PART B

1. A sinusoidal modulating frequency of 50 kHz and amplitude 3V is modulated with a carrier freq of 10 MHz having a peak voltage of 6V. The output is connected to a 600Ω load. Determine

(i) power in carrier freq

(ii) power in each sideband

(iii) total power

(iv) draw amplitude spectrum

Sol.

$f_m = 50\ kHz$; $V_m = 3V$; $f_c = 10\ MHz$; $V_c = 6V$

$R = 600\ \Omega$

(i) $P_c = ?$

$$P_c = \frac{V_c^2}{2R} = \frac{\cancel{6}^{1} \times \cancel{6}^{3}}{\cancel{2}\times\cancel{600}\,100} = 0.03\ W$$

(ii) $P_{LSB} = P_{USB} = ?$

$$m = \frac{V_m}{V_c} = \frac{3}{6} = \frac{1}{2} = 0.5$$

$$P_{LSB} = P_{USB} = \frac{m^2}{4}\frac{V_c^2}{2R} = \frac{0.5 \times 0.5}{4} \times \frac{6 \times 6}{2 \times 600} = \frac{9}{4800}$$

$$P_{LSB} = P_{USB} = 1.875 \times 10^{-3}\ W = 1.875\ mW$$

(iii) $P_t = ?$

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

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

$$P_t = 0.03\left(1 + \frac{0.5 \times 0.5}{2}\right) = 0.03\left(\frac{2.25}{2}\right)$$

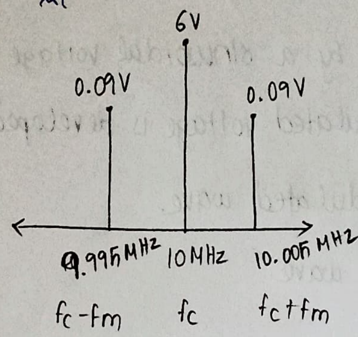
$$P_t = 0.03375\ W = 33.75\ mW$$

(iv) amplitude spectrum

$$f_c + f_m = 10 \times 10^6 + 0.005 \times 10^6 = 10.005\ MHz$$

$$f_c - f_m = 9.995\ MHz$$

$$\frac{mV_c}{2} = \frac{0.03 \times \cancel{6}^{3}}{\cancel{2}} = 0.09\ V$$

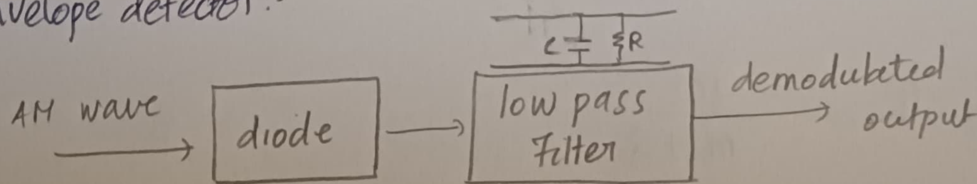


Demodulation of AM signal :-

The process of extracting an original message signal from the modulated wave is known as detection or demodulation. The circuit that demodulates the modulated wave is known as demodulator.

2 - i

Envelope detector :-



- → It is used to detect high level AM wave.
- → The detector consists of a diode and a lowpass filter. The diode is the main detecting element
- → Envelope detector is also called the diode detector.
- → The low pass filter consists of parallel combination of the resistor and capacitor.

→ In the positive half of the AM wave, the diode conducts and the capacitor charges to the peak value of AM wave

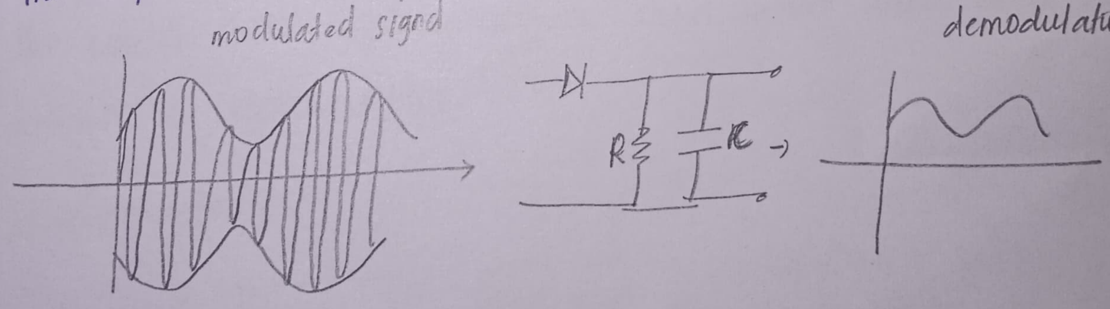
→ when the value of AM wave is less than this value, the diode will be reverse biased.

→ Thus the capacitor discharges through R till the other positive half cycle.

→ Select the components in such a way that the capacitor charges very quickly & discharges slowly.

→ As a result, the capacitor voltage waveform will be same as that of the envelope of AM wave. Thus the message signal can be retrieved.

→ the capacitor gets charged during the rising edge and discharges through the resistor R in the falling edge. Thus the capacitor helps in giving an envelope of the input as the output.



Steps in an envelope detector:-

① Rectification :

The received signal is first passed through a rectifier circuit, typically a diode. This causes the negative portion of the signal to be clipped off leaving only the positive envelope of the signal.

② Filtering :-

After rectification, the signal is passed through a low pass filter. This filter removes any high frequency component ~~incl~~ including carrier frequency leaving behind only the slowly varying envelope.

③ Output :-

The output of the envelope detector is the recovered message signal m(t), which is the original baseband signal that was modulated.

2. In an FM system, when the audio frequency (AF) is 500 Hz and the AF voltage is 2.4 V, the deviation is 4.8 kHz. If AF voltage is now increased to 7.2 V, what is the new deviation? If AF voltage is raised to 10 V while AF is dropped to 200 Hz, what is the deviation? Find modulation index in each case?

Sol. $\Delta f = 4.8 \text{ kHz}$; $V_m = 2.4 \text{ V}$; $f_m = 500 \text{ Hz}$; $V_m' = 7.2 \text{ V}$

$$\Delta f = K_f V_m$$

$$K_f = \frac{\Delta f}{V_m} = \frac{4.8 \times 10^3}{2.4} = 2 \times 10^3$$

2 - ii

$$\Delta f' = 2 \times 10^3 \times 7.2 = 14.4 \text{ kHz}$$

$$m = \frac{\Delta f}{f_m} = \frac{4.4 \times 10^3}{500} = 28.8$$

(ii) $A\,V_m' = 10\,V$; $f_m = 200\ Hz$

$$\frac{V_m'}{V_m} = \frac{10}{2.4} = 4.166$$

new AF voltage is 4.166 times greater than original voltage then deviation be

$$\Delta f' = 4.8 \times 10^3 \times 4.166 = 19.99\ kHz \approx 20\,kHz$$

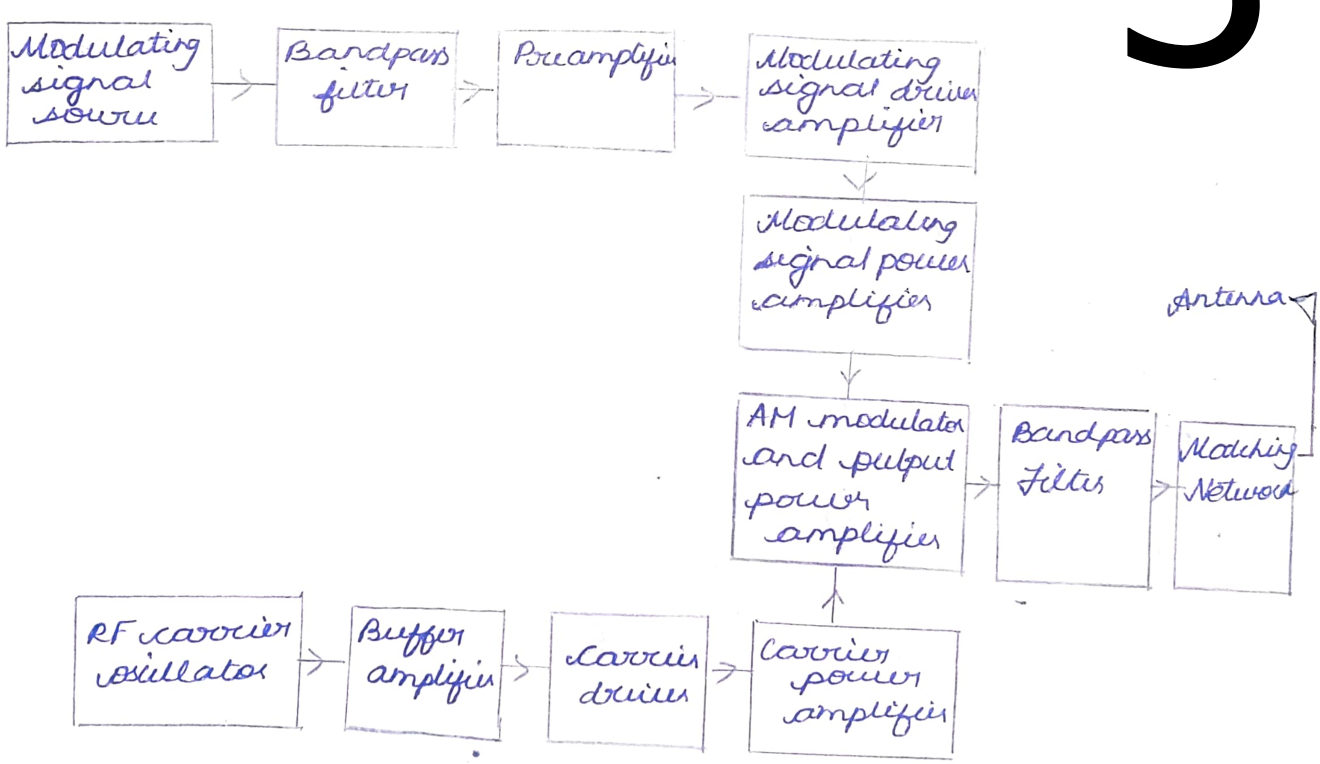
$$m = \frac{\Delta f}{f_m} = \frac{20 \times 10^3}{200} = 100$$

Draw the block diagram of high level transmitter and explain the operation of various blocks.

AM transmitter takes the audio/any signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted.

In high level modulation, modulation takes place in the final stage of amplification, and therefore the modulation circuitry has to handle high power.

3



i) Modulating Signal Path:

⇒ Modulating Signal Source provides the original signal to be transmitted. Since it has low power, it requires amplification.

⇒ Bandpass Filter removes unwanted noise and frequencies outside the required bandwidth. Only the necessary frequency components of the signal is used for modulation.

⇒ Preamplifier amplifies the filtered modulating signal. It provides impedance matching.

⇒ Modulating signal driver amplifies the information signal to an adequate level to drive the modulator.

⇒ The power amplifies amplifies the modulating signal to a high power level. This ensures that sufficient modulation index is achieved.

ii) Carrier Signal Path:

⇒ RF carrier oscillator generates a stable high-frequency radio carrier signal.

⇒ Buffer amplifier prevents variations in the oscillator circuit. It matches impedance levels between circuit.

⇒ Carrier Driver amplifies the carrier signal before sending it to power amplifier. It ensures the carrier has sufficient power for modulation.

⇒ The power amplifier amplifies the carrier signal to a high power level.

iii) Modulation and Transmission:

⇒ AM modulator modulates the signal. The modulating signal and carrier signal are combined. The amplitude of the carrier varies according to the modulating signal.

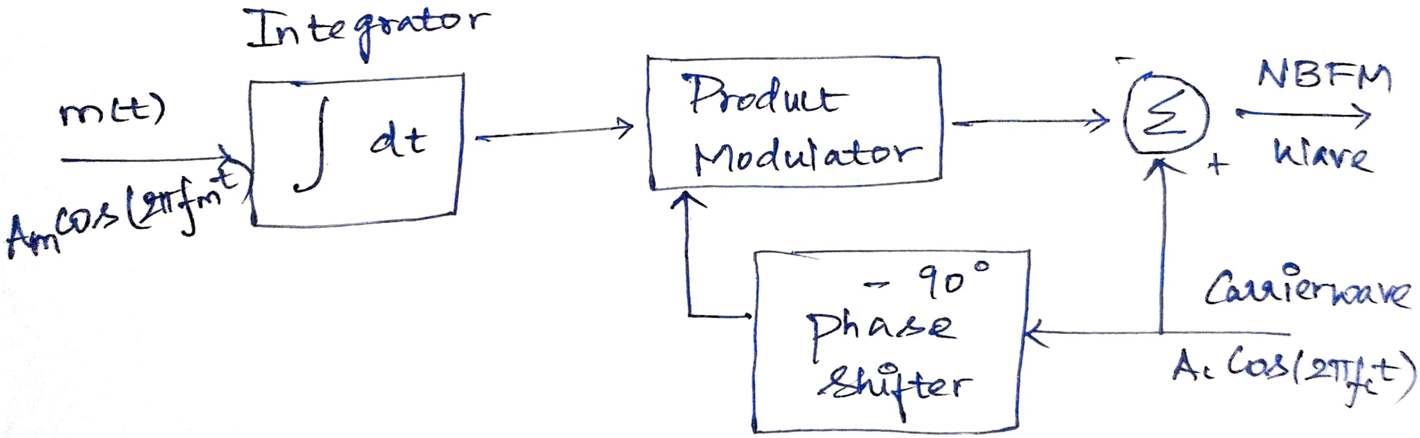
⇒ The bandpass filter, filters out unwanted harmonics.

⇒ The matching network matches the impedance between the transmitted output and the antenna.

⇒ Antenna converts the electrical signal into electromagnetic waves.

4) NBFM :-

4



When the value of modulating index β is small, then the bandwidth of FM is narrow and it is equal to $2f_m$.

Let the FM wave,

$$s(t) = A_c \cos[\omega_c t + \beta \sin \omega_m t]$$

$$\therefore \cos[A+B] = \cos A \cos B - \sin A \sin B$$

$$= A_c \left[\cos w_c t \cos(\beta \sin w_m t) - \sin w_c t \sin(\beta \sin w_m t) \right]$$

$$= A_c \cos w_c t \cos(\beta \sin w_m t) - A_c \sin w_c t \sin(\beta \sin w_m t)$$

$\therefore$ When θ is small $\Rightarrow$ $\cos \theta = 1$

$\sin \theta = \theta$

Assuming modulating index (β) is small

$\cos(\beta \sin w_m t) \simeq 1$

$\sin(\beta \sin w_m t) \simeq \beta \sin w_m t$

The simplified equation for FM representing NBFM

$$s(t) = A_c \cos w_c t - A_c \sin w_c t \, \beta \sin w_m t$$

$$\therefore \sin A \sin B = \frac{\cos(A-B) - \cos(A+B)}{2}$$

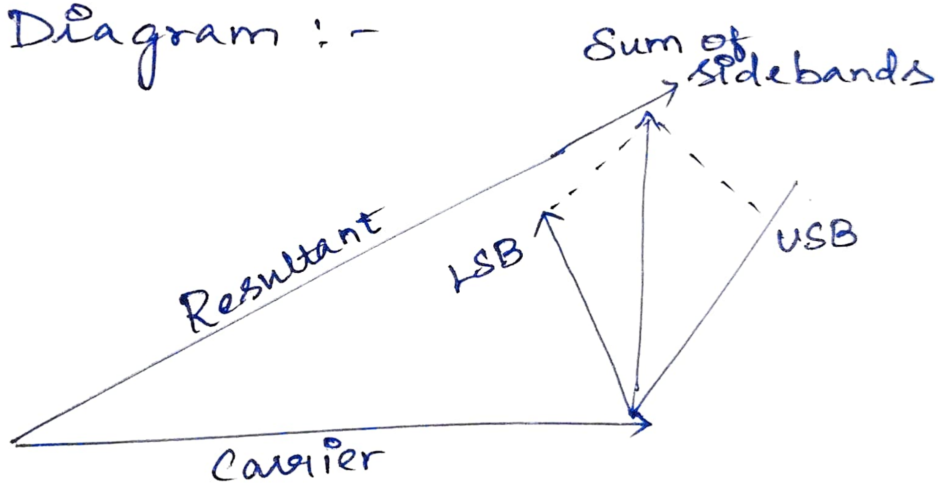
$$= A_c \cos w_c t - \frac{\beta A_c}{2} \left[\cos(w_c t - w_m t) - \cos(w_c t + w_m t) \right]$$

$$S_{AM}(t) = A_c \cos w_c t - \frac{\beta A_c}{2} \cos(w_c t - w_m t) + \frac{\beta A_c}{2} \cos(w_c t + w_m t)$$

This modulator involves splitting the carrier wave into two paths. one path is direct and Another path Contains 90° phase shifting network and a product modulator.

The combination of two waves generates a DSBSC modulated wave.

Phasor Diagram :-



5) PPL method of FM generation & detection:

Phase Locked Loop Direct FM transmitter:

The transmitter is used to generate high index Wide band FM signal.

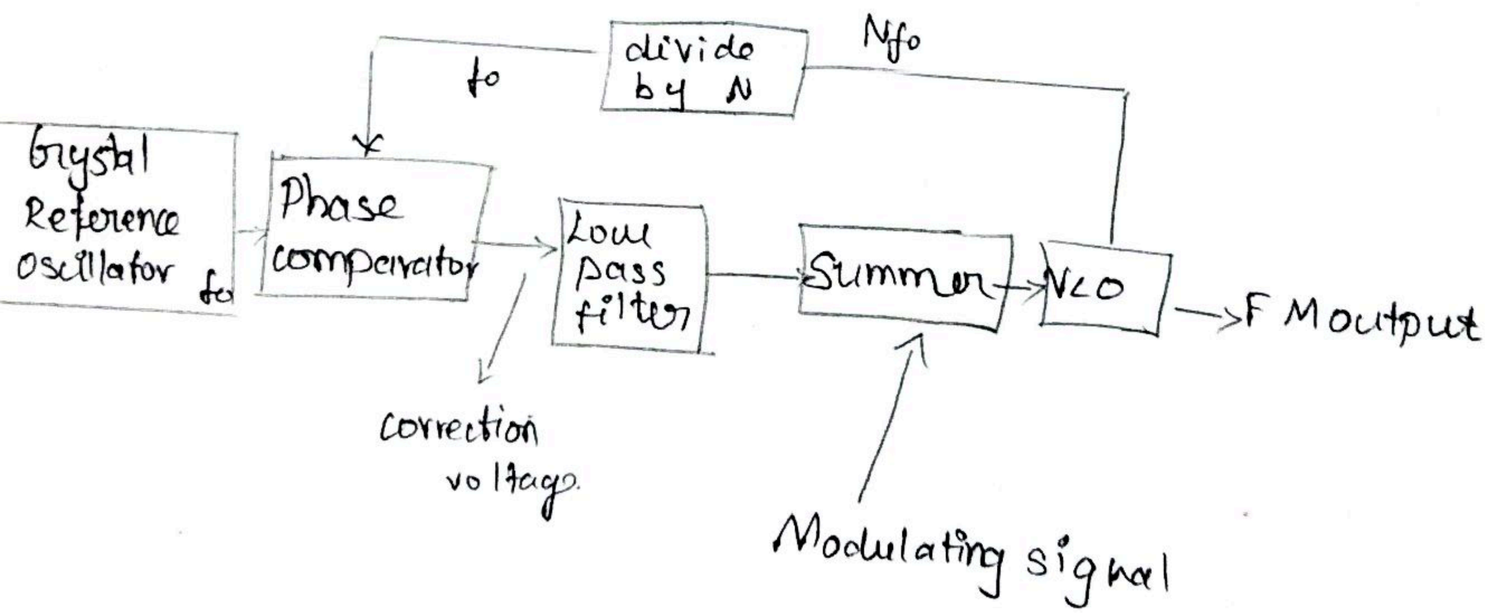
When the input frequencies of phase comparator is same, then they are locked to each other. Phase comparator output is zero.

It is passed through lowpass filter to the summer.

The summer has modulating signal as another input. This modulating signal is used to control the output frequency of VCO.

The output of VCO is FM signal whose frequency depends upon the modulating signal.

The output of VCO is divided by N & given to the phase comparator.



① Phase comparator:

The PLL compares the phase of two signals. Here, the phase comparator compares the output frequency of the voltage-controlled Oscillator (VCO) (divided by N) with the input reference signal [which could be the frequency of a local oscillator]

When both frequencies are the same, the phase comparator's output is zero, indicating the system is 'locked'.

② Low pass filter:

The output of the phase comparator [which could have a varying frequency due to phase differences] is passed through a low pass filter.

This filter removes the high-frequency components, leaving only the DC or low-frequency error signal, which controls the VCO's frequency.

③. Summer:

The error signal from the lowpass filter is summed with the modulating signal. The modulating signal is the information signal that will be used to vary the output frequency of the ~~vco~~ VCO.

④ Voltage Controlled Oscillator (VCO):

The VCO is responsible for generating the FM signal. Its output frequency is determined by the modulating signal, which directly influences how much the VCO frequency shifts.

The VCO's frequency is varied based on the combined input of the error signal from the PLL loop and the modulating signal.

⑤ Frequency Division by N:

The output of VCO is then divided by N before being feed back into the phase comparator.

This division is typically used to match the frequency range or provide a specific output frequency.

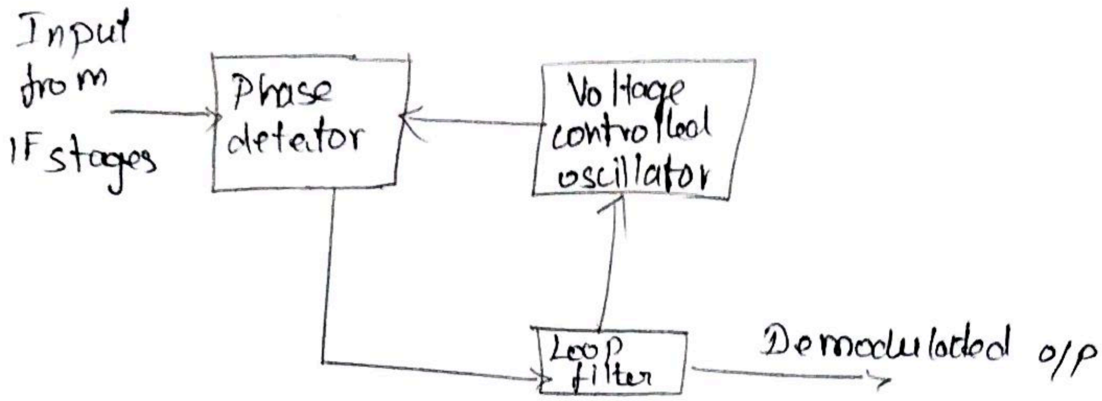
Detection of FM: [Demodulator]:

The basic components are,

1) Phase detector

2) Voltage control oscillator

3) Low pass filter



- A PLL frequency demodulator requires no ~~d~~ tuned circuits.
- It automatically compensates for changes in the carrier frequency due to instability in transmit oscillator.

Working:

The VCO natural frequency is equal to IF center frequency.

If the PLL input is deviated FM signal & the VCO natural frequency is equal to IF center frequency

The correction voltage produced at the output of the phase comparator & fed back to the i/p of the VCO is proportional to the frequency deviation, thus, the demodulated information signal.

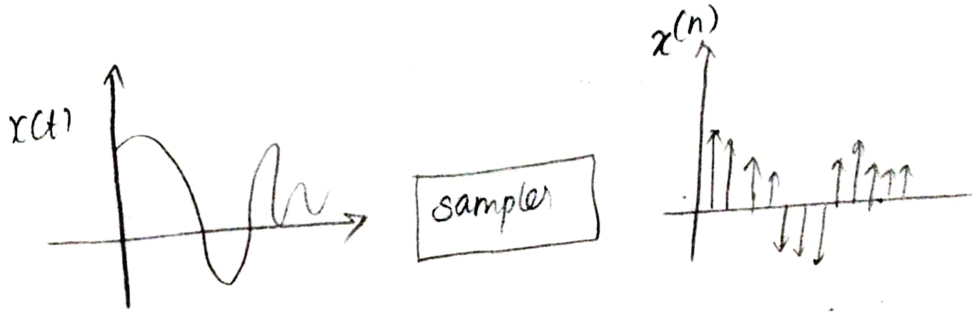
6) Describe the process of sampling in detail and derive the expression for a sampled signal and the corresponding reconstructed waveform.

Sampling :

Sampling is the process of converting a continuous-time signal into a discrete time signal.

The signal can be recovered if the sampling frequency f_s is greater than or equal to twice the highest frequency component of message signal.

6



Sampling theorem :-

→ The band-limited signal which has no frequency component higher than f_m, may be completely recovered or described by its sample values less than or equal to $\frac{1}{2f_m}$ second apart

→ $f_s \geq 2f_m$ (satisfactory reconstruction of signal from its samples.

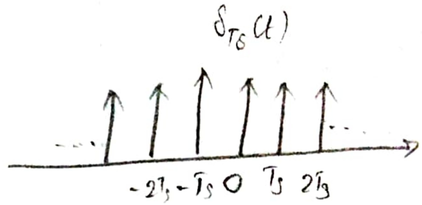
Process :-

Let $x_B(t)$ → Baseband signal
$C_S(t)$ → Periodic train of pulses

The two signals $g(t)$ & $c(t)$ are applied to the multiplier ($g(t) c(t)$).

The output is $g(t)$ sampled at the occurrence of each pulse.

When pulse occurs, multiplier output has same value of $g(t)$ and at all times it is zero.



$$\delta_{T_S}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

$$\delta_{T_S}(t) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} e^{jn\omega_s t}$$

$$\delta_{T_S}(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \cdots\right]$$

$$F[c(t)] = 2\pi\, \delta(\omega - n\omega_s)$$

$$F[c(t)] = \frac{2\pi}{T_s} \sum_{n=-\infty}^{\infty} \delta(\omega - n\omega_s)$$

Sampled signal,

$$F[g(t)] = F[x(t) \cdot c(t)]$$

$$G(\omega) = X(\omega) \cdot C(\omega)$$

$$= \frac{2\pi}{T_s} \sum_{n=-\infty}^{\infty} X(\omega - \omega_s)$$

Nyquist rate:

$$f_s = 2 f_m$$

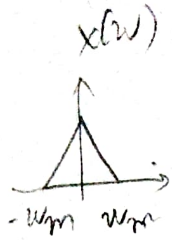
Minimum sampling used to sample the band limited signal can be ~~to~~ reconstructed from the samples

Cases :-

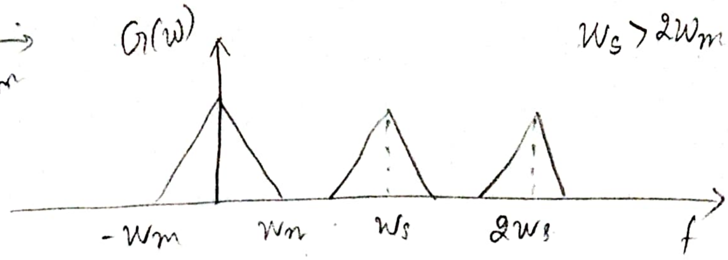
$f_s = 2f_m \rightarrow$ critical sampling

$f_s > 2f_m \rightarrow$ Over sampling

$f_s < 2f_m \rightarrow$ Under sampling

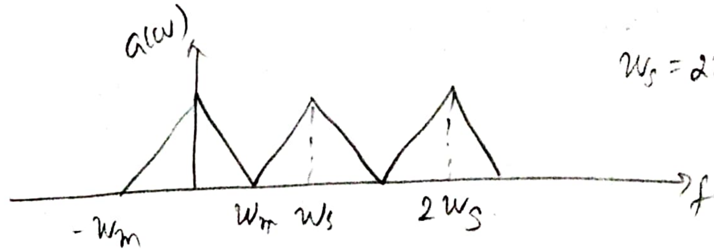


$W_s > 2W_m$ / $f_s > 2f_m$



$W_s < 2W_m$ / $f_s < 2f_m$

$W_s = 2W_m$ / $f_s = 2f_m$.

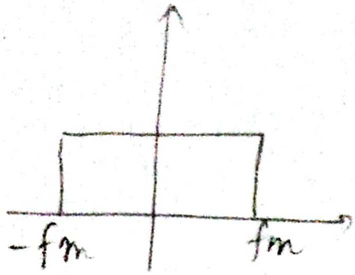


When the continuous time signal is sampled at Nyquist rate ($f_s = 2f_m$), the sampled spectrum ($G(w)$) contains non-overlapping $G(w)$ repeating periodically.

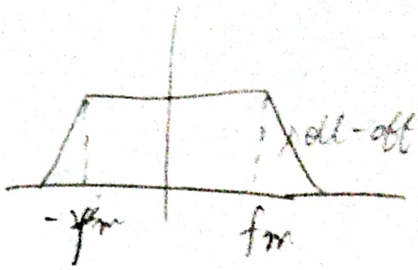
<u>Reconstruction filter</u>:

→ The low-pass filter is used to recover original signal from its samples.

→ This is also known as interpolation filter.



Ideal low pass filter



Practical low-PF

→ Ideal LPF is not possible practically. In case of practical LPF, the amplitude response decrease slowly to become zero.

$$T_s = \frac{1}{2f_m}$$

$$2T_s f_m = 1 \quad \Rightarrow H(w) = T_s \, rect\left(\frac{w}{4\pi f_m}\right)$$

$$h(t) = F^{-1}[H(w)]$$

$$h(t) = 2 f_m T_s \, sinc\,(2\pi f_m t)$$

Each sample produces a sinc pulse of height equal to the strength of sample. Addition of all sinc pulses results in $x(t)$.

$$x(t) = \sum_k x(kT_s)\, h(t - kT_s)$$

$$= \sum_k x(kT_s)\, sinc\,[2\pi f_m (t - kT_s)]$$

$$x(t) = \sum_k x(kT_s)\, sinc\,(2\pi f_m - k\pi)$$

8. Coherent Detector

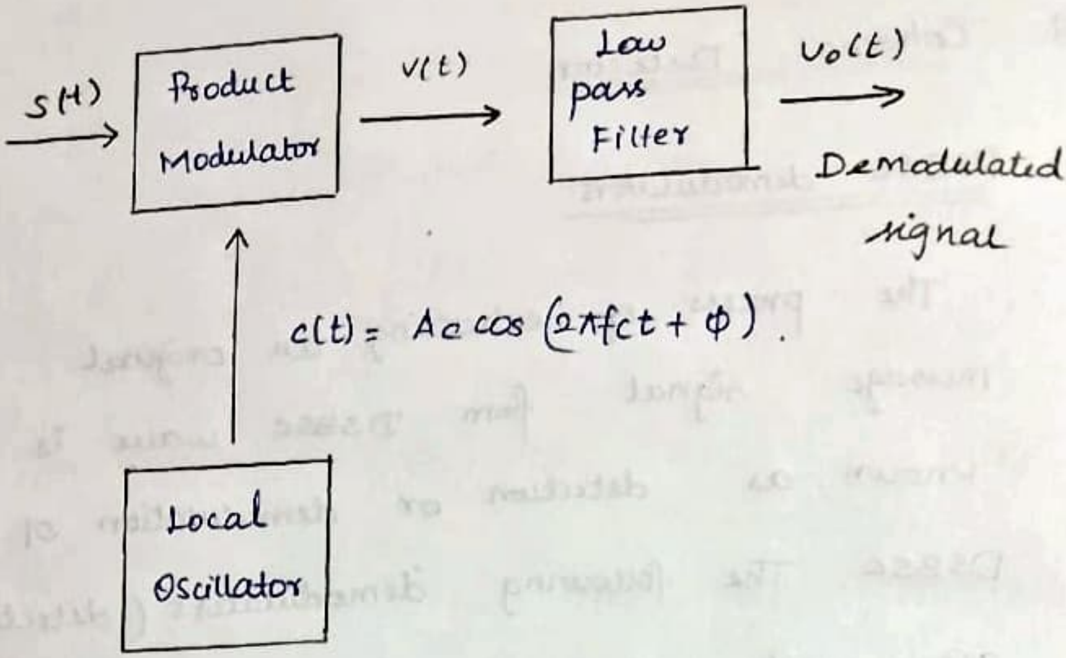
DSBSC demodulators:

The process of extracting an original message signal from DSBSC wave is known as detection or demodulation of DSBSC. The following demodulators (detectors) are used for demodulating DSBSC wave

- Coherent Detector.
- Costas Loop.

Coherent Detector:

The same carrier signal (which is used for generating DSBSC signal) is used to detect the message signal. Hence, this process of detection is called as Coherent or synchronous detection. Following is the block diagram of the coherent detector.



- In this process, the message signal can be extracted from DSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in DSBSC modulation. the resulting signal is then passed through a Low Pass Filter. Output of this filter is the desired message signal.

- Let the DSBSC wave be,

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

- The output of the filter is the desired message signal.

- Let the DSBSC wave be

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

- The output of the local Oscillator is

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

where ϕ is the phase difference between the local oscillator signal and the carrier signal, which is used for DSBSC modulation.

- The output of product modulator as

$$v(t) = s(t)\, c(t).$$

Subs, $s(t)$ and $c(t)$ values in the above equation.

$$\Rightarrow v(t) = A_c \cos(2\pi f_c t)\, m(t)\, A_c \cos(2\pi f_c t + \phi)$$

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

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

$$V(t) = \frac{A_c^2}{2} \cos\phi \, m(t) + \frac{A_c^2}{2} \cos(4\pi f_c t + \phi) m(t)$$

- The first term is the scaled version of the message signal, extracted by passing the above signal through a low pass filter.

The output of low pass filter is

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

- The demodulated signal amplitude will be maximum, when $\phi = 0^\circ$.

- The local oscillator signal and the carrier signal should be in phase, i.e, there should not be any phase difference between these two signals.

- The demodulated signal amplitude will be zero, when $\phi = \pm 90^\circ$. This effect is called as Quadrature null effect.

4) With neat sketches, explain about the VSB modulation system and mention its advantages over SSB modulation.

9

Soln:

Vestigial Sideband Modulation:

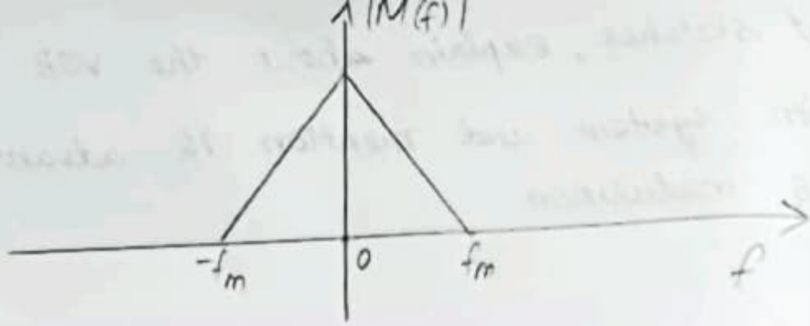
⇒ In VSB modulation, one sideband is passed almost completely whereas just a trace or vestige, of other sideband is retained.

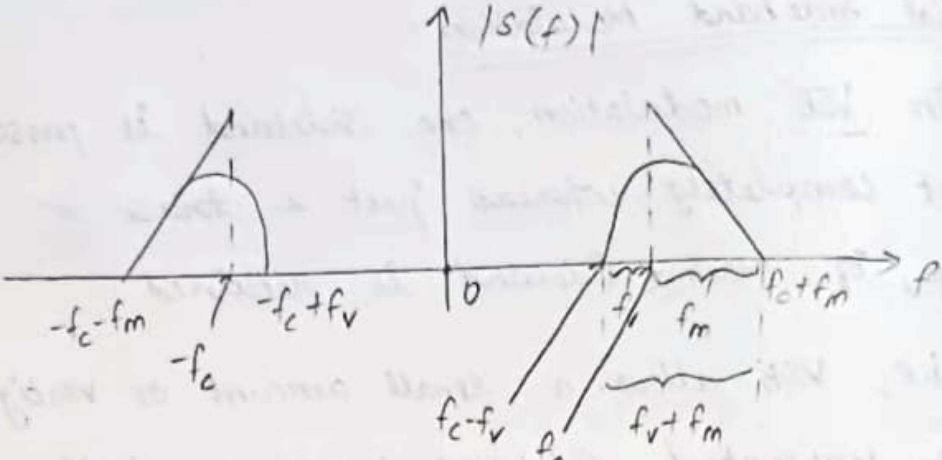
⇒ i.e, VSB allow a small amount or vestige, of the unwanted sideband to appear at the output of an SSB modulator.

⇒ The resulting signal has a bandwidth greater the bandwidth of the modulating (baseband) signal but less than the DSB signal bandwidth.

⇒ in VSB, Transmission bandwidth,

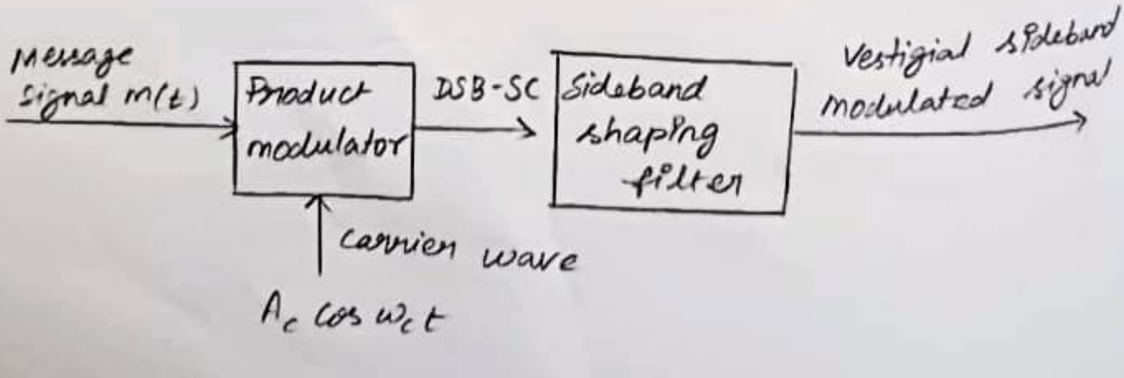
$f_T = f_m + f_v$, where 'f_v' is the vestigial of frequency band



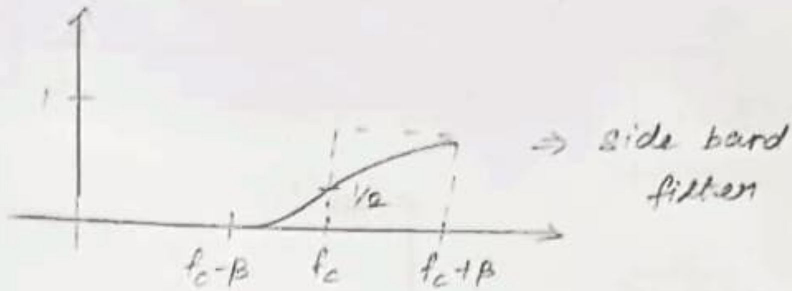


⇒ The above diagram illustrates the spectrum of a VSB modulated wave $s(t)$ in relation to that of the message signal $m(t)$ assuming that the lower sideband is modified into vestigial sideband.

⇒ To generate a VSB modulated wave, we pass a DSB-SC modulated wave through a sideband shaping filter.



⇒ The key to VSB is the ~~[illegible]~~ sideband filter, a typical transfer function is shown as,



⇒ While the exact shape of the response is not crucial, it must have odd symmetry above the carrier frequency and a relative response of 1/2 at f_c.

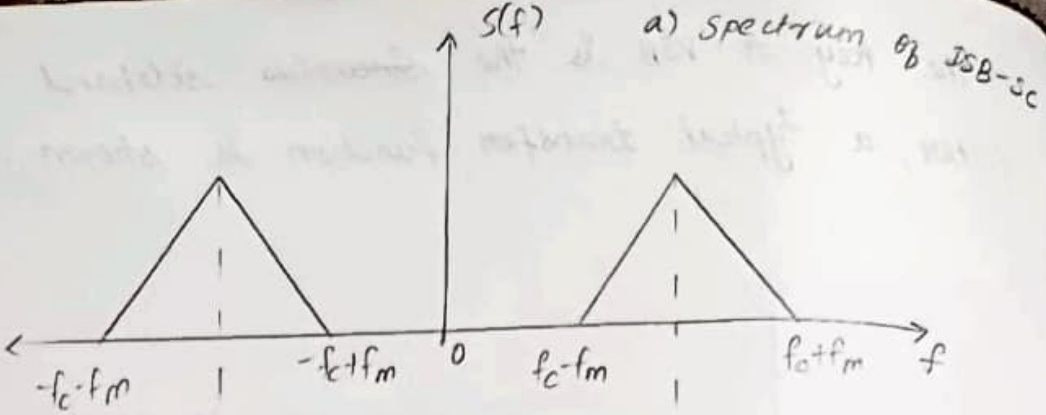
⇒ The filter response is designed so that the original message spectrum $M(\omega)$ is reproduced on demodulation.

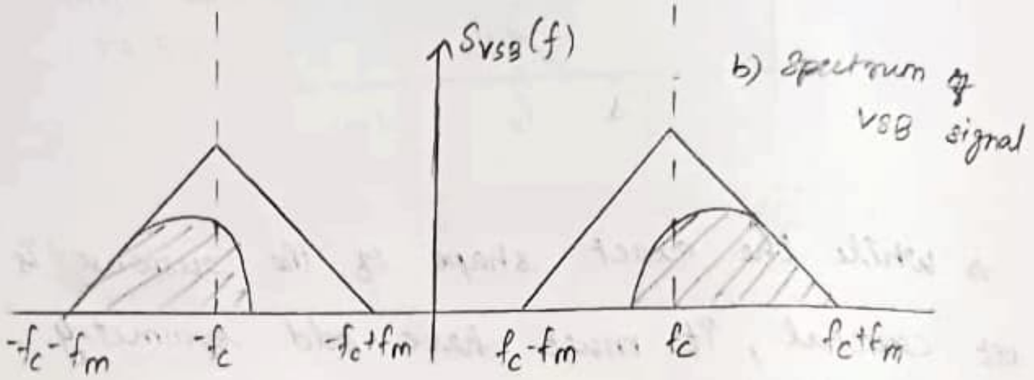
On demodulation,

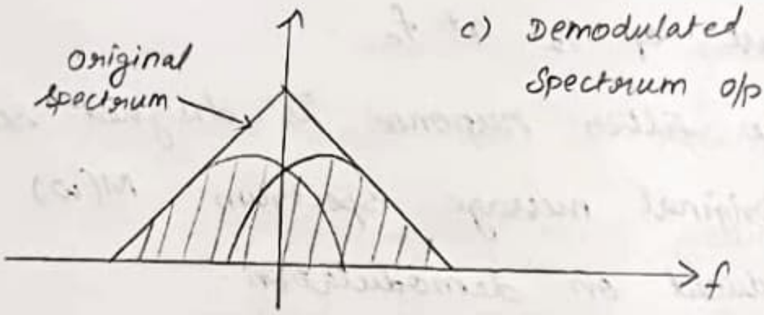
The positive frequency part of $S(\omega)$ [spectrum of transmitted signal $s(t)$] is shifted downward in frequency by f_c.

The negative frequency part of $S(\omega)$ is shifted upward in frequency by f_c.

In effect, a reflection of the vestige of the lower sideband makes up for the missing part of the upper sideband.







Advantages of VSB over SSB

⇒ Generation of an SSB signal is difficult in SSB

⇒ Selective filtering is to be done to get the original signal back in SSB

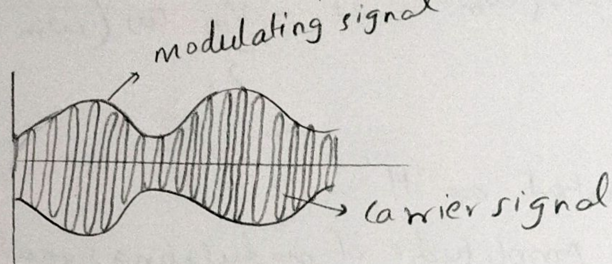
⇒ Phase shifter should be exactly tuned to 90° in SSB

These can be overcome in VSB

10) With necessary waveforms, derive the expression of AM and power relation in AM for sinusoidal modulation.

A:- **Amplitude Modulation** : (DSBFC - Double side Band Full carrier)

The Amplitude of high frequency carrier signal is varied in accordance with instantaneous value of baseband modulating signal's amplitude.



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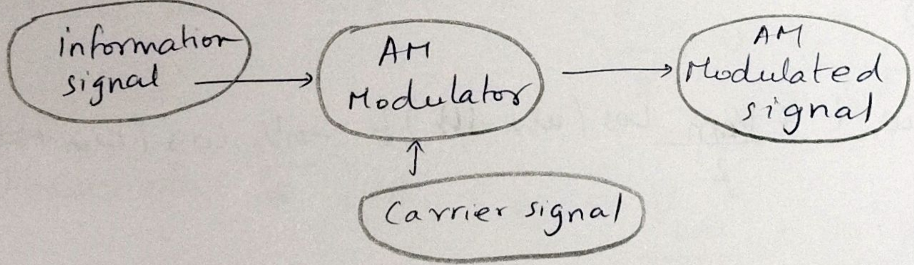
Derivation:

$v_c(t) = V_c \sin w_c t$ (carrier signal)

$v_m(t) = V_m \sin w_c t$ (modulating signal) → information



$V_{AM}(t)$ → AM Modulated signal

$$V_{AM}(t) = (V_c + v_m(t)) \sin w_c t$$

$$= (V_c + V_m \sin w_m t) \sin w_c t \qquad [v_m(t) = V_m \sin w_m t)$$

$$\Rightarrow V_c \sin w_c t + V_m \sin w_m t \sin w_c t$$

Since : $\sin\omega_m t \sin\omega_c t$ in the format of $\sin A \sin B$

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

Here $A = \omega_m$; $B = \omega_c$

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

$$V_{AM}(t) = V_c \sin\omega_c t + \frac{V_m}{2}\left[\cos(\omega_m - \omega_c)t - \cos(\omega_m + \omega_c)t\right]$$

$$V_{AM}(t) = V_c \sin\omega_c t + \frac{V_m}{2}\cos(\omega_m - \omega_c)t - \frac{V_m}{2}\cos(\omega_m + \omega_c)t \quad \rightarrow (1)$$

Modulation index :-

Modulation index denoted as 'M'.

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

$V_m \rightarrow$ Amplitude of modulating signal

$V_c \rightarrow$ Amplitude of carrier.

Rearranging to get V_m :-

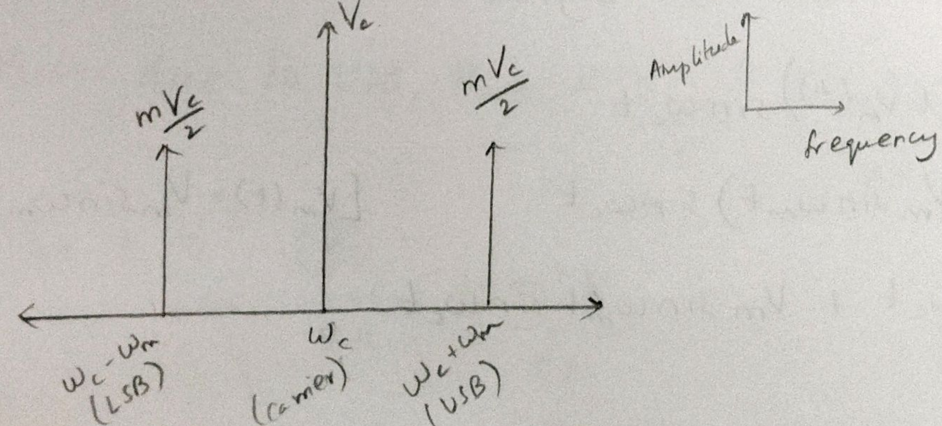
$$\boxed{V_m = m V_c} \longrightarrow (2)$$

note : $m \leq 1$ (for proper modulation) in AM

Sub (2) in (1)

$$V_{AM}(t) = V_c \sin\omega_c t + \frac{V_c m}{2}\cos(\omega_m - \cancel{\omega_c})t - \frac{mV_c}{2}\cos(\omega_m + \omega_c)t$$

Spectrum of AM wave :



Bandwidth: (B_{AM}):

The total Bandwidth of AM is obtained by their difference between 2 side bands.

$$B_{AM} = (f_c + f_m) - (f_c - f_m) = 2 f_m$$

Power relation of AM wave (P_{AM})

$$P_{AM} = \frac{V_{car}^2}{R} + \frac{V_{LSB}^2}{R} + \frac{V_{USB}^2}{R}$$

V_{car}^2, V_{LSB}^2, $V_{USB}^2 \rightarrow$ RMS values

$$V_{car}^2 = \left(\frac{V_c}{\sqrt{2}}\right)^2, \quad V_{LSB}^2 = \left(\frac{mV_c}{2\sqrt{2}}\right)^2, \quad V_{USB}^2 = \left(\frac{-mV_c}{2\sqrt{2}}\right)^2$$

Power due to carrier, $P_c = \frac{V_{car}^2}{R} \Rightarrow \frac{V_c^2}{2}\left(\frac{1}{R}\right) \Rightarrow \frac{V_c^2}{2R}$

power due to LSB, $P_{LSB} = \frac{V_{LSB}^2}{R} = \frac{m^2 V_c^2}{8}\left(\frac{1}{R}\right) = \frac{m^2 V_c^2}{8R}$

Power due to USB, $P_{USB} = \frac{V_{USB}^2}{R} = \frac{m^2 V_c^2}{8R}$

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

$$= \frac{V_c^2}{2R}\left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right) = \frac{V_c^2}{2R}\left(1 + \frac{m^2}{2}\right)$$

As we knowfrom $P_c = \frac{V_c^2}{2R}$

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

$$\boxed{\frac{P_{AM}}{P_c} = 1 + \frac{m^2}{2}}$$

* Transmission efficiency:

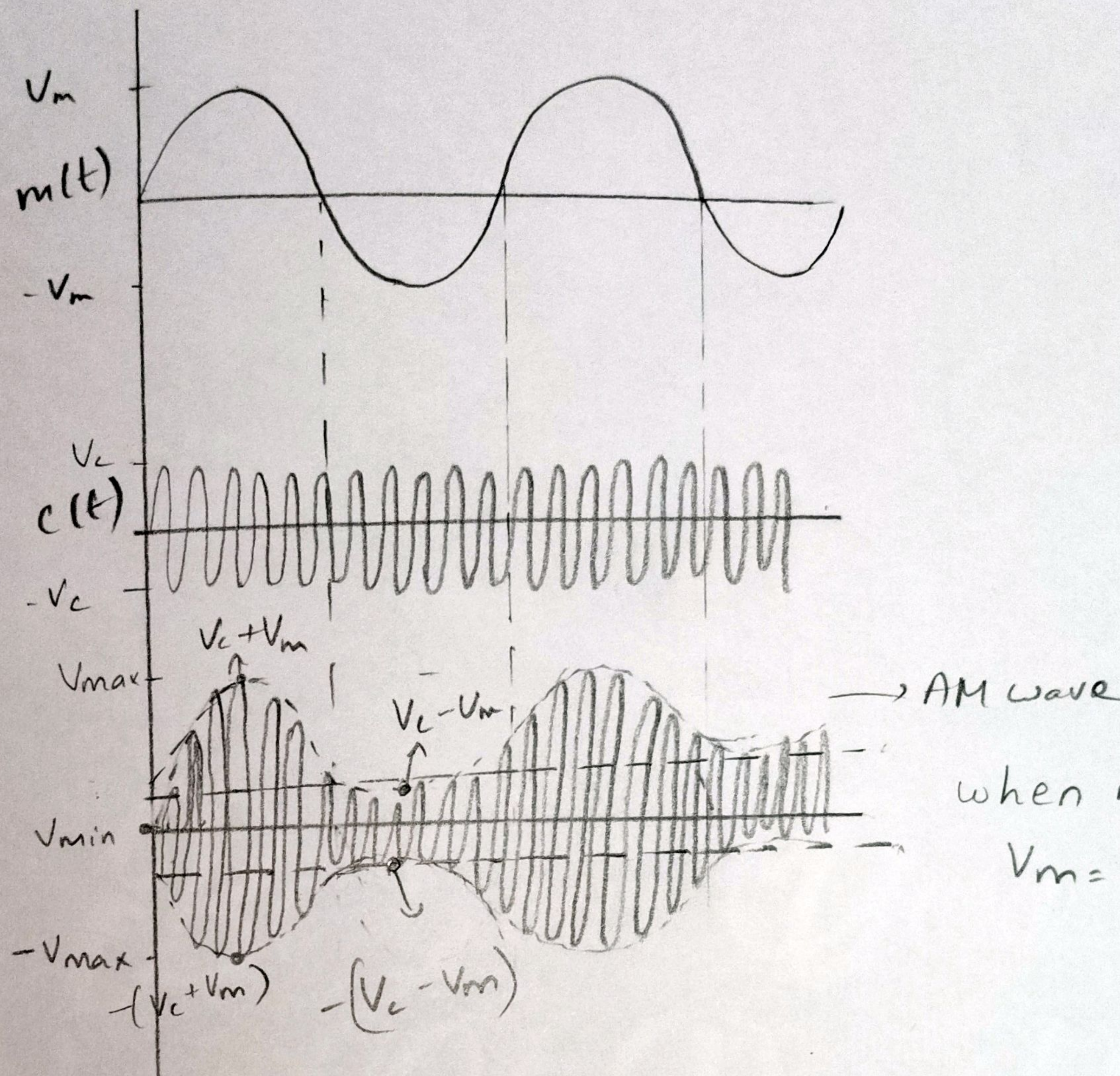
It is the ratio of power of comibined in both sidebands to the total power transmitted.

$$\frac{P_{USB} + P_{LSB}}{P_T} = \eta$$

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

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

Generation of AM waveform



when $m=1$

$V_m = V_c$

11. A sinusoidal carrier amplitude of 10 V and freq 10 kHz. It is amplitude modulated by a sinusoidal voltage of amplitude 3 V and frequency 1 kHz. Modulated voltage is developed across 50 Ω resistance.

a) write eqn for modulated wave

b) plot modulated wave

c) modulation index

d) spectrum of AM wave

11

sol. $V_c = 10\,V$; $f_c = 10\,kHz$; $R = 50\,\Omega$; $V_m = 3\,V$; $f_m = 1\,kHz$

c) $m = \frac{V_m}{V_c} = \frac{3}{10} = 0.3$

$\omega_m = 2\pi f_m = 2\pi \times 10^3$

$\omega_c = 2\pi f_c = 20\pi \times 10^3$

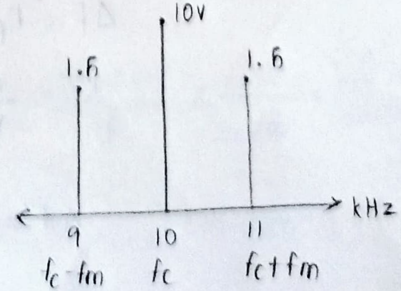
a) $V_{AM} = V_c(1 + m \sin \omega_m t) \sin \omega_c t$

$V_{AM} = 10(1 + 0.3 \sin(2\pi \times 10^3)t) \sin(20\pi \times 10^3)t$

d) $f_c + f_m = 11\,kHz$

$f_c - f_m = 9\,kHz$

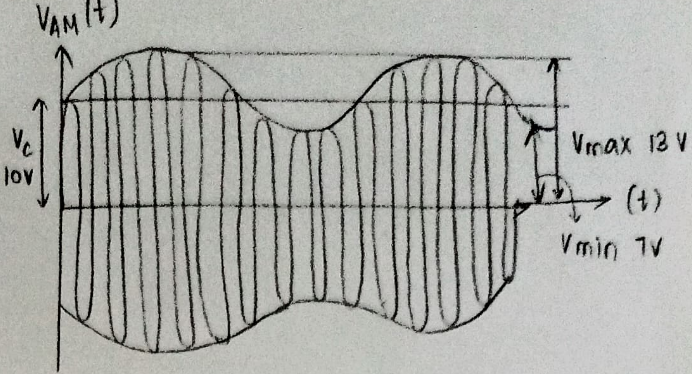
$\frac{mV_c}{2} = \frac{0.3 \times 10}{2} = 1.5\,V$



freq. spectrum of AM wave.

b) $V_{max} = V_c + V_m = 13\ V$

$V_{min} = V_c - V_m = 7\ V$



12. Explain the working of Super heterodyne reciever with a neat diagram.

A superheterodyne reciever uses frequency mixing to convert a recieved signal to a fixed intermediate Frequency (IF) which is conveniently processed than the original carrier frequency.

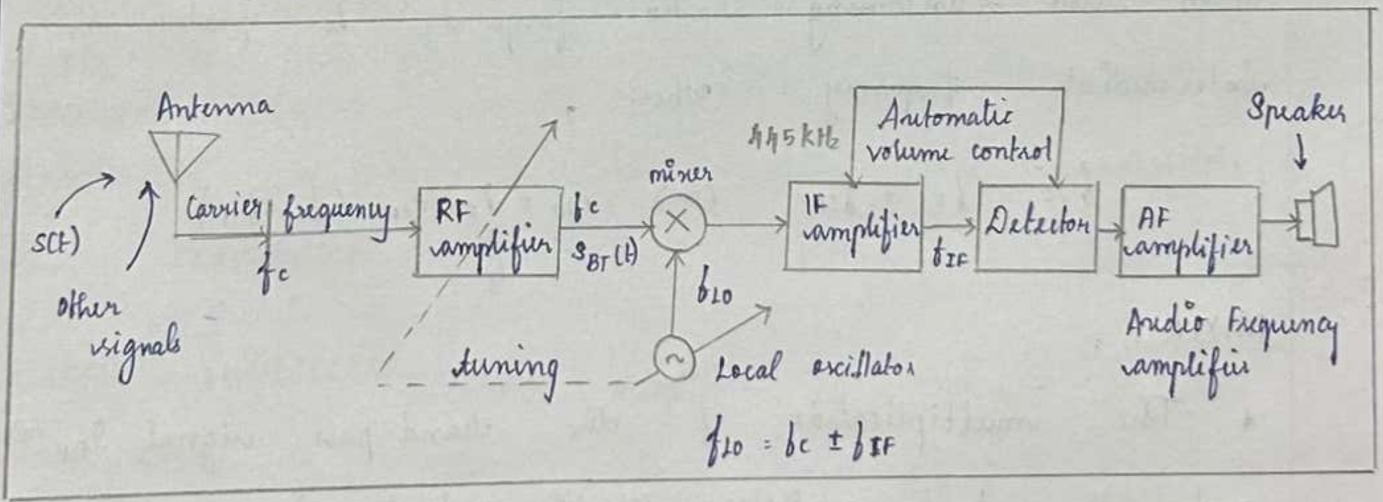
It satisfies three function of a reciever:

→ Carrier - frequency tuning

→ Filtering

→ Amplification.

12



The Superheterodyne reciever consists of:

<u>ANTENNA</u>: The incoming amplitude modulated wave is picked up by the recieving antenna

RADIO FREQUENCY (RF) TUNED AMPLIFIER :

* The modulated signal recieved is given to RF amplifier.
* The RF amplifier has a band-pass characteristics that passes the desired signal (Pre-selector) and provides amplification, rejecting adjacent signals and noise.
* It also tunes the Local oscillator and RF filter at the same time.

MIXER :

LOCAL OSCILLATOR : It generates a signal that mixes with the incoming radio frequency to produce an intermediate frequency. where,

$$f_{LO} = f_c + f_{IF} \quad (or) \quad f_{LO} = f_c - f_{IF} \; (f_c > f_{IF})$$

MIXER :

* The multiplication of the band-pass signal $S_{BP}(t)$ and the local oscillator signal output is called mixing.
* The incoming signal is converted to a pre-determined fixed intermediate frequency (f_{IF}) (lower than f_c)

$$f_{IF} = f_{LO} - f_c$$

f_{LO} → local oscillator frequency
f_c → carrier frequency
f_{IF} → intermediate frequency (neither input frequency nor

final ~~band~~ baseband frequency)

* In superheterodyne receiver, every AM radio signal is converted to a common IF of $f_{IF} = 455 kHz$

* This allows for the use of single tuned IF amplifier for signals from any radio station.

*

INTERMEDIATE FREQUENCY AMPLIFIER:

* The IF section provide most of the amplification and selectivity in the receiver.

* The IF filter is a Band pass filter that provides most of the gain.

* The IF amplifier is designed to have a
ENVELOPE DETECTI
bandwidth of 5kHz, which matches the bandwidth of the transmitted signal.

ENVELOPE DETECTOR: The output is applied to a demodulator to recover the base band signal.

AUDIO FREQUENCY AMPLIFIER & LOUD SPEAKER:

* The final operation is the power amplification of the recovered message signal.

* Loud speaker to convert electrical audio signal into sound wave.

13) $u(t) = 100 \cos[2\pi f_c t + 4 \sin 200 \pi t]$

$f_c = 10\ MHz$

a) $P = \frac{A_c^2}{2} = \frac{100^2}{2} = \frac{10000}{2} = 5000\ W$

b) $u(t) = 100 \cos[\theta(t)] \rightarrow (1)$

$\theta(t) = w_c t + \phi_c + K_p V_m \cos w_m t \rightarrow (2)$

maximum deviation $\delta_p = K_p V_m$

By Comparing (1) and (2)

$\delta_p = 4$

13

c)

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

$\frac{d\,\theta(t)}{dt} = 2\pi f_c + 4[2000\pi \cos(200\pi t)]$

$f_i(t) = f_c + \frac{\cancel{4}^{2}[2000 \cancel{\pi} \cos(2000\pi t)]}{\cancel{2}\cancel{\pi}}$

$f_i(t) = f_c + 4000 \cos(2000\pi t) \rightarrow (3)$

$f_i(t) = f_c + K_f m(t) \rightarrow (4)$

Comparing (3) and (4)

$\Delta f_p = 4000$ Hz where $\cos(2000\pi t) = \pm 1$

d)

$\theta(t) = 2\pi f_c t + 4 \sin(2000\pi t)$

since phase deviation is directly proportional to message signal $m(t) = \sin(2000\pi t)$

this corresponds to Phase Modulation (PM)