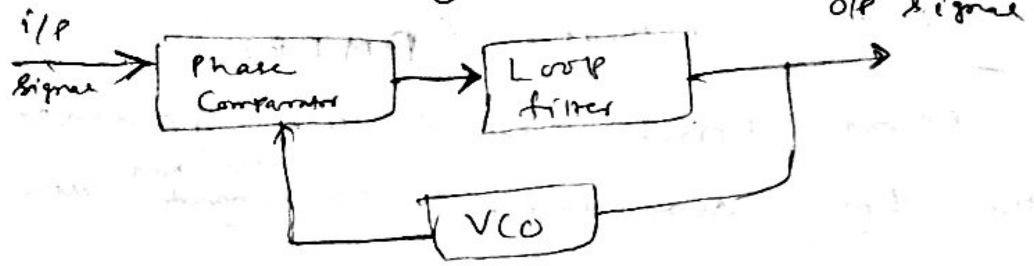
In this approach, a system in which the FM discrimination is placed on the feedback branch of a feedback system that employs a VCO path. The BW of discriminator and the subsequent lowpass filter is designed to match the BW of message signal m(t). The o/p of the lowpass filter is the desired message signal. This type of FM demodulator is called an FM demodulation with feedback (FMFB).

3rd Method:- (Use of PLL)

The i/p to the PLL (Phase locked Loop) is the angle-modulated signal.

$$u(t) = A_c \cos[2\pi f_c t + \phi(t)] \quad \text{—— (1)}$$

(1)



Where, for FM,

$$\phi(t) = 2\pi k_f \int_{-\infty}^{t} m(\tau)\, d\tau \qquad \text{—— (2)}$$

The VCO generates a sinusoid of a fixed frequency, in this case the carrier frequency f_c, in the absence of an i/p control voltage.

Now, suppose that the control voltage to the VCO is the O/P of the loop filter, denoted by $v(t)$. Then, the instantaneous frequency of the VCO is

$$f_v(t) = f_c + K_v\, v(t) \qquad \text{—— (3)}$$

Where K_v is a deviation constant with units Hz/volt. Consequently the VCO o/p may be expressed as

$$y_v(t) = A_v \sin[2\pi f_c t + \phi_v(t)] \qquad \text{—— (4)}$$

where $\phi_v(t) = 2\pi K_v \int_{0}^{t} v(\tau)\, d\tau \qquad \text{—— (5)}$

The phase comparator is basically a multiplier and filter that rejects the signal components centered at $\underline{2f_c}$. Hence, its o/p may be expressed as

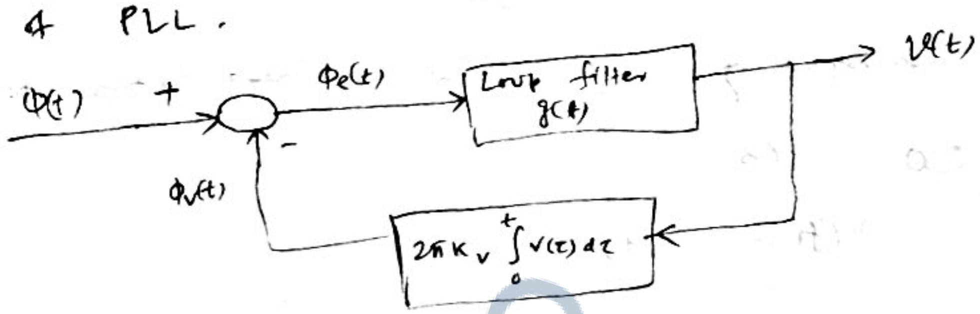
$$e(t) = \frac{1}{2} A_v A_c \sin[\phi(t) - \phi_v(t)] \qquad \text{—— (6)}$$

Where the difference, $\phi(t) - \phi_v(t) \equiv \phi_e(t)$, constitutes the phase error. The signal $e(t)$ is the i/p to the loop filter.

Lets assume the PLL is in lock, so that phase error is small then

$$\sin[\phi(t) - \phi_v(t)] \simeq \phi(t) - \phi_v(t) = \phi_e(t)$$

Under this condⁿ, we may deal with the linearized model of PLL.



ϕ_e may be expressed as,

$$\phi_e(t) = \phi(t) - 2\pi K_v \int_0^t v(\tau)\,d\tau$$

differentiating,

$$\frac{d}{dt}\phi_e(t) = \frac{d}{dt}\phi(t) - 2\pi K_v\, v(t)$$

$$\Rightarrow \frac{d}{dt}\phi_e(t) + 2\pi K_v \int_0^{\infty} \phi_e(\tau)\, g(t-\tau)\,d\tau = \frac{d}{dt}\phi(t)$$

$$[\because v(t) = \phi_e(t) * g(t)]$$

Taking F.T both the sides.

$$\Rightarrow j2\pi f\, \phi_e(f) + 2\pi K_v\, \phi_e(f).\, G(f) = (j2\pi f)\, \phi(f).$$

$$\left[\because F\left[\frac{d}{dt}x(t)\right] = j\omega\, X(\omega) = j2\pi f\, X(f),\quad F[\phi_e(t) * g(t)] = \phi_f \cdot G(f)\right]$$

~~$\Rightarrow jf\,\phi_e(f) = jf\,\phi(f)$~~

$$\Rightarrow \phi_e(f)\,[jf + K_v G(f)] = jf\,\phi(f)$$

$$\Rightarrow \phi_e(f) = \frac{\cancel{jf}\,\phi(f)}{\cancel{jf}\left[1 + \left(\frac{K_v}{jf}\right).G(f)\right]} = \frac{\phi(f)}{\left[1 + \frac{K_v}{jf}\cdot G(f)\right]}$$

We design $G(f)$ such that, $\left|\frac{K_v}{jf}G(f)\right| \gg 1$

$$\Rightarrow \quad \Phi_e(f) = \frac{\Phi(f)}{\frac{K_v}{jf} \cdot G(f)}$$

The corresponding eqn for control voltage to the VCO is

$$V(f) = \Phi_e(f) \cdot G(f)$$

$$= \frac{\Phi(f)}{\frac{K_v}{jf} \cdot \cancel{G(f)}} \cdot \cancel{G(f)}$$

$$V(f) = \frac{jf \; \Phi(f)}{K_v}$$

$$= \frac{j 2\pi f \; \Phi(f)}{2\pi K_v}$$

Taking I.F.T,

$$v(t) = \frac{1}{2\pi K_v} \cdot \frac{d}{dt}\left[\phi(t)\right] \qquad \left[F\left[\frac{d}{dt}\phi(t)\right] \leftrightarrow j2\pi f \; \Phi(f) \right]$$

$$v(t) = \frac{1}{2\pi K_v} \cdot m(t) \cdot 2\pi \cdot K_f = \frac{K_f}{K_v} \cdot m(t)$$

Since control voltage of VCO is proportional to message signal, $v(t)$ is the demodulated signal.

__Advantage of using feedback:-__

To improve the SNR (Signal to Noise ratio) of the signal.

Radio & T.V broadcasting:-

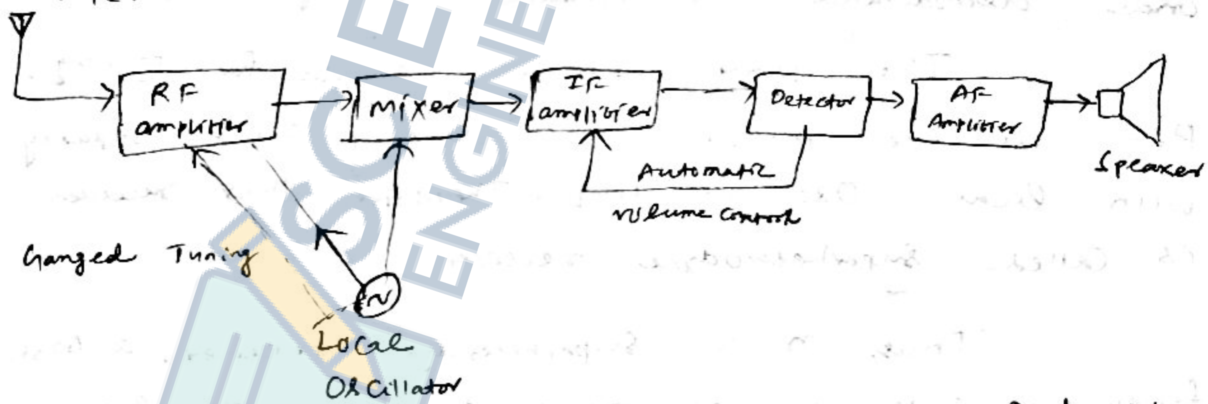
A.M Radio broadcasting:-

→ In AM radio broadcasting freq band → 535-1605 KHz for transmission of voice & music

→ Carrier frequency 540-1600 KHz with 10-KHz spacing.

→ Radio stations employs conventional AM for signal transmission. The baseband message m(t) is limited to a bandwidth of approximately 5 KHz. Since there are billions of receivers & relatively few radio transmitters, the use of conventional AM for broadcast is justified from an economic standpoint.

The most commonly used A.M radio receiver broadcast is the so called super heterodyne receiver. shown in fig. below.



It consist of a radio frequency (RF) tuned amplifier, a mixer, a local oscillator, an intermediate (IF) amplifier, an envelope detector, an audio frequency amplifier and a loud speaker.

Tuning for the desired radio frequency is provided by a variable capacitor which simultaneously tunes the RF amplifier and frequency of the local oscillator.

In superheterodyne receiver, every AM radio signal is converted to a common IF frequency $\boxed{f_{IF} = 455\text{ KHz}}$.

The IF amplifier is designed to have BW of 10 KHz, which matches the BW of the transmitted signal.

The frequency conversion to IF is performed by the combination of the RF amplifier and mixer. The frequency of local oscillator is

$$f_{LO} = f_c + f_{IF} \qquad (955\text{ KHz} < f_{LO} < 2055\text{ KHz})$$

where f_c is the carrier frequency of the desired AM radio signal.

$$\Rightarrow \boxed{f_{IF} = f_{LO} - f_c}$$

The IF freq signal is now amplified and demodulated to produce the original signal.

The word heterodyne stands for mixing. Here we have mixed the incoming signal frequency with local oscillator freq. Therefore this receiver is called superheterodyne receiver.

Thus, in a superheterodyne receiver, a const. frequency difference is maintained between the local oscillator signal frequency and incoming RF signals freq through capacitance tuning in which the capacitances are ganged together and operated by a common control knob.

The IF amplifier generally contains a number of transformers each consisting a pair of mutually coupled tuned ckts. Thus, with this large number of ~~[illegible]~~ double-tuned ckts, operating at a specially chosen freq, the IF amplifier provides most of gain (i.e. sensitivity) and band width requirements (selectivity) of the receiver.

Also, since the characteristics of the IF amplifiers are independent of the incoming frequency to which the receiver is tuned, the selectivity and sensitivity of the superheterodyne receiver are quite uniform throughout its tuning range.

Since IF amplifier works at a fixed I.F frequency, the design of this system is quite easy to provide high gain and const. BW.

After the I.F amplifier, the signal is arrived at the i/p of the demodulator which extracts the original modulating signal. This audio signal is amplified by an audio amplifier to get a particular voltage level. This amplified audio signal is further amplified by a power amplifier to get a specified power level so that it may activate the loud speaker.

The loud speaker is a transducer which converts this audio electrical signal into audio sound signal and thus the original signal is reproduced.

Advantage of superheterodyning :-

1) No variation in BW. BW remains const. over the entire operating range.

2) High sensitivity and selectivity

3) High adjacent channel rejection.

Note :- The automatic volume control (AVC) is provided by a feedback control loop which adjusts the gain of IF amplifier based on the power level of the signal at the envelope detector.

Image frequency :-

The superheterodyne receiver suffers from a major drawback known as Image frequency problem. This problem of Image frequency arises because of superheterodyne principle.

We know,

$$f_0 = f_s + f_i \quad \text{where}$$

$f_0 \to$ Local oscillator freq.

$f_s \to$ desired incoming freq.

$f_i \to$ IF freq.

$$\Rightarrow f_i = f_0 - f_s$$

But if a freq f_{si} manages to reach mixer such that

$$f_{si} = f_0 + f_i$$

then this freq f_{si} would also produce f_i $(-f_i)$ when it is mixed with f_0. This undesired or spurious IF signal will also amplified by I.F stage and thus would cause interference. This has a effect of 2 sources or stations being received simultaneously. This situation is obviously undesirable.

The rejection of an image freq. signal by a single tuned ckt. may be defined as the

(69) ratio of the gain at the signal frequency to the gain at the image freq. This is given as

$$\boxed{\alpha = \sqrt{1+Q^2\rho^2}}$$

where $\boxed{Q = \text{Quality factor}}$ of the tuned ckt.

$$\boxed{\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}}$$

$f_{si} \rightarrow$ Image freq.

$$f_{si} = f_o + f_i = (f_s + f_i) + f_i = f_s + 2f_i$$

$$\therefore \boxed{f_{si} = f_s + 2 \cdot IF}$$

FM Radio broadcasting:-

Commercial FM radio broadcasting utilizes the freq band 88-108 MHz for transmission of voice & music signals. The carrier frequencies are separated by 200 KHz & the peak-freq deviation is fixed at 75 KHz. Pre-emphasis is generally used, to improve the demodulator performance in the presence of noise in the received signal.

The receiver most commonly used in FM radio broadcast is a superheterodyne type. The block diagram of such receiver is shown in fig(1). The common tuning betn the RF amplifier and the local oscillator allows the mixer to bring all FM radio signals to a common IF BW of 200 KHz, centered at $f_{IF} = 10.7$ MHz. Since the message signal m(t) is embedded in the frequency of the

(70) Carrier, any amplitude variations in the received signals are a result of additive noise and interference.

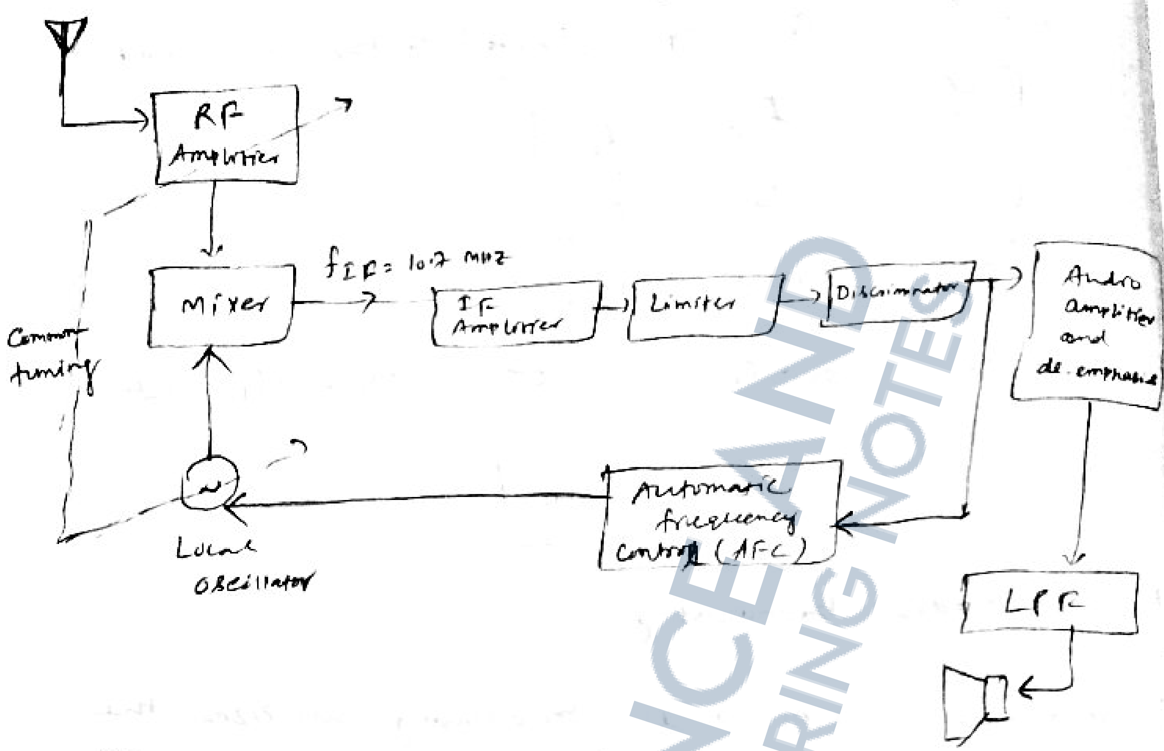


Fig 1: Block diagram of superheterodyne FM radio receiver.

The amplitude limiter removes any amplitude variations in the received signal at the o/p of the IF amplifier by band-limiting the signal.

A bandpass filter centered at $f_{IF} = 10.7$MHz with BW of 200 KHz is included in the limiter to remove higher order frequency components introduced.

A balanced frequency discriminator is used for frequency demodulation. The resulting message signal is then passed to the audio freq amplifier, which performs the functions of de-emphasis and amplification. The o/p of audio freq. amplifier is further filter by LPF to remove out-of-band noise & its o/p is used to drive loudspeaker.

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Signals

(21) Television Broadcasting:- (Black & white):-

The first step in T.V signal transmission is to convert a visual image into an electrical signal. The two-dimensional image or picture is converted to one-dimensional electrical signal by sequentially scanning the image and producing an electrical signal that is proportional to the brightness level of the image. The scanning is performed in a T.V camera, which optically focuses the image on a photo cathode tube that consist of photosensitive surface.

The scanning of the image is performed by an electron beam that produces an o/p current or voltage which is proportional to the brightness of the image. The resulting electrical signal is called a video signal.

The scanning of electron beam is controlled by 2 voltages applied across the horizontal & vertical deflection plates. These two voltages are shown in fig (a) & (b).

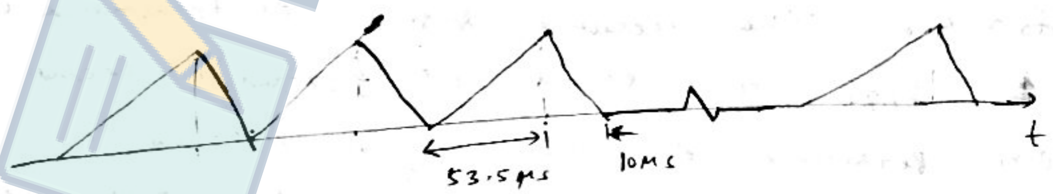


fig:-a

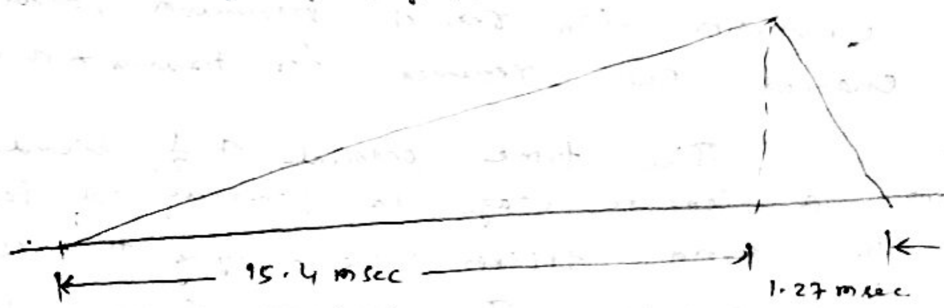
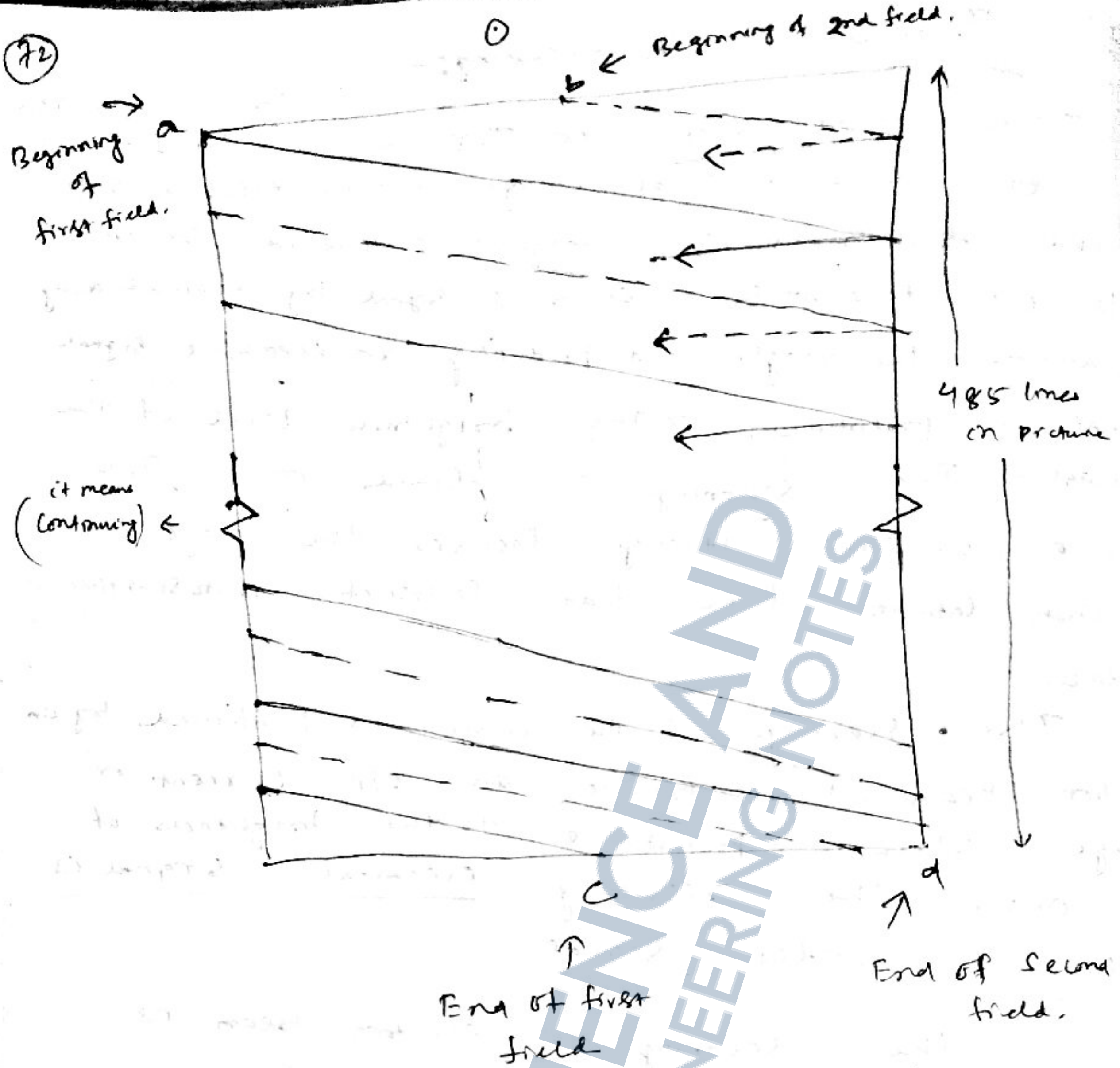


fig (b)

Signals waveforms applied to horizontal (a) & vertical (b) deflection plates.



(fig:2:- Interlaced scanning pattern)

In this scanning method, the image is divided into 525 lines that defines a frame. The resulting signal is transmitted in 1/30 of a second (i.e 30 frames per second, 1/30 of a second). our persistence of vision $= \frac{1}{16}$ of a second

The number of lines determine the picture resolution and in combination with rate of transmission, determine the channel BW required for transmission of image.

The time interval of $\frac{1}{30}$ second to transmit a complete image is generally not fast enough to avoid flicker (i.e annoying to eyes of the avg. viewer). To overcome flicker, the

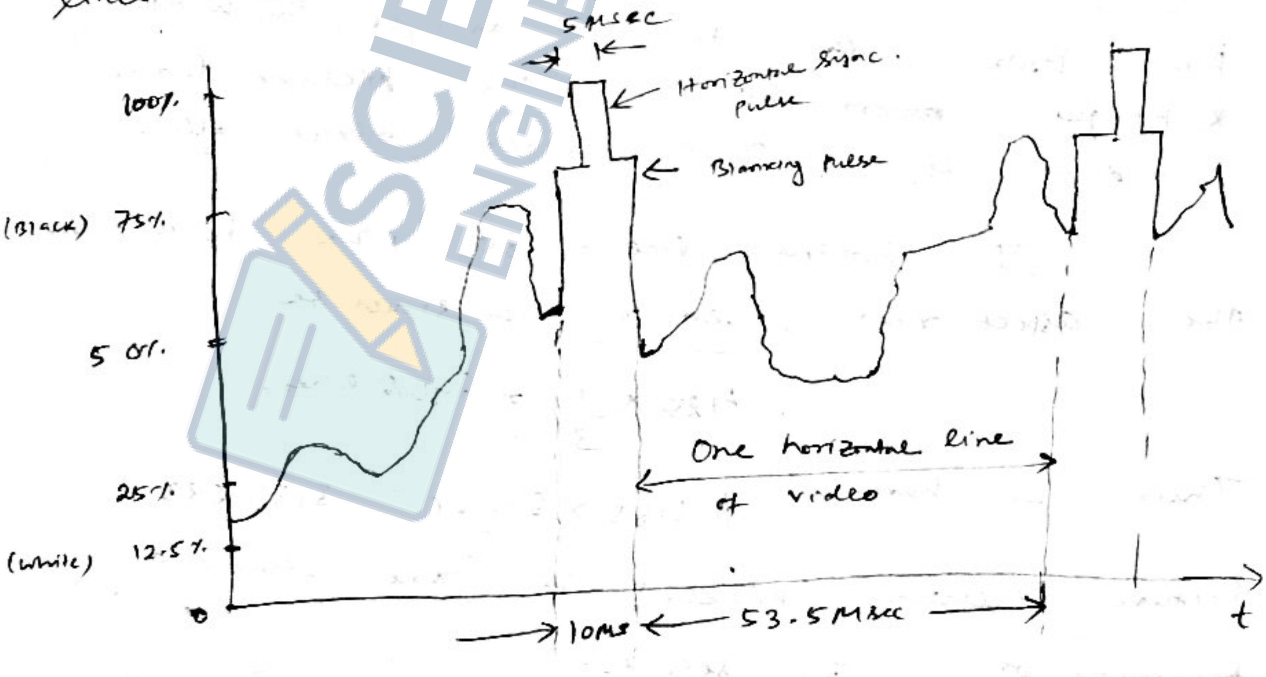
1

(73) Scanning of the image is performed in a interlaced pattern shown in fig (2).

The interlaced pattern consists of 2 fields, each consisisting of 262.5 lines. Each field is transmitted in $\frac{1}{60}$ of the seconds, which exceeds the flicker rate that is observed by the avg. eye.

The first field begins at point 'a' and terminates at point 'c'. The second field begins at point 'b' and terminates at point 'd'.

A horizontal line is scanned in 53.5 μs as indicated by the sawtooth signal waveform applied to the horizontal deflection plates. The beam has 10 μsec to move to the next line. During this interval a blanking pulse is inserted to avoid the appearance of retrace lines across the T.V receiver.



(Typical video signal for one horizontal sweep)

A 5 μsec pulse is added to the blanking pulse to provide synchronization for the horizontal sweep ckt at the receiver.

After the transmission of one interlaced field, the vertical sawtooth signal waveform applied to the vertical deflection plates is reset to zero. The retrace interval of 1.27 msec corresponding 20 lines scans, allows the beam to move from the bottom to the top of the picture.

A vertical blanking pulse is inserted during the interval to avoid the appearance of retrace line at the receiver. When we allow for vertical retrace (twice per frame) the actual number of horizontal lines in the image is 485.

To calculate BW of video signal:-

For proper view to human eyes, the width & height ~~matrix~~ ratio of the picture frame should be 4:3. It is called aspect ratio.

If horizontal lines = 485, then to maintain the aspect ratio, vertical lines should be

$$485 \times \frac{4}{3} = 646.66 \text{ lines.}$$

Thus we have $(485 \times 646.66 \simeq 313,633)$ picture elements (pixels) per frame, which are transmitted in $\frac{1}{30}$ second.

This is equivalent to sampling rate of 10.5 MHZ, which is sufficient to represent a signal as large as 5.25 MHZ.

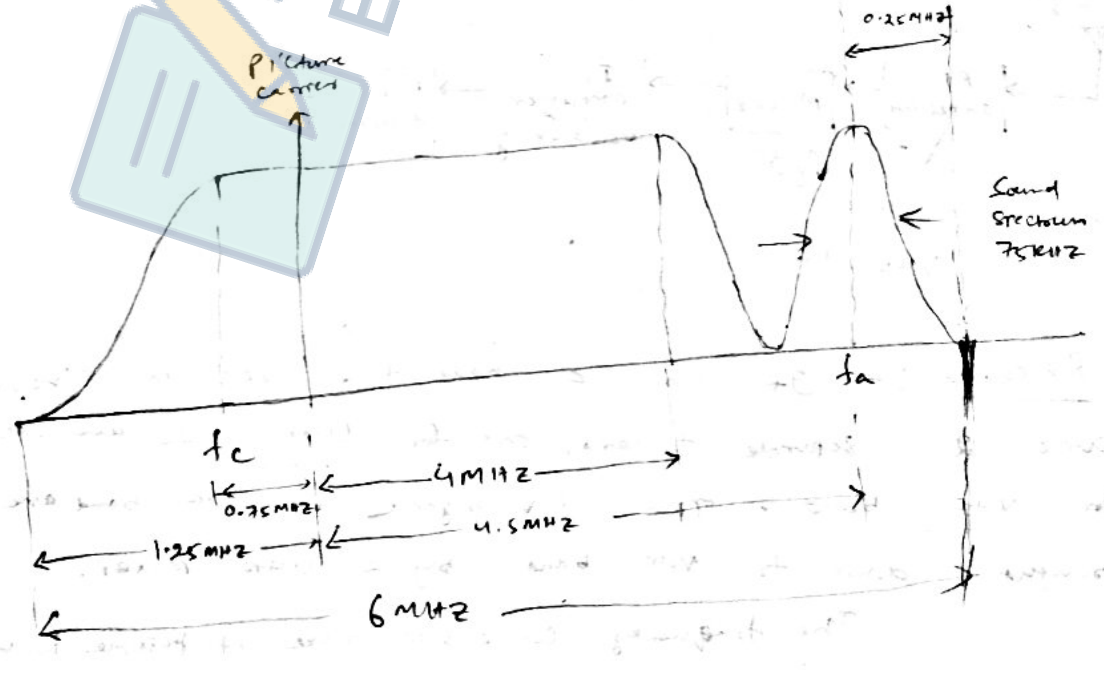
However the light intensity of adjacent pixels in an image is highly correlated. Hence, the BW of video signal is less than 5.25 MHz. In commercial T.V broadcasting BW of video signal is limited to $W = 4.2$ MHz.

Generally VSB modulation is used for T.V picture transmission.

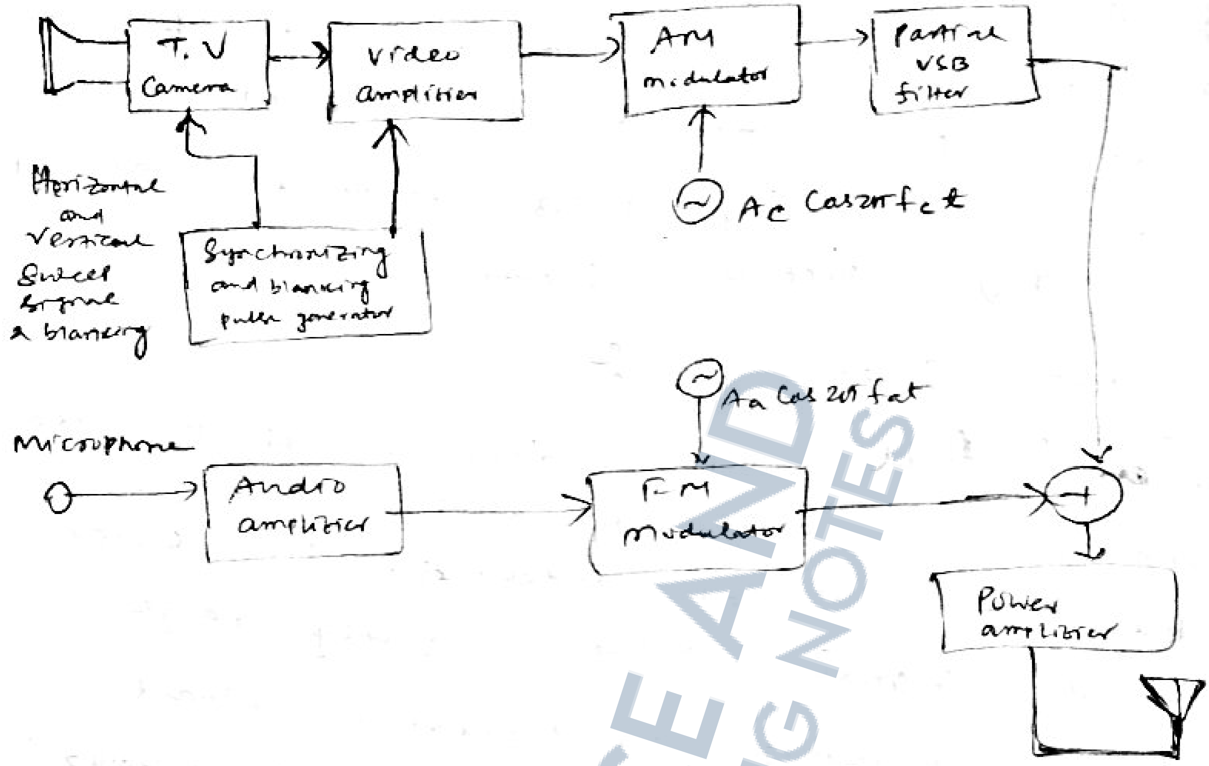
Complete Channel BW :-

In T.V transmission [redacted], full upper sideband of the video signal is transmitted along with a portion (1.25 MHz) of lower sideband. The lower sideband signal in the frequency range f_c and $f_c - 0.75$ MHz is transmitted without attenuation. The frequencies range $f_c - 1.25$ MHz to $f_c - 0.75$ MHz are attenuated.

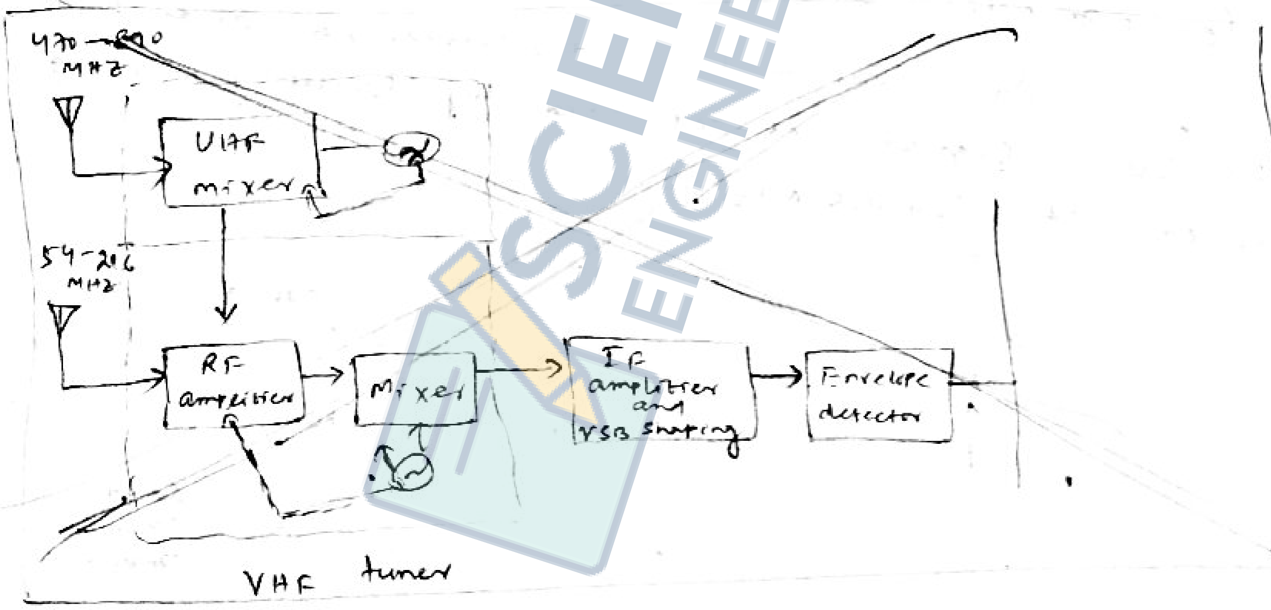
In addition to video signal, a audio portion of the T.V signal is transmitted by frequency modulating a carrier at $f_c + 4.5$ MHz.



Black & white T.V Transmitter & Receiver :-



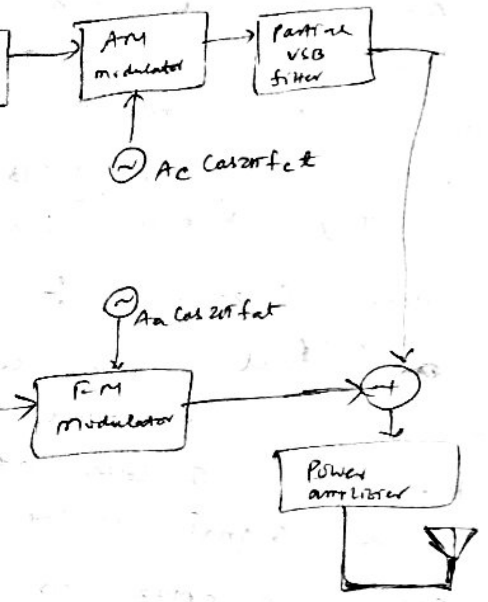
(Block diagram of B&W T.V transmitter)



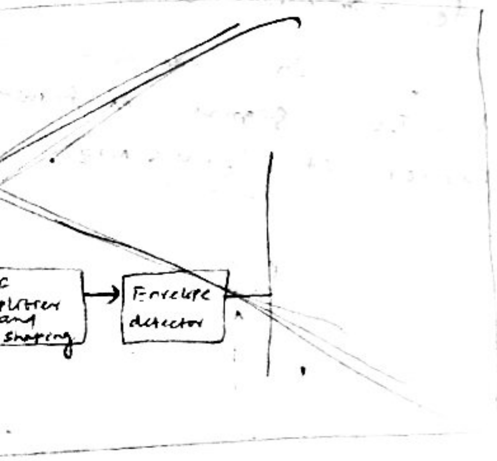
Receiver :- It is a heterodyne receiver. There are 2 separate tuners, one for UHF band and one for VHF band. The T.V signal in UHF band are brought down to VHF band by a UHF mixer.

The frequency conversion make it possible to use

Transmitter & Receiver :-



B&W T.V transmitter)



a heterodyne receiver. There
, one for UHF band and one
T.V signal in UHF band are
and by a UHF mixer.
Conversion make it possible to use

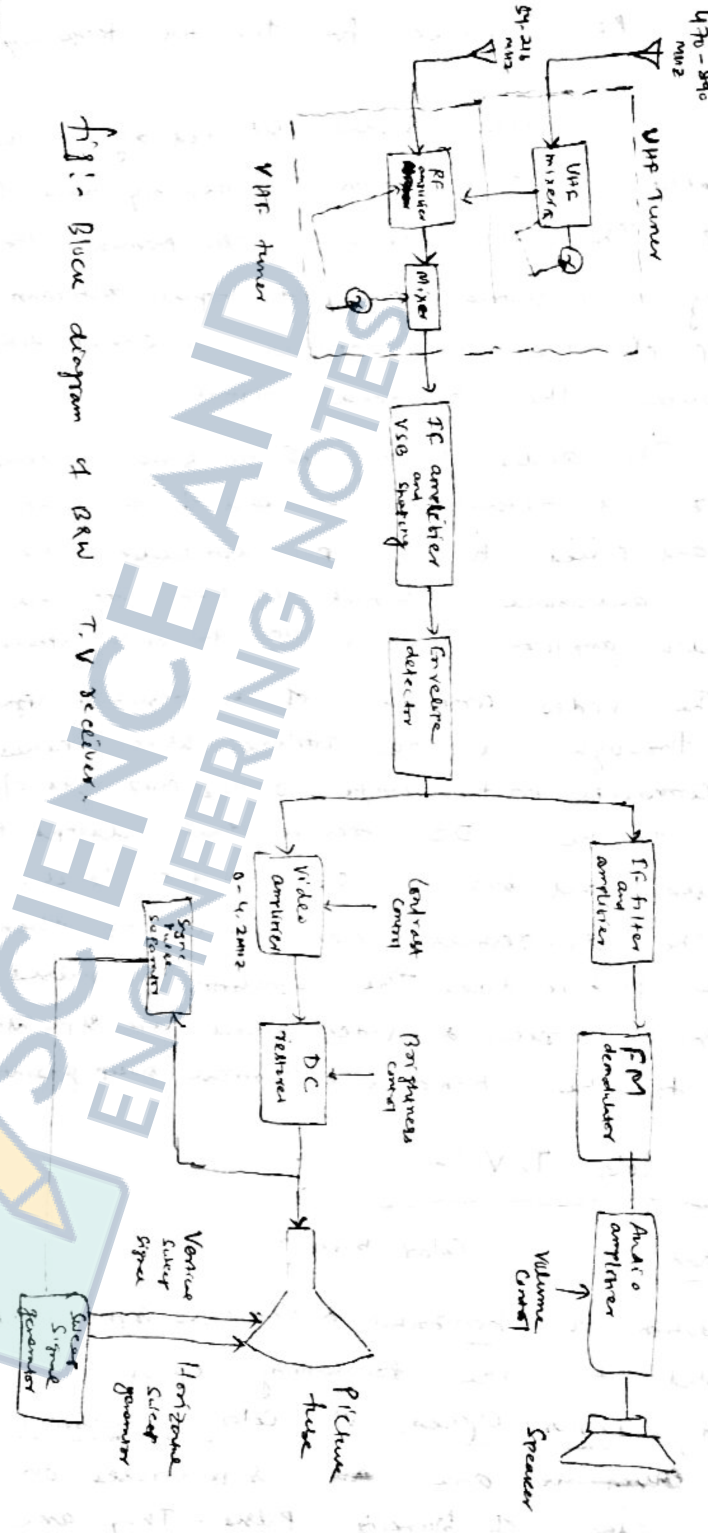


fig:- Block diagram of B&W T.V receiver.

a common RF amplifier for the two frequency bands.

Then the video signal selected by the tuner is translated to a common IF frequency band of 41-47 MHz. The IF amplifier also provides the VSB shaping required prior to signal detection. The o/p of the IF amplifier is envelope detected to produce the baseband signal.

The audio portion of the signal centered at 4.5 MHz is filtered out by means of an IF filter amplifier and passed to the FM demodulator. The demodulated audioband signal is then amplified by an audio amplifier and its o/p drives the speaker.

The video component of the baseband signal is passed through a video amplifier which passes frequency components in the range 0-4.2 MHz. Its o/p is passed to the DC restore that clamps the blanking pulses and sets the correct d.c level.

The dc-restored video signal is then fed to the picture tube. The synchronizing pulses contained in the received video signal are separated and applied to the horizontal & vertical sweep generators.

Compatible Color T.V :-

2 mark 1) What is Color burst?

In addition to horizontal & vertical synchronization pulses added to the transmitting signal at the transmitter, eight cycles of color subcarrier $A_c \cos 2\pi f_{sc} t$, ~~[illegible]~~ are ~~[illegible]~~ superimposed on the trailing edge of blanking pulses. They are are called 'Color burst'. The purpose is to provide frequency

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(7) a signal for subcarrier phase synchronization at the receiver.

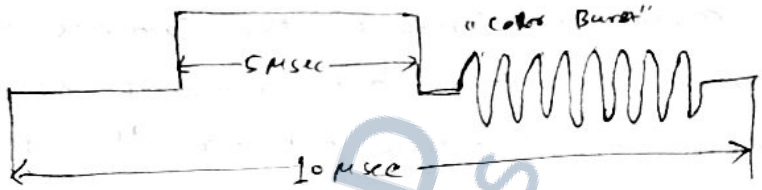

fig:- Blanking pulse with Color subcarrier.