Generate DSBSC by product modulator.

→ To generate the DSBSC wave we need to take the product of $x(t)$ and carrier $c(t)$. So we need to use a device called product modulator for the generation of DSBSC waves.

There are two forms of product modulators as under:

I. linear modulator and II. Non-linear Modulator.

I. linear modulator for DSBSC generation

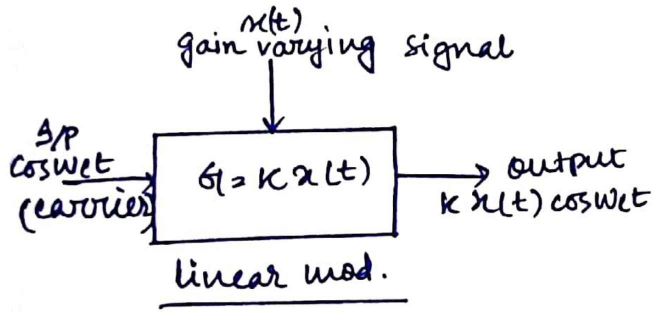
A linear modulator is a system whose gain or transfer function can be varied with time by applying a time varying signal at certain point. The gain is proportional to signal $f(t)$.

$$G = Kf(t)$$

where, G = gain & K = constant of proportionality.

$f(t)$ = gain varying signal.

Let us consider the diagram in which the carrier signal $\cos\omega_c t$ is applied at the input terminal and $x(t)$ is applied as the gain varying signal.



The gain of the modulator is $G = Kx(t)$.

Therefore, the output is given by

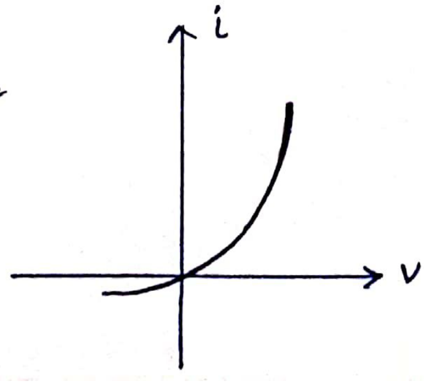
$$V_0 = G \cdot \text{input}$$

$$= Kx(t) \cos\omega_c t$$

The modulation of the signal is called DSBSC.

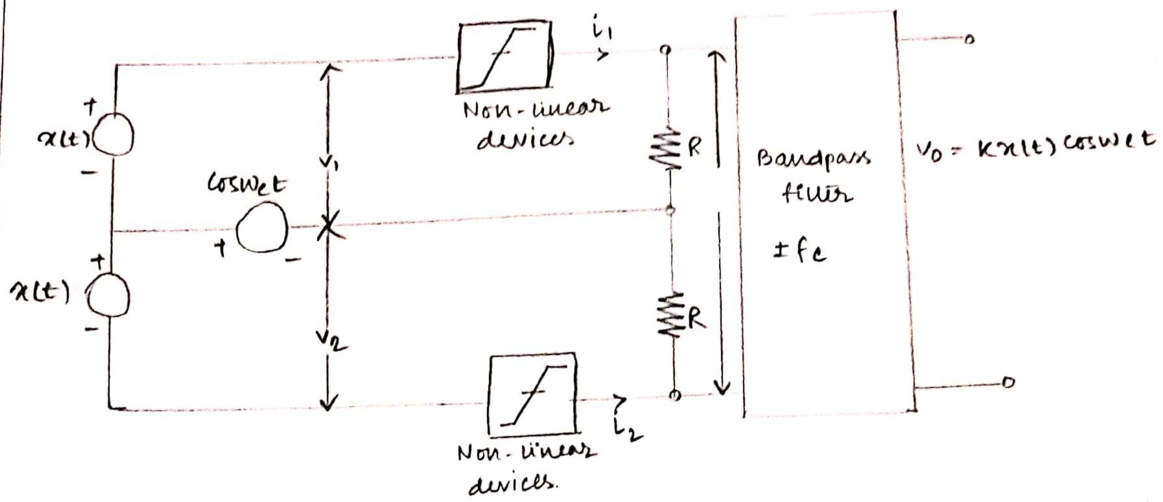
II. Modulation using Non-linear Devices.

The modulation can also be acheived by using the non-linear devices. The graph shows the typical non-linear device characteristics. A semiconductor diode is a good example of a non-linear device. The relation between the voltage across a device (v) and the device current (i) as non-linear as shown in the graph, and it can be mathematically expressed as ~~linear~~



$$i = av + bv^2$$

where a and b are constants.



DSBSC modulator using non-linear devices

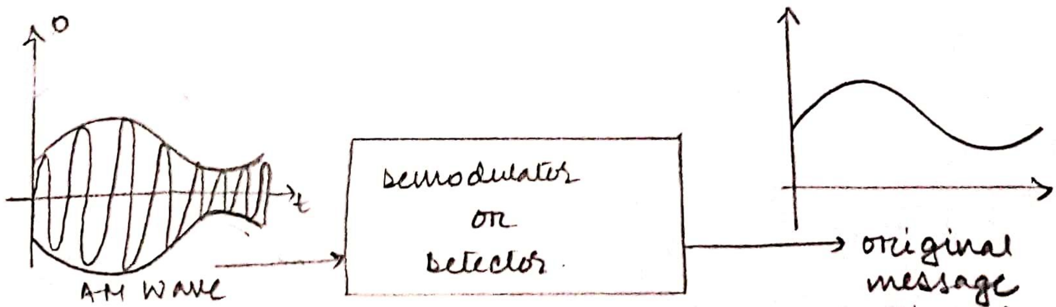
This diagram shows a possible arrangement for the use of non-linear elements for DSBSC modulator.
Such an arrangement is called as Balanced modulator.

What are the different types of demodulators used in the AM detection?

→ The process of detection or demodulations 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.

Types of AM detectors-

I. Square law detector
II. Envelop detector.



Process of demodulation

What is frequency modulation ? Give the mathematical expression of frequency modulation ?

→ The frequency modulation is a type of angle modulation in which the instantaneous frequency $f_i(t)$ is varied in linear proportion with the instantaneous magnitude of the message signal $x(t)$. This is expressed mathematically as follows :

For F.M. $f_i(t) = f_c + k_f x(t)$

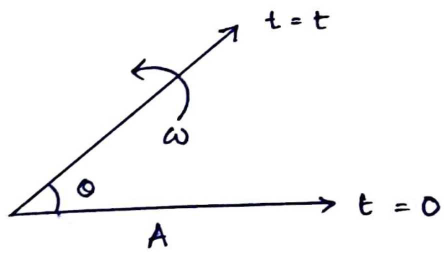
$f_{(i)}(t)$ = Instantaneous frequency

f_c = unmodulated carrier frequency

k_f = Frequency sensitivity Hz/volt.

Mathematic expression for FM :

We know that, the FM wave is a 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, the FM wave is represented ~~as~~ by.

$$s(t) = E_c \sin[F(\omega_c, \omega_m)] \quad \ldots\ldots (\text{I})$$

$$s(t) = E_c \sin\theta(t) \quad \ldots\ldots (\text{II})$$

where , $\theta(t) = F(\omega_c, \omega_m) \quad \ldots\ldots (\text{III})$

As shown in the figure, $E_c \sin\theta(t)$ is a rotating vector. If E_c is rotating at a constant velocity 'ω', then we could have written that $\theta(t) = \omega t$. But in FM, this velocity is not constant. Infact, it is changing continuously. The angular velocity of FM wave is given as,

$$\omega = [\omega_c + k E_m \cos\omega_m t]$$

Hence, to find $\omega(t)$, we must integrate ω with respect to time.

Therefore, $\theta(t) = \int \omega dt = \int [\omega_c + k E_m \cos\omega_m t] dt$

or, $\theta(t) = \omega_c \int \left[1 + \frac{KE_m}{\omega_c} \cos \omega_m t\right] dt$

$$= \omega_c \left[t + \frac{KE_m \sin \omega_m t}{\omega_c \, \omega_m}\right] = \omega_c t + \frac{KE_m \, \omega_c \sin \omega_m t}{\omega_m \, \omega_c}$$

or, $\theta(t) = \omega_c t + \frac{KE_m \sin \omega_m t}{f_m}$ ..... (IV)

As per the definition, $\Delta f = KE_m$

Thus, $\theta(t) = \omega_c t + \frac{\Delta f \sin \omega_m t}{f_m}$

Substituting this value of $\theta(t)$ in the equation we get the eqn for the FM wave as under:

$$s(t) = E_c \sin \left[\omega_c t + \frac{\Delta f}{f_m} \sin \omega_m t\right]$$

But, $\frac{\Delta f}{f_m} = m_f$ i.e. the modulation index of FM wave. Hence, the eqn for FM wave is given as under:

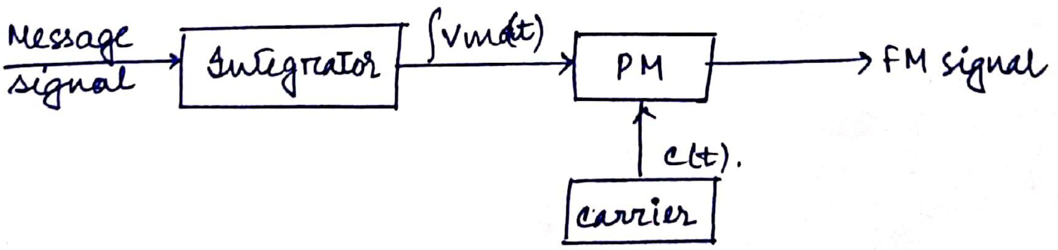
$$\boxed{s(t) = E_c \sin [\omega_c t + m_f \sin \omega_m t]}$$

– This is the expression for a FM wave, where m_f represents the modulation index.

Convert phase modulation wave to frequency modulation wave and vice versa.

→ Generation of FM using Phase modulator.

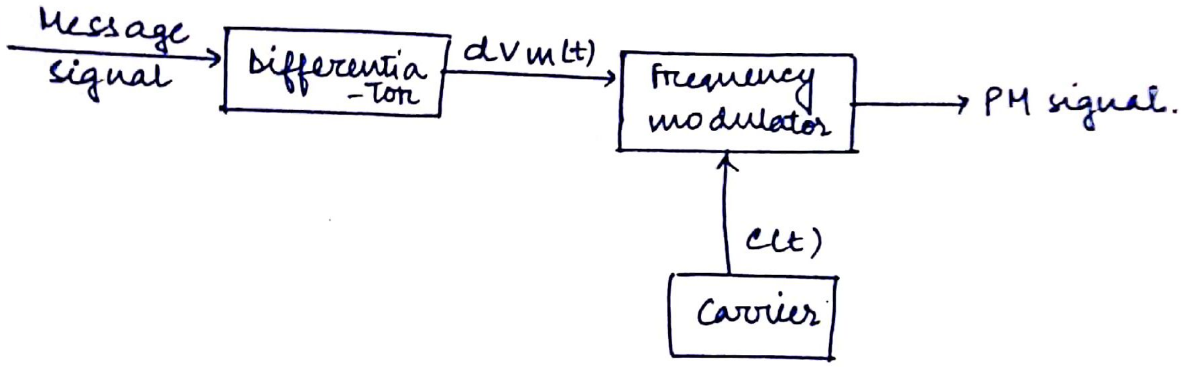
FM wave is actually a PM wave having a modulated signal $\int_0^t x(t)\,dt$ instead of $x(t)$.



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

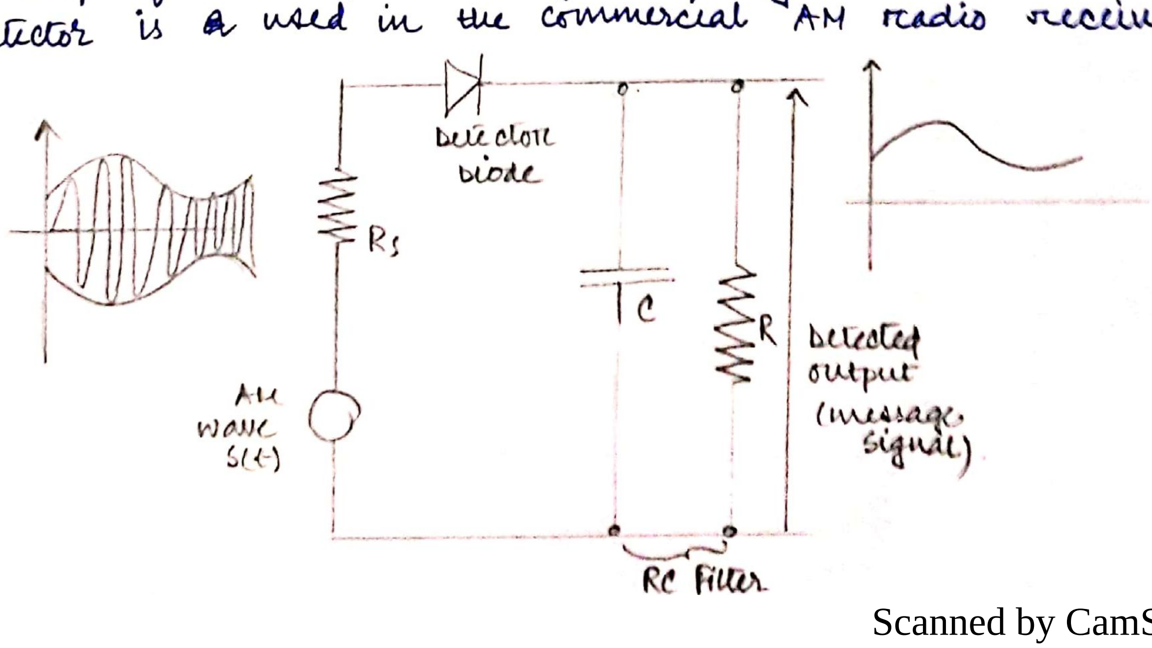
Generation of PM using a frequency modulator

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



Explain the working principle of Envelop Detector.

→ The envelop detector is a simple and very ~~different~~ efficient device which is suitable for the detection of a narrowband AM signal. A narrowband AM wave is the one in which the carrier frequency f_c is much higher as compared to the bandwidth of the modulating signal. An envelop detector produces an output signal that follows the envelop of the input AM signal exactly. The envelop detector is ~~a~~ used in the commercial AM radio receivers.



Working operation

The standard AM wave is applied at the input of the detector. In every positive half cycle of the input, the detector diode is forward biased. It will charge the filter capacitor C connected across the load resistance R to almost the peak value of the input voltage. As soon as the capacitor charges to the peak value, the diode stops conducting. The capacitor will discharge through R between the positive peaks. The discharging process continues until the next positive half cycle. When the input signal becomes greater than the capacitor voltage, the diode conducts again and the process repeats itself.

Nyquist rate

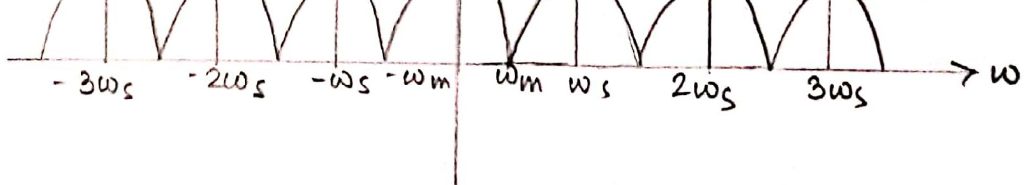
→ When the sampling rate becomes exactly equal to $2f_m$ samples per second, then it is called Nyquist rate. Nyquist rate is also called the minimum samping rate. It is given by,

$$f_s = 2f_m$$

Similarly, maximum sampling interval is called Nyquist interval. It is given by,

$$\text{Nyquist interval } T_s = \frac{1}{2f_m} \text{ seconds.}$$

When the continuous-time band-limited signal is sampled at Nyquist rate ($f_s = 2f_m$), the sampled-spectrum $G(\omega)$ contains non-overlapping $G(\omega)$ repeating periodacally. But the successive cycles of the $G(\omega)$ touch each other. Therefore, the original spectrum $X(\omega)$ can be recovered from the sampled spectrum by using a low pass filter with a cut-off frequency ω_m.



Sampled spectrum at Nyquist rate

State Nyquist Theory of Sampling

→ The Nyquist Theorem, also known as the sampling theorem, is a principle in the digitization of analog signals. For analog-to-digital conversion (ADC) to result in a faithful reproduction of the signal, slices called samples, of the analog waveform must be taken frequently. The number of samples per second is called the sampling rate or sampling frequency.

Any analog signal consists of components at various frequencies. The simplest case ~~of~~ is the sinewave, in which all the signal energy is concentrated at one frequency. In practice, analog signals usually have complex waveforms, with components at many frequencies. The highest frequency component in an analog signal determines the bandwidth of that signal. The higher the frequency, the greater the bandwidth, if all other factors are held constant.

Suppose, the highest frequency components, in Hz, for a given analog signal is f_{max}. According to the Nyquist theorem, the sampling rate must be at least $2f_{max}$ or twice the highest analog frequency component. The sampling in an analog-to-digital converter is actuated by a pulse generator (clock). If the sampling rate is less than $2f_{max}$, some of the highest frequency components in the analog input signal will not be correctly represented in the digitized output. When such a ~~a~~ digital signal is converted back to analog, false frequency components appear that were not in the original analog signal. This undesirable condition is a form of distortion called aliasing.

Write short note on aliasing.

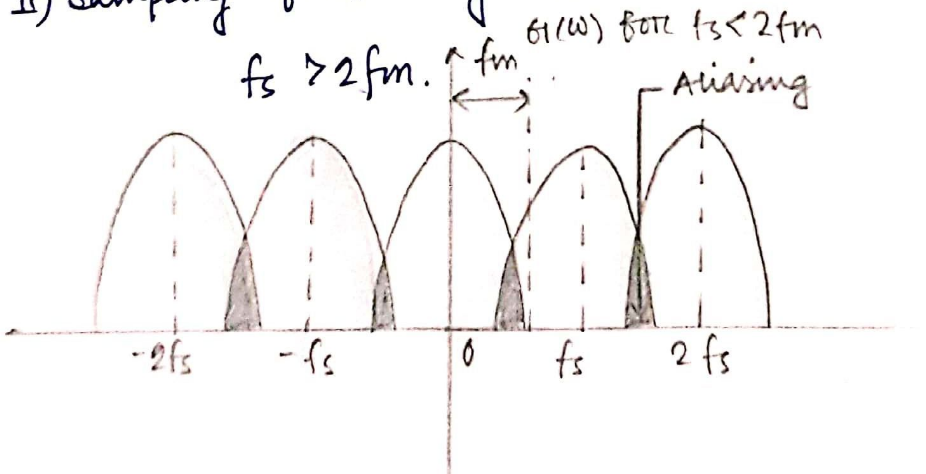
→ When a continuous-time band-limited signal is sampled at a lower rate lower than Nyquist rate $f_s < 2f_m$, then the successive cycles of the spectrum $G(\omega)$ of the sampled signal $g(t)$ overlap with each other.

Hence, the signal is under-sampled, in this case ($f_s < 2f_m$) and some amount of aliasing is produced in this under-sampling process. In fact, aliasing is the phenomenon in which a high frequency component in the frequency-spectrum of the signal takes identity of a lower-frequency component in the spectrum of the sampled signal. It's clear that because of the overlap due to aliasing phenomenon, it is not possible to recover original signal $x(t)$ from sampled signal $g(t)$ by low-pass filtering since the spectral components in the overlap regions add and hence the signal is distorted. Since any information signal contains a large no. of frequencies so, to decide a sampling frequency is always a problem. Therefore, a signal is first passed through a low-pass filter. This low pass filter blocks all the frequencies which are above f_m Hz. This process is known as band limiting of the original signal $x(t)$. This low pass filter is called prealias filter, because it is used to prevent aliasing effect. After band-limiting, it becomes easy to decide sampling frequency since the maximum frequency is fixed at f_m Hz.

In short, to avoid aliasing:

I) Prealias filter must be used to limit band of frequencies of the signal to f_m Hz.

II) Sampling frequency 'f_s' must be selected such that $f_s > 2f_m$.



Spectrum of the sampled signal for the case $f_s < 2f_m$

Prove the relation : $P_t = P_c\left(1+\frac{m_a^2}{2}\right)$

→ Let us consider that a carrier signal $A\cos\omega_c t$ is amplitude-modulated by a single-tone modulating signal $x(t) = V_m\cos\omega_m t$.

Then the unmodulated or carrier power,

P_c = mean square (ms value)

$$P_c = \overline{(A\cos\omega_c t)^2} = \frac{A^2}{2}$$

The sideband power, $P_s = \frac{1}{2}\overline{x^2(t)} = \frac{1}{2}\overline{(V_m\cos\omega_m t)^2} = \frac{1}{2}\frac{V_m^2}{2} = \frac{1}{4}V_m^2$

We know that the total modulated power P_t is the sum of P_c and P_s.

Therefore, $P_t = P_c + P_s = \frac{A^2}{2} + \frac{1}{4}\cdot V_m = \frac{A^2}{2}\left[1+\frac{1}{2}\left(\frac{V_m}{A}\right)^2\right]$

But, $\frac{V_m}{A} = \frac{\text{max. baseband amplitude}}{\text{max. carrier amplitude}} = m_a$ = modulation index for AM.

Hence, $P_t = \frac{A^2}{2}\left[1+\frac{1}{2}\cdot m_a^2\right]$

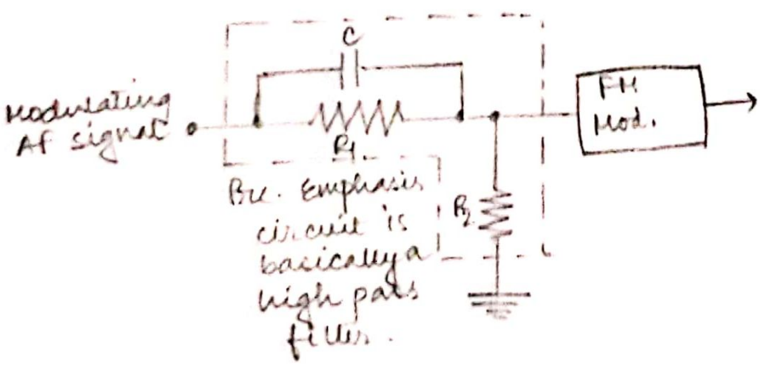
But, $\frac{A^2}{2} = P_c$ = carrier power

Therefore, $\boxed{P_t = P_c\left(1+\frac{m_a^2}{2}\right)}$. – Proved.

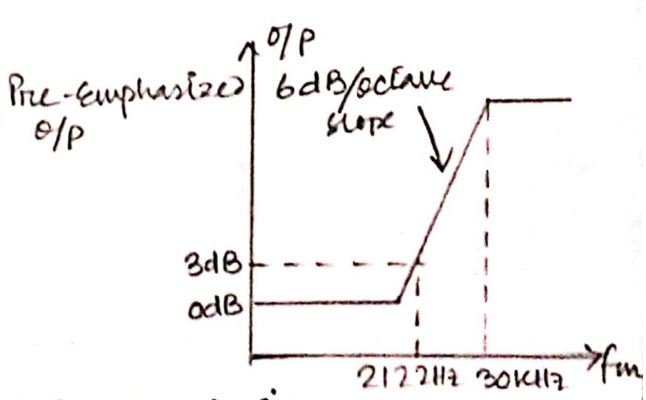
Write short note on Pre-Emphasis and De-Emphasis.

→ Pre-emphasis

Noise occurs primarily at high frequencies; therefore, noise interferes more with high modulating frequencies. Interference from high-frequency noise can be minimized by boosting the amplitude of high-frequency modulating signals prior to modulation. This is called pre-emphasis.

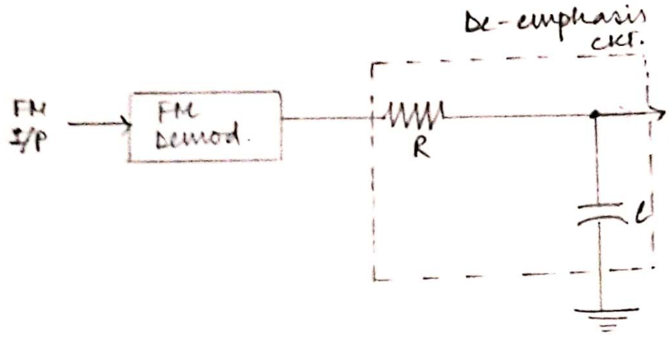


(a) Typical pre-emphasis circuit

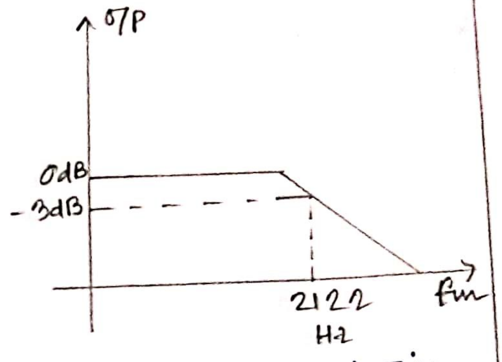


(b) Pre-emphasis characteristics.

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



(a) De-emphasis circuit



(b) Its characteristics

Write down the expression of NBFM? What are the differences of NBFM and AM?

→ NBFM wave and mathematical derivation

The expression for FM wave is,

$$V = E_c \cos[\omega_c t + m_f \sin\omega_m t] = E_c[\cos\omega_c t \cos\{m_f \sin\omega_m t\} - \sin\omega_c t \sin\{m_f \sin\omega_m t\}]$$

Assuming the modulation index to be compared to 1 radian then we may use following approximations, $\cos\{m_f \sin(\omega_m t)\} = 1$ and $\sin\{m_f \sin(\omega_m t)\} = m_f \sin(\omega_m t)$

So the above expression can be written as,

$$V = E_c \cos\omega_c t - E_c \sin\omega_c t \cdot m_f \sin\omega_m t = E_c \cos\omega_c t - \frac{m_f E_c}{2}[2\sin\omega_c t \sin\omega_m t]$$

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

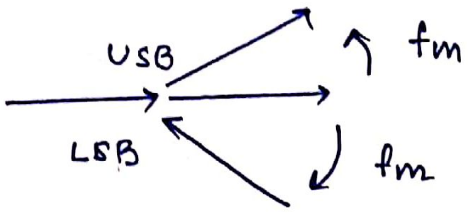
$$= E_c \cos\omega_c t - \frac{m_f E_c}{2}\cos(\omega_c - \omega_m)t + \frac{m_f E_c}{2}\cos(\omega_c + \omega_m)t.$$

The above expression is an approximate expression for a narrow band FM produced by base band signal,

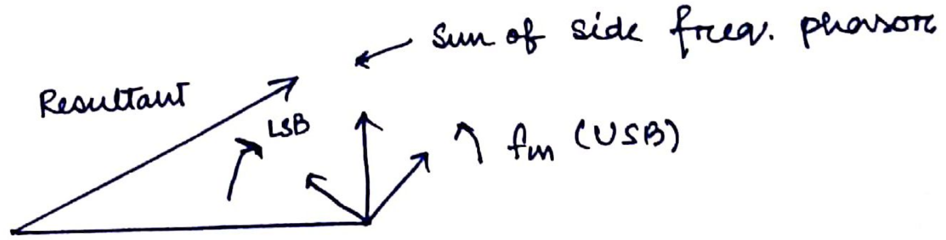
$$e_m(t) = E_m \cos(\omega_m t).$$

In case of AM, the resultant of two side frequency phasors ~~is~~ is always in phase with carrier phasor, whereas in NBFM, resulting phasor, the carrier phasor is out of phasor with respect to it. Both of them are represented in the following fig.

phasor diagram of AM -



phasor diagram of NBFM -



What is delta modulation? Write down its limitations.

→ Delta modulation (DM or δ- modulation) is an analog to digital and digital-to-analog signal ~~modulation~~ conversion technique used for tranmission of voice information where quality is not of primary importance. DM is the simpliest form of differential pulse-code modulation (DPCM) where the differences between successive samples are encoded into n-bit data streams.

Rather than quantizing the absolute value of the input analog waveform, delta modulation quantizes the difference between the current and the previous step, as shown in the diagram.

■ **Limitations of Delta modulation** –

The Delta modulation has two major drawbacks as under:

i) slope overload distortion;

ii) Granular or idle noise

How can we overcome the limitation of delta modulation?

→ Adaptive delta modulation (ADM) or continuously variable slope delta modulation (CVSD) is a modification of DM in which the step size is not fixed. Rather, when several consecutive bits have the same direction value, the encoder, and decoder assume that the slope overload is occuring, and the step size becomes progressively larger. Otherwise, the step size becomes gradually smaller over time. ADM reduces slope error, at the expense of increasing quantizing error. This error can be reduced by using a low pass filter. The block diagram of ADM –

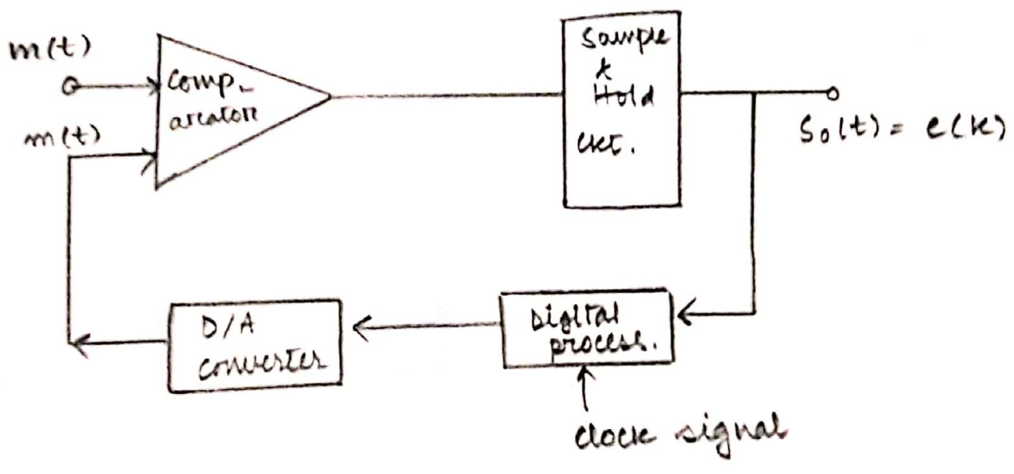


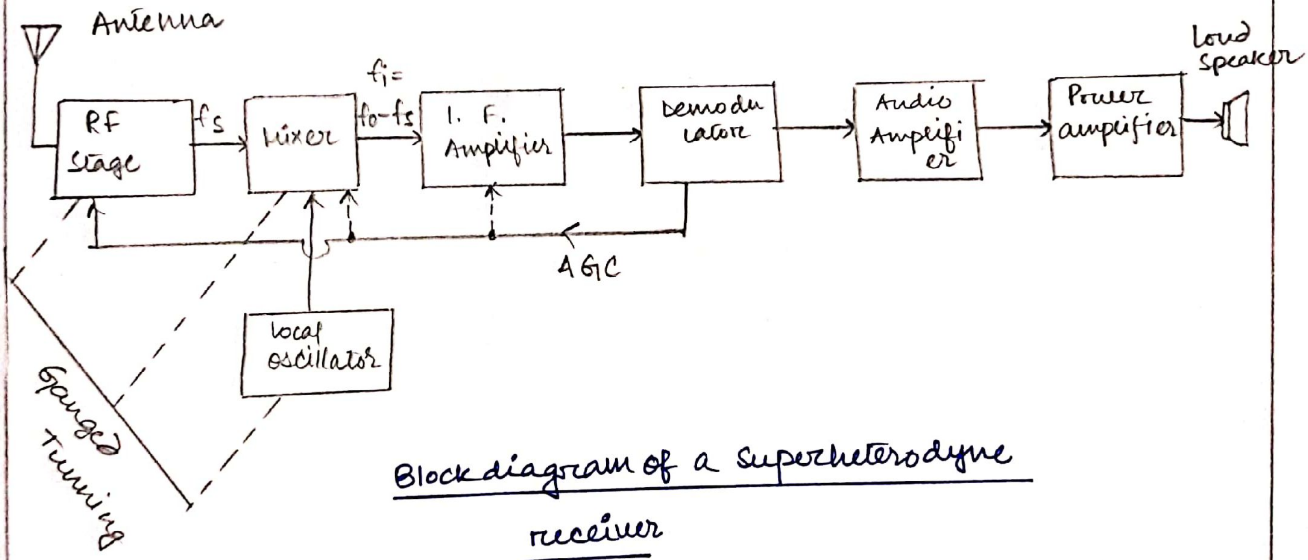
Fig. ADM.

Draw and explain the block diagram of super heterodyne receiver.

→ The figure shows the block diagram of a superhetrodyne receiver. All the drawbacks in TRF receiver have been removed in a superhetrodyne receiver. The basic superhetrodyne receiver is most widely used. This means that the superhetero dyne principle is used in all types of receiver like Television receiver, radar, receiver, etc.

In a superhetrodyne receiver, the incoming RF signal frequency is combined with the local ~~or~~ oscillator signal frequency through a mixer and is converted into a signal of lower fixed frequency. This lower frequency is known as intermediate frequency. However, the intermediate frequency signal contains the same modulation as the original signal. This ~~signal~~ intermediate frequency signal is now amplified and demodulated to reproduce the original signal.

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



Block diagram of a superheterodyne receiver

- **Advantages of Superheterodyning**

I) No variation in bandwidth. The BW remains constant over the entire operating range.

II) High sensitivity and selectivity.

III) High adjacent channel rejection.

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