Analog Communication (Modulation)

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Modulation →

- The process of changing one characterestics (phase, frequency or amplitude) of carrier signal in accordance with the instantaneous value of message signal and making the signal suitable for transmission is called analog modulation.
- Any wave has three significant characterestics viz amplitude, frequency and phase, and modulation is a process of impressing information to be transmitted on a high frequency wave, called the carrier wave, by changing it's one of the characteristics.
- Carrier wave is a high frequency, constant amplitude, constant frequency and non-interrupted wave generated by radio frequency oscillators. These waves are inaudible ~~by~~ i.e. by themselves they are not able to produce any sound in the speaker.

Needs of Modulation →

Basic aspects →

A question may be asked as, when the baseband signals can be transmitted directly why to use the modulation?

The answer is that the baseband transmission has many limitations which can be overcome using modulation. It may be explained below.

In the process of modulation, the baseband signal is translated i.e. shifted from low frequency to high frequency. This frequency shift is proportional to the frequency of carrier.

Hence, below are some points, to needs of modulation →

(Need of modulation/ advantages of modulation)

(1) Reduction in Height of Antenna →

For the transmission of radio signals, the antenna height must be multiple of ($\lambda/4$). Here λ is the wave length, $\lambda = c/f$ where c is the velocity of light

f is the frequency of signal to be transmitted.

The minimum antenna height required to transmit a baseband signal of $f = 10\,kHz$ is calculated as:

$$\text{Minimum height of antenna} \rightarrow \frac{\lambda}{4} = \frac{c}{4f} = \frac{3*10^8}{4*10*10^3}$$

$$= 7500 \text{ meters}$$

$$= 7.5\,Km$$

The antenna of this height is practically impossible to install. Now, let us consider a modulated signal at $f = 1\,MHz$. The minimum antenna height is given as,

$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3*10^8}{4*10^6*1} = 75m.$$

This antenna can be easily installed ~~pro~~ practically. Thus, modulation reduces the height of the antenna.

(2) Increase the Range of communication →

The frequency of baseband signal is low, and the low frequency signals can not travel a long distance when they are transmitted. They get heavily attenuated (suppressed). The attenuation reduces with increase in frequency of the transmitted signals, and they travel longer distance.

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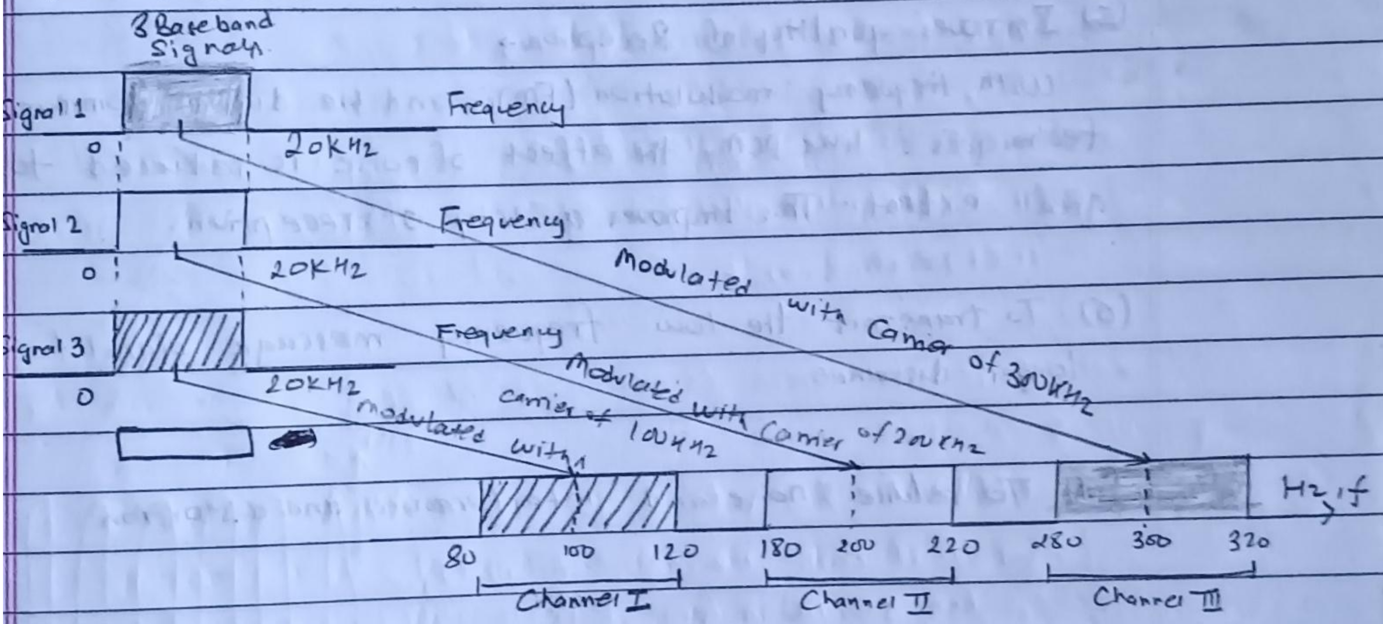
The modulation process increases the frequency of the signal to be transmitted. Therefore, it increases the range of communication

(3) Avoids Mixing of Signals →

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 KHz. Therefore, all the signals get mixed together and a receiver cannot separated them from each other.



Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals.

(4) Multiplexing is possible →

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously. This is possible only with modulation. The multiplexing allows the same channel to be used by many signals. Hence, many TV channels can use the same frequency range without getting mixed with each other, or different frequency signals can be transmitted at the same time.

(5) Improves quality of Reception →

With frequency modulation (FM), and the digital communication techniques like PCM, the effect of noise is reduced to a great extent. This improves quality of reception.

(6) To transmit the low frequency message signal to the longer distance.

(7) To reduce noise and interference and distortion.

Classification of Modulation →

- Modulation
  - Analog
    - Amplitude (AM) Modulation
      - DSB-FC
      - DSB-SC
      - SSB-SC
      - ISB
      - VSB
      - QAM
    - Angle Modulation
      - PM
      - FM
    - Pulse Modulation
      - PAM
      - PWM
      - PPM
  - Digital
    - ASK
      - 1. BASK
      - 2. QASK / QAPSK / QAM
    - Angle
      - PSK
        - 1. BPSK
        - 2. DPSK
          - (a) DEPSK
        - 3. QPSK
          - (a) OQPSK - $\pi/2$
          - (b) non-offset QPSK - π
          - (c) $\pi/4$ - QPSK
      - FSK
        - 1. BFSK
        - 2. CPFSK (MSK)
          - (a) GMSK
    - Pulse Modulation
      - PCM
        - RZ
        - NRZ
      - DM

Amplitude Modulation →

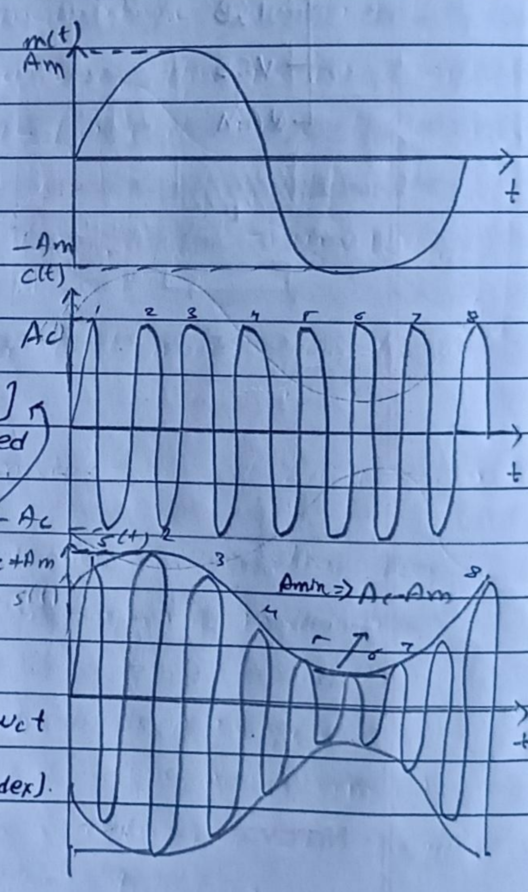
- The modulation in which the amplitude of carrier wave is varied in accordance with the instantaneous value of the message signal is known as amplitude modulation. The frequency and phase remain constant.
- The system in which the maximum amplitude of the carrier wave is made proportional to the instantaneous value (amplitude) of the modulating or base band signal.

(1) DSB-FC also called AM.

→ Time domain Representation,

Let us consider a modulating signal $m(t)$ of sinusoidal wave is given as,

$$m(t) = A_m \sin\omega_m t$$



Let a carrier wave, (sinusoidal) $c(t)$ is given as,

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

[Amplitude of carrier is varied as per $m(t)$]

Hence, the standard form of modulated signal $s(t)$ is given as,

$$s(t) = A' \sin\omega_c t$$
$$= [A_c + m(t)]\sin\omega_c t$$
$$= [A_c + A_m \sin\omega_m t]\sin\omega_c t$$
$$= A_c\left[1 + \frac{A_m}{A_c}\sin\omega_m t\right]\sin\omega_c t$$

Here, $\frac{A_m}{A_c} = \mu$ (modulation Index).

$K_a = 1/A_c$

K_a ⇒ amplitude sensitivity.

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→ Frequency domain Representation,

The amplitude modulation signal is given as,

$$s(t) = A_c\left[1 + \frac{A_m}{A_c}\sin\omega_m t\right]\sin\omega_c t$$

$$= A_c[1 + \mu\sin\omega_m t]\sin\omega_c t$$

$$= A_c\sin\omega_c t + \mu A_c\sin\omega_m t\,\sin\omega_c t$$

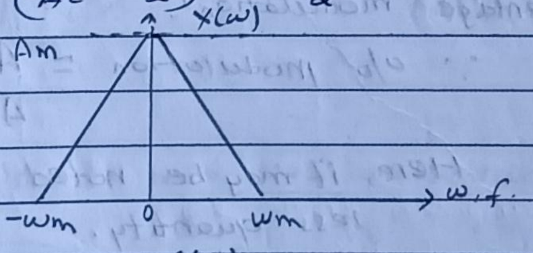
$$= A_c\sin\omega_c t + \frac{A_c\mu}{2}\cos(\omega_c - \omega_m)t - \frac{A_c\mu}{2}\cos(\omega_c + \omega_m)t$$

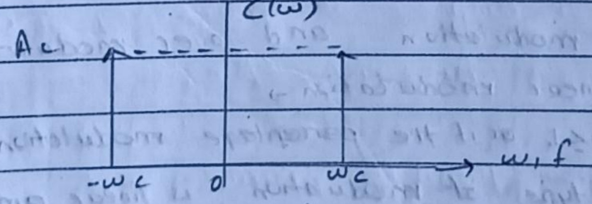
$$= \underbrace{A_c\sin\omega_c t}_{} + \underbrace{\frac{A_c\mu}{2}\cos(\omega_c - \omega_m)t}_{} - \underbrace{\frac{A_c\mu}{2}\cos(\omega_c + \omega_m)t}_{}$$

→ this signal is having three frequency component, i.e. ω_c, $\omega_c + \omega_m$ and $\omega_c - \omega_m$, here, $\omega_c + \omega_m$ & $\omega_c - \omega_m$ are side bands.

→ sideband amplitude

$$\frac{\mu A_c}{2} = \left(\frac{A_m}{A_c} \times \frac{A_c}{2}\right) = \frac{A_m}{2}$$

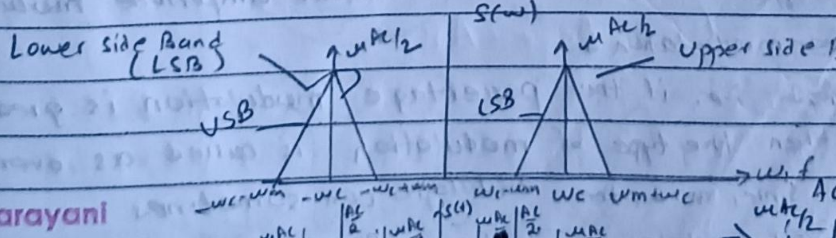


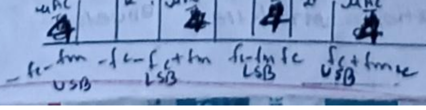


Bandwidth (BW)

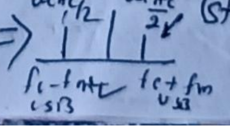
$\Rightarrow f_c + f_m - f_c + f_m$

$\Rightarrow 2f_m$.





$\Rightarrow$



waveform (Standard i.e eq^n)

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Modulation Index and Modulating Factor of AM wave →

In AM wave, the modulation index (m) is defined as the ratio of amplitudes of the modulating and carrier waves as under:

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

When, $A_m \leq A_c$, the modulation index 'm' has values 0 & 1 and no distortion is introduced in the AM waves. But, if $A_m < A_c$, then m is greater than 1. This will distort the shape of AM signal. The distortion is called as 'over modulation'. The modulation index is also called as modulation factor, modulation coefficient or degree of modulation. However, this modulation index is expressed as percentage and it is called as 'percentage modulation'.

$$\therefore \text{ \% modulation} = \frac{A_m}{A_c} * 100. = m * 100\%.$$

Here, it may be noted that m is a dimensionless quantity.

⇒ Linear modulation and over modulation of AM wave →

(1) Linear modulation →

If $m \leq 1$ or if the percentage modulation is less than 100, then the type of modulation is linear amplitude modulation.

(2) over modulation →

If $m > 1$, i.e. if the percentage modulation is greater than 100%, then the type of modulation is called as overmodulation. Since, $m > 1$, the envelope can sometimes reverse the phase. Over modulation introduces envelope distortion.

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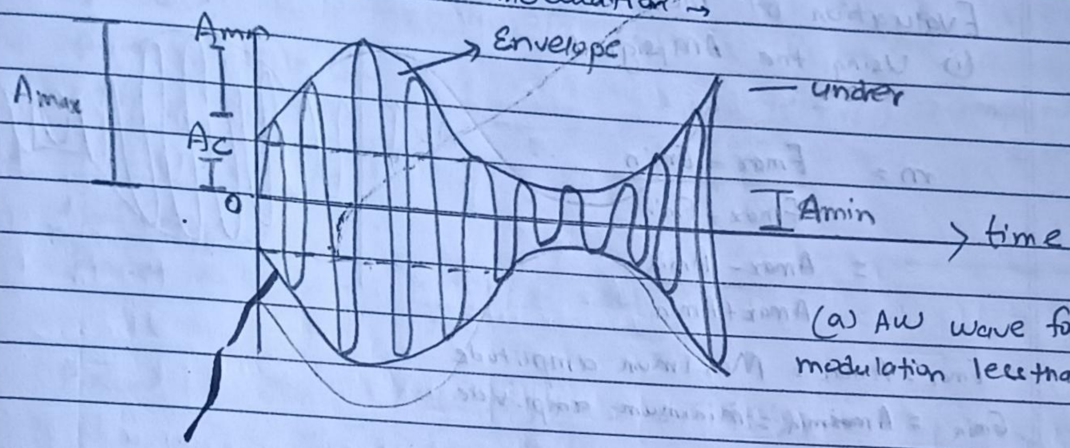
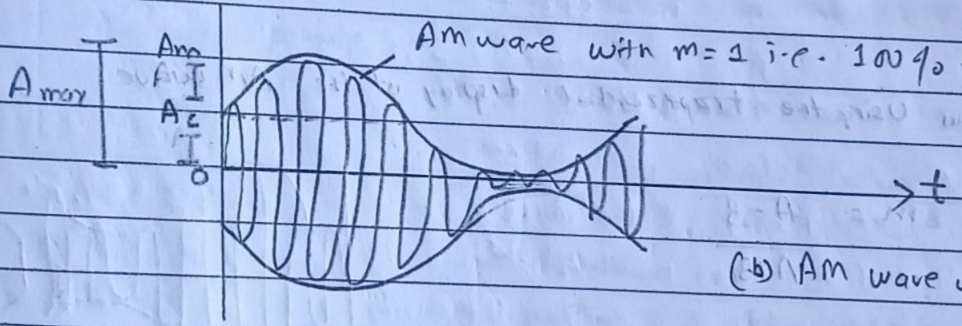
wave form of Linear & over modulation →



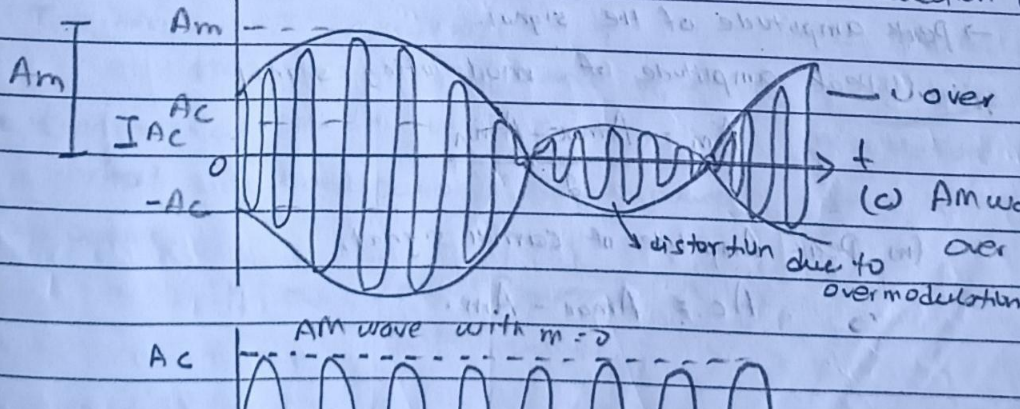
(a) AW wave for %age modulation less than 100%



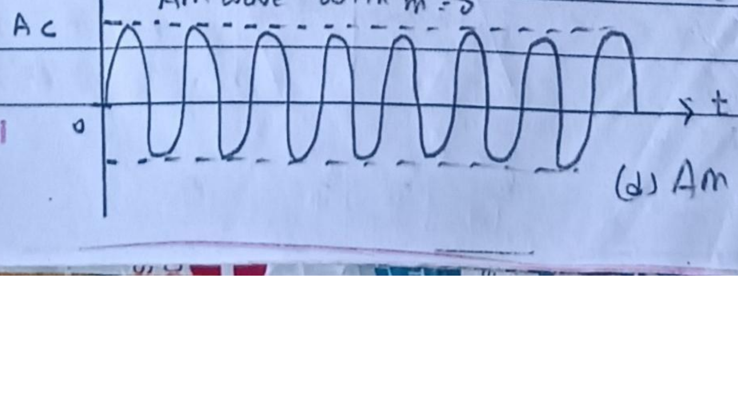
Am wave with m=1 i.e. 100% modulation Ac=Am

(b) AM wave with 100% modulation

Am wave with m>1 i.e. overmodulection (Am>Ac).



(c) Am wave with over modulation



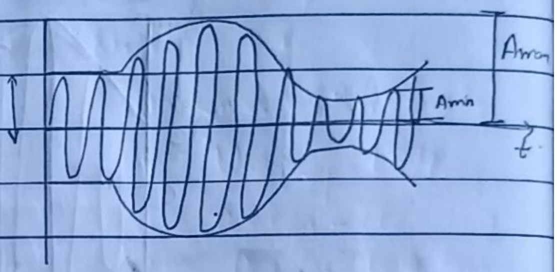
(d) AM wave with m=0.

Evaluation of modulation index.

(i) Using the AM signal.

$$m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$$

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

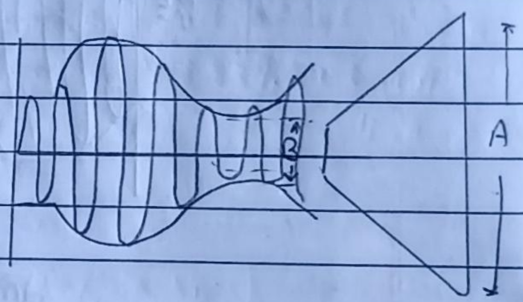


$E_{max} = A_{max}$ = Maximum amplitude

$E_{min} = A_{min}$ = minimum amplitude

(ii) Using the trapezoidal display of the AM wave.

$$m = \frac{A - B}{A + B}$$



→ Peak amplitude of the signal.

(i) Peak amplitude of modulating signal

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

(ii) Peak Amplitude of carrier signal,

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

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Power of AM:-

we have, $P = V_{rms}^2/R = (V_m/\sqrt{2})^2/R = V_m^2/2R$

power (P) = $V^2/2R$

$R = 1$ & $V = A_c$.

$(\mu A_c/2) \Rightarrow$ amplitude of side band.

$\boxed{(P_c) \Rightarrow A_c^2/2}$

Similarly,

Power in USB = $\left(\frac{\mu A_c}{2}\right)^2/2 = \mu^2 A_c^2/8$

Power in LSB = $\left(\frac{\mu A_c}{2}\right)^2/2 = \mu^2 A_c^2/8$

The total power of AM is given as,

Total power $(P_T) = P_c + P_{USB} + P_{LSB}$

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

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

$$P_T = \frac{A_c^2}{2}\left[1 + \frac{\mu^2}{2}\right] \Rightarrow P_c\left(1 + \frac{\mu^2}{2}\right).$$

Transmission efficiency (η):

The transmission efficiency of a AM wave is the ratio of transmitted power which contains the information (i.e. the total side band power) (useful power) to the total transmitted power.

$$\eta = \frac{\text{Useful Power}}{\text{Total power}}$$

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$= \dfrac{P_{USB} + P_{LSB}}{P_T}$

$= \dfrac{\frac{\mu^2 A_c^2}{8} + \frac{\mu^2 A_c^2}{8}}{\frac{A_c^2}{2}\left(1+\frac{\mu^2}{2}\right)}$

$= \dfrac{\mu^2}{2+\mu^2} \quad 0 \le \mu \le 1$

Note,

Transmission Bandwidth,

$BW = USB_{freq} - LSB_{frequency}$

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

$= f_c + f_m - f_c + f_m$

$= 2f_m.$

When, $\mu = 1$.

$\eta = 1/3 = 33.33\%$. $\{\mu \rightarrow maximum\ i.e.\ \mu = 1\}$.

$\therefore \eta_{max} = \dfrac{1}{2+1} \times 100\% = 33.33\%$

the maximum transmission efficieny is 33.33%, this implies 1/3rd of the total power is carried by side band and two third is wasted which is carried by carrier.

Redudancy : $\Rightarrow 1 - \eta$

% of power saving $\Rightarrow 1 - \dfrac{\mu^2}{2+\mu^2}$

$\Rightarrow \dfrac{2}{2+\mu^2}$

Single tone amplitude modulation and ~~double~~ multible tone amplitude modulation.

Single tone → one freq components

multiple-tone → more than one frequency components

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Single Tone AM and Multiple Tone - AM →

In this section, we shall discuss amplitude modulation in which the modulating or baseband signal consists of only one (single) frequency i.e. modulation is done by a signal frequency or tone. This type of amplitude modulation is known as single tone amplitude modulation. (DSB - SC or AM)

A multiple - tone amplitude modulation is that type of modulation in which the modulating signal consists of more than one frequency components.

Let we have carrier signal →

$$c(t) = A_c \sin\omega_c t.$$

and multiple - tone modulating signal →

$$m(t) = A_{m_1} \sin\omega_1 t + A_2 \sin\omega_2 t + A_3 \sin\omega_3 t.$$

So, AM signal will be →

$$s(t) = A' \sin\omega_c t$$

$$= (A_c + A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + A_3 \sin\omega_3 t) \sin\omega_c t$$

$$= \left(A_c + \frac{A_1}{A_c} \sin\omega_1 t + \frac{A_2}{A_c} \sin\omega_2 t + \frac{A_3}{A_c} \sin\omega_3 t\right) \sin\omega_c t$$

$$\mu_1 = \frac{A_1}{A_c}, \quad \mu_2 = \frac{A_2}{A_c}, \quad \mu_3 = \frac{A_3}{A_c}$$

$$\therefore s(t) = (A_c + \mu_1 \sin\omega_1 t + \mu_2 \sin\omega_2 t + \sin\omega_3 t) \sin\omega_c t$$

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Total Transmitted Power;

$P_t = P_c + P_{sideband}$

Carrier power $= \frac{A_c^2}{2}$

Sideband power is given as,

$$P_s = \frac{1}{4} A_c^2 \mu_1 + \frac{1}{4} A_c^2 \mu_2^2 + \frac{1}{4} A_c^2 \mu_3^2$$

$$= \frac{1}{2} A_c^2 \left(\frac{1}{2}\left[(\mu_1^2 + \mu_2^2 + \mu_3^2)\right]\right)$$

$$= \frac{P_c}{2} (\mu_1^2 + \mu_2^2 + \mu_3^2)$$

$$= \frac{P_c'}{2} \mu^2$$

$\because$ where, $\mu^2 = \mu_1^2 + \mu_2^2 + \mu_3^2$

Total transmitted power,

$$P_T = P_c + P_s$$

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

$$P_T = P_c\left(1 + \frac{1}{2}(\mu_1^2 + \mu_2^2 + \mu_3^2)\right)$$

$\therefore$ Hence, this is the required condition for multitone - AM.

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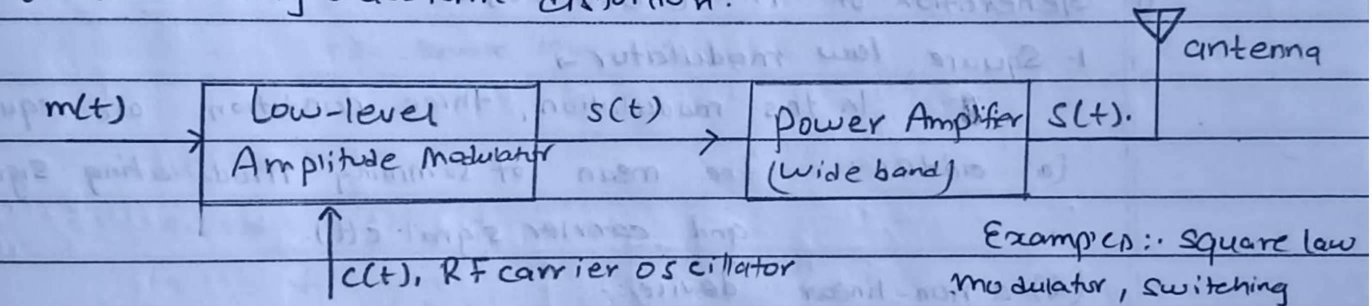
Generation of AM→

The generating circuits for AM wave are called as amplitude modulator circuits.

(1) Low-level modulation → (direct method)→

The generation of AM wave takes place at a low power level. The generated AM signal is then amplified using a chain of linear amplifiers. The linear amplifiers are required in order to avoid any waveform distortion.



Examples:- square law modulator, switching modulator.

Fig: Low-level modulation

(2) High-level modulation → (Indirect ~~mod~~ method)→

In this method, the generation of AM wave takes place at high power levels. The carrier and the modulating signal both are amplified first to an adequate power level and the modulation takes place in the last RF amplifier stage of the transmitter. Highly efficient class C amplifiers are used in high level modulation. Therefore, the efficiency of high levels modulation is higher than that of low level modulation.

Example:- Collector modulator.

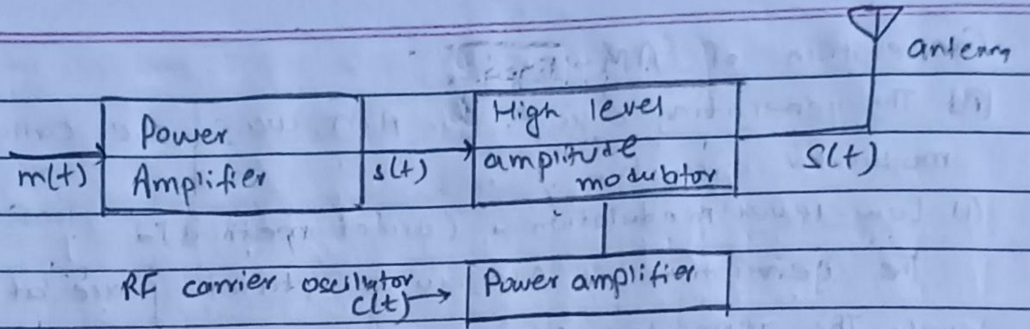


Fig: High level AM.

Generation of AM (DSB-FC) →

1. Square law modulator →

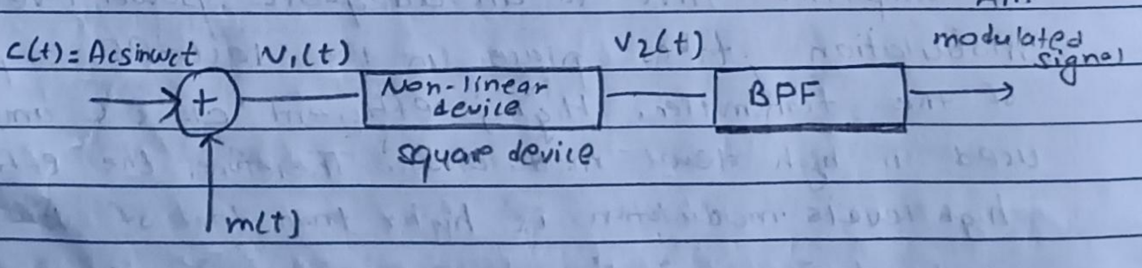
In this modulation, three condition are required →

(a) adder :- the mean of summing modulating signal m(t) and carrier signal c(t).

(b) a non-linear device :-

(c) Band pass filter tuned at f_c.

i.e. semi conductor diode and transistor are most common non-linear device. for filtering the single-double tuned bandpass filter is used.



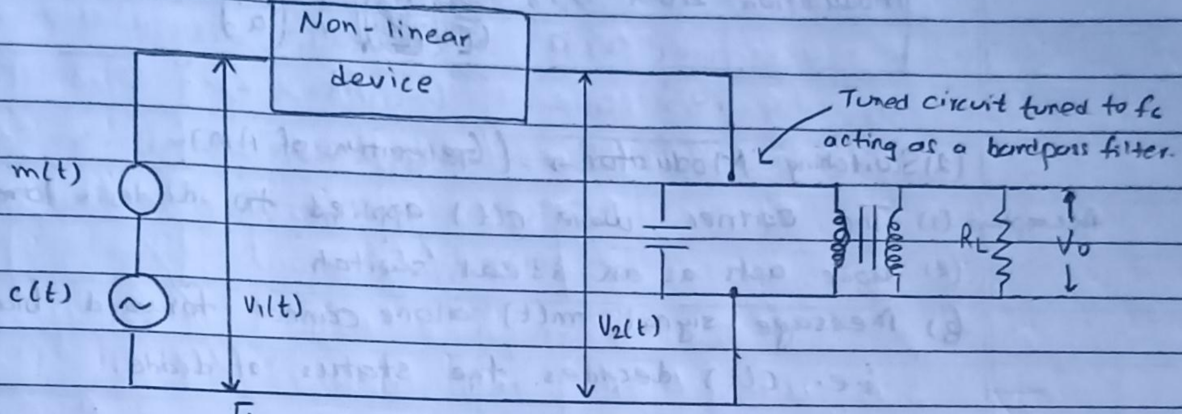


Fig: square law modulator

(1) After adder,

$$V_1(t) = m(t) + c(t)$$

$$\boxed{V_1(t) = A_c \sin w_c t + m(t)}$$

(2) After passing non-linear device,

$$V_2(t) = aV_1(t) + bV_1^2(t).$$

$$= a(A_c \sin w_c t + m(t)) + b(A_c \sin w_c t + m(t))^2$$

$$= aA_c \sin w_c t + am(t) + bA_c^2 \sin^2 w_c t + bm^2(t) + 2bA_c \sin w_c t \cdot m(t).$$

$$= \underbrace{\underset{(I)}{am(t)} + \underset{(II)}{bA_c^2 \sin^2 w_c t} + \underset{(III)}{bm^2(t)}}_{\text{unwanted signal}} + \underbrace{\underset{(IV)}{aA_c \sin w_c t} + \underset{(V)}{2bA_c m(t) \sin w_c t}}_{\text{required signal}}$$

(3) After BPF,

$$s(t) = aA_c \sin w_c t + 2bA_c m(t) \sin w_c t.$$

$$\boxed{s(t) = aA_c\left(1 + \frac{2b}{a} m(t)\right) \sin w_c t.}$$

$$\left(\mu = \frac{2b}{a}\right)$$

$\boxed{\text{Modulation Index} \Rightarrow \frac{2b}{a}} \Rightarrow \left(\frac{2b}{a}\right)$

(2) Switching Modulator → (Generation of AM).

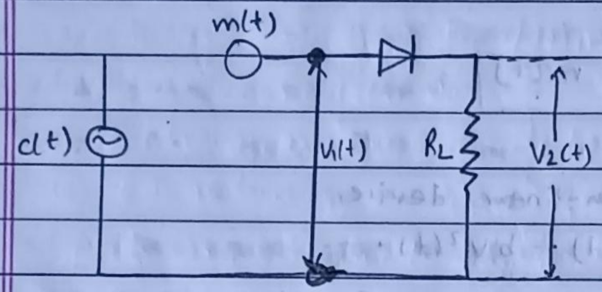
Assumption: (1) The carrier wave c(t) applied to diode is larger.

(2) diode acts as an ideal switch

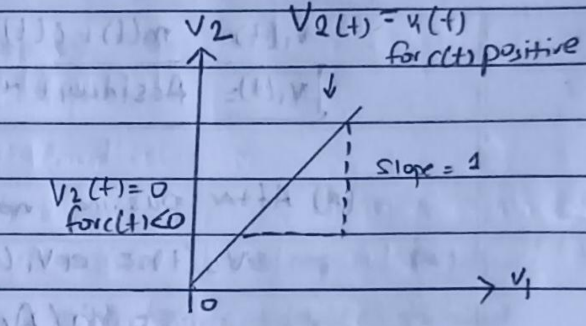
(3) Message signal m(t) alone cannot forward bias the diode.

i.e. c(t) decides the status of diode.

– the schematic diagram of switching modulator is shown in figure below →



(a) Switching modulator



(b) Idealized input-output relation

we have,

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

where, $m(t) << A_c$ then $V_1(t) = c(t)$.

and $V_2(t) = v_1(t)$ $c(t) > 0$ (+ve).

$= 0$ $c(t) < 0$ (−ve).

Hence, the voltage $V_2(t)$ varies periodically between $V_1(t)$ and zero at the rate equal to carrier frequency f_c. In other words, $V_2(t)$ is multiplication of $V_1(t)$ and unit length rectangular pulse triangle $g_p(t)$.

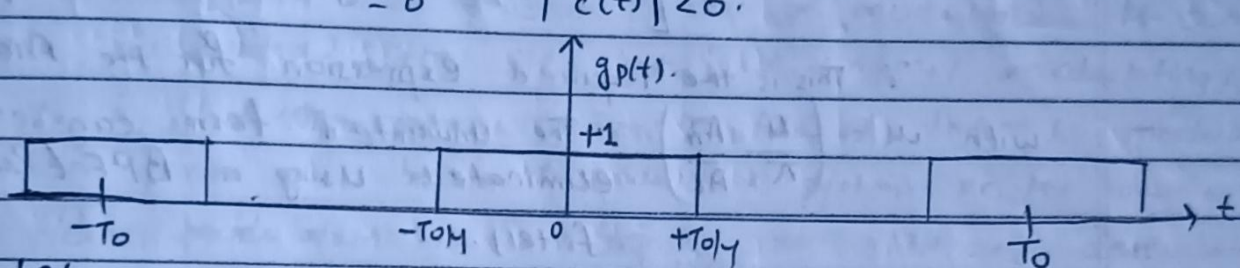
i.e. $V_2(t) = V_1(t) * 1$, $c(t) > 0$.

$= V_1(t) * 0$, $c(t) < 0$.

i.e. $V_2(t) = V_1(t) * g_p(t)$ —— ①

where, $g_p(t) = 1$ $|c(t)| > 0$.

$= 0$ $|c(t)| < 0$.



Let $g_p(t)$ is periodic signal, it is represented as is periodic pulse train of duty cycle equal to $T_0/2$.

Let us express $g_p(t)$ with the help of Fourier series;

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

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

Substituting $g_p(t)$ into eq^n ①, ② we get,

$$V_2(t) = [m(t) + A_c \cos 2\pi f_c t] * g_p(t)$$

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

③

The odd harmonic components are unwanted signal, and therefore, are assumed to be eliminated.

$$V_2(t) = \underbrace{\frac{1}{2} m(t)}_{\text{modulating signal}} + \underbrace{\frac{1}{2} A_c \cos(2\pi f_c t) + \frac{2}{\pi} m(t) \cos(2\pi f_c t)}_{\text{AM waves}} + \underbrace{\frac{2}{2} A_c \cos^2(2\pi f_c t)}_{\text{second harmonic of carrier}}$$

FC → Full carrier

SC → Suppressed carrier

In this expression, the first and and the forth terms are unwanted terms whereas the second and third terms together represent the AM wave. Clubing the second and third terms together, we obtain,

$$v_2(t) = \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c} m(t)\right] \cos(2\pi f_c t) + \text{Unwanted Terms}$$

∴ This is the required expression for the AM wave with $\mu = \left(\frac{4}{\pi}\frac{A_m}{A_c}\right)$. The unwanted terms can be eliminated using a BPF (Band-pass-filter).

Advantages of DSB-FC / (AM) →

(1) AM transmitters are less complex.

(2) AM receivers are simple, detection is easy.

(3) AM receivers are cost efficient. Hence, even a common person can afford to buy it.

(4) AM waves can travel a longer distance.

(5) Low bandwidth.

disadvantages :-

(1) Less efficient than other amplitude modulation.

(2) there is wastage of precious bandwidth since bandwidth is twice the frequency of message signal.

(3) Westage of transmission power because 2/3rd of total power is carried by carrier signal.

(4) Effected due to noise. (5) Envelope is distorted due to noise.

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applications

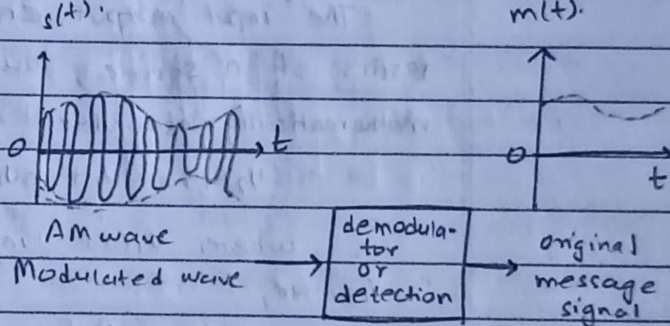
(1) Radio broad casting

(2) Picture transmission in a TV system.

Detection (demodulation) of AM waves →

- The process of detection or demodulation is the process of recovering the message signal from the received modulated signal. This means that the process of detection is exactly opposite to that of modulation.
- Demodulation is the reverse process of modulation. It converts the modulated signal s(t) into the original modulating signal m(t). In every communication system, if there is modulation then their must be a demodulator system on the other end.
- Hence, modulation is for transmission side and demodulation is used for receiver side.

Types of AM detectors.
(i) Square law detector.
(ii) Envelope detector.



(i) Square law detector →

The schematic diagram of Square law detector is shown in figure below →

- DSP-FC / AM / conventional AM wave is applied to the non-linear device output of non-linear device is given to the low pass filter. and output of LPF is required message signal. In square law modulation bandpass filter is used where as demodulation low pass filter is used.

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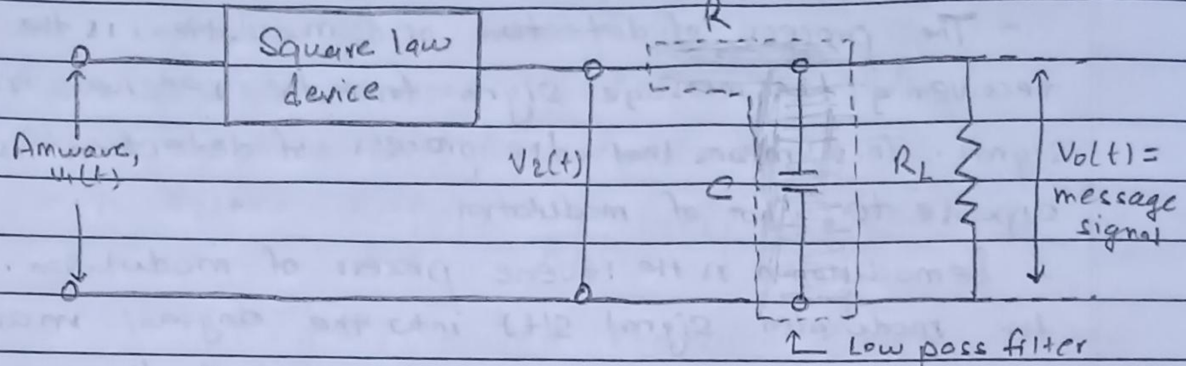

Figure: a square law detector

Working operation →

The input output characterestics i.e. the transfer characterestics of a square law device is non-linear and it is expressed mathematically as under:

$$v_2(t) = a v_1 + b v_1^2(t) \quad -(1)$$

where $v_1(t)$ = input voltage to the detector = Am wave.

Therefore,

$$v_1(t) = A_c[1 + m\, m(t)] \cos(2\pi f_c t) \quad -(2)$$

From eq^n (1) & (2) we get,

$$v_2(t) = a A_c[1 + m.m(t)] \cos 2\pi f_c t + b A_c^2 [1 + m\, m(t)]^2 \cos^2(2\pi f_c t) \quad -(3)$$

But $\cos^2\theta = \frac{1}{2}(1 + \cos 2\theta)$.

Therefore, $\cos^2(2\pi f_c t) = \frac{1}{2}[1 + \cos(4\pi f_c t)]$.

Substituting this we get,

$$v_2(t) = a A_c[1 + m\, m(t)] \cos(2\pi f_c t) + \frac{b A_c^2}{2}[1 + 2m\, m(t) + m^2 m^2(t)][1 + \cos(4\pi f_c t)] \quad -(4)$$

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out of these terms, the only desired term is $bA_c^2\, m\, m(t)$ which is due to the bv_1^2 term. Hence, the name of this detector is square law detector.

This desired term is extracted by using a low pass filter (LPF) after the diode as shown figure above. Thus, after passing low pass filter we get,

$$\boxed{v_o(t) = (bA_c^2\, m)\, m(t)}$$

This means that we have recovered the message signal ~~red~~ $m(t)$ at the output of the detector.

distortion in the detector output,

Another term which passes through the low pass filter (LPF) to the load resistence R_L. is as under: $\frac{1}{2} bA_c^2 m^2 m^2(t)$.

This is an unwanted signal and gives rise to a signal distortion. The ratio of desired signal to the undesired one is given by,

$$\text{Ratio} = (\text{desired output}) / (\text{undesired output})$$

$$= \frac{bA_c^2\, m\, m(t)}{\frac{1}{2} \cdot bA_c^2\, m^2 m^2(t)} = \frac{2}{m \cdot m(t)} \quad - (8).$$

Hence, this the required expression of square law detector.

(II) Envelope detector →

– this method is simply and highly effective for demodulation of narrow band AM ($f_0 >> f$) for which percentage of modulation is less than 10%.

– It provides output signal that follows the envelope of input signal waveform, exactly hence name envelope detector.

– the schematic diagram of envelope detector is shown in figure below →

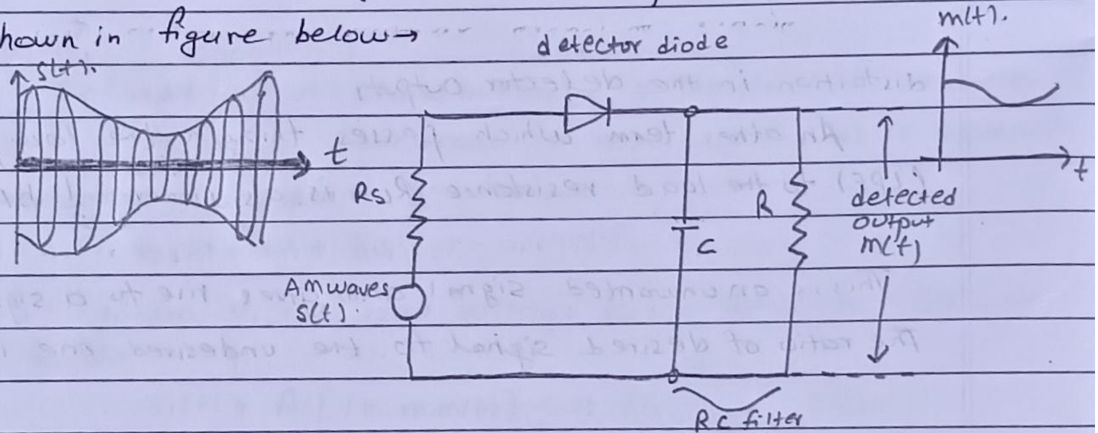

Fig → Envelope detector for Am wave

Assumption →

(1) diode ideal i.e. zero impedence when forward biased & Infinite impedence when reverse biased.

(2) charging time constant R_sC is much short compared to carrier period. (low) $R_sC << 1/RC$

(3) discharging time constant R_LC is larger.

$$\frac{1}{f_c} << R_LC << \frac{1}{f_m}.$$

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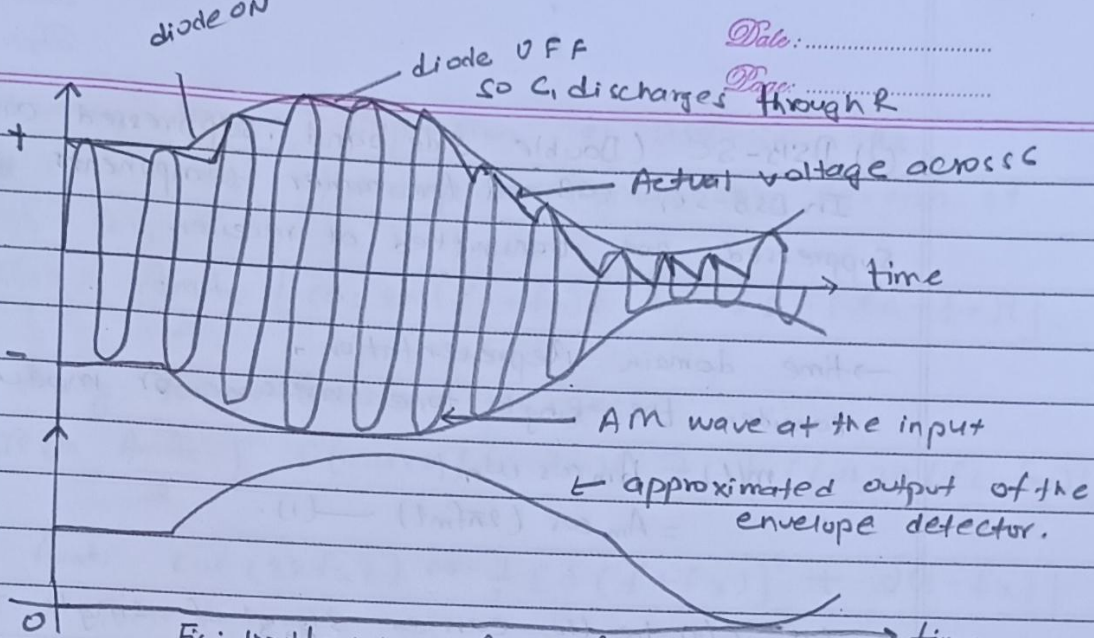

Fig: Input/output waveforms for an envelope detector.

-operation,

It consists of diode and resistor / capacitor (RC) filter. the Standard amplitude modulation wave is applied at i/p of detector. On the positive half cycle of input signal ~~~~ s(t). the diode is forward biased and capacitor C charges up rapidly to the peak value, the diode stops conducting, the i/p signal falls below which makes the diode reverse biased. the Capacitor C now discharges slowly through the resistor R_L. the discharging process continues until the next positive half cycle. When the i/p signal becomes greater than Capacitor voltage, the diode conducts again and the process is repeated.

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(2) DSB-SC (Double side band, Suppressed carrier) →

In DSB-SC, ~~ESB and~~ the carrier components is fully suppressed and transmitted at receiver.

→ time domain Representation,

consider, the single tone message or modulating signal,

$$m(t) = A_m \cos \omega_m t$$
$$= A_m \cos(2\pi f_m t) \quad \text{—(1)}.$$

Let $c(t)$ be the carrier signal of single tone,

$$c(t) = A_c \cos(2\pi f_c t) \quad \text{—(2)}.$$

Hence, the time domain expression of DSB-SC is given as, product of $m(t)$ & $c(t)$.

$$s(t) = m(t) * c(t)$$
$$= A_m A_c \cos 2\pi f_c t \cdot \cos 2\pi f_m t$$
$$= \frac{A_m A_c}{2} \cdot 2 \cos 2\pi f_c t \cdot \cos 2\pi f_m t$$

→ (fc is carrier freq. high carrier)

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

∴ Hence, this is the required expression for time domain ~~of~~ of DSB-SC.

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→ frequency domain expression of DSB-SC, as,

we know, the amplitude modulation equation of DSB-SC as,

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

Taking fourier transform we get,

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

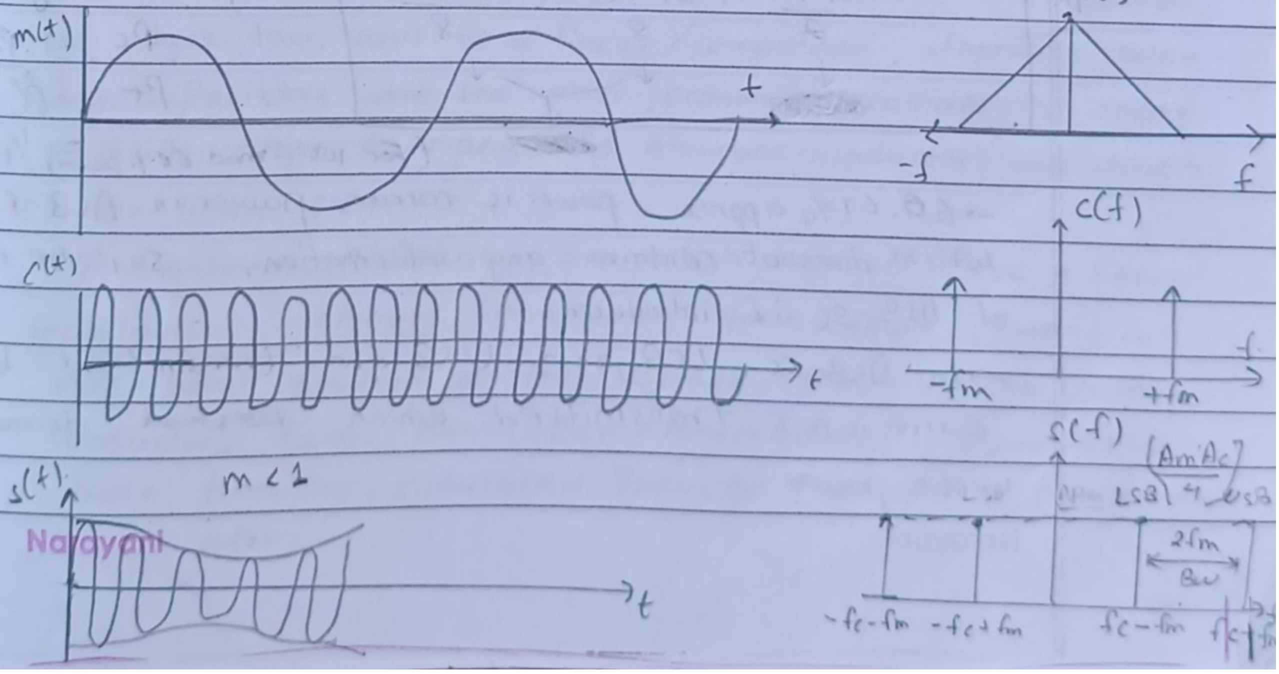
But, $\cos(2\pi f_x t) \leftrightarrow \frac{1}{2}[\delta(f+f_x)] + \delta(f-f_x)]$

There fore,

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

∴ Hence, this is the required expression of DSB-SC (frequency domain).

Gr: Wave form of DSB-SC →



→ Transmission power in DSB-SC,

We have,

In DSB-~~FC~~, carrier power is zero, only the USB & LSB power is useful to certain value.

In DSB-SC no carrier power i.e. $P_c = 0$.

$\therefore$ Power of Carrier $(P_c) = A_c^2/2$

$\therefore$ Power of USB $= (\mu^2 A_c^2)/8$.

$\therefore$ Power of LSB $= (\mu^2 A_c^2)/8$.

(Hence, Total power (P_T) is given as, for DSB-FC)

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

$$= \frac{A_c^2}{2} + \frac{\mu^2 A_c^2}{8} + \frac{\mu^2 A_c^2}{8} \longrightarrow P_c\left(1 + \frac{\mu^2}{2}\right).$$

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

Here, Total side band power,

$\Rightarrow (\mu^2 A_c^2)/4$

& total $P_T \Rightarrow$

$\frac{\mu^2 A_c^2}{4}$.

DSB-SC

• No use DSB, use SC.

For DSB-FC standard,

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

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

[redacted]% [redacted]%

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

$P_c = P_c$.

$\therefore$ power saving $\Rightarrow$

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

(For 100% mod 66% power)

→ 66.67% approx power is carrier power in DSB-FC which doesnot contain any information, so, the concept of DSB-SC is introduced.

→ In DSB-SC, LSB and USB are transmitted but carrier is not transmitted which ~~doesnot~~ donot waste power.

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→ Transmission efficiency,

$$\eta = \frac{P_{USB} + P_{LSB}}{P_{Total\ power}} \quad \left(\because \frac{\text{Useful power}}{\text{Total power}} \right)$$

$$= \frac{\mu^2 A_c^2}{8} + \frac{\mu^2 A_c^2}{8} \Big/ \frac{\mu^2 A_c^2}{4}$$

$$= 1, \text{ i.e. } 100\%$$

∵ Hence, η of DSB-SC is equal to 1.

~~# Gen~~

~~# Generation of DSB-SC~~

~~non-linear elements~~

Linear and non-linear elements for generation of DSB-SC;

(1) Linear Elements →

In an electrical circuit, a linear element is an electrical element with linear relationship between current and voltage. Theoretically, we can say any component which will follow the ohm's law, will be a linear component otherwise non-linear. Resistors are the most common examples of linear element. other examples like air-core inductor, capacitor, etc.

(2) Non-linear Elements →

A non-linear element is one which does not have a linear input/output relation. In a diode for example current is a non-linear function of the voltage. Most of the semi-conductors have non-linear characteristics. Examples - diode, transistor, saturated inductor (with core), transformer, etc.

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Generation of DSB - SC;

(1) Balanced Modulator →

The balanced modulator is used for the generation of DSB-SC AM signal →

- the schematic diagram of Balanced modulator is shown in figure below →

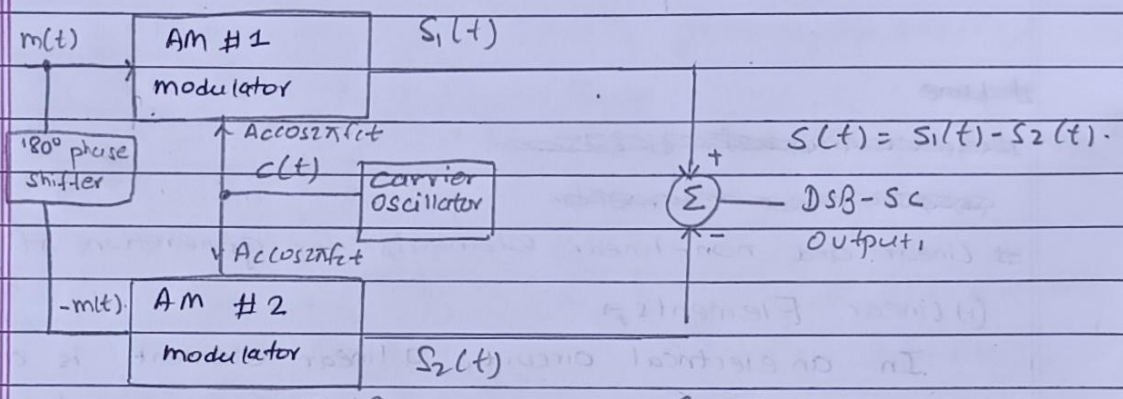

Fig: Balanced modulator for DSB-SC generation.

It consists of two AM modulator in balanced mode. So, as to support the carrier wave. Let us assume that the two modulator, are identical except the sign $[+m(t), -m(t)]$.

Standard wave of two modulator is given as,

$S_1(t) = A_c[1 + k_a m(t)] \cos 2\pi f_c t$. , $K_a = 1/A_c$ amplitude sensitivity

$S_2(t) = A_c[1 - k_a m(t)] \cos 2\pi f_c t$.

Now,

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The output is given as,

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

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

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

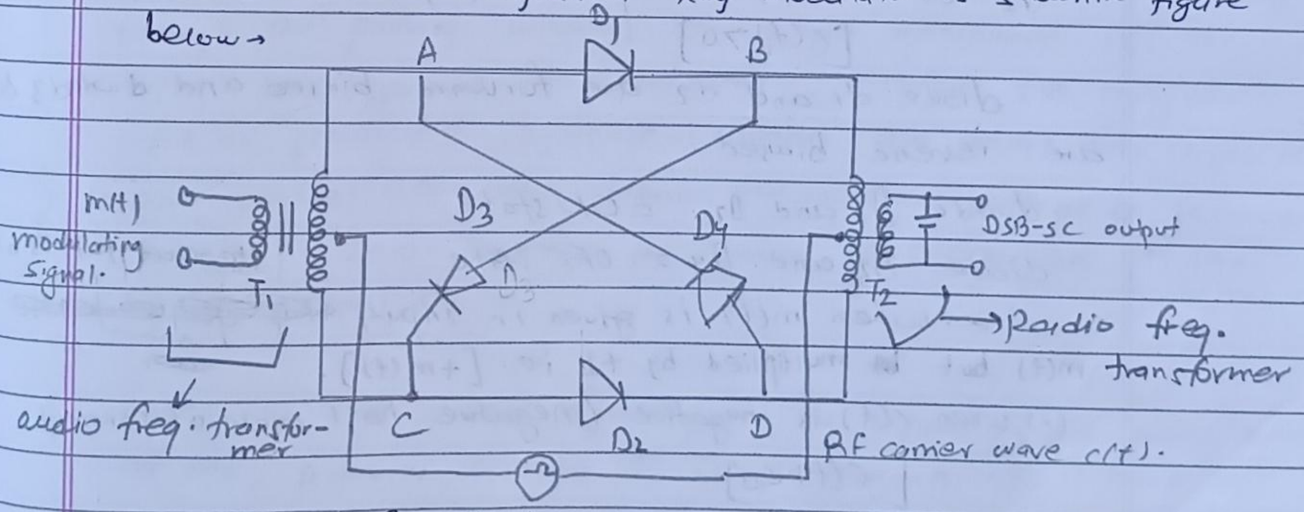
$$= A_c \cos 2\pi f_c t + A_c k_a m(t)\cos 2\pi f_c t - A_c \cos 2\pi f_c t + A_c k_a m(t)\cos 2\pi f_c t$$

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

∴ Here, except scaling factor $2k_a$ the output of Balanced modulator is same as DSB-SC.

(2) Ring Modulator (or chopper modulator) →

- It is also called lattic or double balanced modulator.
- the schematic diagram of Ring modulator is shown in figure below →



It consists of four diodes, an audio frequency transformer T_1 and an RF transformer T_2. The carrier signal is assumed to be a square wave with frequency f_c and it is connected between

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the centre taps of the two transformers. The DSB-SC output is obtained at the secondary of the RF transformer T_2.

Working:-

The operation is explained with the assumptions that the diodes act as perfect switches and that they are switched ON & OFF by the RF carrier signal. This is because the amplitude and frequency of the carrier is higher than that of the modulating signal.

Mode1: Carrier suppression:- (without presence of m(t))

m(t) = 0, modulating signal is zero.

(i) when c(t) is positive (positive half cycle of carrier) [c(t) > 0]

diode d_1 and d_2 are forward biased and diode d_3 & d_4 are reverse biased.

diode D_1 and D_2 = ON state

diode D_3 and D_4 = OFF state

i.e. when m(t) is given in input, the output is also m(t) but multiplied by +1 i.e. [+m(t)].

(ii) when c(t) is negative (negative half cycle of carrier) [c(t) < 0]

diode d_1 and d_2 are reversed biased and diode d_3 & d_4 are forward biased.

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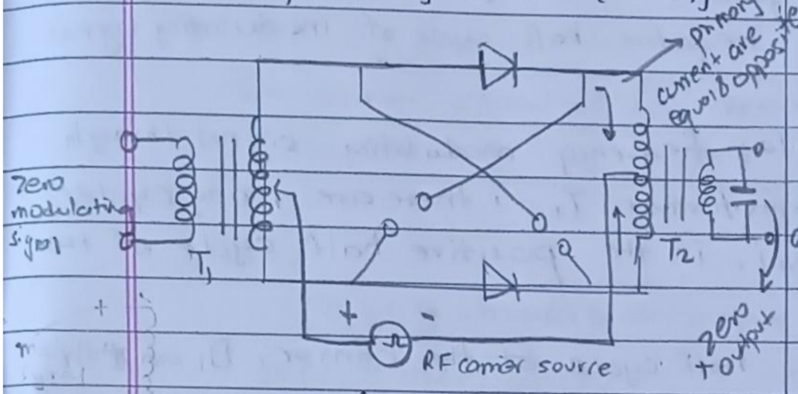
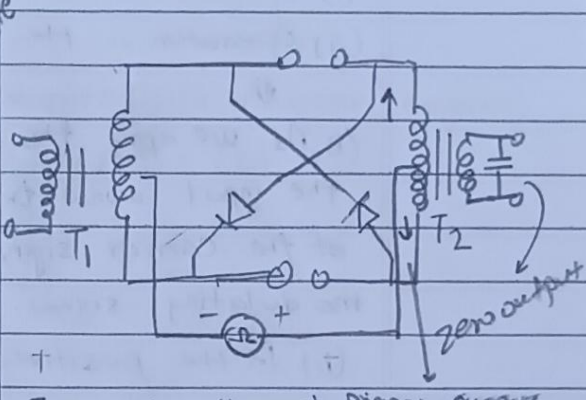
diode D_1 and D_2 = OFF state
diode D_3 and D_4 = ON state
i.e. when m(t) is given in input, the output is also m(t) but multiplied by −1 i.e. [−m(t)].



mode1 : (i)



mode1 : (ii) + primary currents are equal and opposite

mode1 : (i); let us observe the directions of currents flowing through the primary windings of output transformer T_2, they are equal and opposite to each other. Therefore, the magnetic field produced by these currents are equal and opposite and cancel each other. Hence, the induced voltage in secondary winding is zero. Therefore, the carrier is supported in the positive half cycle.

mode1 : (ii) the current flowing in the upper and lower halves of the primary winding of T_2 are again equal and in opposite directions. This is going to cancel the magnetic fields are explained in mode I. Thus, the output voltage in this mode also is zero. Thus the carrier is suppressed in the negative half cycle as well.

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Mode 2: Operation in presence of modulating signal →

Now, let us discuss the operation, when RF carrier and the modulating signal both are applied.

(i) operation in the positive half cycle of modulating signal.

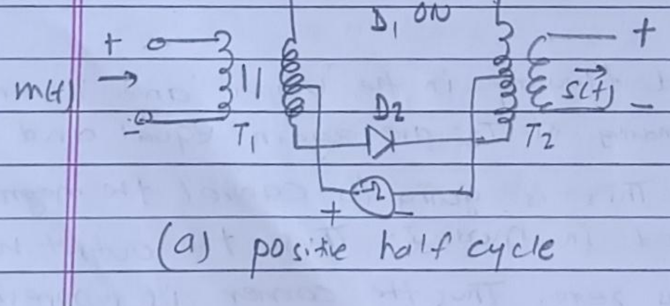
(ii) Operation in the negative half cycle of modulating signal.

⇓

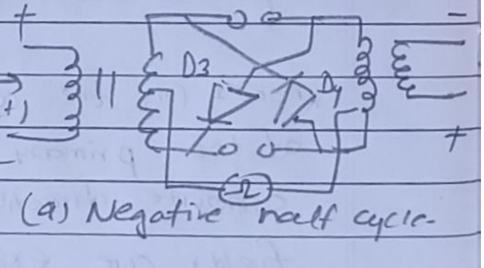
(i) As we apply the low frequency modulating signal through the input audio transformer T_1, there are many cycles of the carrier signal, in the positive half cycle iof the modulating signal.

(ii) In the positive half cycle of the carrier, D_1 and D_2 are on and secondary of T_1 is applied as it is across the primary of T_2. Hence, during the posite half cycle of carrier, the output of T_2 is positive.

(iii) In the negative half cycle of the carrier, D_3 & D_4 are turned on and the secondary of T_1 is applied in a reversed manner across the primary of T_2. Thus, the primary voltage of T_2 is negative and output voltage also becomes negative.



(a) positive half cycle



(a) Negative half cycle

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From this two case, the ring modulator in its Ideal form is produced is a product modulator.

Here,

$$c(t) = g(t) = \begin{cases} +1 & c(t) > 0 \\ -1 & c(t) < 0 \end{cases}$$

So, carrier signal $c(t)$ is represented in fourier series,

i.e. $c(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos(2\pi f c t (2n-1))$.

Now,

The ring modulator output is,

$$s(t) = m(t) * c(t)$$

$$= \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} * [\cos(2\pi f c t (2n-1)) * m(t)]$$

Taking, when $n = 1$, excluding all values $n > 1$.

$\therefore s(t) = \frac{4}{\pi} \cos 2\pi f c t * m(t)$ + Higher order term

Using suitable BPF;

$$\boxed{s(t) = \frac{4}{\pi} \cos 2\pi f c t * m(t)}$$

$\therefore$ Hence, this is the required expression of DSB-SC generator using Ring modulator.

Synchronous (coherent) detector for DSB-SC,

The experimental diagram of synchronous detector is shown in figure below →

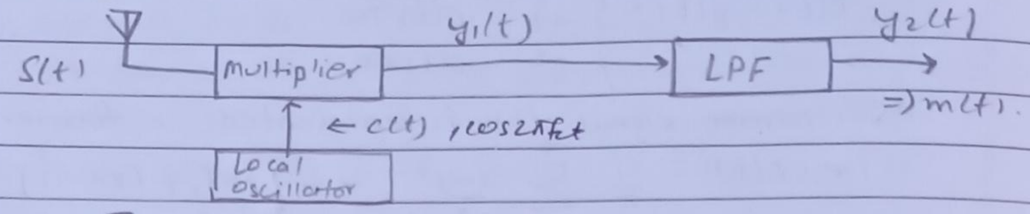

Fig: demodulation of DSB-SC.

We know,

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

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

Then, the output of multiplier $y_1(t)$ is,

$$y_1(t) = m(t) \cdot A_c \cos 2\pi f_c t \cdot \cos 2\pi f_c t$$

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

$$= \frac{A_c}{2} m(t) \cdot (1 + \cos 4\pi f_c t).$$

$$= \frac{A_c}{2} \cdot m(t) + \frac{A_c}{2} m(t) \cdot \cos 4\pi f_c t.$$

$y_1(t)$ is passed to LPF to block the unwanted wave signal.

$$\boxed{y_2(t) = \frac{A_c}{2} m(t)}$$

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Characterestics:-
(1) Efficiency is greater (no power is wasted).
(2) Modulation and demodulation process are complex and expensive.
(3) Bandwidth is twice that of message signal. So wastage of precious bandwidth.

"If two signals at different frequencies are passed through a non-linear resistance then at the o/p, we get an AM signal with ~~freques~~ suppressed carrier."
→ device with non-linear resistance → diode, JFET ~~&~~ or BJT.

Applications & uses:-
(1) FM and TV broadcasting to transmit two-channel stereo signal.
(2) Some types of PSK (phase-shift keying). which is used for transmitting binary data.

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(3) SSB-SC (Single Side Band - Suppressed Carrier) →

- In Amplitude Modulation and double-sideband suppressed-carrier (DSB-SC) modulation are wasteful of bandwidth, since they both need a transmission bandwidth equal to twice the message signal bandwidth. In either case one half of the transmission bandwidth is occupied by the upper side band of the modulated signal whereas the other half is occupied by the lower sideband.

- However, the lower and upper sidebands are uniquely related to each other by virtue of their symmetry about the carrier frequency, if amplitude & phase spectra of either sideband is given, we can uniquely determine the other. This means that as far as the transmission of information is concerned, only one sideband is necessary. Thus, if the carrier and one of the two sidebands are suppressed at the transmitter, no information is lost.

- Modulation of this type which provides a single sideband with suppressed carrier is known as single sideband suppressed carrier (SSB-SC) system. Thus, SSB-SC system reduces the transmission bandwidth by half. This means that in a given frequency band we can accommodate twice the number of channels by using a single sideband in place of both the sidebands.

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we know, the transmission bandwidth of standard AM as well as DSB-SC modulated wave is 2W Hz. i.e. twice the message bandwidth W. Therefore, both these systems are bandwidth inefficient systems. In both these systems, one half ~~is occupied~~ of the transmission bandwidth is occupied by the upper sideband (USB) and the other half is occupied by the lower sideband (LSB).

But the most important thing is that the information contained in the USB is exactly identical to that carried by the LSB. So, by transmitting both the side bands we ~~can~~ are transmitting the same information twice. Hence, we can transmit only one sideband (USB or LSB) without any loss of information. So, it is possible to suppress the carrier and one side band completely. When only one sideband is transmitted, the modulation is referred to as Single Sideband ~~Communication~~. modulation. It is also called as SSB or SSB-SC modulation.

→ Time domain expression;

Proof:- Mathematical Expression,

Let, $m(t)$ be the modulating signal of DSB-SC wave,

$$m(t) = A_m \sin \omega_m t \quad \text{or} \quad A_m \cos \omega_m t$$

$c(t)$ be the carrier signal then,

$$c(t) = A_c \sin \omega_c t \quad \text{or} \quad A_c \cos \omega_c t$$

∴ $s(t)$ of SSB-SC is given as,

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

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

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

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

On taking Upper Side Band (USB) only we get,

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

$$= \frac{A_m A_c}{2} \{ \cos 2\pi f_c \cdot \cos 2\pi f_m t - \sin 2\pi f_c t \cdot \sin 2\pi f_m t \}$$

$$= \frac{A_c}{2} \{ A_m \cos 2\pi f_c t \cdot \cos 2\pi f_m t - A_m \sin 2\pi f_c t \cdot \sin 2\pi f_m t \}.$$

$$= \frac{A_c}{2} \{ \underbrace{m(t) \cdot \cos 2\pi f_c t}_{\text{Inphase component}} - \underbrace{\hat{m}(t) \cdot \sin 2\pi f_c t}_{\text{Quadrature Component}} \}.$$

∴ Hence, this is the required expression of SSB-SC.

→ In this system transmission power is saved. and transmission efficiency is higher.

Power Saving SSB-SC

⇒ Single Side band and Carrier suppressed.

i.e.

$$\% \text{ power saving} \Rightarrow \frac{\text{Power in Carrier} + \text{Power in one Side band}}{\text{total power}}$$

$$\Rightarrow \frac{\left(1 + \frac{\mu^2}{4}\right)}{\left(1 + \frac{\mu^2}{2}\right)}$$

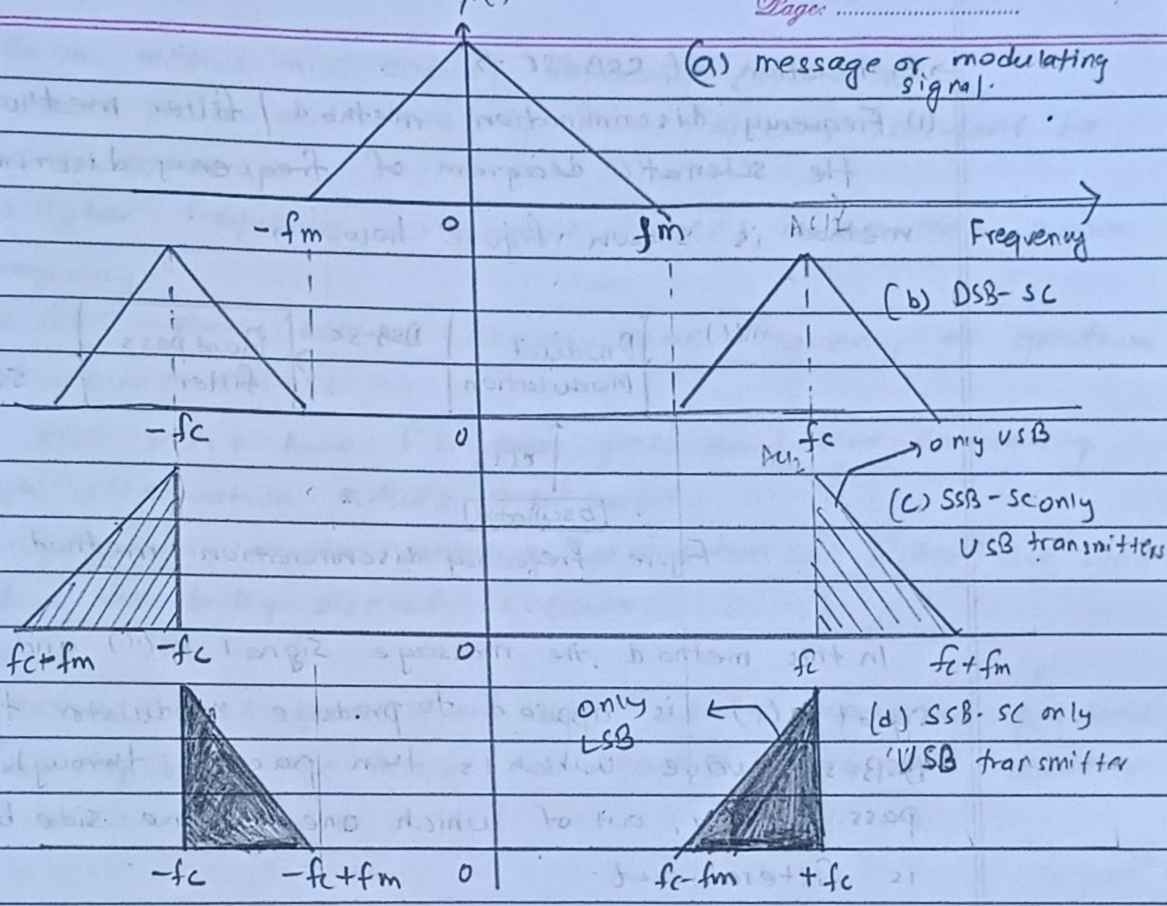

Fig:- Frequency spectrum of SSB-SC.

Power of SSB-SC:

$P_{total} = P_{USB} = P_{LSB}$.

$$P = \frac{V_{rms}^2}{R} = \frac{(V_m/\sqrt{2})^2}{R} = \frac{V_m^2}{2R} = \frac{V_m^2}{2}$$

$$\Rightarrow \frac{A_m^2 A_c^2}{2\times 4} = \frac{A_m^2 A_c^2}{8}$$

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→ Generation of SSB-SC →

(1) Frequency discrimination method / filter method →

the schematic diagram of frequency discrimination method is shown figure below →

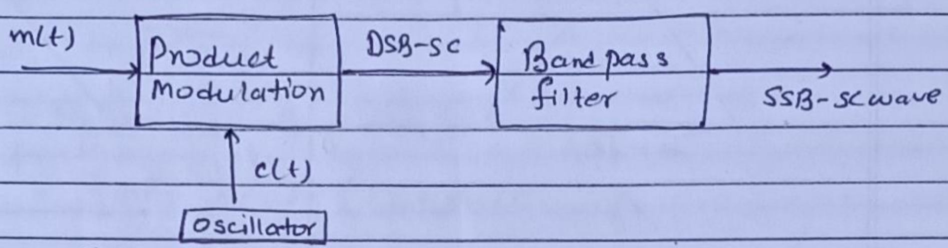

Figure: frequency discrimination method.

In this method the message signal m(t) and carrier signal c(t) is applied to produce modulator to generator DSB-SC wave which is then passed through band pass filter, out of which one of the side band and is filtered out.

(1) the frequency discrimination method is useful only if the band signal is restricted c(t) its lower edge due to which upper and lowerside band is non-over lapping. However this system is not useful for video communication where the message signal or baseband signal starts from DC.

(2) For this method m(t) satisfy following conditions,

(a) m(t) should not have any low frequency content i.e. 0 to 300 Hz.

(b) Higher frequency component of m(t) is less than carrier frequency.

(c) BPF should occupy exactly same frequency as spectrum of desired SSB-SC

(d) BPF must have flat ~~ban~~ pass-band and extremly high attenuation outside pass band.

(e) Generally, crystal ceramic (or mechanical filter) are used due to bulky size of LC filter.

~~ep~~.

- The width of ~~gaid~~ guard band which separates passband from stop band be twice the lowest frequency content of message signal.

- Limitation,

(1) Accurate BPF design is complex.

(2) not suitable for video communication.

(2) Phase discrimination method / phase shift method →

The diagram of phase shift method is shown below →

+ve for LSB } adder used
- ve for USB } subtractor used

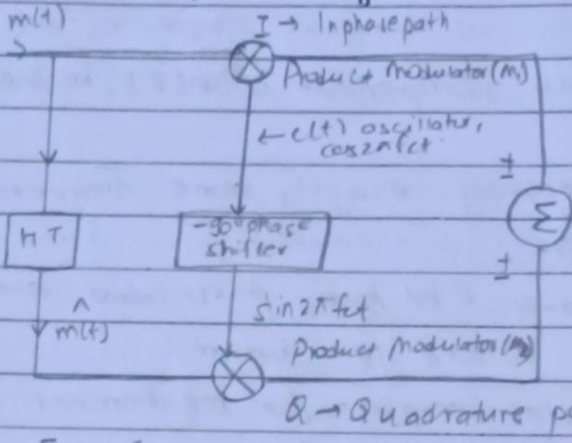

Fig: Generation of SSB-SC wave (phase discrimination method).

- this method involves two separate simultaneous modulation process and subsequent combination of resulting modulation products. the components required are DSB-SC modulation (M_1 and M_2), adder / subtractor, Hilbert Transform.

- Operation,

the Incoming signal m(t) is applied to product modulator (I), which produce DSB-SC wave i.e. (m(t) Accos 2πfct) ~~dee~~. i.e. m(t) Accos2πfct and $\hat{m}(t)$ is produced by Hilbert Transform of m(t) which is feed to adder / Subtractor which give the required SSB-SC wave. If adder is used LSB can be obtained and subtractor is used for USB.

- In this method,
(i) can generate SSB - signal at any frequency.
(ii) use low frequency as modulating signal.
(iii) easy to switch from one side band to the other.

→ advantages

1. Less BW (bandwidth) requirements, as SSB requires a BW of fm. This will allow more number of signals to be transmitted in the same frequency range.
2. Lots of power saving. this is due to the transmission of only one sideband component, at 100% modulation the percent power saving is 83.33%
3. Reduced interference of noise. this is due to the reduced bandwidth. as the bandwidth increases, the amount of noise added to the signal will increase.

→ disadvantages

1. the generation and reception of SSB signal ~~is~~ is complicated.

→ 2. the SSB transmitter and receiver need to have an excellent frequency stability. A slight change in frequency will hamper the quality of transmitted and received signal. therefore, SSB is not generally used for the transmission of good quality music.

It is used for speech transmission.

Synchronous (coherent) detector for ~~SSB~~ SSB-SC →

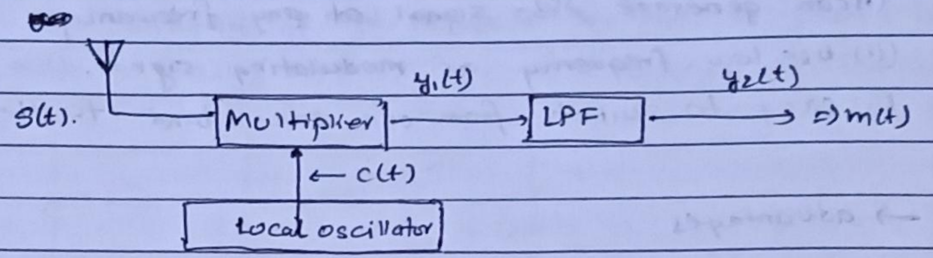

Fig: demodulator of SSB-SC

We know,

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

and, $c(t) = \cos 2\pi f_c t.$

Then, the output of multiplier $y_1(t)$ is,

$$y_1(t) = s(t)\cdot c(t)$$

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

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

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

∵ $y_1(t)$ is passed to LPF to block the unwanted wave signal.

$$\boxed{\therefore y_2(t) = \frac{A_c}{4} m(t)}$$

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- applications of SSB →

(1) SSB transmission is used in the applications where the power saving and low bandwidth requirements are important.

(2) the application areas are land and air mobile communication telemetry, military communications, navigation and amature radio. Many of these applications are point to point communication applications.

Note, Both DSB and SSB signals are more efficient in terms of power usage, the power wasted in the useless carrier is saved, thereby allowing more power to be put into the side bands.

7) Demodulation of SSB-SC →

(a) Re-insertion technique →

when large carrier is introduced, asynchronous detection is not good to recover message signal, the carrier re-insertion technique is used. The expression for SSB-SC with large carrier is given as,

$$s(t) = A_c \cos 2\pi f_c t + [m(t) \cdot \cos 2\pi f_c t + \hat{m}(t) \cdot \sin 2\pi f_c t] \quad —(1)$$

Here, large carrier is inserted (added) not multiplied →

$$s(t) = [A + m(t)] \cos 2\pi f_c t + \hat{m}(t) \sin 2\pi f_c t \quad —(2)$$

$$s(t) = \underbrace{e(t) \cos(2\pi f_c t + \phi)}_{\text{envelope}}.$$

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where,

$$e(t) = \left\{ [A + m(t)]^2 + \hat{m}^2(t) \right\}^{1/2}$$

$$= \left\{ \{A^2 + 2Am(t) + m^2(t)\} + \hat{m}^2(t) \right\}^{1/2}$$

$$= A^{2 \times 1/2} \left\{ 1 + \frac{2Am(t)}{A} + \frac{m^2(t)}{A^2} + \frac{\hat{m}^2(t)}{A^2} \right\}^{1/2}$$

$$= A \left\{ 1 + \frac{2m(t)}{A} + \frac{m^2(t)}{A^2} + \frac{\hat{m}^2(t)}{A^2} \right\}^{1/2}$$

If $A >> |m(t)|$ then $A >> |\hat{m}(t)|$. So, higher term is ignored thus,

$$e(t) = A \left[1 + \frac{2m(t)}{A} \right]^{1/2}$$

$$= A \left[1 + \frac{m(t)}{A} \right] \quad \{\text{Using Binomial expression}\}$$

$$\boxed{e(t) = A + m(t)}$$

∴ Hence, this is the required expression of reinsertion technique in SSB-SC.

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(4) ISB (Independent Side Band) →

- It is not widely used. It can be generated by two identical modulator. DSB-SC modulators, two side band filters and summing device are used for ISB signal generator.

- the schematic diagram of ISB system is shown in figure below →

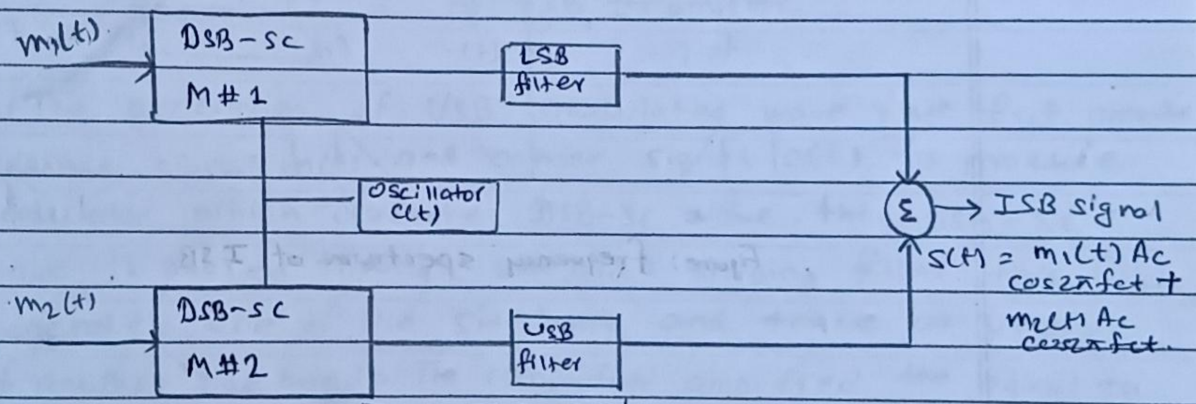

Figure: ISB system

Message signal are given to two DSB-SC wave generation followed by SSB filters i.e. one generator with LSB filter and while other for USB filter is. So, as to generate lower and upper side band respectively.

the output of the SSB-SC filters are summed upto give out the ISB wave which is given as:-

$$s(t) = m_1(t) A_c \cos 2\pi f_c t + m_2(t) A_c \cos 2\pi f_c t.$$

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The frequency domain spectrum is shown in figure below →

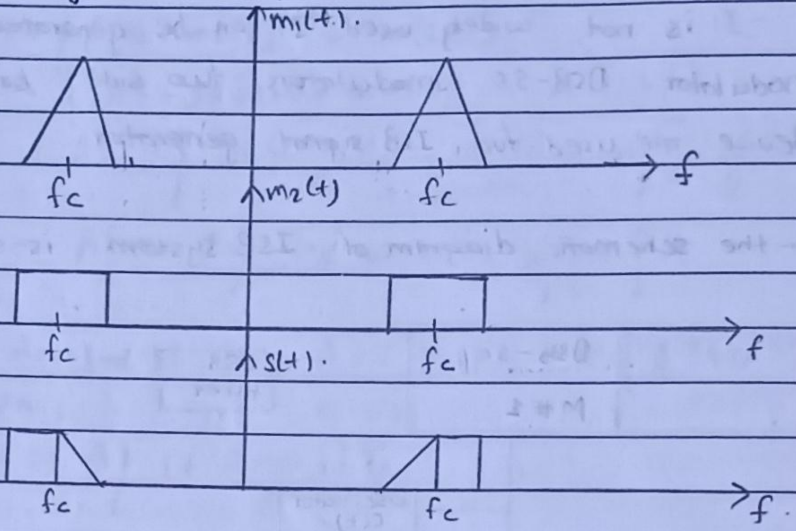

Figure: frequency spectrum of ISB.

(5) VESTIGIAL SIDEBAND (VSB) →

The stringent frequency-response requirements on the sideband filter in SSB-SC system can be relaxed by allowing a part of the unwanted sideband (called as Vestige) to appear in the output of the modulator. Due to this, the design of the sideband filter is ~~amplified~~ simplified to a great extent. But the bandwidth of the system is increased slightly.

There is difficulty in isolating sidebands in SSB-AM. So, the modulation technique is envolved that pass one side band and vestige ~~and~~ or trace of another side band known as VSB-

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– It is compromise between DSB-SC and SSB-SC.

The schematic diagram for Vestigial side band (VSB) is shown in figure below →

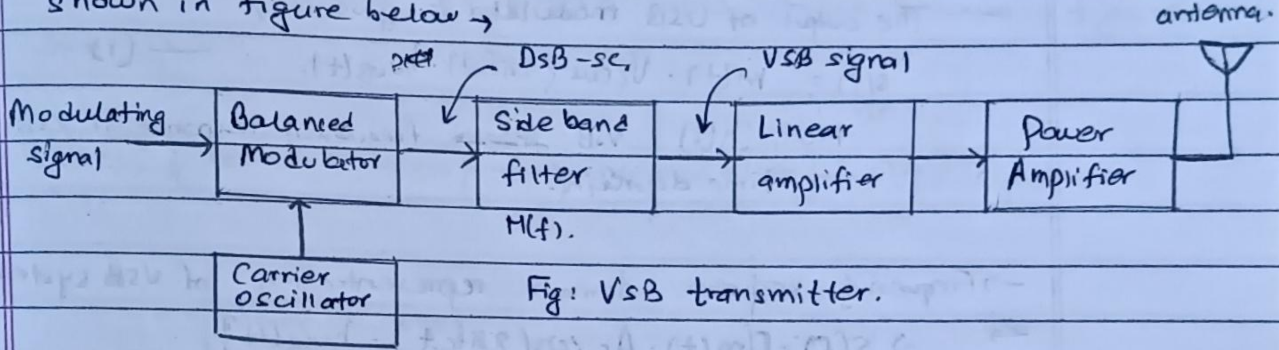

Fig: VSB transmitter.

For the generation of VSB modulated wave, we first provide message signal m(t) and carrier signal c(t) to produce modulator which generates DSB-SC wave. this DSB-SC wave is passed through side band shaping filter. this filter generates one of the sideband and trace or vestige of another side band. The amplifier amplified the signal to the antenna.

— Examples:-

- In television broadcasting VSB (full carrier) and 30% of LSB is transmitted.
- VSB modulation has been standard for transmission of television signals, where good phase characterestics and transmission of low frequency component are important.

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→ Time domain representation of VSB system →

The output of VSB modulator is given as,

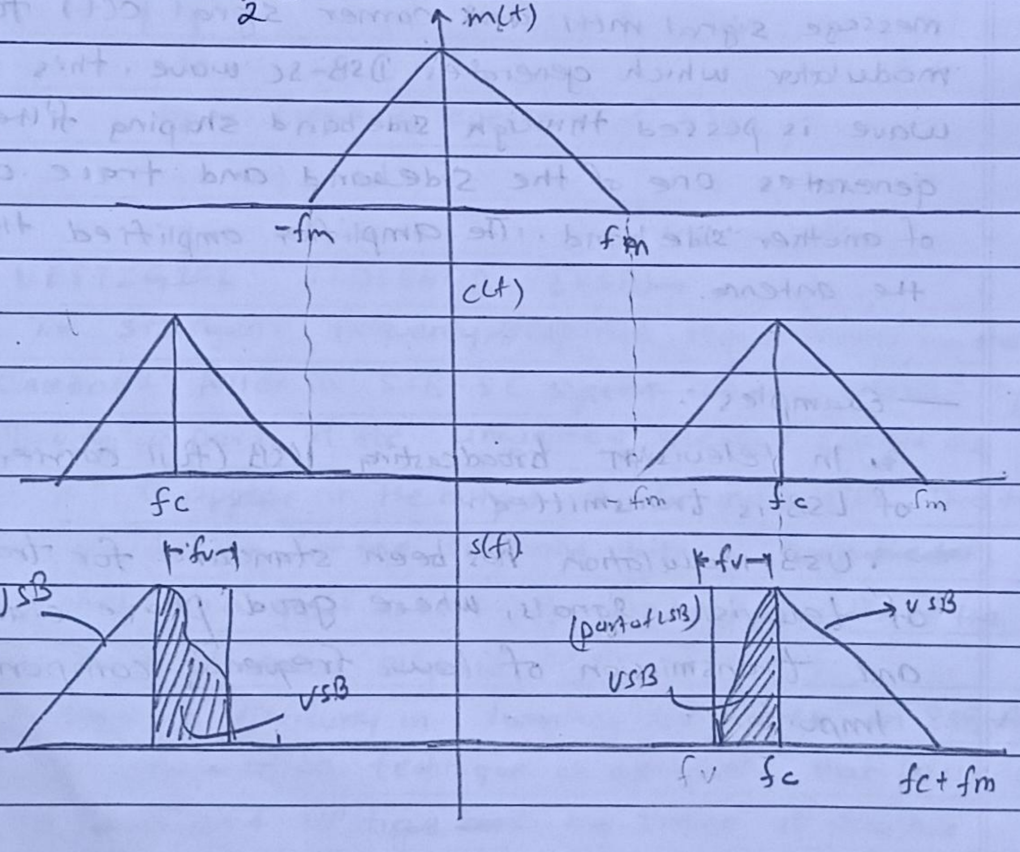
$$s(t) = m(t) \cdot V_c \cos(2\pi f_c t) \cdot h_{VSB}(t). \quad \text{—— (1).}$$

Here, $h(t)$, VSB ~~[illegible]~~ frequency response of VSB in time domain.

→ Frequency ~~response~~ domain representation of VSB system →

$$\Rightarrow S(f) = F[m(t) \cdot A_c \cos(2\pi f_c t) \cdot h_{VSB}(t)]$$

$$= \frac{A_c}{2}[m(f - f_c) + m(f - f_c)] \cdot H_{VSB}(f) \quad \text{—— (11)}$$



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→ Generation of VSB modulated wave →

The block diagram of a VSB modulator is shown in figure below →

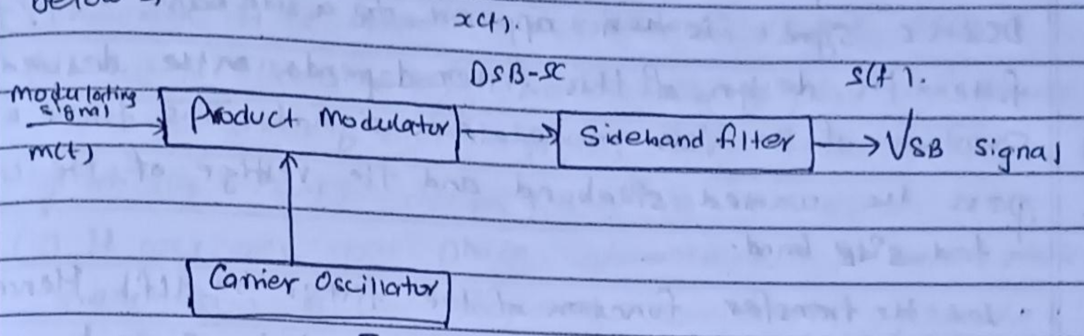

Fig: Generation of VSB signal.

The modulating signal m(t) is applied to the product modulator. the output of the carrier oscillator is also applied to the other input of the product modulator. the output of the product modulator is given as,

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

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

After passing $x(t)$, DSB-SC to VSB filter we get,

$$s(t) = m(t) \cdot V_c \cos(2\pi f_c t) \cdot h_{VSB}(t) \quad \text{---(1)}.$$

→ transmission Bandwidth →

From frequency spectrum, it is evident that the transmission bandwidth of VSB modulated wave is given by,

$$B = (f_m + f_v) Hz$$

where, $f_m \Rightarrow$ message frequency

$f_v \Rightarrow$ vestigial sideband frequency.

$$BW \Rightarrow (\cancel{f_c} + f_m - \cancel{f_c}) + f_v$$

$$\Rightarrow f_m + f_v.$$

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~~The modulating ...~~:

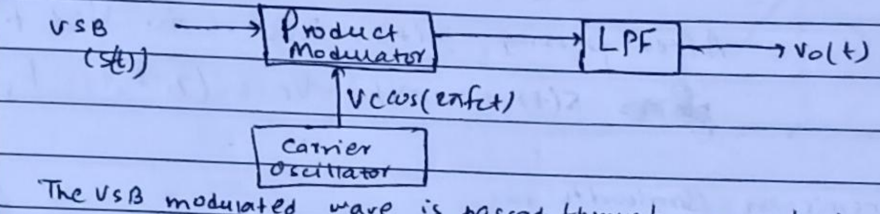
This represents a DSB-SC modulated wave. this DSB-SC signal is then applied to a side band shaping filter. the design of this filter depends on the desired spectrum of the VSB modulated signal. This filter will pass the wanted sideband and the vestige of the unwanted side band.

· let the transfer function of the filter be $H(f)$. Hence, the spectrum of the VSB modulated signal is given by,

$$S(f) = \frac{V_c}{2}\left[X(f-f_c) + X(f_1+f_c)\right] H(f).$$

demodulator of VSB →

The synchronous detector for the detection of VSB modulated wave is



The VSB modulated wave is passed through a product modulator where it is multiplied with the locally generated synchronous carrier. Hence, the output of the product modulator is given by,

$$m(t) = s(t)\,x_c(t) = s(t) * V_c \cos 2\pi f_c t.$$

Taking Fourier transform,

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

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

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

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

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→ advantages of VSB →

(1) the main advantage of VSB modulation is the reduction in bandwidth. It is almost as efficient as the SSB.

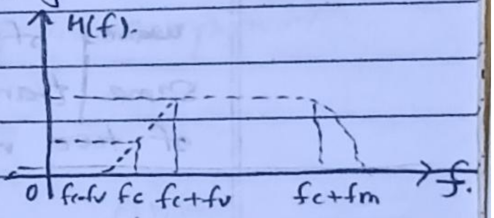
(2) due to allowance of transmitting a part of lower sideband, the constraint on the filters have been relaxed. So, pratically, easy to design filters can be used.

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

→ applications of VSB,

(1) VSB modulation has become standard for the transmission of Television signals. Because the video signals need a large transmission bandwidth if transmitted using DSB-FC or DSB-SC techniques.

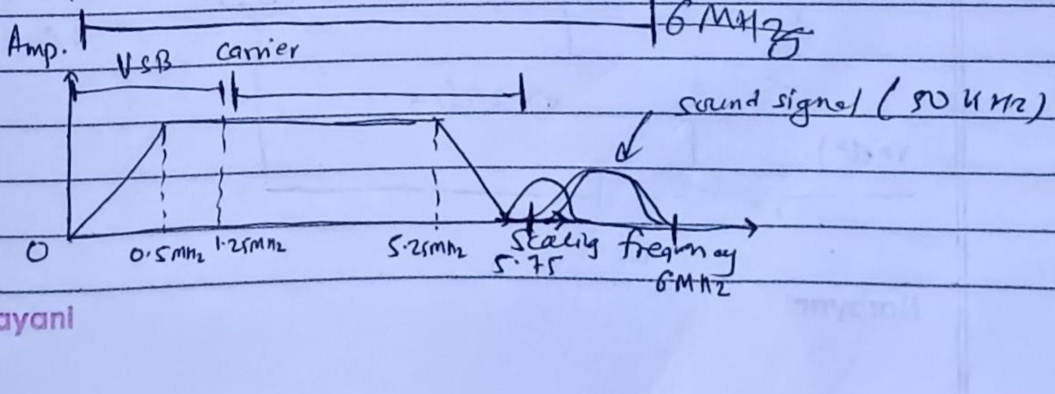
i.e.



(a) In TV transmission.

(b) In TV transmission ⇒ Audio + Video

- BW of video ⇒ 4.2MHz
- By DSB-FC = 2(4.2) + guard band + audio ⇒ 9MHz
- By VSB ⇒ 6MHz



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Note,

VSB modulation requires a channel bandwidth that is between that required for SSB and DSB-SC systems, and the saving in bandwidth can be significant if modulating signals with large bandwidths are being handled, as in the case of television signals and high-speed data.

(6) QAM (Quadrature Amplitude Modulation) →

- The QAM is similar to DSB-SC, the only difference is that QAM sends two message signals over the same spectrum. It is used for the Bandwidth conversation scheme. It allows two DSB-SC modulated waves of two Independent message to occupy the same transmission bandwidth and allows separation of two messages at the receiver output.

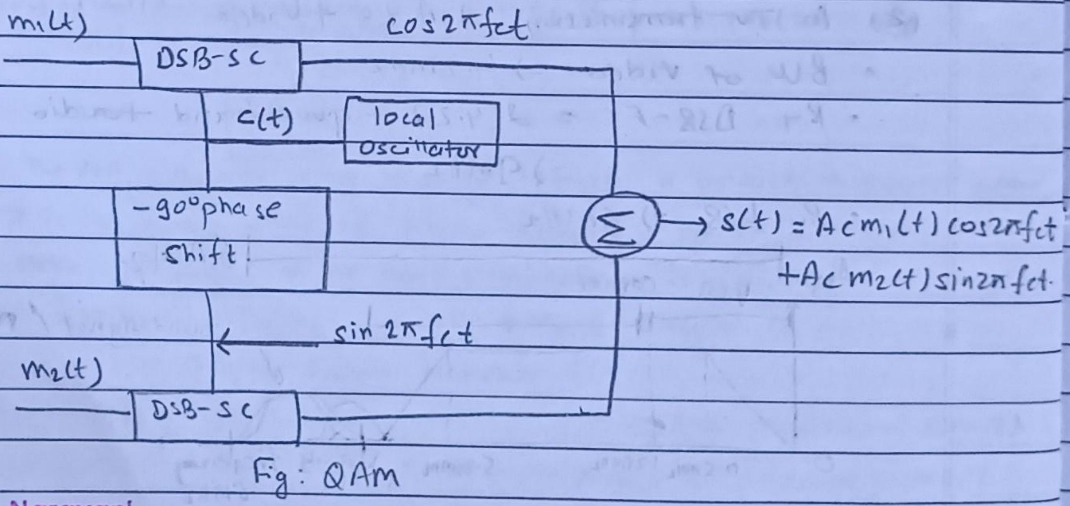

Fig: QAM

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QAM has two DSB-SC modulators which are supplied with different message signal and two carrier waves different message signal and two carrier waves of same frequency but quadrature in phase to eachother.

- the output of DSB-SC are fed to adder producing,

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

- multiplexed signal ~~equ~~ occupies transmission bandwidth of $2W$, centred at carrier frequency f_c. where W is the bandwidth of $m_1(t)$ and $m_2(t)$. which ever is the largest. Hence, multiplixed signal consists of Inphase component

$A_c m_1(t) \cos 2\pi f_c t$ and quadrature phase component. $A_c m_2(t) \sin 2\pi f_c t$.

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Synchronous (coherent) detector for DSB-FC wave →

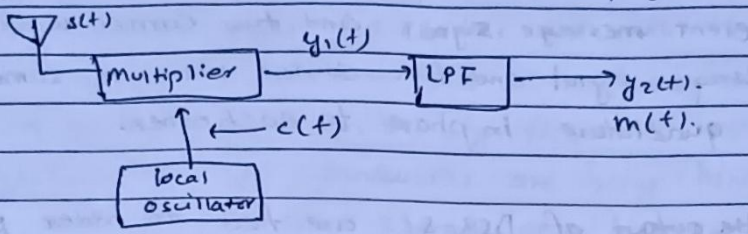

Figure: demodulation of DSB-FC.

Here, we have, $s(t) = A_c[1 + K_a m(t)] \cos 2\pi f_c t$.

and $c(t) = \cos 2\pi f_c t$.

Then the output of multiplier $y_1(t)$ is,

$$y_1(t) = s(t) \cdot c(t)$$

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

$$= A_c/2 [1 + K_a m(t)] * \{\cos 4\pi f_c t + 1\}.$$

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

$$= \underbrace{\frac{A_c}{2}}_{DC} + \underbrace{\frac{A_c}{2} \cos 4\pi f_c t}_{Noise} + \underbrace{\frac{A_c}{2} K_a m(t)}_{message} + \underbrace{\frac{A_c}{2} K_a m(t) \cos 4\pi f_c t}_{Noise}.$$

$y_1(t)$ is passed to LPF to block the unwanted wave signal.

$$y_2(t) = \frac{A_c}{2} K_a m(t).$$

Numerical

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(Q) The carrier wave is represented by the equation, $e_c = 10\sin\omega t$. Draw the waveform of an AM wave for $m = 0.5$.

Solution,

(i) Given, $e_c = 10\sin\omega t$, $E_c = 10 \text{ Volts} = A_c$.

(ii) let us evaluate E_m from E_m.

Since, $m = \frac{E_m}{E_c}$, therefore,

$E_m = m \times E_c = 0.5 \times 10 = 5$ volt.

$E_{max} = E_c + E_m = 10 + 5 = 15$ volt.

$E_{min} = E_c - E_m = 10 - 5 = 5$ volt.

(iii) The AM wave for $m = 0.5$ has been shown below;

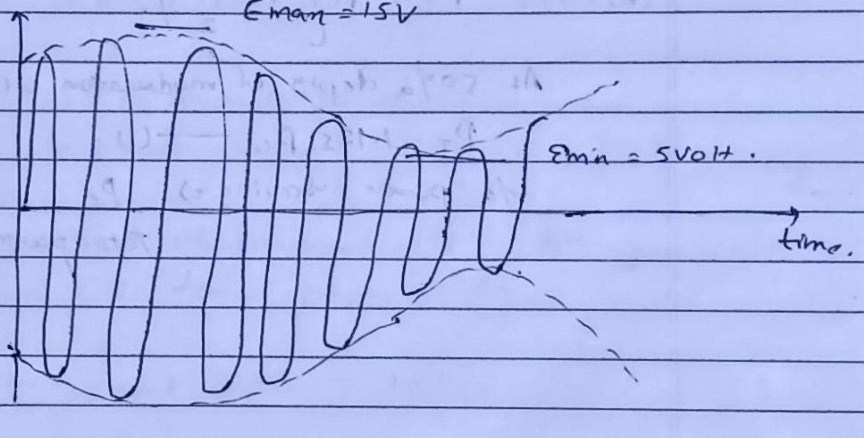


Fig: AM wave for (0.5) m.

Q. Calculate the percent power saving for a DSB-SC signal for the percent modulation of (a) 100 % and (b) 50 %.

Solution,

(a) Total power of AM $\Rightarrow P_T = P_c\left(1 + \frac{\mu^2}{2}\right)$

At 100 % depth of modulation $\mu = 1$.

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

$$P_T = 1.5P_c \quad \text{——(1).}$$

(Here carrier power is saved).

% power saving $\Rightarrow \frac{P_c}{\text{Total power}} \times 100\% = 66.66\%$

(b) $P_T = P_T \Rightarrow P_c\left(1 + \frac{\mu^2}{2}\right)$.

At 50% depth of modulation $\mu = 0.5$

$$P_T = 1.125\,P_c \quad \text{——(1).}$$

% power saving $\Rightarrow \frac{P_c}{\text{Total power}} \times 100\% = \frac{\cancel{P_c}}{1.125\cancel{P_c}} \times 100$

$\Rightarrow 100/1.125$

$\Rightarrow 88.88\%$.

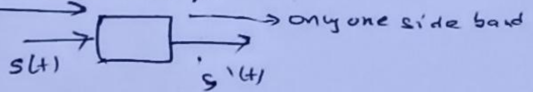
Q. Calculate the percent power saving for the SSB signal if the AM wave is modulated to a depth of (a) 100 % and 50 %.

Solution,

carrier and one side band is suppressed (SSB-SC).

(Carrier + SSB (one side band)

saved



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Power saving in SSB signal $\Rightarrow$ Carrier and one sideband are suppressed.

therefore only one side band is transmitted.

$$\% \text{ power saving} \rightarrow \frac{\text{Power in carrier} + \text{Power in one side band}}{\text{Total Power}}$$

$$\rightarrow \frac{P_c(1+m^2/4)}{P_c(1+m^2/2)}$$

$$\Rightarrow \frac{1+m^2/4}{1+m^2/2}$$

power of side band $\left(\frac{m^2 A_c^2}{8}\right)$

Carrier power $\left(A_c^2/2\right)$

At 100% modulation, $m=1$,

$$\% \text{ power saving} = \frac{1.25}{1.5} = 83.333\%$$

At 50% modulation, $m = 0.5$

$$\% \text{ power saving} = \frac{1.0625}{1.125} = 94.44\%. \text{ Ans}$$

(Q) If the upper side band power of 100% AM signal is 22W, Find carrier and total power of AM.

Solution,

Modulation percent = 100%

$\therefore m = 1.$

Upper side band power (P_{USB}) = 22W.

Carrier power = ?

total power = ?

P_{LSB} = 22W.

$$P_{LSB} = \frac{m^2 P_c}{4}$$

$$22 = \frac{(1)^2 \cdot P_c}{4}$$

$$22 \times 4 = P_c$$

$\therefore P_C = 88$ watt. (Carrier power).

$\therefore$ Total power $= P_c + P_{USB} + P_{LSB}$

$= 88 + 22 + 22$

$= 132$ watt.

(8) For an AM wave with $\mu = 0.3$, amplitude of carrier 10V and carrier power 900W calculate.

(i) amplitude of message signal ✓

(ii) Power of sideband

(iii) Percentage of total power carried by sideband.

Solution,

Modulation index (μ) = 0.3

Amplitude of carrier wave (A_c) = 10V.

Carrier power (P_c) = 900W.

we have,

$$\text{Modulation index } (\mu) = \frac{A_m}{A_c}$$

$$\therefore A_m = A_c \cdot \mu$$

$$= 10 * 0.3$$

$$= 3V$$

percentage of modulation = 30%. ($\mu \times 100$).

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

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

$$= 900\left(\frac{2+0.3^2}{2}\right)$$

$$= 940.5 \text{ watt.}$$

$$P_{LSB} = \frac{\mu^2}{4} * P_c$$

$$= \frac{0.3^2}{4} * 900$$

$$= 20.25 \text{ watt.}$$

$$P_{USB} = 20.25 \text{ watt.}$$

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

$$= 900 + 20.25 + 20.25$$

$$= 940.5.$$

% of sideband ⇒

$\frac{40.5}{940.5} * 100 \Rightarrow 4.3\%$.

Angle Modulation →

- In angle modulation, either phase or frequency of carrier signal is varied in accordance with the instantaneous value of message signal.

- Amplitude of the carrier kept constant and it is non-linear modulation.

Angle Modulation

- Frequency Modulation → Frequency of the carrier is varied according to the message signal
- Phase Modulation → Phase angle of the carrier is varied according to the message signal.

Frequency Modulation as well as phase modulation are forms of angle modulation. Angle modulation has several advantages, over the amplitude modulation such as reduction, improved system fidelity and more efficient use of power.

But there are some disadvantages too such as increased bandwidth and use of more complex circuits.

Angle modulation is being used for the following applications:

(i) Radio broadcasting
(ii) TV sound transmission
(iii) Two way mobile radio
(iv) Cellular radio
(v) Microwave and satellite communication

The principle of angle modulation can be stated as,
the standard equation of Angle modulation is given as,

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

$= A_c \cos(\theta(t))$ → phase or freq. mod.

where,

A_c = Carrier amplitude, constant

ω_c = angular velocity

ϕ = phase angle

$\theta(t) = \omega_c t + \phi$ ⇒ angle of modulation.

Phase Modulation →

– In phase modulation, phase of carrier signal changes with respect to modulating signal $m(t)$.

If we have carrier signal $c(t)$, we have modulating signal, $m(t)$

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

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

$= A_c \cos(\theta(t))$

$\theta(t)$ changes as $\phi(t)$ changes

→ For phase modulation,

$\theta(t) = 2\pi f_c t + K_p m(t)$.

∴ PM wave is given as,

$S_{PM}(t) = A_c \cos(2\pi f_c t + K_p m(t))$

K_p: phase sensitivity (rad/volt).

Frequency Modulation →

In frequency modulation, frequency of carrier signal $c(t)$ changes with respect to modulating signal $m(t)$.

- If we have carrier signal,

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

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

$= A_c \cos \theta(t)$.

- We have modulating signal $m(t)$.

- In frequency modulation,

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

- FM wave is given as;

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

$K_f \Rightarrow$ frequency sensitivity (Hz/volt).

Also,

$f_i(t) = f_c + K_f\, m(t)$. $f_i(t) \rightarrow$ instantaneous frequency,

On Integrating, both sides we get,

$\int f_i(t) \delta t = \int f_c \delta t + \int K_f\, m(t) \delta t$

$\int 2\pi f(t) \delta t = \int f_c\, 2\pi \delta t + \int K_f \cdot 2\pi \cdot m(t) \cdot \delta t$

$\int \omega \cdot \delta(t) = 2\pi f_c t + 2\pi K_f \int m(t)\, dt$.

$\theta(t) = 2\pi f_c t + 2\pi K_f \int m(t) \cdot dt$.

Relationship between FM and PM wave →

PM wave ⇒ $A_c \cos[2\pi f_c t + K_p m(t)]$

FM wave ⇒ $A_c \cos[2\pi f_c t + \underbrace{2\pi k_f \int m(t)\,dt}_{\text{deviation of frequency}}]$.

Generation of FM using Phase Modulator

The schematic diagram for generation of FM using PM Modulator (Block diagram) is shown below →

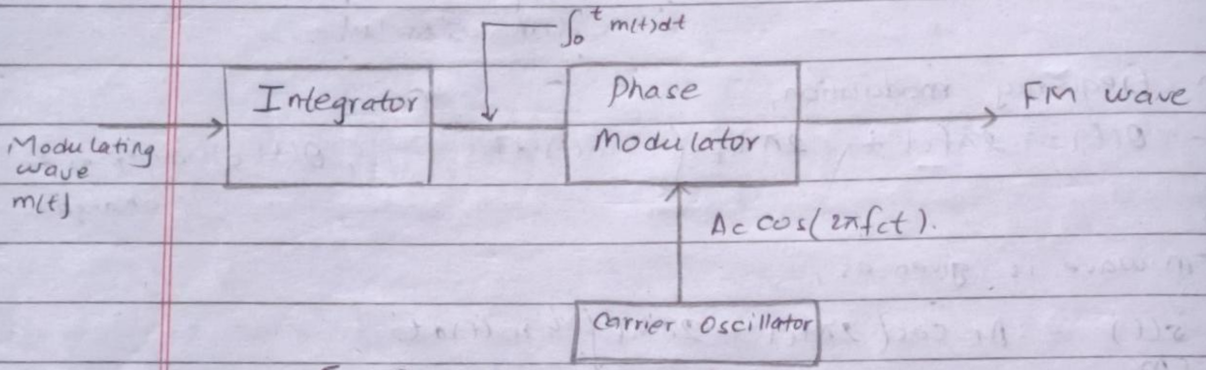

Fig: Generation of FM from phase modulator

Comparing these expressions, we can conclude that an FM wave is actually a PM wave having a modulating signal $\int_0^t m(t)\,dt$ instead of $m(t)$.

This means that we can generate FM wave by applying the integrated version of ~~m(t)~~ $m(t)$ to a phase modulator as shown above.

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Generation of PM using Frequency Modulator

The block diagram for generation of PM using FM is shown in figure below →

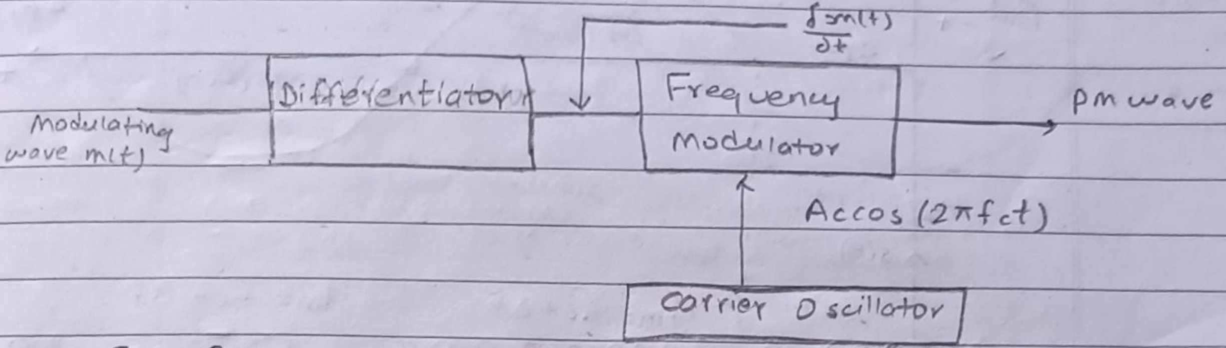


Fig : Generation of PM from frequency Modulator

It is also possible to generate a PM wave using a frequency modulator as shown in figure. The modulating signal is first passed through a differentiator and then applied to a frequency modulator.

Hence, output of FM will be,

$$s(t) = A_c \cos\left\{ 2\pi f_c t + 2\pi k_f \int_0^t \left(\frac{dm(t)}{dt} \right) \right\}$$

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

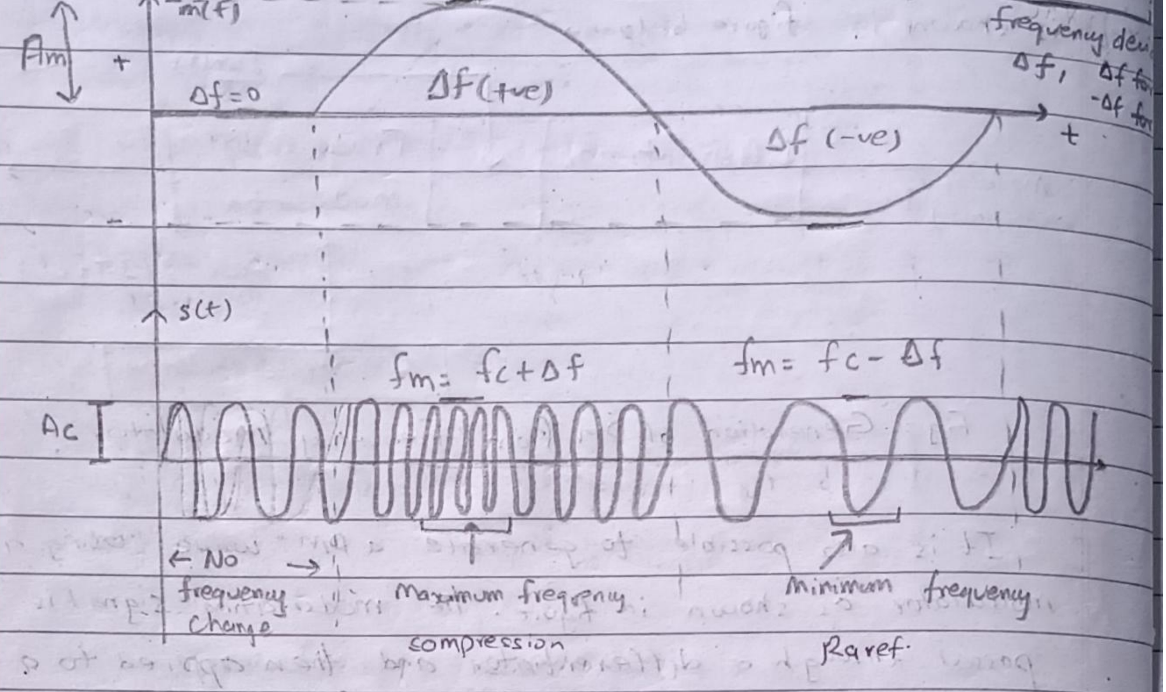
Substituting $2\pi K_f = K_p$, we get, (on comparing above eq^n to PM signal we get) $K_p = 2\pi K_f$.

$$S(t) = A_c \cos(2\pi f_c t + K_p m(t)).$$

which is the expression of PM wave.

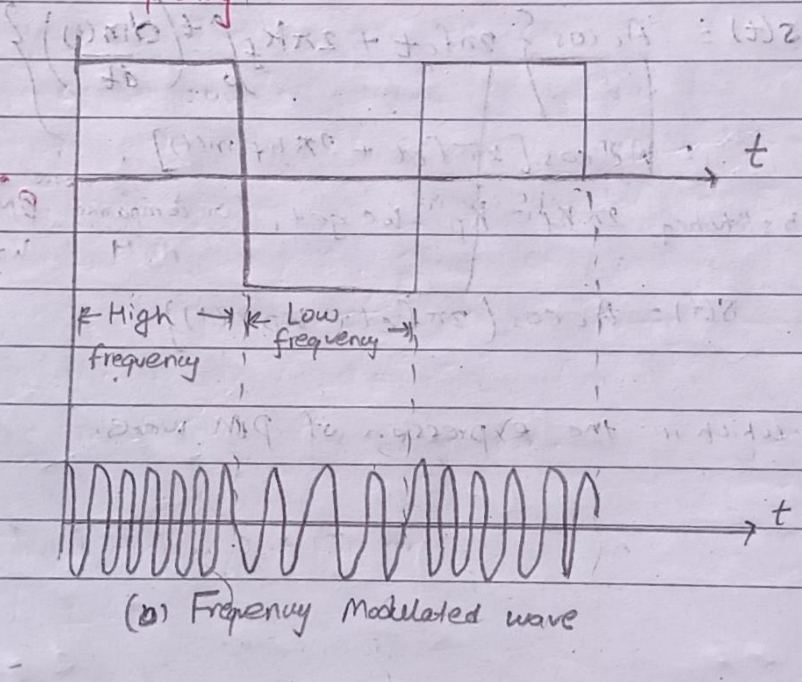
Sinusoidal Frequency Modulation →

$$s_{FM}(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t)\right)$$



(a) Sinusoidal Frequency Modulation

Squared Frequency Modulation →



(b) Frequency Modulated wave

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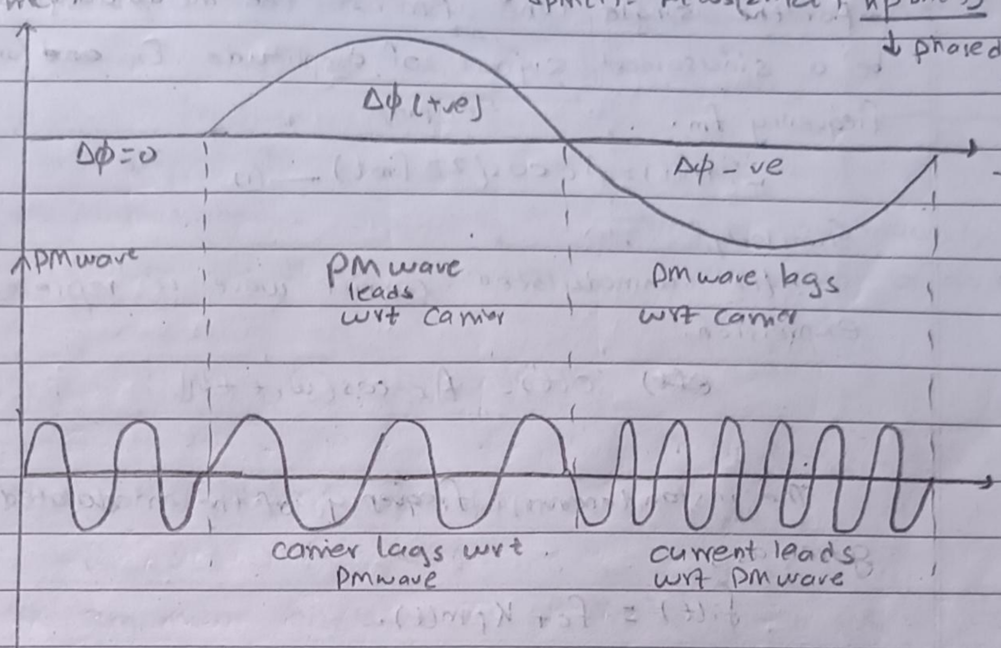
Sinusoidal Phase Modulation →

$S_{pm}(t) = A_c \cos[2\pi f_c t + \underbrace{K_p m(t)}]$

↓ phase deviation

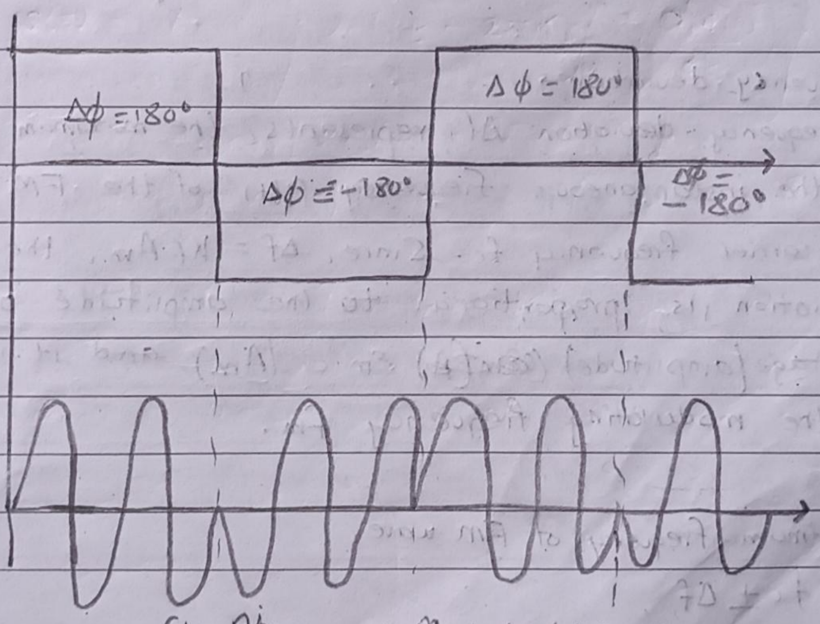
+ve → $\Delta\phi$ ⇒ +ve

−ve → $\Delta\phi$ ⇒ −ve



(a) Sinusoidal Phase Modulation

Squared Phase Modulation →



(b) Phase modulated wave

Single Tone Frequency Modulation →

For the single tone FM i.e. the modulating signal $m(t)$ be a sinusoidal signal of amplitude Em (or Am) and frequency fm.

$$\because m(t) = A_c \cos(2\pi f_m t) \quad \text{---}(1)$$

Similarly,

the unmodulated carrier wave is represented by the expression:

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

The instantaneous, frequency of the modulated wave is given as,

$$f_i(t) = f_c + K_f m(t).$$

$$f_i(t) = f_c + K_f A_m \cos(2\pi f_m t).$$

$$f_i(t) = f_c + \Delta f \cos 2\pi f_m t \quad \text{---}(2).$$

where, $\Delta f = K_f A_m$ = frequency deviation.

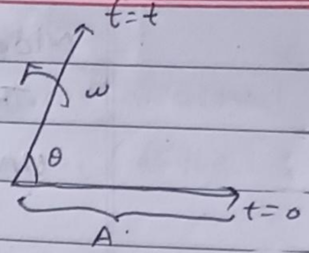
Frequency deviation →

Frequency deviation Δf represents the maximum departure of the instantaneous frequency $f_i(t)$ of the FM wave from the carrier frequency f_c. Since, $\Delta f = K_f A_m$, the frequency deviation is proportional to the amplitude of modulating voltage (amplitude) E_m or (A_m) and it is independent of the modulating frequency f_m.

Maximum frequency of FM wave,

$$f_{max} = f_c \pm \Delta f$$

Mathematical expression for FM



We know that the FM wave is sinewave having a constant amplitude and a variable instantaneous frequency.

As the instantaneous frequency is changing continuously, the angular velocity ω of an FM wave is the function of ω_c and ω_m.

Therefore, FM wave is represented by,

$s(t) = A_c \cos \theta t$

$s(t) = A_c \cos [f(\omega_c, \omega_m)]$

$\theta(t) \Rightarrow f(\omega_c, \omega_n)$.

Now, the argument angle $\theta(t)$ of FM is given as,

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

$\Delta f = K_f \cdot A_m$.

$$= 2\pi \int_0^t (f_c + \Delta f \cos 2\pi f_m t)\,dt$$

$$= 2\pi f_c t + \frac{2\pi \Delta f}{2\pi f_m} \cdot \sin 2\pi f_m t$$

$$= \underbrace{2\pi f_c t}_{\omega_c t} + m_f \cdot \sin \underbrace{2\pi f_m t}_{\omega_m t}$$

where, m_f = modulation index of FM.

$$\boxed{s(t) = A_c \cos [\omega_c t + m_f \sin \omega_m t]}$$

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Modulation Index of FM (m_f)

The modulation ~~ra~~ index of an FM wave is defined as under:

$$m_f = \frac{\text{frequency deviation}}{\text{modulating frequency}}$$

$$= \frac{\Delta f}{f_m}$$

The modulation index (m_f) is very important in FM because it decides the bandwidth of the FM wave. The modulation index also decides the number of sidebands having significant amplitudes.

In AM, the maximum value of the modulation index m is 1. But for FM, the modulation index can be greater than 1.

Deviation Ratio →

In FM broadcasting, the maximum value of deviation is limited to 75kHz. The maximum modulating frequency is also limited to 15kHz. The modulation index corresponding to the maximum deviation and maximum modulating frequency is called as the deviation ratio.

$$\text{Deviation Ratio} = \frac{\text{Maximum deviation}}{\text{Maximum modulating frequency}}$$

Percentage modulation of FM wave →

The percent modulation is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

Thus, $\% \text{ modulation} = \dfrac{\text{Actual frequency deviation}}{\text{Maximum allowed deviation}}$.

Narrow Band Frequency Modulation and Wide band Frequency Modulation.

[A] Narrow band Frequency Modulation →

A narrow band FM is the FM wave with a small bandwidth. The modulation index m_f of narrowband FM is small, as ~~compared~~ compared to one radian. Hence the spectrum of narrow band FM consists of the carrier and upper side-band and a lower sideband.

The carrier wave $c(t)$ is given as,

$c(t) = A_c \cos(\omega_c t + \phi)$, $m(t) = A_c \cos(2\pi f_m t)$.

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

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

$f_i(t) = f_c + K_f * A_m \cos(2\pi f_m t)$

$f_i(t) = f_c + \Delta f \cos 2\pi f_m t$

$s(t) = A_c \cos \theta(t)$.

on integrating $f_i(t)$ we get $\theta(t)$.

$$\theta(t) = \int f_i(t)dt = \int f_c dt + \int \Delta f \cos 2\pi f_m t$$

$$\Rightarrow \int 2\pi f_i(t)dt = \int_0^t 2\pi f_c t \, dt + \int_0^t 2\pi \Delta f \cos 2\pi f_m t \, dt$$

$$\theta(t) \Rightarrow 2\pi f_c t + \frac{2\pi \Delta f}{2\pi f_m} \sin 2\pi f_m t$$

$$= 2\pi f_c t + \frac{\Delta f}{f_m} \sin 2\pi f_m t$$

$$\therefore s(t) = A_c \cos\left(\underbrace{\omega_c t}_{A} + \underbrace{m_f \sin \omega_m t}_{B}\right)$$

As compare, $\cos(A+B) = \cos A \cos B - \sin A \sin B$

$$s(t) = A_c[\cos \omega_c t \cdot \cos(m_f \sin \omega_m t) - \sin \omega_c t \cdot \sin(m_f \sin \omega_m t)]$$

If modulation index m_f is very small then,

$\{\cos(m_f \sin \omega_m t) \approx 1\}$

$\left\{\begin{array}{l}\sin(m_f \sin \omega_m t) \approx m_f \sin \omega_m t \\ \sin(\alpha) \Rightarrow \alpha\end{array}\right\}$

$$\therefore s(t) = A_c \cos \omega_c t - A_c \sin \omega_c t \cdot m_f \cdot \sin \omega_m t$$

$$= A_c \cos \omega_c t - A_c m_f \sin \omega_c t \cdot \sin \omega_m t$$

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

$$\boxed{\therefore s(t) = \underbrace{A_c \cos \omega_c t}_{\text{carrier}} + \underbrace{\frac{A_c m_f}{2} \cos(\omega_c + \omega_m)t}_{\text{upper side band}} - \underbrace{\frac{A m m_f}{2} \cos(\omega_c - \omega_m)}_{\text{lower side band}}}$$

A_c $\quad$ $\frac{A_c m_f}{2}$ $\quad$ $\frac{A m m_f}{2}$

$\therefore$ This is the required expression of NBFM.

Eular relation;
$(e^{\pm j\theta} = \cos \pm j \sin\theta)$

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[B] Wide Band frequency Modulation →

- For large value of modulation index m_f, the FM wave ideally contains the carrier and an infinite number of sidebands located symmetrically around the carrier. Such a FM wave has infinite bandwidth and hence called as wide band FM.
- The modulation index of wideband FM is higher than 1. The maximum permissible deviation is 75kHz and it is used in the entertainment broadcasting applications such as FM radio, TV, etc.

- frequency spectrum of WBFM, (Bessels coefficient) →

The expression for the FM wave is not simple. It is complex since it is sine of sine function.

The FM wave for sinusoidal signal tone modulation is given as,

$$s(t)_{FM} = A_c \cos(2\pi f_c t + m_f \sin \omega_m t). \quad —(1)$$

Let, $m_f \Rightarrow$ modulation inden of FM wave,
$\Rightarrow \beta$.

$$s(t)_{FM} = A_c \cos(2\pi f_c t + \beta \sin \omega_m t) — (1).$$

This equation can be expressed as trigonometric relation,

$$S_{FM}(t) = A_c \cos(2\pi f_c t) \cdot \cos(\beta \sin 2\pi f_m t) - A_c \sin(2\pi f_c t) \cdot \sin(\beta \sin 2\pi f_m t) \quad —(2)$$

From this expanded form we can see the Inphase and Quadrature components of FM wave $s(t)$ is given as,

$$S_I(t) = A_c \cos[\beta \sin(2\pi f_m t)] — (3)$$

$$S_Q(t) = A_c \cos[\beta \sin(2\pi f_m t)] — (4)$$

Hence, the complex envelope of FM wave is given as,

$$\tilde{s}(t) = s_I(t) + j\, s_Q(t)$$
$$= A_c \cos[\beta \sin(2\pi f_m t)] + j A_c \sin(\beta \sin(2\pi f_m t)].$$
$$= A_c e^{j\beta \sin(2\pi f_m t)} \quad \text{(Euler's relation)}$$
$$= A_c \exp(j\beta \sin(2\pi f_m t)) \quad \text{---(5)}$$

The complex envelope $\tilde{s}(t)$ retains all the information related to modulated process indeed we may radily express FM wave $s(t)$ interms of complex envelope $\tilde{s}(t)$.

$$s(t) = \mathrm{Re}[A_c \exp(j2\pi f_c t + j\beta \sin(2\pi f_m t))].$$
$$= \mathrm{Re}[\tilde{s}(t) \cdot \exp(j 2\pi f_m t)] \quad \text{---(4)}$$

The expression of $\tilde{s}(t)$ in complex fourier series as,

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} C_n \exp(j\, 2\pi f_m t) \quad \text{---(6)}$$

where,

$$C_n = \frac{1}{T_m} \int_{-T_m/2}^{T_m/2} A_c \exp(-j2\pi f_m t) \cdot \underbrace{\exp(+j\beta \sin(2\pi f_m t))}_{\tilde{s}(t)} \cdot dt$$

$$= A_c f_m \int_{-1/2f_m}^{+1/2f_m} \exp[(-j2\pi f_m t) + j\beta \sin(2\pi f_m t)] \, dt \quad \text{---(7)}$$

Let us assume,

$$x = 2\pi f_m t$$
$$\frac{dx}{dt} = 2\pi f_m \cdot \frac{d(t)}{dt}$$
$$dx = 2\pi f_m \, dt.$$
$$\therefore dt = dx/2\pi f_m. \quad \text{---(8)}$$

Note, properties of Bessel function,

1. $J_{-n}(\beta) = \begin{cases} J_n(\beta); & n = \text{even} \\ -J_n(\beta); & n = \text{odd} \end{cases}$

2. $J_{n-1}(\beta) + J_{n-1}(\beta) = \frac{2n}{\beta} J_n(\beta)$.

3. $\sum_{n=-\infty}^{\infty} J_n^2(\beta) = 1$ for all β.

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when, $t = -\frac{1}{2} f_m$ then $x = -\pi$

when, $t = +\frac{1}{2} f_m$ then $x = +\pi$

[$\because x$ ranges from $-\pi$ to $+\pi$]

Substuting the value of dt in eqn (7) we get,

$$C_n = \frac{A_c}{2\pi} \int_{-\pi}^{\pi} \exp(j\beta \sin x - jnx)\, dx$$

$$= \frac{1}{2\pi} A_c \int_{-\pi}^{\pi} \exp(j\beta \sin x - jnx)\, dx$$

$$= A_c \left[\frac{1}{2\pi} \int_{-\pi}^{\pi} \exp(j(\beta \sin x - nx))\right] dx \quad \text{——— (9)}$$

(Bessel's function)

the Integral on right hand side is recognized as the, nth order Bessel function of the first kind and argument β. and it is denoted by symbol $J_n(\beta)$.

$$J_n(\beta) = \frac{1}{2\pi} \int_{-\pi}^{\pi} \exp(j(\beta \sin x - nx))\, dx$$

$$C_n = A_c \cdot J_n(\beta) \quad \text{——— (10)}$$

Hence, this is the required expression of C_n in Bessel's form.

Also we have,

$$e^{j\beta \sin 2\pi f_m t} = \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j\, 2\pi f_m t) \quad \text{——— (11)}$$

From eqn (4) and (11) we get,

$$s(t)_{FM} = Re\left[A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j \cdot 2\pi f_m t)\right]$$

$$= A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[2\pi(f_c + n f_m)t].$$

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Performance comparison of wide band and Narrow band FM

S.N.	Parameter/Characteristics	Wideband FM	Narrowband FM
1.	Modulation Index	Greater than 1	Less than or slightly greater than 1
2.	Maximum deviation	75 KHz	5 KHz
3.	Range of modulating frequency.	30 Hz to 15 KHz	30 Hz to 3 KHz
4.	Maximum modulation index	5 to 2500	Slightly greater than 1.
5.	Bandwidth	Large about 15 times higher than BW of narrowband FM.	Small, Approximately same as that of AM
6.	Applications	Entertainment broad-casting (can be used for high quality music transmission)	FM mobile communication like police wireless, ambulance etc. (This is used for speech transmission).
7.	Pre-emphasis and De-emphasis	Needed	Needed

Side bands and Modulation Index :- (Relation)

We know that any modulation process produces sidebands. We have seen that in AM, two sidebands are produced with frequencies equal to $f_c + f_m$ and $f_c - f_m$.

In FM and PM, also, sum and difference sideband frequencies are produced.

Note: The number of side bands with significant amplitude increase with increase m_f.

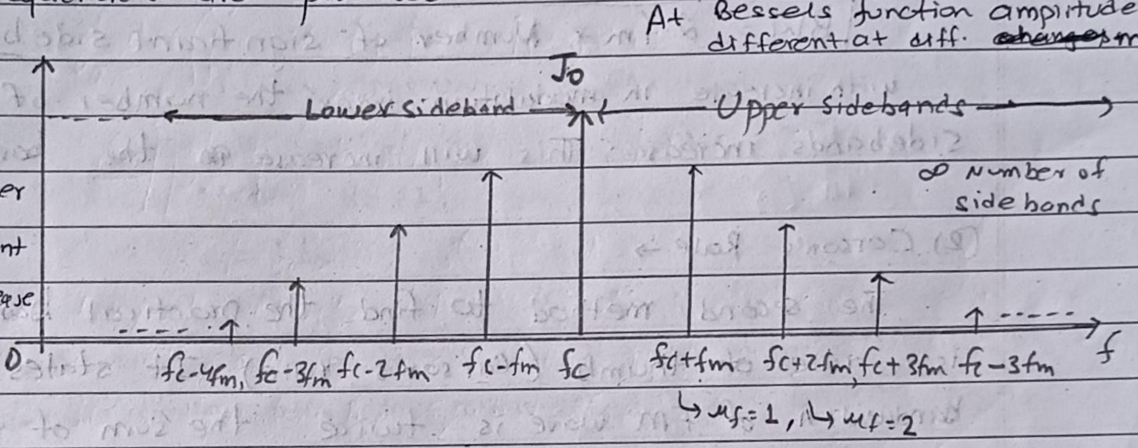

Figure: Ideal frequency spectrum of FM,

In addition theoretically infinite number of pairs of upper and lower sidebands are also generated. Hence, the spectrum of FM or PM signal is generally wider than the spectrum of AM.

Effect of Modulation Index →

As the amplitude of modulating signal varies, the frequency deviation will change, the number of sidebands produced, their amplitudes will change. above figure, illustrates the effects of modulation index on the frequency spectrum of FM. Theoretically, the bandwidth of FM is infinite.

Transmission bandwidth of frequency modulation

(1) Practical Bandwidth →

Theoretically, the bandwidth of the FM wave is infinite. But, practically, it is calculated based on how many sidebands have significant amplitude. The simplest method to calculate the bandwidth is as under:

$$\boxed{BW = 2 f_m \times \text{Number of significant sidebands.}}$$

With increase in modulation index, the number of significant sidebands increase. This will increase the bandwidth.

(2) Carson's Rule →

The second method to find the practical bandwidth is a rule of thumb (Carson's rule). It states that the bandwidth of FM wave is twice the sum of the deviation and the highest modulating frequency.

Thus,

$$BW = 2[\Delta f + f_{m(max)}]$$

The Carson's Rule gives correct results if the modulation index is greater than 6.

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Constant Average Power of frequency modulation →

we know that the envelope of FM wave always has a constant magnitude. Therefore, the average power of such a wave dissipated in 1 Ω resistance will always be constant. The transmitted power in FM is given as,

$$P_t = \frac{(E_c/\sqrt{2})^2}{R_L} \quad [\text{Since } E_c = A_c]$$

$(E_c/\sqrt{2}) \Rightarrow$ RMS value of FM wave,

Hence, $R_L = 1$,

$$P_t = \frac{1}{2} E_c^2$$

Since, E_c is constant, P_t also will be constant. It is possible to express the transmitted power in the form of series expansion as under,

$$P_t = \frac{1}{2} E_c^2 \sum_{n=-\infty}^{\infty} J_n^2(m_f).$$

But, $\sum_{n=-\infty}^{\infty} J_n^2(m_f) = 1$. $\left\{ \begin{array}{l} \text{for } m_f = 1, 2, --- \infty, \\ P_t = 1, 2, -- \infty \end{array} \right\}$.

$$\therefore P_t = \frac{1}{2} E_c^2 = \frac{1}{2} A_c^2.$$

Transmission power in FM wave →

$$P_T = A_c^2/2$$

practically 98% power is transmitted.

$$\text{So, } P_T = 0.98 * \frac{A_c^2}{2}.$$

Multiple- frequency Modulation -

we have already derived the expression for FM wave with a single modulating input signal. Let us now extend this concept to obtain the expression of FM for more than one modulating signals.

Let $m_1(t)$ and $m_2(t)$ be two modulating signals and let,

$$m(t) = m_1(t) + m_2(t)$$

Let, $m_1(t) = A_{m1} \cos \omega_{m1} t. \quad = E_{m1} \cos\omega_{m1} t.$

$m_2(t) = A_{m2} \cos\omega_{m2} t. \quad = E_{m2} \cos\omega_{m2} t$

Therefore,

$$m(t) = A_{m1} \cos \omega_{m1} t + A_{m2} \cos\omega_{m2} t.$$

The instantaneous frequency is given by,

$$\omega_i = \omega_c + K_f m(t)$$

$$= \omega_c + K_f [E_{m1} \cos\omega_{m1} t + E_{m2} \cos\omega_{m2} t]$$

The maximum frequency deviation is given by,

$$\Delta\omega = (E_{m1} + E_{m2}) K_f$$

$$\theta(t) = \int \omega_i \, dt = \omega_c + \frac{E_{m1} K_f}{\omega_{m1}} \sin\omega_{m1} t + \frac{E_{m2} K_f}{\omega_{m2}} \sin\omega_{m2} t$$

Let, $\frac{E_{m1} K_f}{\omega_{m1}} = m_{f1}$ and $\frac{E_{m2} K_f}{\omega_{m2}} = m_{f2}$.

$$\therefore \theta(t) = \omega_c + m_{f1} \sin\omega_{m1} + m_{f2} \sin\omega_{m2} t$$

Hence, the expression for FM wave is given by,

$$s(t) = E_c \sin \theta(t)$$

$$s(t) = E_c \sin [\omega_c t + m_{f1} \sin n\omega_{m1} t + m_{f2} \sin n\omega_{m2} t]$$

Spectrum of this FM wave contains sidebands of frequencies $(f_c \pm nf_{m1})$ and $(f_c \pm nf_{m2})$.

In addition to that, there will be cross modulation terms such as $(\omega_c \pm n\omega_{m1} \pm k\omega_{m2})$.

In AM, each new frequency added gives rise to its own pair of side bands only. There are no cross modulation terms. Hence, AM is called as linear modulation while FM is called as a non-linear modulation.

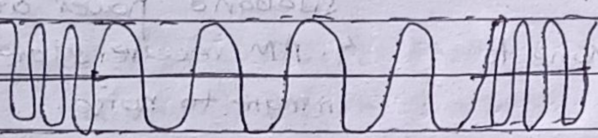
Difference/ Comparison of FM and PM ~~signals~~ systems →

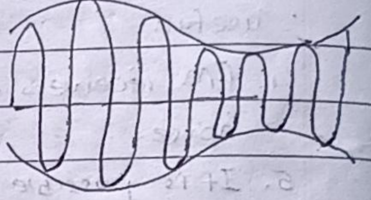
FM	PM
1. $S(t) = V_c \sin[\omega_c t + m_f \sin\omega_m t]$	1. $s(t) = V_c \sin[\omega_c t + m_p \sin\omega_m t]$
2. Frequency deviation is proportional to modulating voltage.	2. Phase deviation is proportional to the modulating voltage.
3. Associated with the change in f_c, there is some phase change.	3. Associated with the changes in phase, there is some change in f_c.
4. m_f is proportional to the modulating voltage as well as the modulating frequency f_m.	4. m_p is proportional only to the modulating voltage.
5. It is possible to receive FM on a PM receiver.	5. It is possible to receive PM on FM receiver.
6. Noise immunity is better than AM and PM.	6. Noise immunity is better than AM but worse than FM.
7. Amplitude of the FM wave is constant.	7. Amplitude of the PM wave is constant.
8. Signal to noise ratio is better than that of PM.	8. Signal to noise ratio is inferior to that of FM.
9. FM is widely used.	9. PM is used in some mobi systems.
10. In FM, the frequency deviation is proportional to the modulating voltage only.	10. In PM, the frequency devi is proportional to both the modulating voltage and modulating frequency.

Performance Comparison of FM and AM System →

S.N.	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.	Hence, transmitted power remains constant. It is independent of m_f.	Transmitted power is dependent on the modulation index.
3.	All the transmitted power is useful.	Carrier power and one sideband power are useless.
4.	FM receivers are immune to noise.	AM receivers are not immune to noise.
5.	It is possible to decrease noise further by increasing deviation.	This feature is absent in AM.
6.	Bandwidth $= 2[\Delta f + f_m]$. The BW depends on modulation index.	$BW = 2f_m$. It is not dependent on the modulation index.
7.	BW is large. Hence, wide channel is required.	BW is much less than FM.
8.	Space wave is used for propagation. So, radius of transmission is limited to line of sight.	Ground wave and sky wave propagation is used. Therefore, larger area is covered than FM.
9.	Hence, it is possible to operate several transmitters on same frequency.	Not possible to operate more channels on same frequency.

10. FM transmission and reception equipment are more complex.	10. AM equivalents are less complex.
11. The number of sidebands having significant amplitudes depends on modulation index m_f.	11. Number of sidebands in AM will be constant & equal to 2.
12. The information is contained in the frequency variation of the carrier.	12. The information is contained in the amplitude variation of the carrier.
13. FM wave:	13. AM wave.





14. Applications:
Radio, TV broadcasting, police wireless, point to point communications.

14. Applications:
Radio and TV broadcasting

Advantages of FM over AM*

(1) FM receivers may be fitted with amplitude limiters to remove the amplitude variations caused by noise.

(2) It is possible to reduce noise still further by increasing the frequency - deviation.

(3) Standard frequency allocation's provide a guard band between commercial FM stations. due to this, there is less adjacent-channel interference than in AM.

(4) FM broadcasts operate in the upper VHF and UHF frequency ranges at which there happens to be less noise than in the MF and HF ranges occupied by AM broadcasts.

(5) The amplitude of FM wave is constant.
In FM all the transmitted power is useful.

Disadvantages of FM over AM:-

(i) A much wider channel typically 200 KHz is required in FM as against only 10KHz in AM broadcast. This forms serious limitation of FM.

(ii) FM transmitting and receiving equipments particularly used for modulation and demodulation tend to be more complex and hence more costly.

Generation of FM wave →

Methods of FM Generation

- Direct Methods
  - Reactance modulator
  - Varactor diode modulator
- Indirect Methods
  - Armstrong method

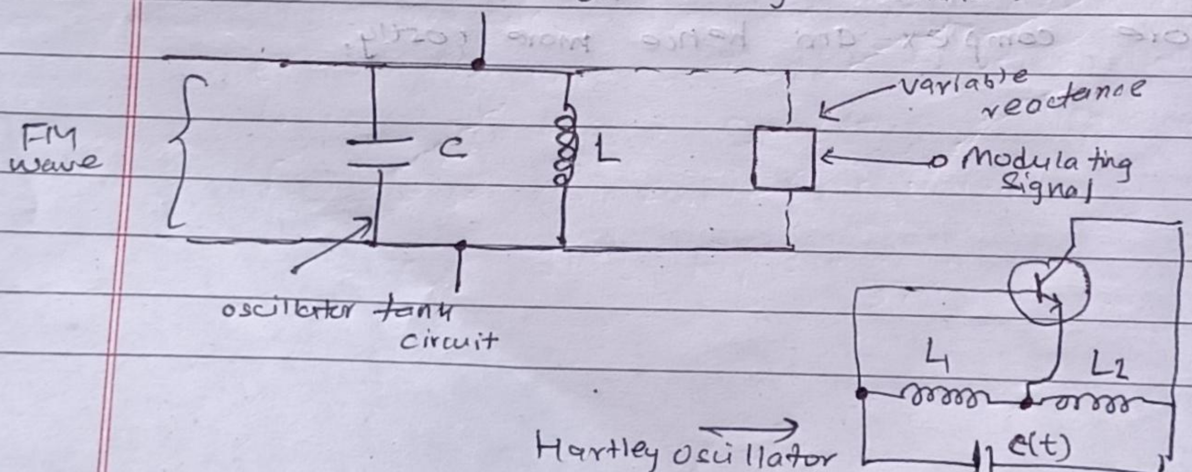
(i) the direct method or parameter variation method.
(ii) the Indirect method or the Armstrong method.

(I) Direct Method or Parameter Variation Method →

In direct method or parameter variation method, the baseband or modulating signal directly modulates the carrier. The carrier signal is generated with the help of an oscillator circuit. This oscillator circuit uses a parallel tuned L-C circuit. Thus the frequency of oscillation of the carrier generation is governed by the expression.

$$\omega_c = \frac{1}{\sqrt{LC}}$$

(1) Reactance Modulator or Hartley Oscillator →



In direct FM generation, the instantaneous frequency of the carrier is changed directly in proportion with the message signal. For this, a device called voltage controlled oscillator (VCO) is used.

A VCO can be implemented by using a sinusoidal oscillator with a tuned circuit having a high value of Q.

The frequency of this oscillator is changed by incremented variation in the reactive components involved in the tuned circuit. If L or C of a tuned circuit of an oscillator is changed in accordance with the amplitude of modulating signal then FM can be obtained across the tuned circuit as shown above.

A two or three terminal device is placed across the tuned circuit. The reactance of the device is varied proportional to modulating signal voltage. This will vary the frequency of the oscillator to produce FM. The device used are FET, transistor or varactor diode.

An example of diode FM is shown above figure (b) which uses a Hartley oscillator along with a varactor diode. The varactor diode is reverse biased.

Its capacitance is dependent on the reverse voltage applied across it. This capacitance is shown by the ~~capacitor~~ capacitor C(t) in figure (b).

Frequency of oscillator of the Hartley Oscillator is given as,

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)C(t)}} \quad —(1)$$

where, C(t) = fixed capacitor shunted by voltage variable capacitor.

$= C + C_{varactor}$

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Let the relation between the modulating voltage $m(t) = 0$. and the capacitance $C(t)$ be represented as under:

$$C(t) = C - K_C m(t) \quad \text{---} (2).$$

where, C = total capacitance when $m(t)$ and K_C is the sensitivity of the varactor capacitance to change in voltage.

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

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

$$= \frac{1}{2\pi\sqrt{(L_1+L_2)C}\left[1 - \frac{K_C m(t)}{C}\right]^{1/2}}$$

But let, $\frac{1}{2\pi\sqrt{(L_1+L_2)C}} = f_0$, which is the oscillator frequency in absence of the modulating signal $[m(t) = 0]$.

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

If the maximum change in the capacitance corresponding to the modulating wave is assumed to be small as compared to the unmodulated capacitance C then above eq^n for, $f_i(t)$ can be approximated as under:

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

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$$f_i(t) = f_0 + \frac{f_0 k_c}{2C} m(t).$$

Now, let us define,

$$\frac{f_0 k_c}{2C} = k_f.$$

Therefore we have,

$$\boxed{f_i(t) = f_0 + k_f m(t).}$$

where, k_f is called as the frequency sensitivity of the modulator.

(d) Varactor diode Method →

the schematic diagram of varactor diode method is shown below →

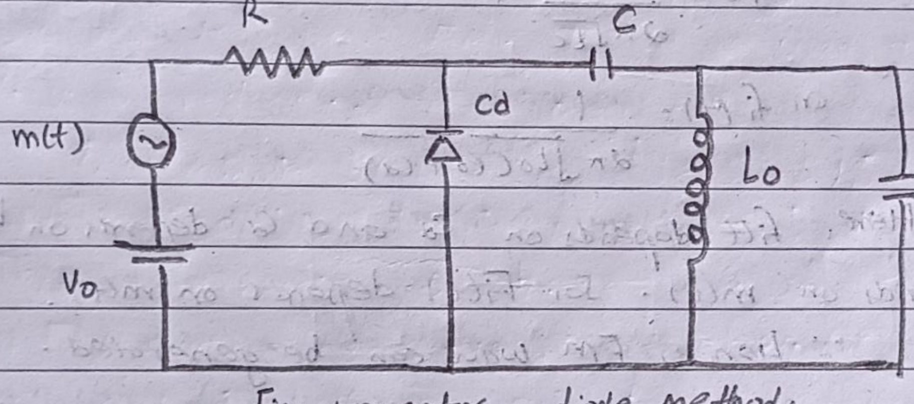

Fig: varacator diode method.

- In varacator diode, capacitance (Cd) changes with DC. bias, voltage. In this method, capacitor C is made much small than varactor diode capacitance Cd, so that RF voltage from oscillator across diode is small as compared to reverse bias dc voltage.

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- In addition reactance of capacitor 'c' at highest modulating frequency is made large enough compare to resistor 'R'.
- So that shunting baseband signal $m(t)$, through tuned circuit (L_0C_0) may be checked. The capacitance of C_d of varactor diode is,

$$C_d = \frac{k}{\sqrt{V_d}} = K[V_d]^{-1/2} \quad (1)$$

Here, V_d is Instantaneous voltage across diode and is given as,

$$V_d = V_0 + m(t) \quad (2)$$

The oscillator frequency is given as,

$$f_c = \frac{1}{2\pi\sqrt{LC}}$$

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

Here, $f_i(t)$ depends on C_d and C_d depends on V_d & V_d depends on $m(t)$. So, $F_i(t)$ depends on $m(t)$.

∴ Hence, FM wave can be generated.

[2] Indirect Method →

- FM generator or Armstrong modulation called as Indirect method.

- In this method, the modulating wave / message signal is used to,

(a) generate the NBFM wave.

(b) frequency multiplication is performed to increase the frequency deviation to the desired level WBFM.

- Now,

(1) To generate NBFM, first we begin with FM wave $s(t)$, For general case of modulating wave $m(t)$.

$$s_1(t) = A_1 \cos(2\pi f_1 t + \phi_1(t)] \text{ --- (1)}.$$

where,

f_1 = carrier frequency

A_1 = amplitude

$$\phi_1 = 2\pi k_1 \int_0^t m(t)\,dt \text{ --- (2)}.$$

Assume, provided that $\phi_1(t)$ is small compared to one radiation for all t;

$\phi(t) << 1$ for all t;

$\cos[\phi(t)] = 1$ --- (3)

$\sin[\phi(t)] = \phi[(t)]$ --- (4).

Now, equation (1) becomes,

$$s_1(t) = A_1 \cos 2\pi f_1 t * \cos \phi_1(t) - A_1 \sin 2\pi f_m t * \sin \phi_1(t)$$

$$= A_1 \cos 2\pi f_1 t - A_1 \sin 2\pi f_m t \cdot \phi_1(t).$$

$= A_1 \cos 2\pi f_1 t - A_1 \sin 2\pi f_1 t \times k_1 2\pi \int_0^t m(t)\,dt \rightarrow (5)$

$\therefore s(t) = A_1 \cos 2\pi f_1 t - A_1 \sin 2\pi f_1 t \cdot k_1 2\pi \int_0^t m(t)\,dt.$

the schematic diagram is shown in figure below →

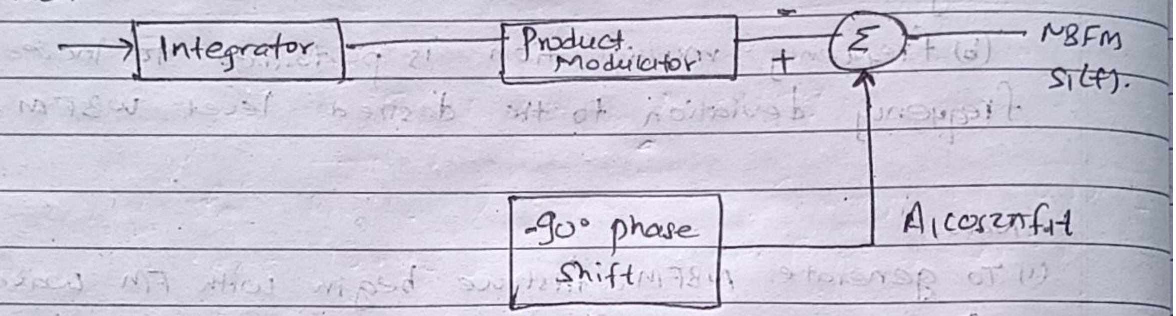
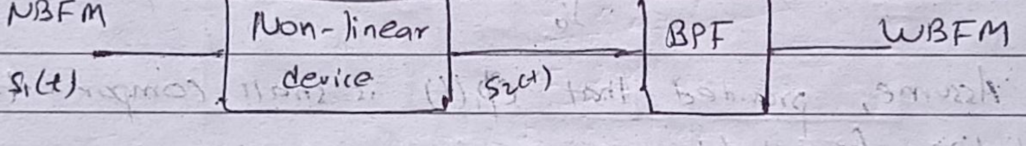


Fig: NBFM modulator.

(2) Now, the next step is to perform frequency multiplication. It consists of non-linear device followed by Bandpass filter.



- the memoryless non-linear device is represented as,

$s_2(t) = a_1 s_1(t) + a_2 s_1^2(t) + a_3 s_1^3(t) + \cdots a_n s_1^n(t)$

where,

$a_1, a_2, \cdots, a_n$ are constant coefficiants, Substuting the value of $s_1(t)$ in equation above then expanding and collecting required term, we find that o/p $s_2(t)$ has dc component and 'n' frequency modulated wave with carrier frequency,

$f_1, 2f_1, 3f_1, ---- nf_1$

and frequency deviation, $\Delta f_1, 2\Delta f_1, 3\Delta f_1, ---n\Delta f_1$

The value of f_1 is determined by frequency sensitivity K_1 of NBFM and maximum amplitude of $m(t)$.

the BPF is used to design;

(1) to pass FM wave centered of carrier frequency (nf_1). and with frequency deviation $n\Delta f_1$!

(2) to supress all other FM spectra.

the schematic diagram of WBFM is shown in figure below→

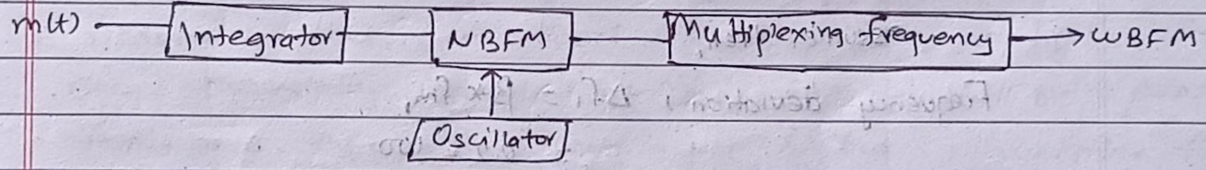


fig: WBFM.

Note:-

(Example of Armstrong Modulator)→

the schematic diagram of Armstrong Modulation is shown in figure below→

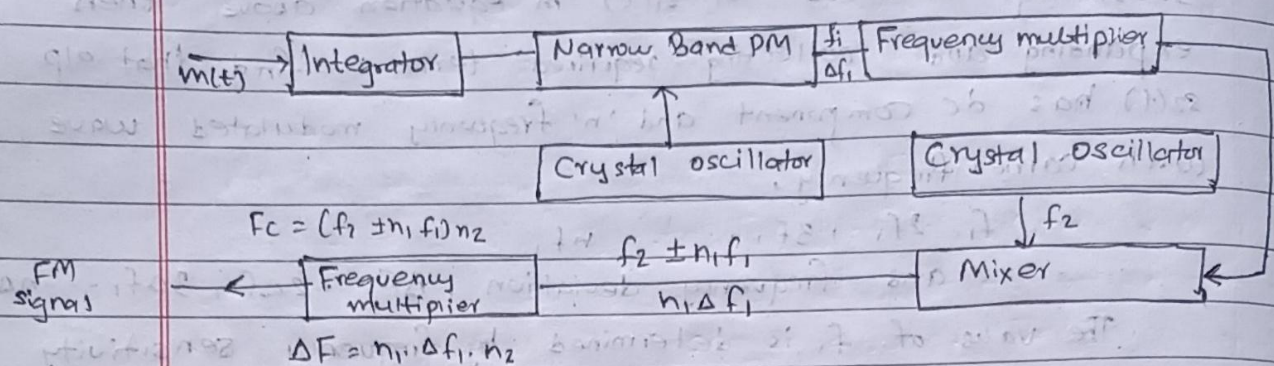


Fig: Armstrong modulator.

Let $f_{m_1} = 100\,Hz$, $f_{m_2} = 15\,KHz$ } assume NB frequency $(f_1) = 0.1\,MHz$, Carrier frequency $(f_2) = 9.5\,MHz$ }

Frequency deviation ; $\Delta f = 75\,KHz$ (actual) | $F_c = 100\,MHz$

To limit the distortion $\beta \leq 0.3$, so, take $\beta = 0.2$ } assume, then,

Frequency deviation, $\Delta f_1 = \beta * f_{m_1}$
$= 0.2 * 100$
$= 20\,Hz$

$\Delta f_2 = \beta * f_{m_2}$
$= 0.2 * 15\,KHz = 3\,KHz$

Now,

Frequency multiplier $= \frac{\Delta f}{\Delta f_1} = \frac{75 * 10^3}{20} = 3750$

$\therefore n = n_1 n_2 = 3750$ —(1)

Also,

$f_c = (f_a \pm n_1 f_1) n_2$

$100 = (9.5 \pm 0.1 n_1) \times 2$

$\therefore 9.5 - 0.1 n_1 = 50$ — (2)

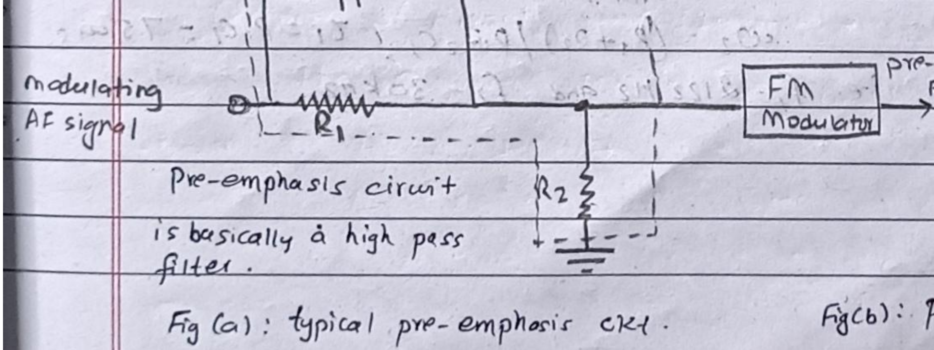
From eqn (1) and (2) we get,

$n_1 = 75$ and $n_2 = 50$.

Pre-Emphasis and de-Emphasis –

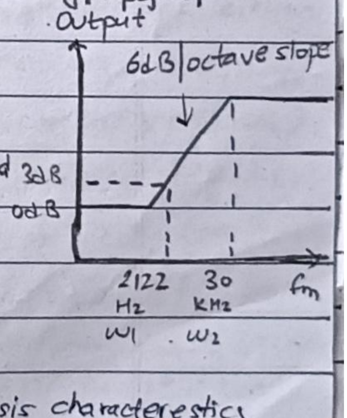
Pre-Emphasis –

It has been proved that in FM, the noise has a greater effect on the higher modulating frequencies. this effect can be reduced by increasing the value of modulation index, for higher modulating frequencies (f_m). this can be done by increasing the deviation Δf and Δf can be increased by increasing the amplitude of modulating signal at higher modulating frequencies. thus, if we boost the amplitude of higher frequency modulating signals artifically then it will be possible to improve the noise immunity at higher modulating frequencies. The artificial boosting of higher modulating frequencies is called pre-Emphasis.



Pre-emphasis circuit is basically a high pass filter.

Fig (a): typical pre-emphasis ckt.



Fig(b): Pre-emphasis characterestics

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the modulating AF signal is passed through a high pass RC filter, before applying it to the FM modulator.

As fm increases, reactance of C decreases and modulating voltage applied to FM modulator goes on increasing. The frequency response characterestics of the RC high pass network is shown above. The boosting is done according to this pre arranged curve. the amount of pre-emphasis in US FM transmission and sound transmission in TV has been standardized at 75 μsec.

- The pre-emphasis circuit is basically a high pass filter. The pre-emphasis is carried out at the transmitter. The frequency for the RC high-pass network is 2122 Hz Hence, the pre-emphasis circuit is used at the transmitter.

Here, Mathematically,

$$H_p(j\omega) = \frac{R_2}{R_1+R_2}\left(\frac{1+j\omega/\omega_1}{1+j\omega/\omega_2}\right)$$

where, $\omega_1 = 1/R_1C_1$

$\omega_2 = (R_1+R_2)/R_1R_2C_1$, $\tau_1 = R_1C_1 = 75\,\mu s$.

i.e. $f_1 = 2122$ Hz and $f_2 = 30$ kHz.

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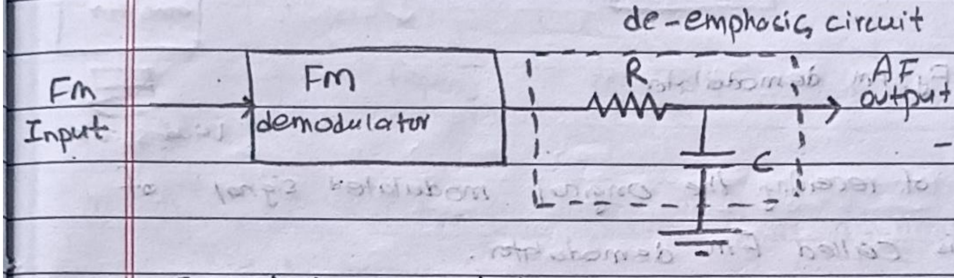
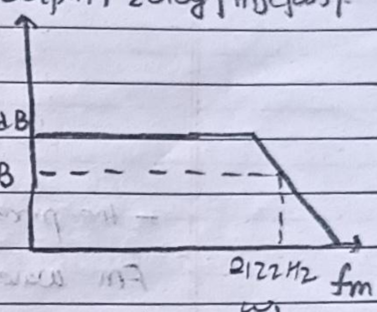
(2) De- Emphasis →

The artificial boosting given to the higher modulating frequencies in the process of pre-emphasis is nullified or compensated at the receiver by a process called De-emphasis. the artifically boosted high frequency signals are brought to their original amplitude using the de-emphasis circuit. The 75μsec. de-emphasis circuit is standard.



Fig(a): typical de- emphasis



Fig(b): de-emphasis characterestics.

It shows that it is a low pass filter. 75 sec de-emphasis corresponds to a frequency response curve that is 3dB down at a frequency whose RC time constant is 75 μsec.

- the demodulated FM is applied to the De-emphasis circuit which increase in fm the reactance of C goes on decreasing and the output of de-emphasis circuit will also reduce.

Here, mathematically,

$$H_D(j\omega) = 1/1 + j\omega/\omega_1$$

where,

$$\omega_1 = 1/RC \qquad \text{i.e. } f = \frac{1}{2\pi RC} = \frac{1}{2\pi \times 75 \times 10^{-6}}$$

$$\tau = RC = 75\,\mu sec. \qquad = 2122\ Hz.$$

FM demodulator →

The block diagram of Fm demodulator is shown in figure below ↓

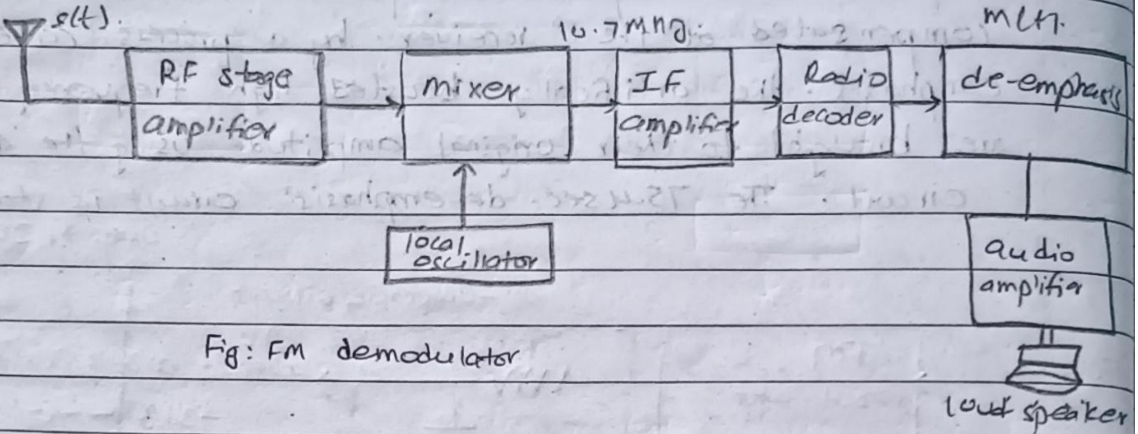


Fig: FM demodulator

– the process of receiving the original modulated signal of Fm wave is called Fm demodulator.

– The RF amplifies the received Fm modulated wave; the amplitude signal is applied to mixer. the second input to mixer comes from local oscillator. The two input frequenci of the mixer generates fixed if signal to 10.7MHz this signal is amplified by IF amplifier.

Radio detector is used to demodulate the signal.

The De-emphasis is performed which block the higher frequency to bring them back to original amplitude. Then audio amplifier amplifies o/p message and fed to loud speaker.

Advantages:-

(1) No variation in Bandwidth

(2) High sensitivity

(3) High selectivity

FM Demodulator

- Direct type
  - Frequency discriminators
  - Zero crossing detector
- Indirect type
  - Phase locked loop

(1) Demodulator of FM wave using Linear discriminator method →
(Non-synchronous method).

In this method, the extraction of message signal Involves →

(a) the demodulation circuit that connects FM signal to AM signal with the help of frequency discriminator.

(b) The original message signal is achieved by using envelope detector.

The FM signal is,

$$S_{FM}(t) = A_c \cos\left\{2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right\} \quad \text{---(1)}.$$

the extraction of $m(t)$ from eqn (1) can be done in 3-step.

(a) amplitude limit

(b) discriminator + differenter

(c) Envelope detector

The block diagram for demodulation of FM wave using Limiter discriminator method is shown in figure below →

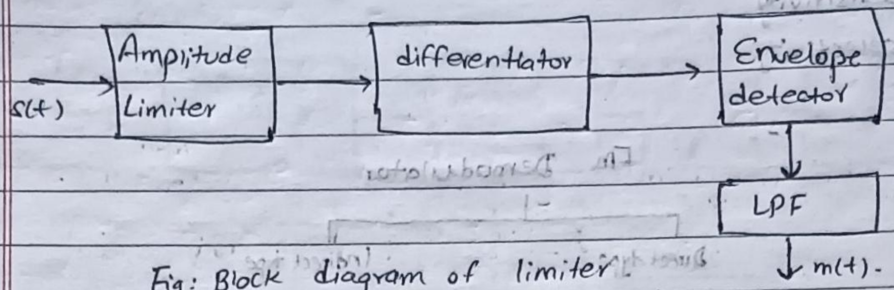


Fig: Block diagram of limiter discriminator method.

(1) Amplitude Limiter limits the amplitude of the signal to reduce the effect of fading and noise.

(2) the o/p of differentier be :-

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

$$= -A_c\,[2\pi f_c + 2\pi k_f m(t)] \cdot \sin(2\pi f_c t + \phi_1(t)] \quad \text{—(2)}.$$

This is equivalent to standard AM signal. This process is differentiation in this case is known as FM to AM conversion.

– From this AM signal message m(t) can be recovered by using envelope detector from eq^n (2).

i.e. $-A_c\,[2\pi f_c t + 2\pi k_f m(t)]$ —(3).

By passing the eq^n (3) in LPF we got,

$= 2\pi k_f m(t) \cdot A_c$

$= K m(t)$.

where, $K = 2\pi k_f \cdot A_c$

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∴ Hence, this is the required message signal i.e. demodulation of FM wave using linear discriminator method.

The schematic diagram of conversion for FM to AM (envelope) is shown in figure below →

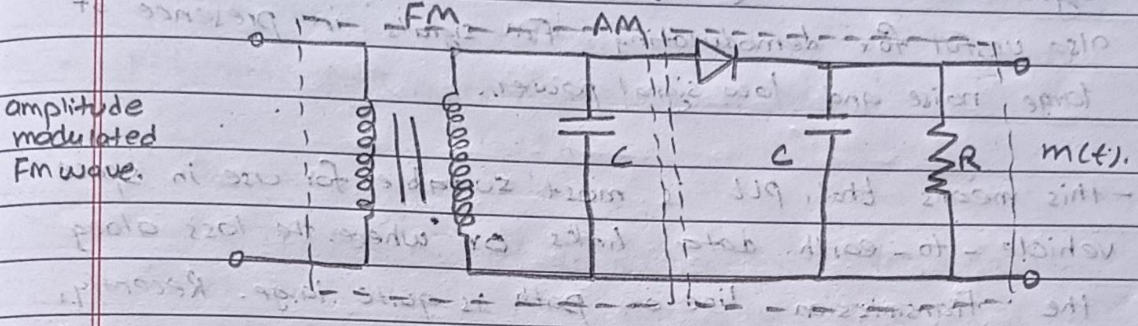

Fig: conversion for FM to AM (envelope).

$$s(t) = A_c\{2\pi f_c + 2\pi k_f m(t)\}$$
$$= A_c 2\pi f_c + A_c 2\pi k_f m(t).$$

(2) Phase locked Loop →

(FM). - It is negative feedback system →

the diagram of PLL is shown figure below →

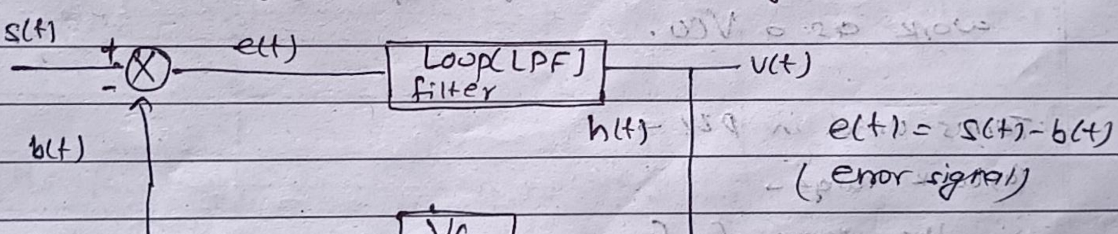

Fig: PLL for demodulator

- A PLL is primarily used in tracking the phase and frequency of the carrier component of an incoming FM signal. PLL is also useful for synchronous demodulation of AM-SC (i.e. Amplitude modulation with suppressed carrier) signals or signals with few cycles of pilot carrier. Further, PLL is also useful for demodulating FM signals in presence of large noise and low signal power.

- this means that, PLL is most suitable for use in space vehicle-to-earth data links or where the loss along the transmission line or path is quite large. Recently, it has found application in commercial FM receivers. A PLL is basically a negative feedback system. It consists of three major components. These components are multiplier, a loop filter and a voltage controlled oscillator (VCO) connected together in the form of a feedback loop. A VCO is a sine wave generator whose freqiency is determined by the voltage applied to it from an external source. It means that any frequency modulator can work as a VCO.

Terms used in PLL →

(a) Lock range →

the range of frequency over which PLL is still in lock mode.

(b) Locked mode →

the condition when frequency and phase of Vco is locked or synchronised with frequency and phase of incoming signal.

(c) Maximum lock sweep range →

The maximum rate of change of i/p frequency for which PLL will still remain in lock mode.

(d) Poll in range →

If PLL is not in locked mode, thin frequency range over which PLL will always enter into lock mode.

(e) Complexity →

It is define by order of loop filter (LPF). Generally, second order filter are used.

– the schematic diagram of PLL as in FM modulation is shown in figure below –

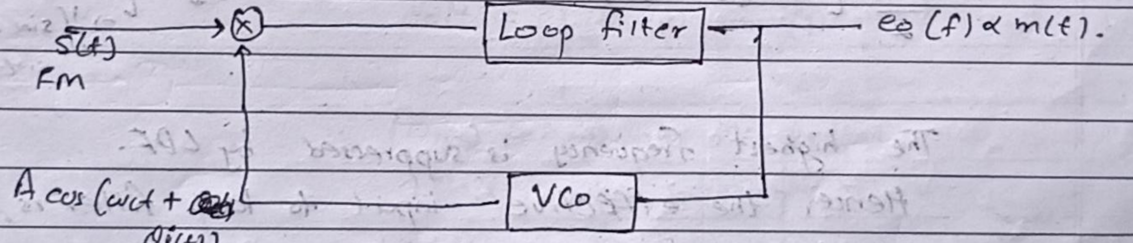

Fig :- PLL as in FM demodulation

The VCO is set at carrier frequency ω_c.
The Instantaneous frequency is given as,

$\omega_{co} \to \omega_c t \; c e_o(t) = 0$ —— (1)

If VCO output is $\beta \sin[\omega_c + \theta_o(t)]$.
then,
Derivation Instantaneous frequency is,
$[\omega_c + \dot{\theta}_o(t)] = 0$ —— (2)

From eq^n (1) and (2) we get,
$\dot{\theta}_o(t) = c e_o(t)$ —— (3).

Let, Incoming signal is $A_c \cos(\omega_c t + \theta_i(t))$.
where, $s(t) = A_c \cos(\omega_c t + \theta_i(t))$.
and $\theta_i(t) = 2\pi k_f \int_0^t m(t)\,dt$.

The multiplier output is,
$\Rightarrow AB\cos[\omega_c t + \theta_i] \cdot \sin(\omega_c t + \theta_o)$
$\Rightarrow \frac{AB}{2}[\sin(\theta_o - \theta_i) + \sin(2\omega_c t + (\theta_i + \theta_o))]$. $\left[\sin A \cos B \Rightarrow \frac{1}{2}\{\sin(A+B) + \sin(A-B)\}\right]$

The highest frequency is suppressed by LPF.
Hence, the effective input to loop filter is,
$\frac{AB}{2} \sin[\theta_o(t) - \theta_i(t)]$.

If $h(t)$ is Impulsive response of loop filter, then,

$$e_o(t) = h(t) * \frac{1}{2} AB \sin(\theta_o(t) - \theta_i(t)).$$

$$e_o(t) = \frac{1}{2} AB \int_0^t h(t-x) \cdot \sin[\theta_o(x) \cdot \theta_i(x)] \quad \text{— (2)}$$

From eq^n (1) and (2) we get,

$$\dot{\theta_o}(t) = AK \int_0^t h(t-x) \cdot \sin\theta_c(x) dx \quad \text{— (3)}$$

where, $K = \frac{1}{2} c\theta$.

and $\boxed{\theta_c(x) = \theta_o(x) - \theta_i(x)}$ => phase error — (4).

where, FM carrier is $A\cos[\omega ct + \theta_i(t)]$ then,

$$\theta_i(t) = k_f \int_{-\infty}^{t} m(\alpha) d\alpha \quad \text{— (5)}$$

From eq^n (4) and (5) we get,

$$\theta_o(t) = k_f \int_{-\infty}^{t} m(\alpha) \cdot d\alpha + \theta_c.$$

$$\theta_o(t) = k_f m(t) \quad \text{— (6)}$$

Assuming small error θ_e, then,

$$e_o(t) = \frac{1}{c} \dot{\theta_o}(t) \quad \text{— (7)}$$

From eq^n (6) and (7) we get,

$$\boxed{e_o(t) = \frac{k_f}{c} m(t)}$$

∴ Hence, this is the required expression of PLL acts as FM demodulation.

PLL (Phase Locked Loop) in AM →

- PLL is a circuit togenerate high frequency sinusoidal signal whose phase and frequency are almost equal to phase and frequency of reference signal.
- PLL is equivalent to narrow bandwidth tracking band pass filter.
- PLL consists of multiplier, VCO (Voltage controlled oscillator) and LPF.

- the experimental diagram of PLL is shown in figure below

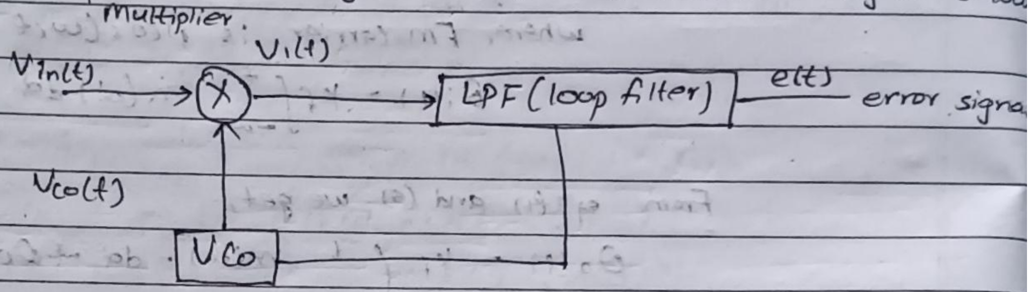

Figure: PLL

Let $V_{in}(t) = A_c\cos(2\pi f_c t + \phi_c)$ and $V_{co}(t) = A\sin(2\pi f_c t + \theta)$.

Assume that frequency of these two signals is same but differs in phase only.

The output of multiplier is,

$V_{in}(t) \cdot V_{co}(t) = V_1(t)$

$$V_1(t) = \underbrace{\frac{A_c A}{2}\sin(\theta - \phi)}_{(I)} + \underbrace{\frac{A_c A}{2}\sin[4\pi f_c t + (\phi_c - \theta)]}_{(II)}.$$

The LPF will block (II) term and only pass (I), term,
so,

$$\boxed{e(t) = \frac{A_c A \sin(\theta - \phi)}{2}}$$

Therefore, the control voltage applied to Vco is proportional to phase difference between two signals.

If input is AM signal with full carrier component, the PLL track carrier component and Vco i/p will be a single to the carrier signal. demodulation of DSB-SC is not possible because there is no carrier signal in DSB-SC.

Radio Receiver→

It is an electronic device which picks the desired signal, rejected unwanted signal, amplify, demodulates, the modulated wave to get original message signal.

(a) Tuned Radio frequency receiver→

The block diagram of tuned RF receiver is shown in figure below→

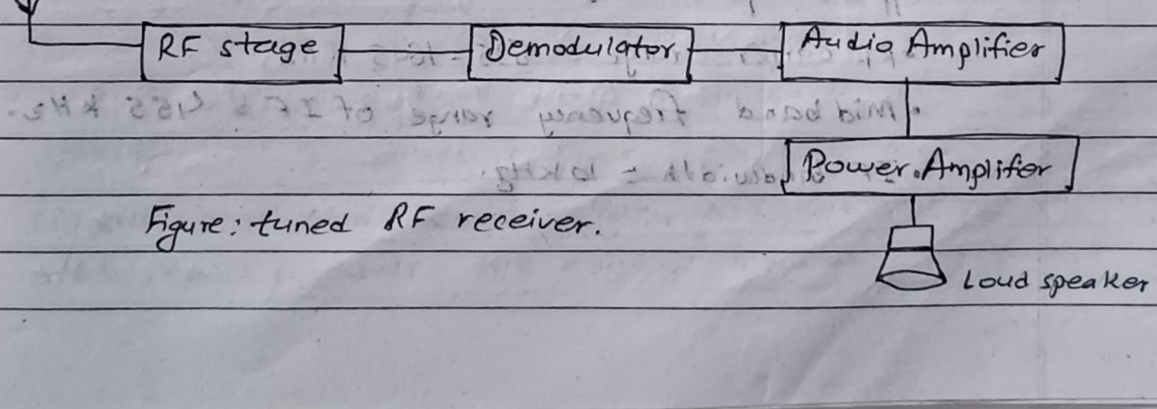

Figure: tuned RF receiver.

- limitations:
  (1) Instability of receiver.
  (2) selectivity is poor.
  (3) bandwidth variation over tunning.

(b) Super heterodyne receiver →

- It is also known as AM receiver. Super heterodyne means, the frequency conversion from variable carrier frequ of RF signal to fixed IF signal.
- the schematic diagram of super heterodyne receiver is shown in figure below →

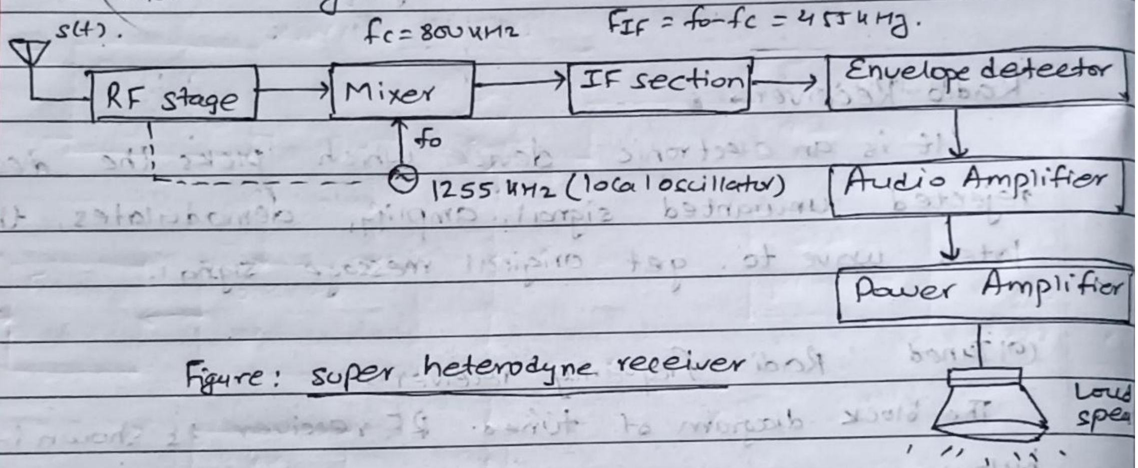

Figure: super heterodyne receiver

- typical parameters
  - RF carrier range = 535 - 1605 kHz
  - Mid band frequency range of IF = 455 kHz.
  - IF bandwidth = 10 kHz.

Stereophonic FM broadcasting:-

Stereophonic means transmission of two independent channels of same source i.e. two separate signal through same carrier, whereas monophonic means transmission of single channel. monophonic receiver will receive only this signal.

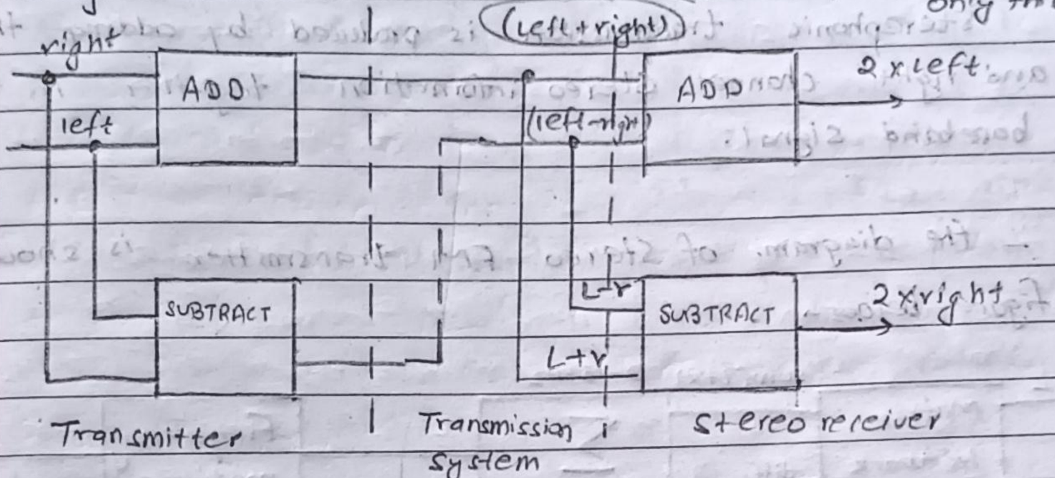

Figure: stereo coding and decoding process

At stereo transmitter a main program channel is transmitted that combines both left and right audio signals together and it can be used by monophonic receiver too, similarly, a stereo program channel is transmitted that can be coupled with the main program channel to produce left and right program material at a stereo receiver. At a stereo receiver, the sum and difference signals are added ~~tog~~ together, and subtracted from each other, this produces the original individual left and right channel signals which can be played over stereo loud speakers or head phones, this stereo coding process is carried out using a stereo generator.

#

(a) Stereo FM transmitter (encoder) →

Sterephonic transmission is produced by adding the left and right channel stereo information together in the baseband signal.

– the diagram of stereo FM transmitter is shown in figure below →

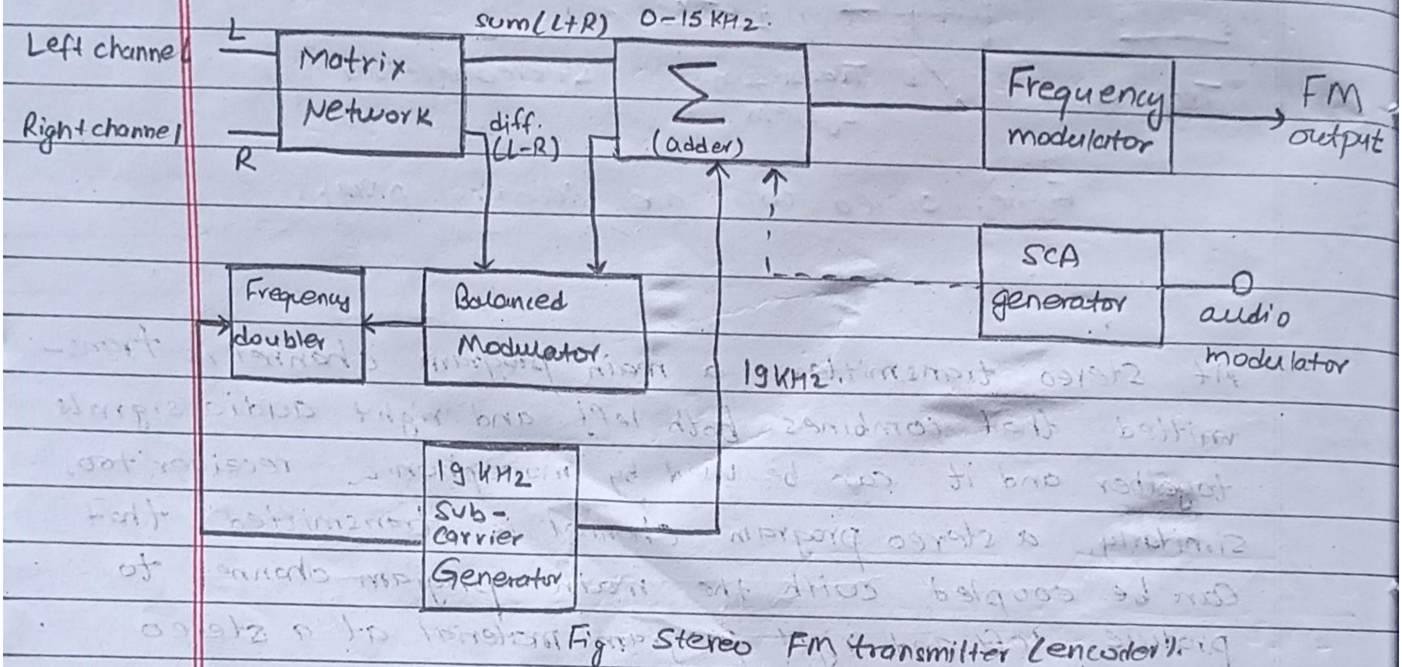

Fig: Stereo FM transmitter (encoder).

The waveform of stereo FM transmitter is shown below figure →

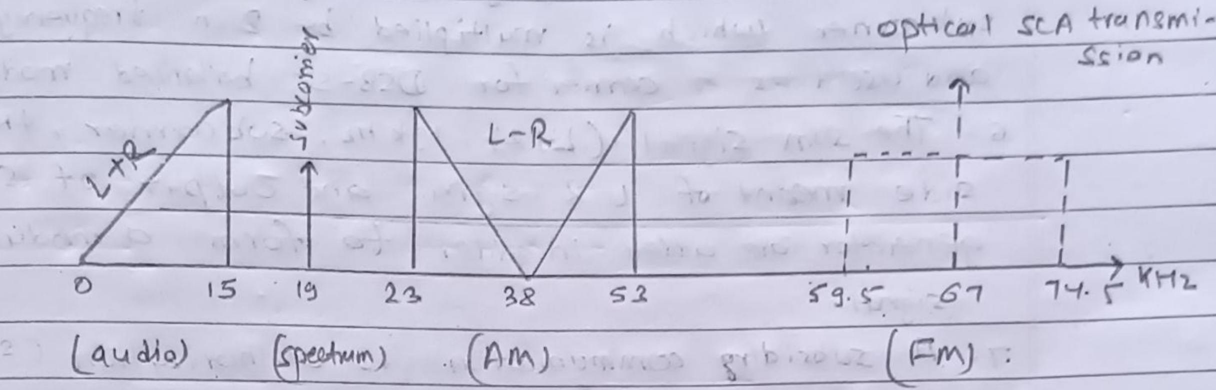

Fig: Stereo FM transmitter waveform.

Working Principle
1. there is two input signal i.e. Right and Left.
2. the two Input channels are passed through matrix network which produced two output namely (L+R) sum component and (L-R) difference component.
3. the sum component L+R modulated carrier in same manner in monophonic signal channel. This signal is demodulated and reproduced by mono FM receiver tunned to stereo channel station.
4. the difference signal (L-R) is applied to balanced modulator, It will suppress 38 KHz, 576 carrier coming from frequencies doubler and produce two side band (DSB-SC). If frequency range of input signal is (50 Hz - 15 kHz) then

two side band produced by balanced modulator will occupied frequency range (23 – 53 KHz).

5. A 19 KHz sub carrier generator generates 19 KHz stable subcarrier which is multiplied by 2 in frequency doubler and used as a carrier for DSB-SC balanced modulator.
6. The sum signal (L+R), 19 KHz, sub carrier, the two side meant of L-R signal and output of SCA generator are added in adder to form a modulating signal.
7. A subsidiry communication is authorization (SCA) optical signal which may be transmitted along with other signal. SCA uses a subcarrier at 67 KHz. The frequency band occupied by SCA signal is from 59.5 to 74.5 KHz.

The schematic diagram of composite FM stereo signal is shown in figure below→

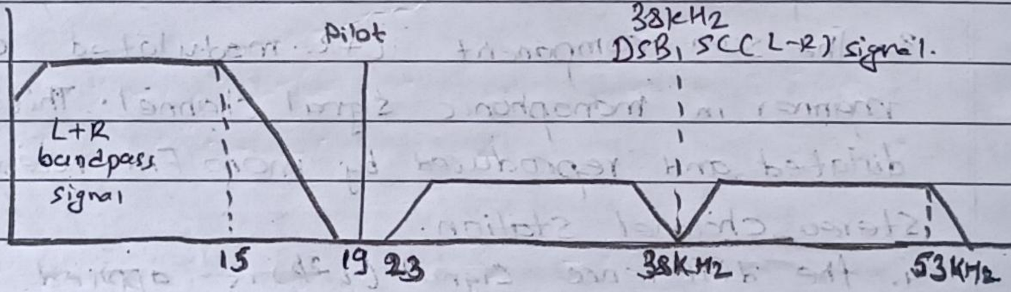

Figure: Composite FM stereo signal.

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- It has two facilities, i.e. Transmitter and Receiver
- In the transmitter, 38 KHz frequency is derived from (left - Right)

channel frequency,

i.e. $\underset{L}{53}-\underset{R}{15} \Rightarrow$ 38 KHz and 23 KHz is derived from

$(38-15) \Rightarrow 23$ KHz.

which is added and frequency ~~multiplied~~ multiplexed on a subcarrier of 38 KHz.

Using DSB-SC modulation →

- An unmodulated 19 KHz subcarrier is derived from 38 KHz subcarrier to provide a synchronous demodulation reference for the sterophonic receiver.
- In the receiver, a synchronous detector at 38 KHz recovers the (L-R) channel information, and then combined with L+R channel information in sum and difference combines to produce the original left-channel and Right channel.
- In stereo broadcast system, a composite FM signal is applied to the FM modulator as shown above.

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(b) Stereo-FM Receiver (decoder) →

Stereo-FM Receiver is produced by adding and subtracting the signals coming from transmitter.

- the schematic diagram of stereo-FM Receiver (decoder) is shown in figure below →

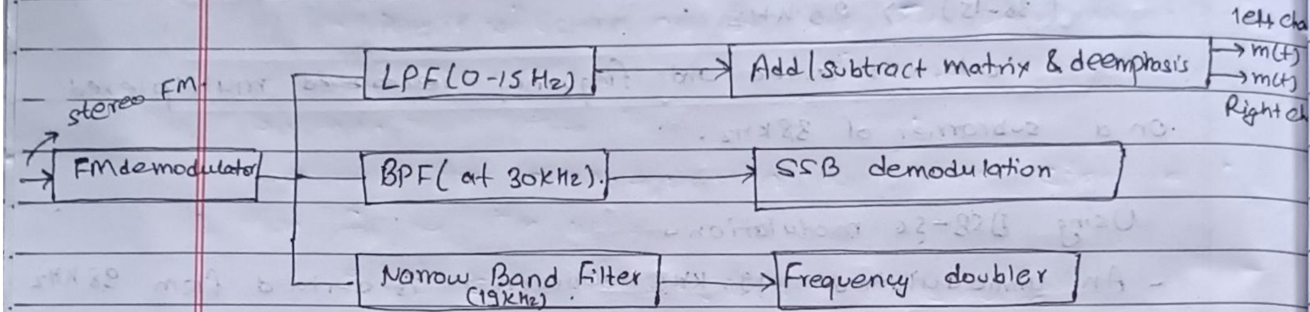

Fig: Stereo-FM Receiver (decoder).

Wave form same as transmitter.

- Working Principle:

(1) the input signal is demodulated by FM demodulator

(2) A locus filter removes all frequency above 15 kHz, to produce L+R signal as its output.

(3) A bandpass filter is used for selecting only two side band of L-R signal which occupies a frequency band from 25 kHz 53 kHz. These side band is applied to SSB demodulator. The carrier signal for this demodulator is obtained from

frequency doubler.

(4) The SSB demodulator produced (L-R) difference signal which occupies to matrix along with L+R signal.

(5) The matrix produces left channel by adding L+R and L-R signal. Similarly, it produces right signal by subtracting same signal. After the de-emphasis this channel are amplified and applied to loud-speaker.

Numerical

Q. An FM wave is given by $s(t) = 10 \sin(5\pi \times 10^8 t + 4 \sin(1250t))$. Find modulating and carrier frequency, frequency deviation, bandwidth, and power dissipated in 5 ohm ~~resistor~~ resictance.

(i) carrier & modulating frequencies

(ii) Modulation index and maximum deviation

(iii) power dissipated by this FM wave in a 5 Ω resistor.

Solution,

comparing the given equation with the standard equation for the FM wave i.e.

$$e_{fm} = s(t) = A_c \sin[\omega_c t + m_f \sin \omega_m t].$$

Comparing we get,

(i) Carrier frequency $(f_c) = \frac{5 * 10^8}{2\pi} = 79.57 MHz.$

(ii) Modulating frequency $(f_m) = \frac{1250}{2\pi} = 199 Hz$

(iii) Modulation index $m_f = 4.$

(iv) Maximum deviation $\Delta f = m_f \times f_m$

$= 4 + 199$

$= 796 Hz.$

(V) Power dissipated in 5 Ω resistance.

$$P = \frac{[\text{RMS value of FM wave}]^2}{R}$$

$$= \frac{\left[\frac{A_c}{\sqrt{2}}\right]^2}{R} = \frac{A_c^2}{2R} = \frac{10^2}{2\times 5} = \frac{10\cancel{0}}{1\cancel{0}}$$

$$= 10\,W.$$

Q) A carrier wave of frequency 91 MHz is frequency modulated by a sine wave of amplitude 10V and 15 kHz. The frequency sensitivity of the modulator is 3 kHz/V.

(i) determine the approximate bandwidth of FM wave using Carson's Rule.

(ii) Repeat part (i), assuming that the amplitude of the modulation wave is doubled and (iii) frequency of the modulating wave is halved.

Solution,

Given, $f_c = 91 \text{ MHz}$.

$A_m \cdot v = 10V$,

$f_m = 15 \text{ kHz}$.

frequency sensitivity $(k_f) = 3 \text{ kHz/V}$

$\Delta f = k_f \times A_m$

$= 3 \times 10 = 30 \text{ kHz}$

(i) Carsons Rule,

$$\text{Bandwidth} = 2(f_m + \Delta f)$$

$$= 2(15 + 30)$$

$$= 90 \text{ kHz}.$$

$$= 2 f_m (1 + m_f)$$

$$= 2 \times 15 (1 + 30)$$

$$= 31 \times 30$$

(ii) Amplitude is doubled & modulating wave is half,

$\Delta F = k_f \times 2A_m$

$= 3 \times 2 \times 10$

$= 60 \text{ kHz}$

$BW = 2(15 + 60)$

$= 150 \text{ kHz}$

(iii) BW? (Frequency of the modulating wave is halved).

$BW = 2(7.5 + 30)$

$= 75 \text{ kHz}$

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Q) A 100 MHz Carrier wave has a peak voltage of 5 volts. The Carrier is frequency modulated (FM) by a sinusoidal modulating signal or waveform of frequency 2 kHz such that the frequency deviation Δf is 75 kHz. The modulated waveform passes through zero and is increasing at $t=0$. Determine the expression for the modulated carrier waveform.

Solution, Because the frequency modulated carrier waveform passes through zero and is increasing at $t=0$, therefore the FM signal must be sine wave (signal). Thus

$$s(t) = A\sin[2\pi f_c t + m_f \sin(2\pi f_m t)].$$

where, m_f = modulation index of FM wave $= \frac{\Delta f}{f_m}$

Given that, f_c = carrier wave frequency $= 100 \times 10^6 = 10^8$ Hertz

Δf = frequency deviation = 75 kHz $= 75 \times 10^3$ Hertz

f_m = modulating frequency = 2 kHz $= 2 \times 10^3$ Hertz

A = peak voltage of carrier wave = 5 volt.

Now,

$$m_f = \frac{\Delta f}{f_m} = \frac{75 \times 10^3}{2 \times 10^3} = 37.5$$

Substituting all the above values in eq^n(i), we get,

$$s(t) = 5\sin[2\pi \times 10^8 t + 37.5 \sin(2\pi \times 2 \times 10^3 t)]$$

$$s(t) = 5\sin[2\pi \times 10^8 t + 37.5 \sin(4\pi \times 10^3 t)].$$

PULSE MODULATION →

-A type of modulation in which pulses are varied in some aspect, such as amplitude, width, position to represent the signal. i.e. the transmission of analog data or speech which is in continuous form is known as pulse modulation.

-The transmission of the voice signal by a carrier is represente as a voltage signal that varies continuously with time, In amplitude modulation and frequency modulation, the carrier signal is varied continuously in analog mannor. this continuous transmission of information in an analog manner is used in the Frequency division multiplex system, while in t time division multiplex system it is not necessary. The ti division multiplex system use pulse modulation. In the pul modulation, the continuous signal is converted into a series of pulses, each proportional to the amplitude of th signal and corresponding in time to it, thus in pulse mode lation, a series of pulses carries the information instead of continuous modulated signal.

-Classification of Pulse Modulation

- Pulse Modulation
  - Analog Pulse Modulation
    - PAM
    - PWM
    - PPM
  - Digital Pulse Modulation
    - PCM
      - RZ
      - NRZ
    - DM

Analog Pulse Modulation →

- Analog Pulse Modulation is the technique to sampled analog information signal. In PM system the carrier is a pulse train instead of sinusoidal carrier as in Analog Modulation. Some characterestics / Parameter of a carrier signal (amplitude, width, position) is changed wrt instantaneous value of message signal. PM system is that it permits the simultaneous transmission of several signals on time sharing basis i.e., PWM, PPM & PAM are the types of analog PM.

[A] Pulse Amplitude Modulation →

The modulation in which the amplitudes of regularly spaced rectangular pulses vary according to instantaneous value of modulating or message signal is called PAM. The pulses in PAM signal may be flat-top, or natural or ideal type. Flat top is most popular and widely used because it has high immunity to noise.

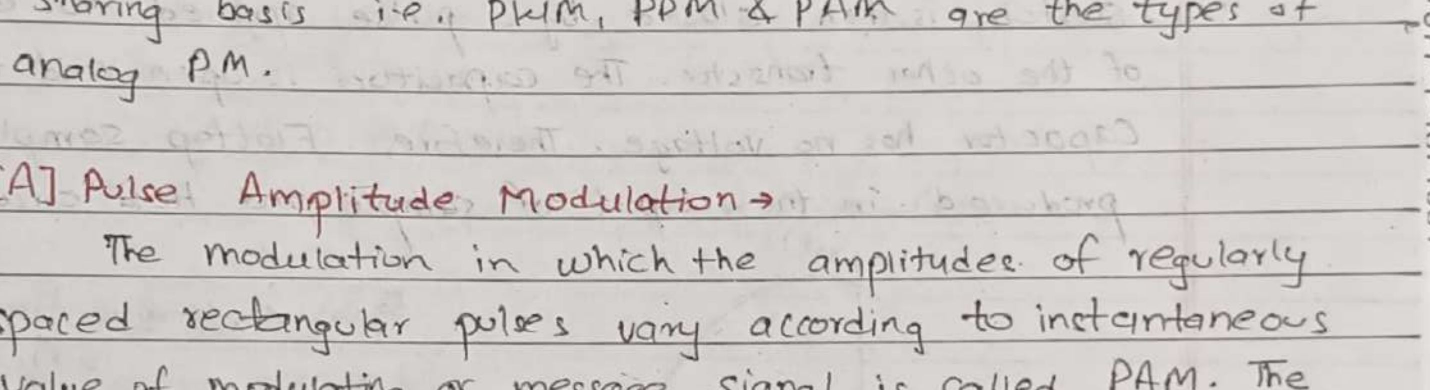
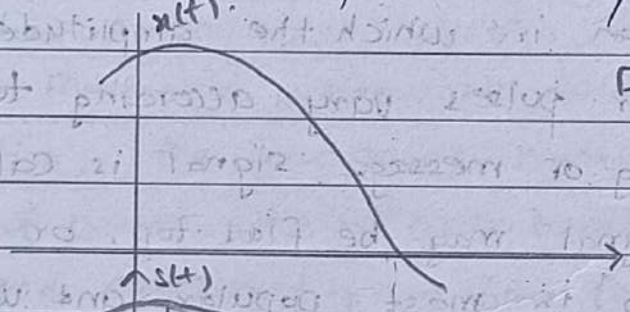
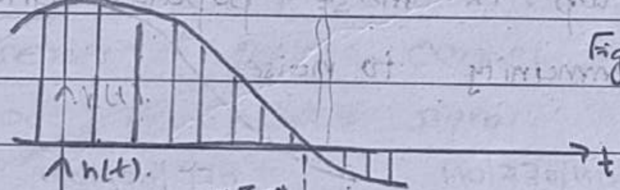


Fig: Sample and hold Circuit.

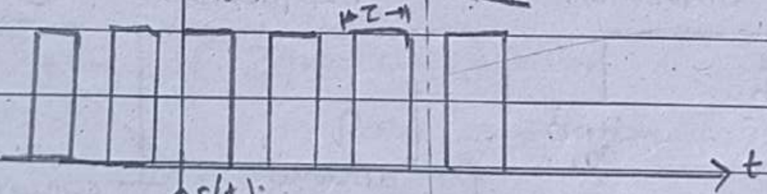
Above figure shows the sample and hold circuit. It consists of two FET switches and a capacitor. The sampling switch (G_1) is closed for short duration by a short pulse applied to the Gate (G_2) of the transistor. During this period the capacitor 'C' is quickly charged upto a voltage equal to the Instantaneous sample valu of the incoming signal $x(t)$. Now sampling switch (G_2) is opened and the capacitor holds the charge. The dischang switch is then closed by a pulse applied to Gate (G_2 of the other transistor. The capacitor is open and thus Capacitor has no voltage. Therefore, Flattop samples is produced in the output of sample and Hold circuit.



Fig(a): continuous time signal $x(t)$.



Fig(b): Instantaneous sample signal $s(t)$.



Fig(c):- Sampling function wave form (periodic pulse train).

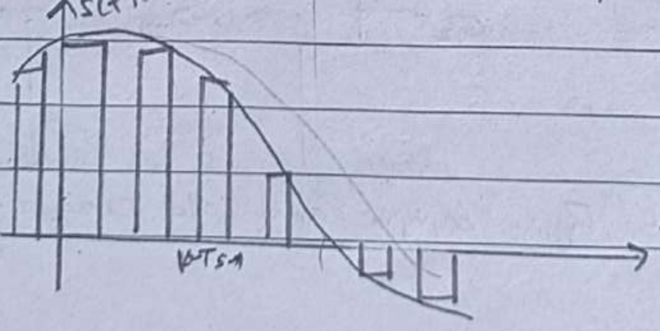


Fig: Flat top sampled Signal.

Let the duration or width of each sample is τ and sampling rate is $f_s = \frac{1}{T_s}$ | $f\{x(t)\}$. is a baseband signal whose instantaneous sampler signal is $s(t)$ and $h(t)$ is the constant width function. Therefore, flat-top sampled signal $g(t)$ is obtained as,

$$g(t) = s(t) \circledast h(t) \quad \text{--- (1)}$$

As the train of impulses is represented as,

$$\delta T_s(t) = \sum_{n=-\infty}^{\infty} \delta(T - nT_s) \quad \text{--- (2)}$$

The $s(t)$ is obtained by multiplication of baseband signal $x(t)$ and $\delta T_s(t)$ i.e.

$$\because s(t) = x(t) \cdot \delta T_s(t)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s) \quad \text{--- (3)}$$

Also we have,

$$g(t) = s(t) \circledast h(t)$$

$$= \int_{-\infty}^{\infty} s(t) \cdot h(t-\tau)\, \delta\tau \quad (\because \text{convolution theorem})$$

$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s) \cdot h(t-\tau)\, d\tau$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \int_{-\infty}^{\infty} h(t-\tau)\, \delta(T - nT_s)\, \delta\tau \quad \text{--- (4)}$$

Taking property of delta function we get,

$$\int_{-\infty}^{\infty} f(t)\, \delta(t - t_0)\, dt = f(t_0)$$

$$\therefore g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t - nT_s)$$

Taking F.T. of eq^n (2) we get,

$$\boxed{G(f) = f_s \cdot \sum_{n=-\infty}^{\infty} X(f - nf_s) \cdot H(f)}$$

→ Transmission Bandwidth of PAM →

In PAM signal the pulse width or duration or length 'τ' is very small in comparision to time period (samplin period) 'T_s' between two sampes.

i.e. $\tau << T_s$ ——(1)

If the maximum frequency of the modulation or message signal $x(t)$ is 'f_m'. According to sampling theorem, the sampling frequency 'f_s' is given as,

$f_s \geq 2f_m$ ——(2)

$\frac{1}{T_s} \geq 2f_m$

$\therefore T_s \leq \frac{1}{2f_m}$ ——(3)

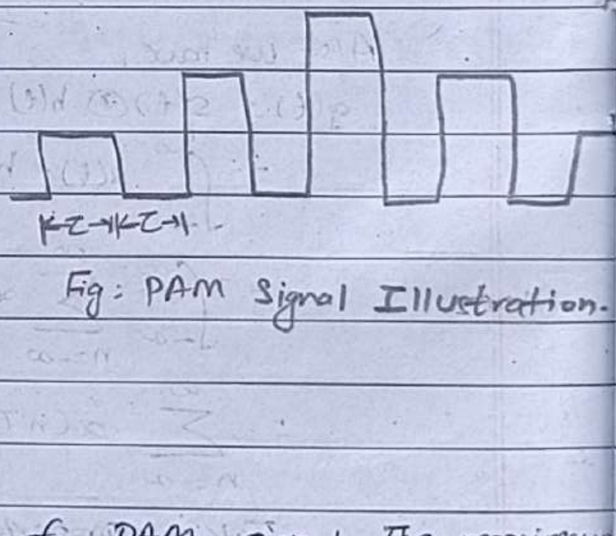


Fig: PAM Signal Illustration.

From eq^n (1) and (3) we get,

$\tau << T_s \leq \frac{1}{2} f_m$ ——(4)

For both ON and OFF time of PAM signal. The maximum frequency of PAM signal is,

$$f_{max} = \frac{1}{\tau + \tau} = \frac{1}{2\tau} \quad \text{——(5)}$$

∴ Therefore, BW requirement for transmission of PAM signal is given as,

$$BW \geq f_{max} \quad \text{--- (6)}$$

$$\therefore BW \geq \frac{1}{2\tau} \quad [\text{From eq}^n\ 5]$$

$$\therefore BW \geq \frac{1}{2\tau} \geq f_m \quad (\text{from eq}^n\ 4)$$

$$\boxed{BW \gg f_m}$$

∴ Hence, this is the required expression of transmission Bw of PAM.

→ Demodulation (Reconstruction) of PAM signal →

- Recovering of the modulating signal from the modulated signal is called demodulation. For PAM signal the demodulation is done by Holding circuit.

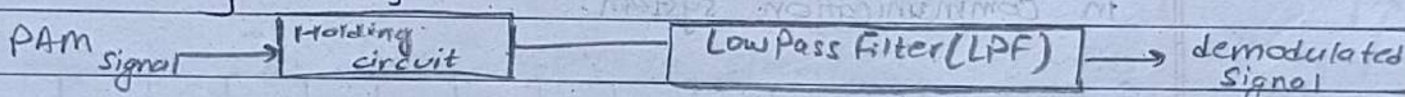
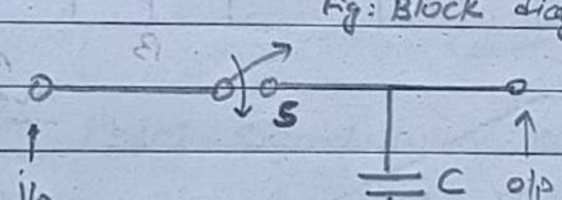
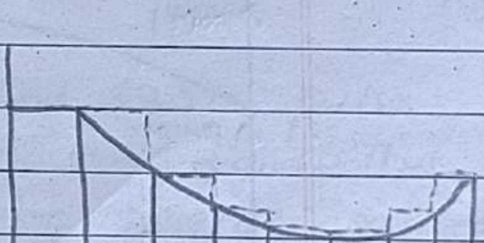


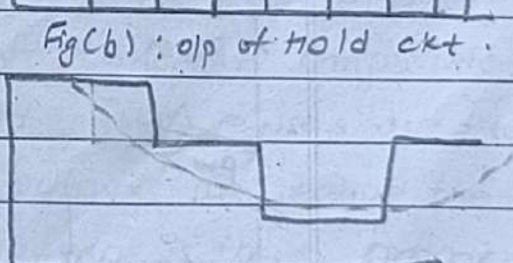
Fig: Block diagram of PAM demodulator



Fig(a): Holding circuit
(zero order holding ckt)



Fig(b): o/p of Hold ckt.



Fig(c): o/p of LPF

Above figure (a), shows the simple holding circuit (zero-order holding circuit because it considers previous sample to decide the value between the two pulses). Here, the switch 'S' is called after the arrival of pulse and opened at the end of pulse. The capacitor 'C' gets charged to the pulse amplitude value and holds for the interval between two pulses. This is then passed through the LPF to reconstructed modulating signal (message signal) which is clearly illustrated by fig (b) and fig (c).

[B] Pulse Width / Length / duration Modulation →

– In PWM the pulse width is changed in proportion to the amplitude of the modulating signal. There are three (3) types of variation in PWM / PDM / PLM. It is fewly important in communication system.

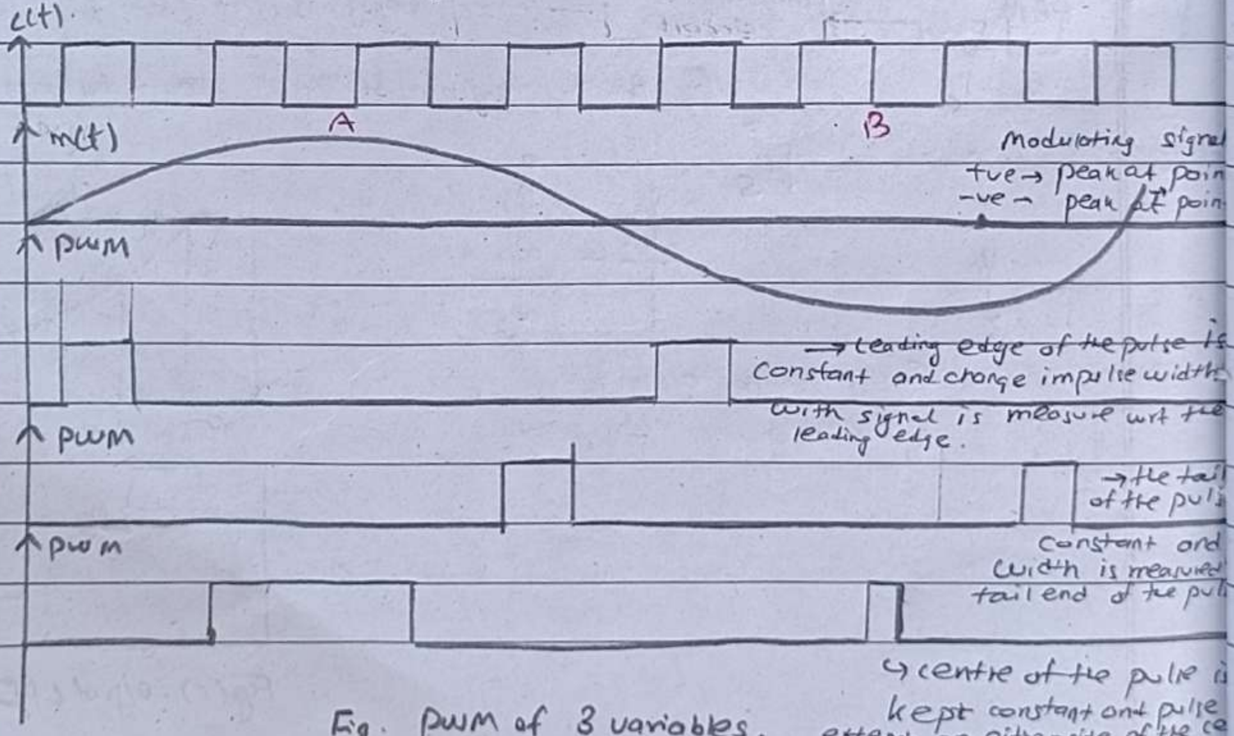


Fig: PWM of 3 variables.

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Above figure shows the frequency spectrum of PWM wave. With a sinusoidal modulating signal of frequency (fm), the spectrum of PWM signal consist the modulating frequency (fm) along with several harmonics.

→ Detection of PWM signal →

The block diagram of detection of PWM signal is shown in figure below →

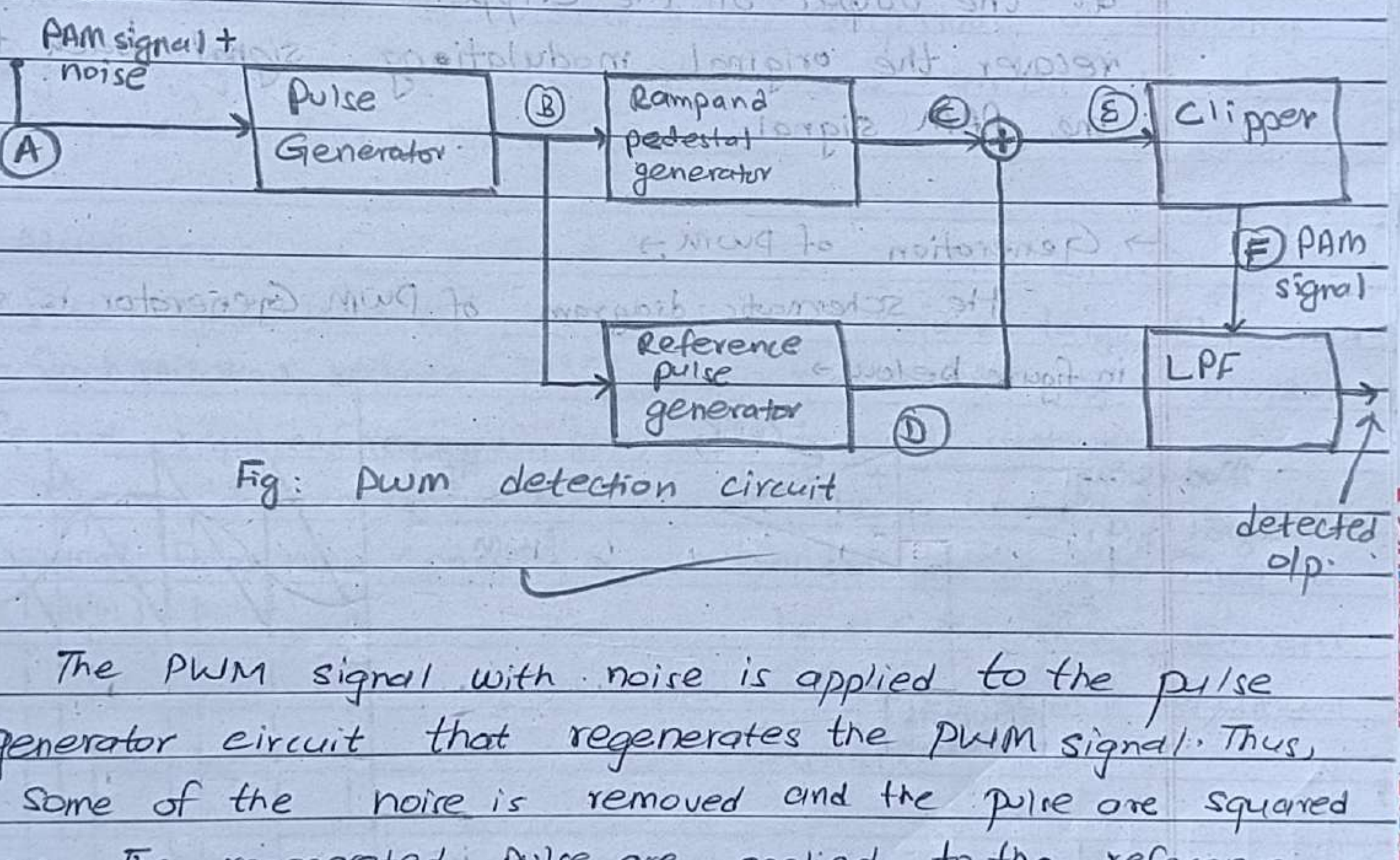

Fig: PWM detection circuit

The PWM signal with noise is applied to the pulse generator circuit that regenerates the PWM signal. Thus, some of the noise is removed and the pulse are squared up. The regenerated pulse are applied to the reference pulse generator. It produces a train of constant amplitude, constant width pulses. The regenerated PWM pulses are also applied to a ramp regenerator. At output we receive the signal of ramp, the height of the ramp is thus propor-

$x(t) = m(t)$
(modulating signal)

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tional to the width of the PWM pulses. At the end of the pulse, a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse. The constant amplitude pulses at the output of reference pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper. A LPF is used to recover the original modulationg signal back from the PAM signal.

→ Generation of PWM →

The schematic diagram of PWM Generator is shown in figure below →

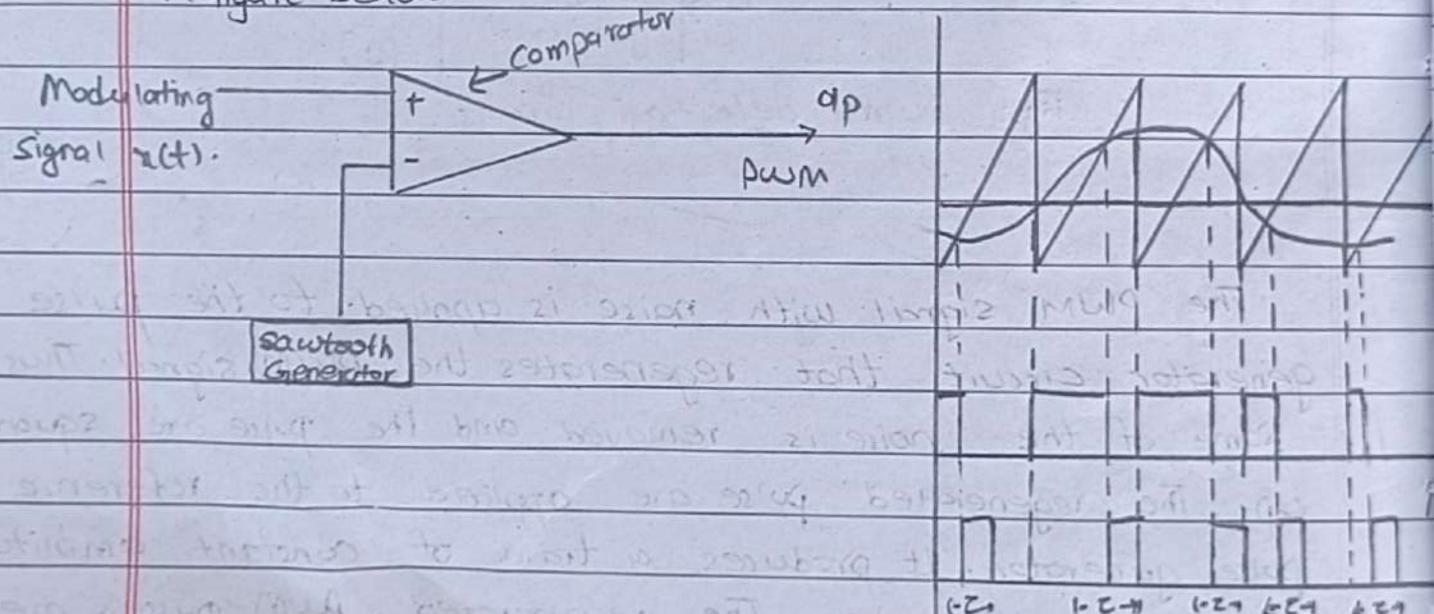


Fig:- PWM Generator

Above figure is a PWM Generator. The sawtooth generator generates the sawtooth signal of frequency 'fs' is a sampling signal and is applied to the inverting terminal of the comparator. The message or modulating signal m(t) is applied to the Non-Inverting input - output of the comparator will be high as long as instantaneous amplitude of m(t) is higher than that of sampling signal producing PWM signal. Likewise, o/p is zero when amplitude of sampling signal is greater than the instantaneous amplitude of modulating signal m(t).

Advantages

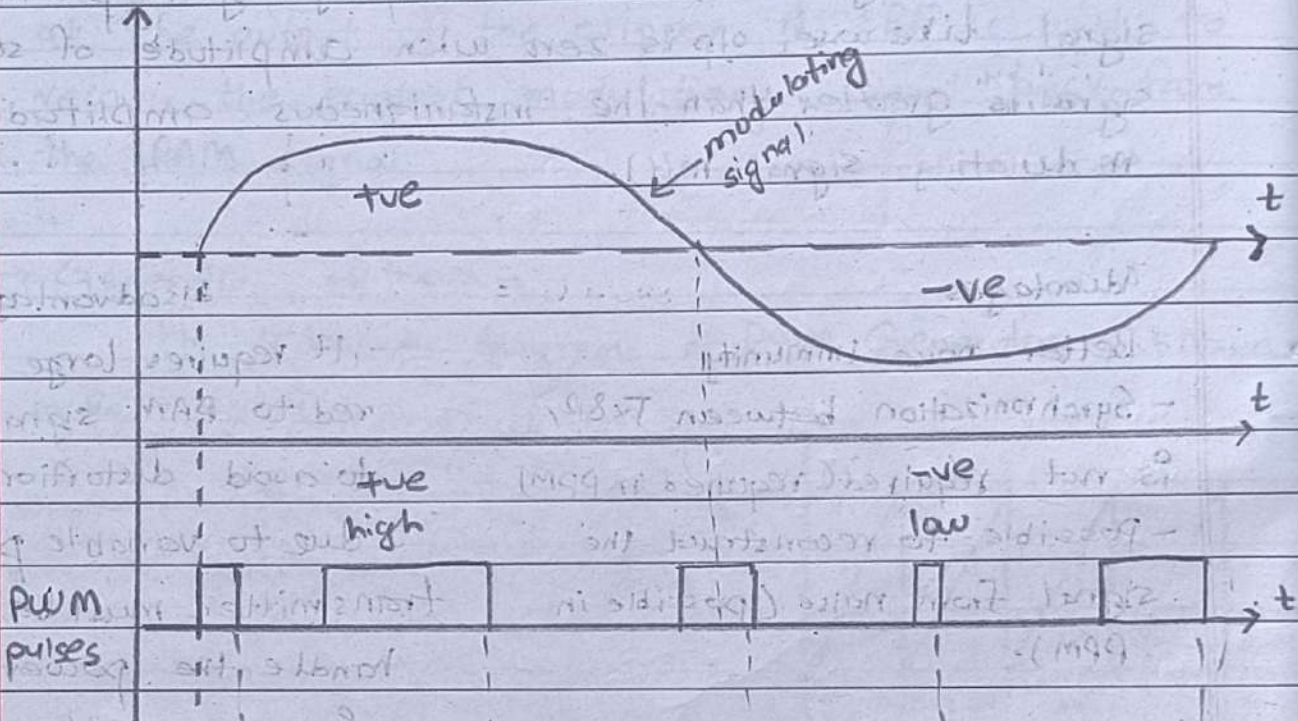
- Better noise Immunity
- Synchronization between Tx & Rx is not required (required in PPM)
- Possible to reconstruct the signal from noise (possible in PPM).

Disadvantages

- It requires large BW compared to PAM signal in order to avoid distortion.
- due to variable pulse width, transmitter must be able to handle the power content of pulse width maximum width.

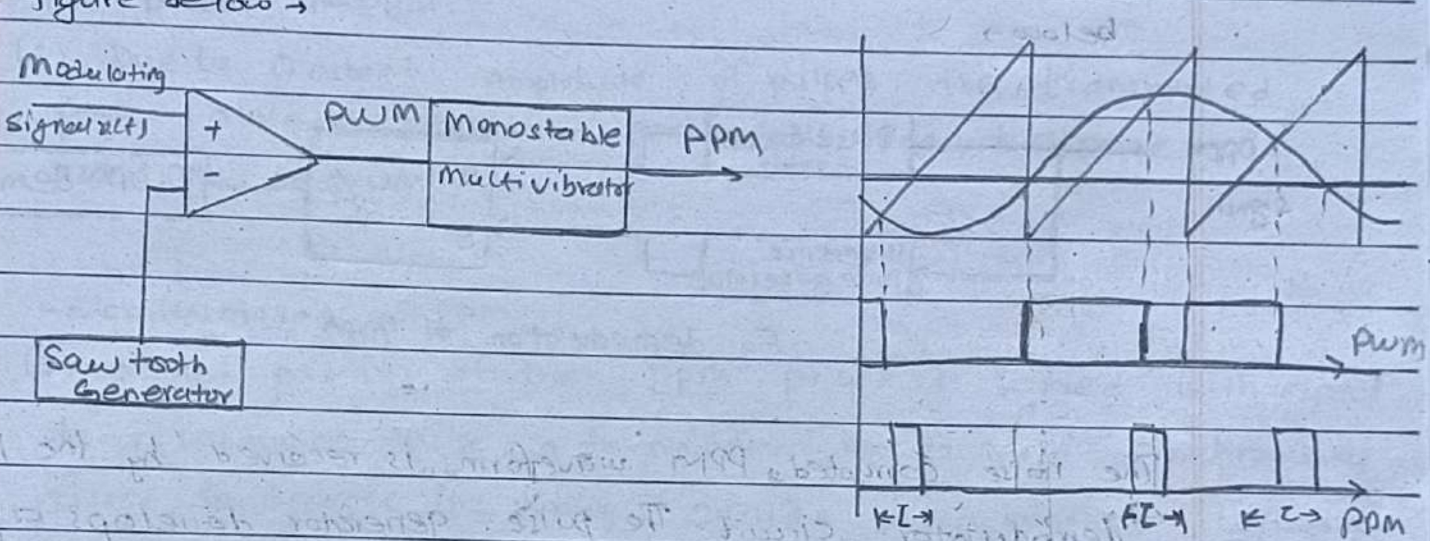
[C] Pulse Position Modulation →

- In PPM the position of each pulse is charged wrt the amplitude of sampled value of modulating signal. In PPM amplitude and width of the pulse are kept constant. It is fewly important than other.



-disadvantages
1. synchronization between Tx & Rx is required.
2. Large Bandwidth is required compared to PAM.

→ Generation of PPM →
- the schematic diagram of Generation of PPM is shown in figure below →



Generation of PPM

Above figure shows the PPM Generator. The pulse width modulation signal is obtained at the output of the comparator which is fed to Monostable Multi-vibrator. The multivibrator is negative edge triggered. Triggering occurs at the trailing (scaling) edge of PWM signal, which generates PPM signal with same or constant width and amplitude of modulating signal.

- advantages,
1. Like PWM, amplitude held constant in PPM results less interference of noise.
2. Signal and Noise seperation are easy.
3. Since pulse width and amplitude is constant, transmission power for each pulse is same.

- Demodulation of PPM →

The diagram (block) of PPM demodulation is shown below →

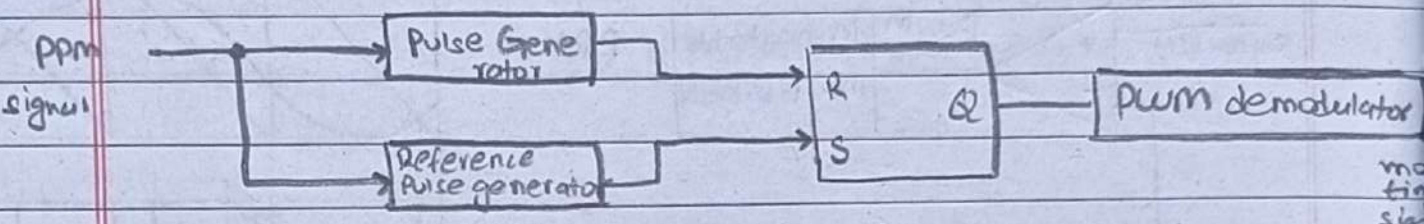


Fig: demodulation of PPM

The noise corrupted PPM waveform is received by the PPM demodulator circuit. The pulse generator develops a pulsed waveform at its output of fixed duration and applies thes pulses to the reset pin (R) of a SR filp-flop. A fixe period reference pulse is generated from the incoming PPM waveform and the SR flip flop is set by the reference pulses. Due to set and reset signals applied t the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator.

- advantages of PPM →

(a) due to constant amplitude of PPM pulses, the infor mation is not contained in the amplitude. Hence, the noise added to PPM signal doesnot distort the informa Thus, it has good noise immunity.

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(b) It is possible to reconstruct PPM signal from the noise contaminated PPM signal. This is also possible in PWM but not possible in PAM.

(c) Due to constant amplitude of pulses, the transmitted power always remains constant. It doesnot change as it used to in PWM.

-disadvantages of PPM:

(a) as the position of the PPM pulses is varied with respect to a reference pulse, a transmitter has to send synchronizing pulses to operate the timing circuits in the receiver. Without them, the demodulation would not be possible to acheive.

(b) Large bandwidth is required to ensure transmission of undistorted pulses.

(Q) For a PAM transmission of voice signal having maximum frequency equal to $f_m = 3 KHz$. Calculate the transmission BW given $f_s = 8 KHz$ and pulse width $\tau = 0.1 T_s$.

Solution,

Given,

$f_m = 3 KHz$, Sampling frequency $(f_s) = 8 KHz$.

pulse width $(\tau) = 0.1 T_s$.

We have,

$$T_s = \frac{1}{f_s} = \frac{1}{8 \times 10^3} = 1.25 * 10^{-4}$$

Now,

$$\tau = 0.1 * 1.25 * 10^{-4} = 12.5 \, \mu s.$$

Therefore,

$$BW >> \frac{1}{2\tau} = \frac{1}{2 * 12.5 * 10^{-6}}$$

$$BW \geq 40 KHz.$$

Note,

$$\tau << T_s \quad \text{— (1)}.$$

$$f_s \geq 2 f_m$$

$$\frac{1}{T_s} \geq 2 f_m$$

$$\therefore T_s \leq \frac{1}{2 f_m} \quad \text{— (2)}$$

$$\tau << T_s \leq \frac{1}{2 f_m}$$

$$\therefore f_m = \frac{1}{\tau + \tau} = \frac{1}{2\tau} \quad \text{— (3)}$$

$$BW \geq f_{max}$$

$$BW \geq \frac{1}{2\tau}$$

$$BW \geq \frac{1}{2\tau} \geq f_m$$

$$\boxed{\therefore BW \geq\!> f_m}$$