# ANALOG COMMINICATION

**JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY: KAKINADA**
**KAKINADA – 533 003, Andhra Pradesh, India**
**DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING**

| II Year-II Semester | | L | T | P | C |
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| **ANALOG COMMUNICATIONS** | | | | | |

**Course Objectives:**

Students undergoing this course are expected to

- Familiarize with the fundamentals of analog communication systems.
- Familiarize with various techniques for analog modulation and demodulation of signals.
- Distinguish the figure of merits of various analog modulation methods.
- Develop the ability to classify and understand various functional blocks of radio transmitters and receivers.
- Familiarize with basic techniques for generating and demodulating various pulse modulated signals.

**UNIT I**

**AMPLITUDE MODULATION :** Introduction to communication system, Need for modulation,Frequency Division Multiplexing , Amplitude Modulation, Definition, Time domain and frequencydomain description, single tone modulation, power relations in AM waves, Generation of AMwaves, square law Modulator, Switching modulator, Detection of AM Waves; Square law detector,Envelopedetector.

**UNIT II**

**DSB & SSB MODULATION:** Double side band suppressed carrier modulators, time domain and frequency domain description, Generation of DSBSC Waves, Balanced Modulators, Ring Modulator, Coherent detection of DSB-SC Modulated waves, COSTAS Loop. Frequency domaindescription, Frequency discrimination method for generation of AM SSB Modulated Wave, Time domain description, Phase discrimination method for generating AMSSB Modulated waves. Demodulation of SSB Waves, Vestigial side band modulation: Frequency description, Generation of VSB Modulated wave, Time domain description, Envelope detection of a VSB Wave pulse Carrier, Comparison of AM Techniques, Applications of different AM Systems, FDM.

**UNIT III**

**ANGLE MODULATION**: Basicconcepts, Frequency Modulation: Single tone frequency modulation, Spectrum Analysis of Sinusoidal FM Wave, Narrowband FM, Wideband FM,Constant Average Power, Transmission bandwidth of FM Wave- Generation of FM Waves, Detection of FM Waves: Balanced Frequency discriminator, Zero crossing detector, Phase lockedloop. Comparison of FM & AM.

**UNIT IV**

**TRANSMITTERS & RECEIVERS: Radio Transmitter** - Classification of Transmitter, AM Transmitter, Effect of feedback on performance of AM Transmitter, FM Transmitter –Variable reactance type and phase modulated FM Transmitter, frequency stability in FM Transmitter. **Radio Receiver** - Receiver Types - Tuned radio frequency receiver, Super heterodyne receiver, RF section and Characteristics - Frequency changingand tracking, Intermediate frequency, AGC, FM Receiver, Comparison with AM Receiver, Amplitude limiting. Communication Receivers, extensions of super heterodyne principle and additional circuits.

**UNITV**

**NOISE:** Review of noise and noise sources, noise figure**,** Noise in Analog communication Systems, Noise in DSB & SSB System, Noise in AM System, Noise in Angle Modulation Systems, Threshold effect in Angle Modulation System, Pre-emphasis & de-emphasis **PULSE MODULATION:** Types of Pulse modulation, PAM (Single polarity, double polarity) PWM: Generation & demodulation of PWM, PPM, Generation and demodulation of PPM, Time Division Multiplexing, TDM Vs FDM

**TEXTBOOKS:**

1. Principles of Communication Systems–HTaub&D.Schilling, GautamSahe, TMH, 3rd Edition, 2007.
2. Principles of Communication Systems-Simon Haykin, John Wiley, 2nd Edition, 2007.
3. Modern Digital and Analog Communication Systems –B.P.Lathi, Zhi Ding, Hari Mohan Gupta, Oxford University Press, 4th Edition, 2017

**REFERENCES:**

1. Electronics & Communication System– George Kennedyand Bernard Davis, TMH 2004.
2. Communication Systems–R.P.Singh, SP Sapre, Second Edition TMH, 2007.
3. Electronic Communication systems–Tomasi, Pearson, fourth Edition, 2007.

**Course Outcomes:**

After undergoing the course, students will be able to

- Differentiate various Analog modulation and demodulation schemes and their spectral characteristics
- Analyze noise characteristics of various analog modulation methods
- Analyze various functional blocks of radiotransmitters and receivers
- Design simple analog systems for various modulation techniques

# 1. AMPLITUDE MODULATION



## Introduction to communication system:

Communication:

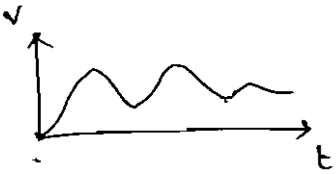
Communication is defined as the basic process of exchanging information.

→ An information signal is transmitted from transmitter to receiver.

→ Based on the nature of signal transmitted, communication system is of 2 types. ① Analog communication system.
② Digital communication system.

① Analog communication system: The message signal used in this system is analog in nature. A signal whose amplitude varies continuously with respect to time period is analog signal.

Eg: sinewave, cos wave,



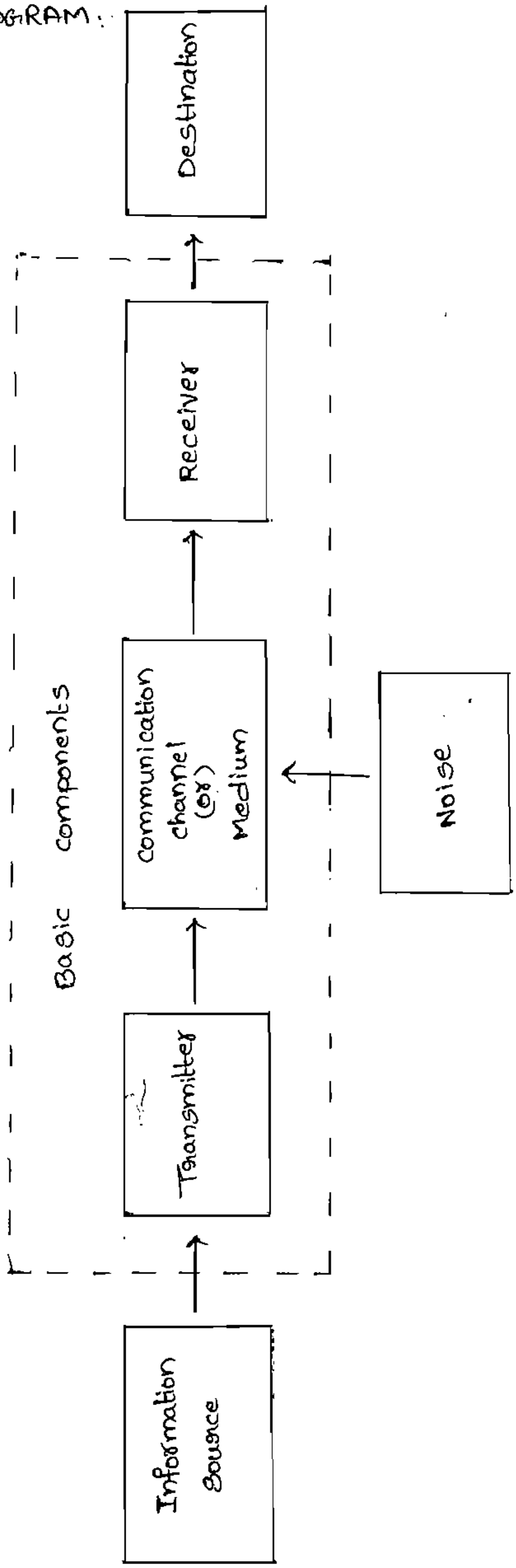

② Digital communication system: The message signal to be transmitted is digital in nature. A signal in which the sequence of numbers represent the amplitude at any instant of time is called digital signal. It is discrete in nature.

Eg:


→ Digital communication systems are preferred over analog communication systems.

Basic elements (or) components of a communication system:

BLOCK DIAGRAM:



The basic components of communication system are transmitter, a communication channel or medium and a receiver.

① Information :

The information source contains message that is to be transmitted. The two main sources of information are the ideas emanating from the human brain and changes in any physical environment. The amount of information contained in any given message is measured in bits or dibs.

② Transmitter :

The transmitter is a collection of electronic circuits designed to convert the information into a signal suitable for transmission over a given communication medium.

Some message signals that comes from information source is non-electrical and it is not suitable for transmission. Inorder to transmit the signal, it should be converted into electrical. The built-in circuitry such as decoders, encoders, transducers etc. in the transmission makes incoming information suitable for transmission and subsequent reception.

③ Communication channel :

The communication channel is the medium by which the electronic signal is transmitted from one place to another. Depending on the type of medium, the communication system can be classified as

- wire communication (or) Line communication
- wireless (or) Radio communication.

④ Noise:

Noise is an unwanted random signal which interfere with the message signal launched by the transmitter. This problem is particularly common in broadcasting, where two or more signals may be picked up at the same time by receiver which inturn will produce different message.

Noise in communication system can be classified as internal noise and external noise. Noise generated by components with in a communication system such as resistors, diode and transistors are referred to as internal noise. The external noise results from sources outside a communication system such as atmospheric and man-made.

⑤ Receiver:

A receiver is a collection of electronic circuits, designed to convert the signal back to the original information. It consists of amplifier, decoder, mixer, oscillator, transducer and so on. The output transducer converts the electrical message signal into its original form.

BASE BAND SIGNAL:

The original message signal whether it may be analog or digital is called a base band signal.

→ The frequency and amplitude of a base band signal is low and hence it can not travel longer distances. These signals should be modulated for travelling long distance.

# MODULATION :

**Def :** The process of changing characteristics (amplitude, frequency, phase) of the carrier (high frequency) signal with respect to instantaneous amplitude values of the modulating (message or low ~~frequ~~ frequency) signal is known as modulation.

## Need for modulation :

Modulation technique is used for the communication of base band signal. The need for modulation (or) advantages of modulation is given below:

1. Reduces the height of antenna.
2. Avoids mixing of signal.
3. Increases the range of communication
4. Allows multiplexing of signals
5. Allows adjustments (or) changes in bandwidth.
6. Improves quality of reception.

### ① Reduces the height of antenna :

The height of antenna required for transmission and reception of radiowaves in radio transmission is a function of wavelength of frequency used. The minimum height of antenna required for perfect (i.e., 100 %) modulation should be

$$h = \lambda/4$$

and $\lambda = \frac{c}{f}$ where 'c' is the velocity of light and f is the frequency.

The aboqve equation said that, at low frequencies, wavelength of the signal is high and the height of antenna also be high.

For example, consider a signal with frequency $f = 15kHz$

height of antenna $h = \frac{\lambda}{4} = \frac{c}{f \times 4} = \frac{3\times10^8}{15k\times4}$

$$= \frac{3\times10^8}{15\times10^3\times4}$$

$= 5000$ meters.

$\Rightarrow$ Antenna height should be 5 km.

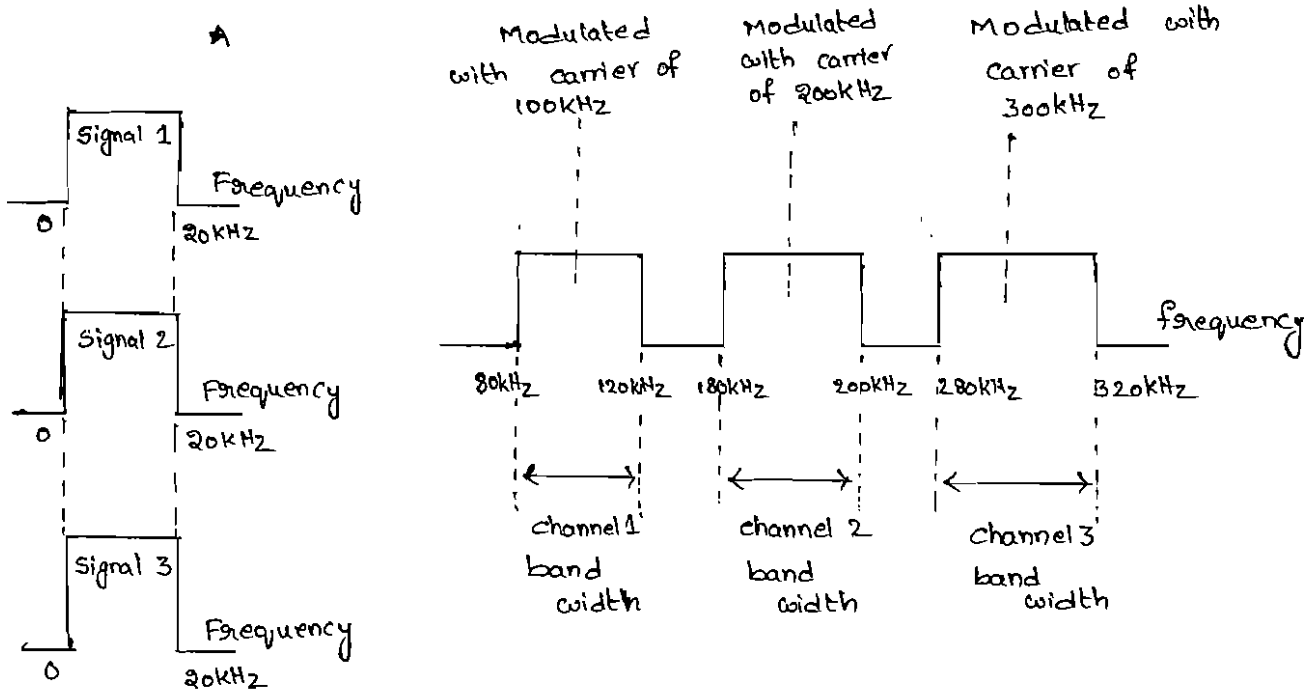
Now consider a signal of frequency $f = 1MHz$

height of antenna $h = \frac{\lambda}{4} = \frac{c}{f\times4} = \frac{3\times10^8}{1M\times4} = \frac{3\times10^8}{1\times10^6\times4}$

$$= \frac{300}{4} = 75 \text{ meters.}$$

$\Rightarrow$ Antenna height is 75 meters which is practically possible and can be installed.

② Avoids mixing of signals:



All sound signals are concentrated within the range from 20 Hz to 20kHz. There is a problem of mixing of signals with same frequency range during transmission. Then it is difficult to separate at the receiver end. this problem can be overcome by modulating carrier signal by different carrier frequencies. once the signals have been transmitted, a tuned circuit at the receiver end selects the ~~so~~ different signals it is tuned for, as shown in the above figure.

③ Increases the range of communication:

At low frequencies radiation is poor and signal gets highly attenuated. Therefore baseband signals cannot be transmitted directly over long distance. Modulation effectively increases the frequency of the signal to be radiated and thus increases the distance over which signals can be transmitted faithfully.

④ Allows multiplexing of signals:

Multiplexing means transmission of two or more signals simultaneously over the same channel. The different signals from different stations can be separated in the receiver since the carrier frequencies for these signals are different. It is commonly known as tuning the receiver to the desired station. By tuning process, the desired signal is selected and at the same time, other unwanted signals are rejected.

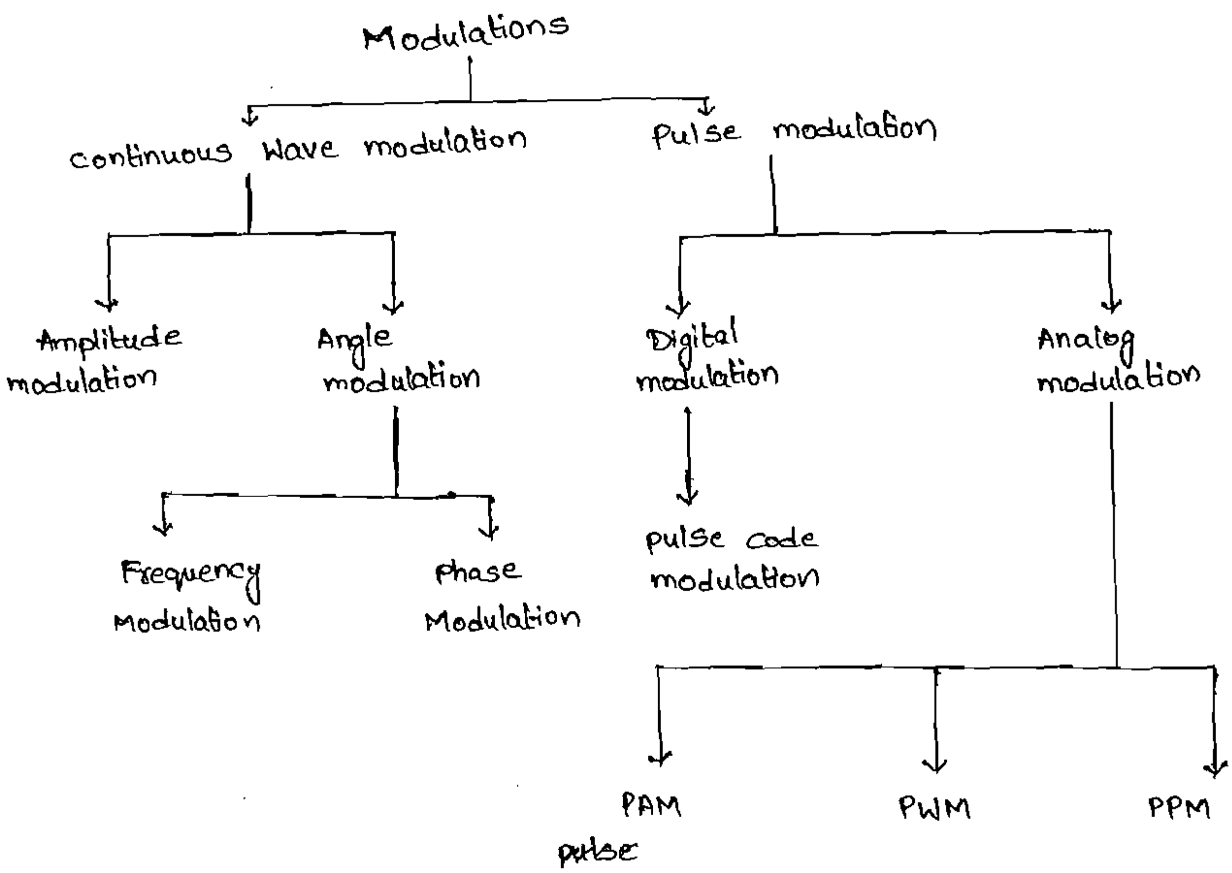
⑤ Allows adjustments in the bandwidth:

Bandwidth of a modulated signal may be made smaller or larger than the original signal. signal to noise ratio in the receiver which is a function of the signal bandwidth can thus be improved by proper control of the bandwidth at the modulating stage.

⑥ Improves quality of reception :

The signal communication using modulation techniques such as frequency modulation, pulse code modulation reduce the effect of noise to great extent. Reduction in noise improves the quality of reception.

## Types of modulation :




PAM - pulse amplitude modulation

PWM - Pulse width modulation

PPM - pulse position modulation

# MULTIPLEXING :

**Def:** Multiplexing is defined as a process of simultaneous transmissio of several messages over a single channel in a way that they do not interfere with each other.

→ At the receiving end, individual signals can be extracted from the multiplexed signal by demultiplexing process.

→ Demultiplexing means one-to-many.

→ Multiplexing means many-to-one.

## Applications :

Multiplexing finds use in most modern day telecommunication processes like i) telephony

ii) satellite communication

iii) wireless communication

iv) telemetry ...etc.

These are mainly two types of multiplexing methods in use.

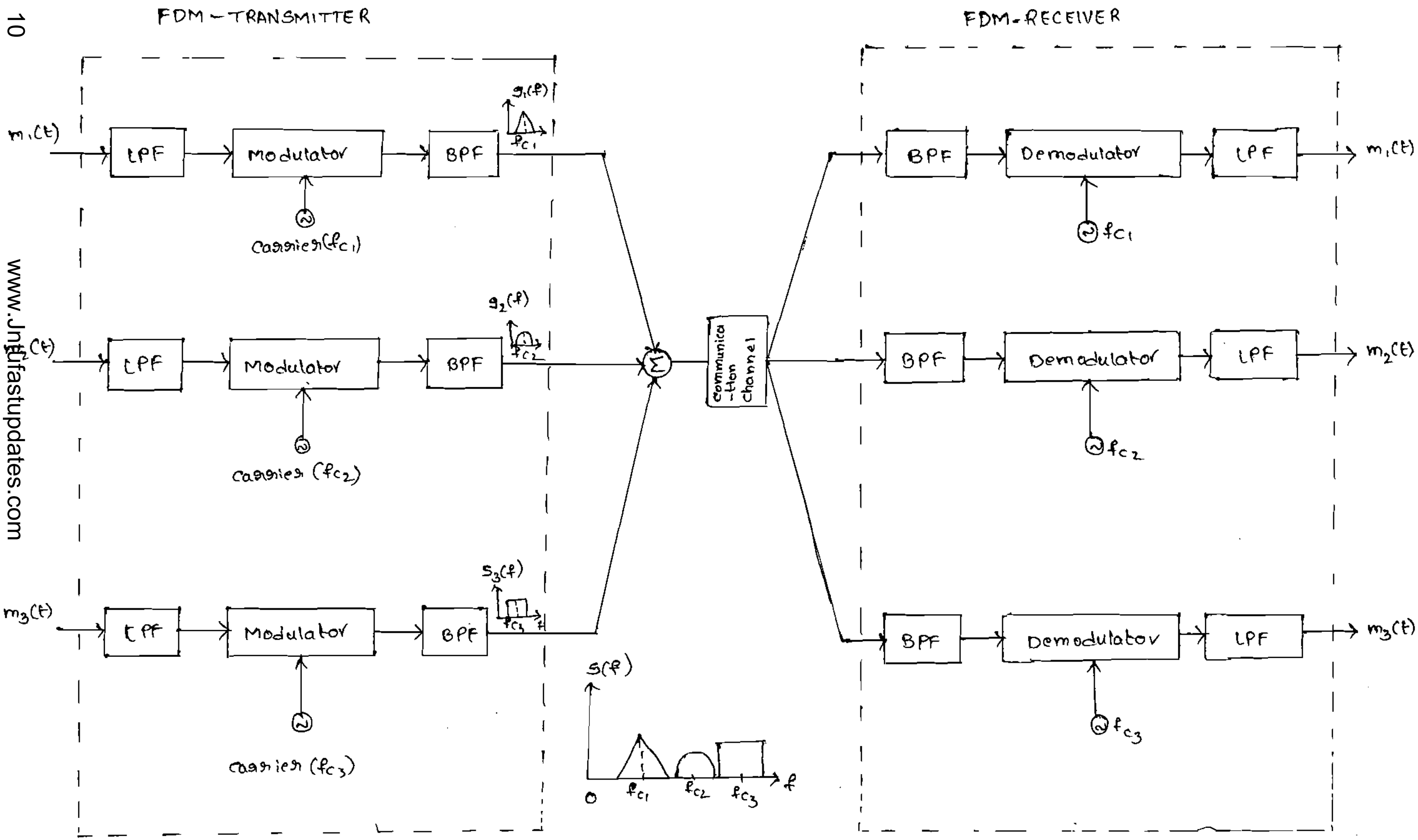
1) Frequency division multiplexing.

2) Time division multiplexing.

## 1) Frequency Division MULTIPLEXING :

In frequency division multiplexing (FDM), the total channe bandwidth available to the system is divided into smaller non-overlapping frequency bands (sub-bands), which enables the receiver to separate them and recover the individual messages.

The block diagram of a typical FDM system is shown in figure. consider three message signals $m_1(t)$, $m_2(t)$, $m_3(t)$.


FDM - TRANSMITTER
FDM - RECEIVER
m1(t)
m2(t)
m3(t)
LPF
Modulator
BPF
carrier (fc1)
carrier (fc2)
carrier (fc3)
g1(f)
g2(f)
S3(f)
fc1
fc2
fc3
Σ
communica-tion channel
BPF
Demodulator
LPF
fc1
fc2
fc3
m1(t)
m2(t)
m3(t)
S(f)
0
fc1
fc2
fc3
f

The Low Pass Filter (LPF) at the transmitter section of FDM system, removes high frequency components of message signal that disturbs the other message signals that share the channel. These filtered signals modulate the carriers having the frequencies $f_{c_1}, f_{c_2}, f_{c_3}$ so that the resulting modulated waves do not have overlapping spectra.

The band Pass filters (BPF) after the modulators are used to restrict the band of each modulated wave to prescribed range. All these signals are now combined to produce a composite signal $s(t)$ which is then transmitted over the channel.

In the receiving end, suitable BPF's, tuned to the carrier frequencies are then employed to recover each of the modulated carriers carrying the individual signals. Each modulated signal is again passed through a demodulator to get back the original message.

→ In an FDM system, the bandwidth allocated to each sub-band is slightly larger than the bandwidth needed by each message. This extra bandwidth, is called a guardband.

Channel bandwidth of FDM system:

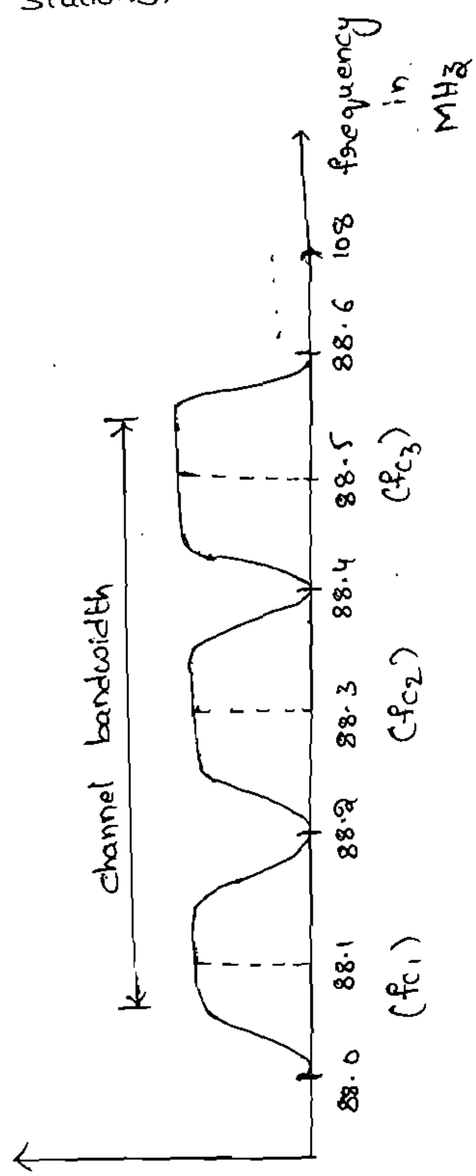
If 'n' message signals, each having a bandwidth equal to 'W' Hertz, then the minimum bandwidth of the channel should be greater than nW Hertz. i.e.,

$$\boxed{\text{Channel B.W of FDM} \geq nW} \text{ Hertz}$$

Where 'n' indicates the number of messages to be multiplexed.

## Applications of FDM :

FDM is used in telephone - system, telemetry, commercial broadcast, television and communication networks. commercial FDM broadcasting stations uses frequency band from 88MHz to 108 MHz. This 88-108 MHz frequency band is divided into sub-bands of bandwidth 200kHz. The FM stations are identified by the center-frequency within their channel (e.g., 88.1 MHz, 88.3 MHz, .... 107.9). This system can provide radio listeners with their choice of upto 100 different radio stations.



# Amplitude Modulation

→ In amplitude modulation, the amplitude of a carrier signal is varied in accordance with the instantaneous values of modulating signal.

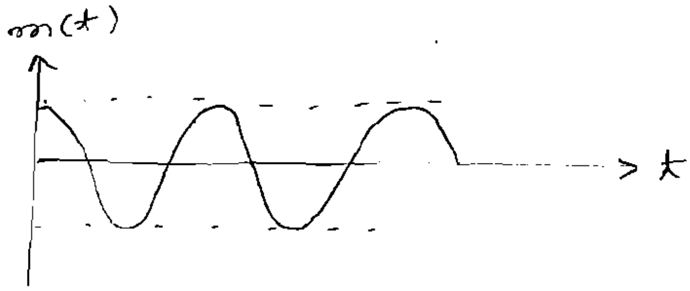
**single tone modulation:**

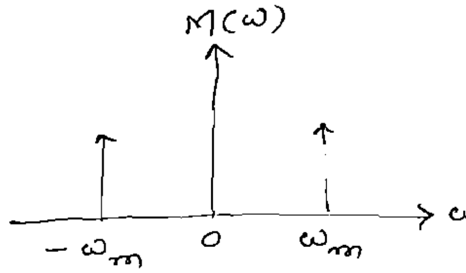
Let us consider the modulating voltage be

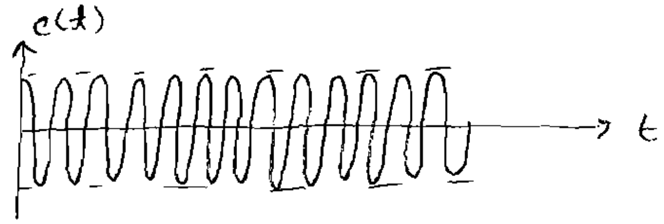
$$m(t) = v_m(t) = V_m \cos \omega_m t$$

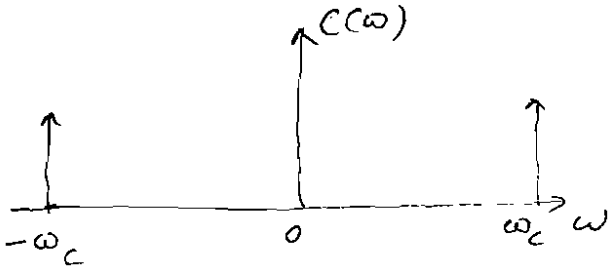
and high frequency carrier signal be

$$c(t) = v_c(t) = V_c \cos \omega_c t$$




→ from the definition of amplitude modulation, the amplitude of the carrier is changed according to the instantaneous values of amplitude of message/modulating signal

$$\therefore \quad V_{AM} = V_c + m(t)$$

$$= V_c + V_m \cos \omega_m t$$

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

$$= V_c [1 + \mu \cos \omega_m t]$$

$$= V_c [1 + k_a m(t)]$$

where $k_a = \frac{1}{V_c}$

where $\frac{V_m}{V_c} = \mu$ called modulation index

% modulation

depth of modulation

modulation factor

$$K_a = \frac{1}{V_c} = \text{amplitude sensitivity}$$

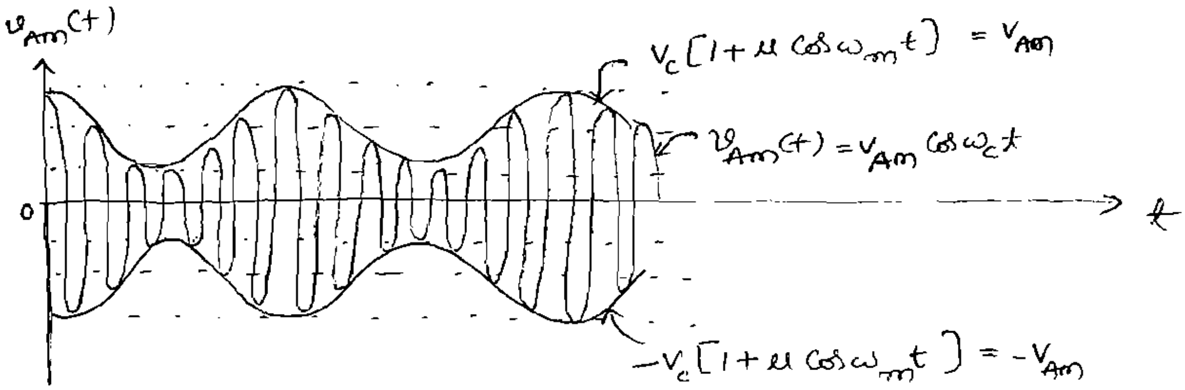
$\therefore$ The instantaneous amplitude of modulated signal is given by

$$v_{Am}(t) = V_{Am} \cos\omega_c t$$

$$= V_c\left[1 + \mu \cos\omega_m t\right] \cos\omega_c t$$

(or)

$$= V_c\left[1 + K_a m(t)\right] \cos\omega_c t$$




Frequency domain description:

$$v_{Am}(t) = V_c\left[1 + \mu\cos\omega_m t\right]\cos\omega_c t$$

$$= V_c\cos\omega_c t + \mu V_c \cos\omega_m t \cos\omega_c t \quad \times\frac{2}{2}$$

$$= V_c\cos\omega_c t + \frac{\mu V_c}{2}\, 2\cos\omega_m t\cos\omega_c t$$

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

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

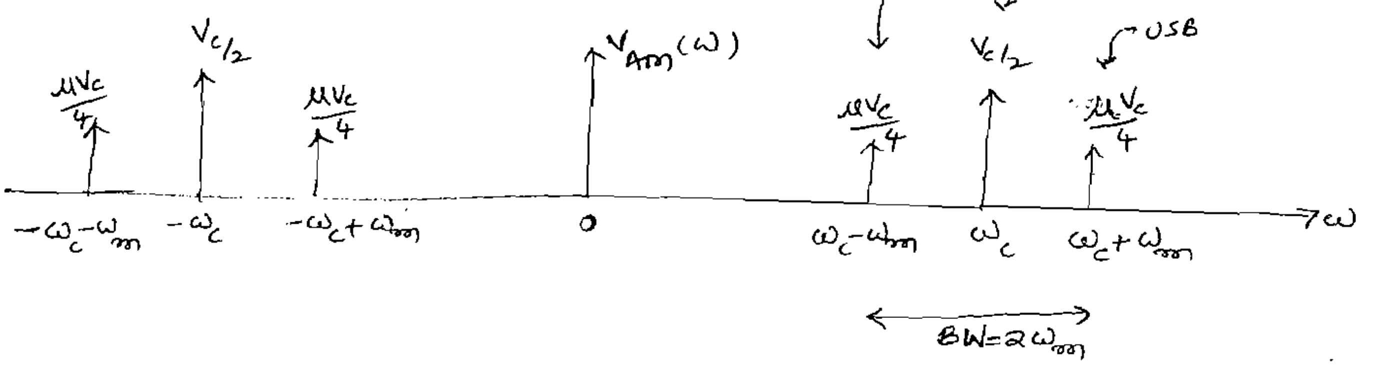
// $2\cos A\cos B = \cos(A+B) + \cos(A-B)$

$$x(t) \longleftrightarrow X(\omega)$$

$$x(t)\cos\omega_0 t \longleftrightarrow \frac{1}{2}\left(X(\omega+\omega_0) + X(\omega-\omega_0)\right)$$

$$\left.\begin{array}{l} x(t)\left[\frac{e^{j\omega_0 t}+e^{-j\omega_0 t}}{2}\right] \\ \frac{1}{2}\left[x(t)\, e^{j\omega_0 t} + x(t)\, e^{-j\omega_0 t}\right] \end{array}\right\} \longleftrightarrow \frac{1}{2}\left[X(\omega+\omega_0) + X(\omega-\omega_0)\right]$$

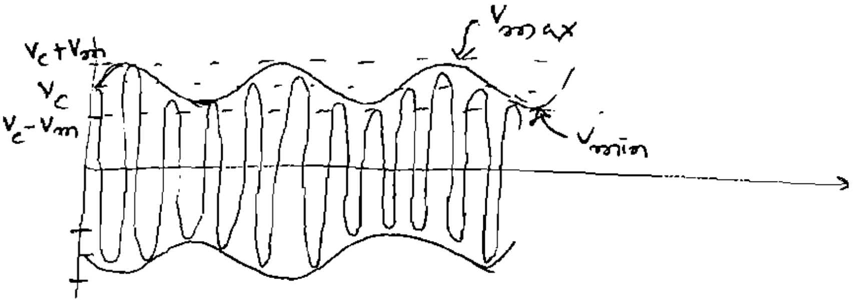
$v_{AM}(t) \longleftrightarrow V_{AM}(\omega)$




// multiplication with cos signal shifts the spectrum of the signal to both sides and makes magnitude half

$$BW = \omega_c+\omega_m-(\omega_c-\omega_m)$$
$$= \cancel{\omega_c}+\omega_m-\cancel{\omega_c}+\omega_m = 2\omega_m$$

## Calculation of modulation index:




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

$$V_{max}-V_{min} = V_c+V_m-(V_c-V_m) = 2V_m$$

$$V_m = \frac{V_{max}-V_{min}}{2}$$

$$V_c = V_{max}-V_m$$
$$= V_{max}-\left[\frac{V_{max}-V_{min}}{2}\right] = \frac{2V_{max}-V_{max}+V_{min}}{2}$$
$$= \frac{V_{max}+V_{min}}{2}$$

II method : from $v_{AM}(t)$ eq

$V_{max} = V_c[1+\mu]$

$V_{min} = V_c[1-\mu]$

$V_{max} = V_c+V_c\mu \rightarrow ①$

$V_{min} = V_c-V_c\mu \rightarrow ②$

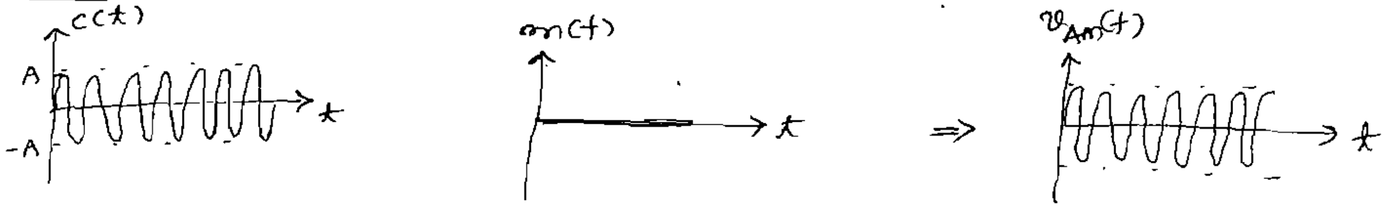
Solving eqns ① & ②

$\mu = \dfrac{V_{max}-V_{min}}{V_{max}+V_{min}}$

$$\therefore \mu = \frac{V_m}{V_c} = \frac{V_{max}-V_{min}}{V_{max}+V_{min}}$$

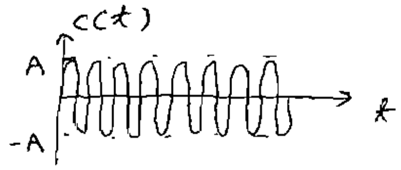
$$\mu = \frac{\text{amplitude change of carrier}}{\text{Normal carrier wave}} = \frac{V_c(max)-V_c}{V_c}$$

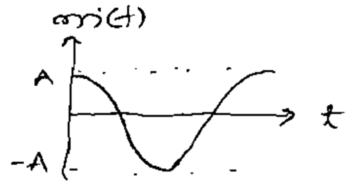
case(i) : No modulation

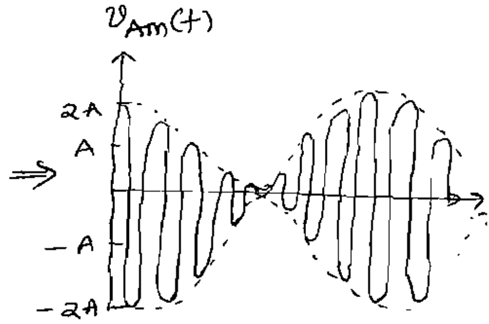



$$\mu = \frac{V_m}{V_c} = \frac{0}{A} = 0$$

case (ii) : Perfect modulation

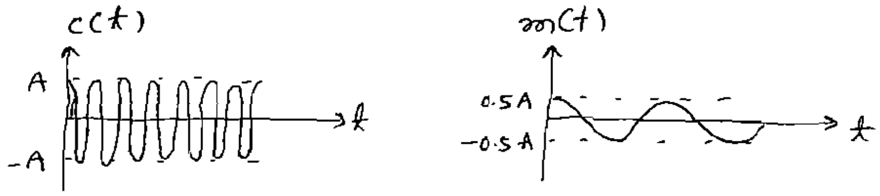


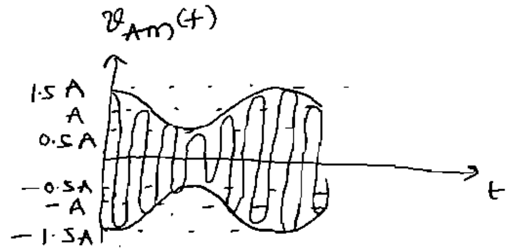




$$\mu = \frac{V_m}{V_c} = \frac{A}{A} = 1 \cong 100\% \quad \cong \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{2A-0}{2A+0} = 1 = 100$$

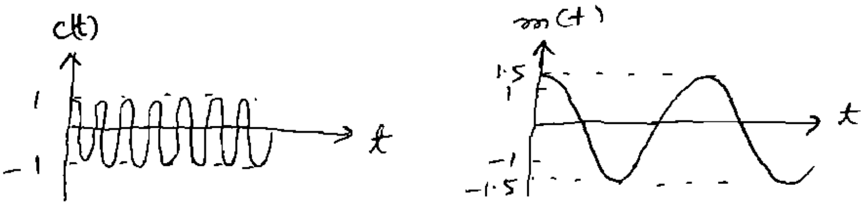
case (iii) : under modulation

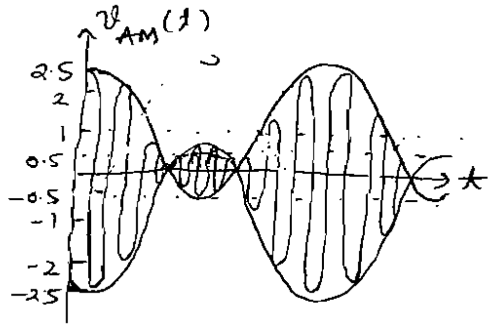



$$\mu = \frac{V_m}{V_c} = \frac{0.5A}{A} = 0.5 \cong 50\% \quad \cong \frac{1.5A - 0.5A}{1.5A + 0.5A} = \frac{1A}{2A} = 0.5 = 50\%$$

case (iv) : over modulation




$$\mu = \frac{V_m}{V_c} = \frac{1.5}{1} = 1.5 = 150\% \quad \cong \frac{2.5-(-0.5)}{2.5+(-0.5)} = \frac{3}{2} = 1.5 \cong 150\%$$

→ $\mu = \frac{V_m}{V_c} \leq 1$ i.e as long as the % modulation is less than unity, there exists a one to one relationship between message signal and envelope of the modulated signal.

Condition 1: $\mu \leq 1 \cong K_a m(t) \leq 1$

condition 2: BW of m(t) << BW of c(t)

## Power relations in AM

$$m(t) = v_m(t) = V_m \cos \omega_m t$$

$$c(t) = v_c(t) = V_c \cos \omega_m t$$

$$v_{AM}(t) = \underbrace{V_c \cos\omega_c t}_{=\text{ carrier}} + \underbrace{\frac{\mu V_c}{2} \cos(\omega_c + \omega_m)t}_{+\text{ USB}} + \underbrace{\frac{\mu V_c}{2} \cos(\omega_c - \omega_m)t}_{+\text{ LSB}}$$

→ Carrier component of the modulated wave has the same amplitude as the unmodulated carrier i.e amplitude of the carrier is unchan-ged.

∴ energy is added or subtracted depending on the other components

→ The modulated wave contains extra energy in the two sidebands.

∴ modulated wave contains more power than the carrier, which had before modulation took place.

→ it is clear from the above equation that, the amplitude of the sidebands depends on the modulation index $\mu \left(= \frac{V_m}{V_c}\right)$, ∴ the total power in the modulated wave will also depend on the modulation index.

total power in the modulated wave is given by

$$P_t = \frac{V^2_{carrier}}{R} + \frac{V^2_{LSB}}{R} + \frac{V^2_{USB}}{R}$$

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

where all 3 voltages are RMS values and R is resistance of antenna through which power is dissi-pated

$$P_c = \frac{V^2_{carrier}}{R} = \frac{(V_c/\sqrt{2})^2}{R} = \frac{V_c^2}{2R}$$

$$P_{LSB} = \frac{V^2_{LSB}}{R} = \frac{\left(\frac{\mu V_c}{2}/\sqrt{2}\right)^2}{R} = \frac{\mu^2 V_c^2}{8R}$$

$$P_{USB} = \frac{V^2_{USB}}{R} = \frac{\left(\frac{\mu V_c}{2}/\sqrt{2}\right)^2}{R} = \frac{\mu^2 V_c^2}{8R}$$

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

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

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

for $\mu = 1$ i.e 100% modulation

$$= \frac{V_c^2}{2R}\left[1+\frac{1}{2}\right] = 1.5\,\frac{V_c^2}{2R} = 1.5\,P_c$$

$$P_{LSB} = P_{USB} = \frac{\mu^2 V_c^2}{8R} = \frac{V_c^2}{2R}\left[\frac{\mu^2}{4}\right] = P_c\,\frac{\mu^2}{4}$$

## Current relations in AM

$$P_t = P_c\left[1+\frac{\mu^2}{2}\right]$$

$$I_t^2 \cdot \cancel{R} = I_c^2 \cdot \cancel{R}\left[1+\frac{\mu^2}{2}\right]$$

$$I_t = I_c\sqrt{1+\frac{\mu^2}{2}}$$

$I_c$ → carrier current

$I_t$ → modulated signal current

## Efficiency

→ It can be defined as the ratio of power in sidebands to total power, because the side-bands are containing the actual information.

$$\% \eta = \frac{\text{Power in sidebands}}{\text{total power}}$$

$$= \frac{P_{LSB} + P_{USB}}{P_t}$$

$$= \frac{\frac{\mu^2 V_c^2}{8R} + \frac{\mu^2 V_c^2}{8R}}{\frac{V_c^2}{2R}\left[1 + \frac{\mu^2}{2}\right]} = \frac{\frac{\mu^2 \cancel{V_c^2}}{\cancel{2}4\cancel{R}}}{\frac{\cancel{V_c^2}}{\cancel{2R}}\left[1 + \frac{\mu^2}{2}\right]}$$

$$= \frac{\frac{\mu^2}{\cancel{2}}}{\frac{2+\mu^2}{\cancel{2}}} = \frac{\mu^2}{2+\mu^2}$$

for $\mu = 1$ i.e 100% modulation

$$= \frac{1}{2+1} = \frac{1}{3} \times 100 = 33.3\ \%$$

i.e only 33.3 % of power is utilized and the remaining power is wasted by the transmiss of carrier along with sidebands.

## Multitone modulation

Let $\quad m(t) = V_m(t) = V_{m_1}\cos\omega_{m_1}t + V_2\cos\omega_{m_2}t$

$\quad c(t) = v_c(t) = V_c\cos\omega_c t$

modulated signal

$$v_{AM}(t) = V_{AM}\cos\omega_c t$$

$$= [V_c + m(t)]\cos\omega_c t$$

$$= [V_c + V_{m_1}\cos\omega_{m_1}t + V_{m_2}\cos\omega_{m_2}t]\cos\omega_c t$$

$$= V_c\left[1 + \frac{V_{m_1}}{V_c}\cos\omega_{m_1}t + \frac{V_{m_2}}{V_c}\cos\omega_{m_2}t\right]\cos\omega_c t$$

$$= V_c[1 + \mu_1\cos\omega_{m_1}t + \mu_2\cos\omega_{m_2}t]\cos\omega_c t$$

where $\mu_1 = \dfrac{V_{m_1}}{V_c}$ & $\mu_2 = \dfrac{V_{m_2}}{V_c}$

$$= V_c\left[1 + \frac{1}{V_c}V_{m_1}\cos\omega_m t + \frac{1}{V_c}V_{m_2}\cos\omega_{m_2}t\right]\cos\omega_c t$$

$$= V_c[1 + K_a m_1(t) + K_a m_2(t)]\cos\omega_c t$$

where $K_a = \dfrac{1}{V_c}$ = amplitude sensitivity

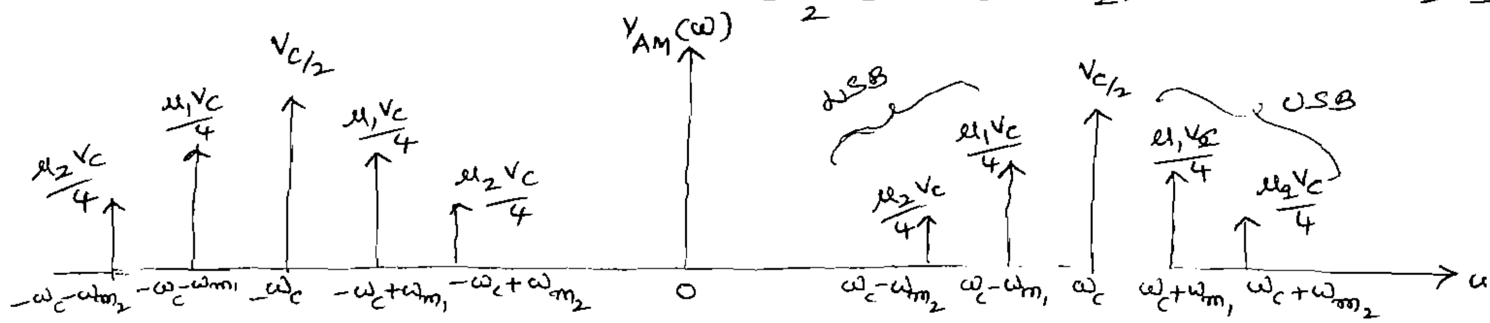
$m_1(t) = V_{m_1}\cos\omega_{m_1}t$

$m_2(t) = V_{m_2}\cos\omega_{m_2}t$

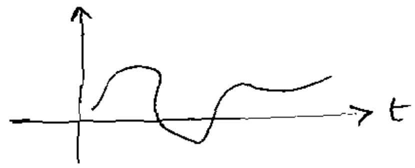
$$= V_c\cos\omega_c t + \mu_1 V_c\cos\omega_{m_1}t\cos\omega_c t + \mu_2 V_c\cos\omega_{m_2}t\cos\omega_c t$$

$$= V_c\cos\omega_c t + \frac{\mu_1 V_c}{2}[\cos(\omega_c+\omega_{m_1})t + \cos(\omega_c-\omega_{m_1})t]$$

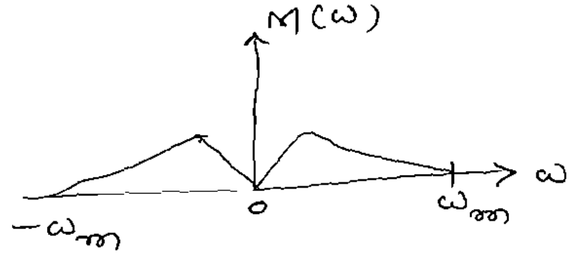
$$+ \frac{\mu_2 V_c}{2}[\cos(\omega_c+\omega_{m_2})t + \cos(\omega_c-\omega_{m_2})t]$$



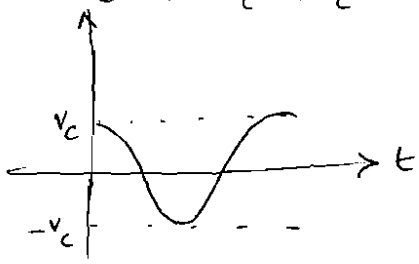
Illy when $m(t)$ is containing infinite no. of waves, the spectrum of modulated signal is as follows



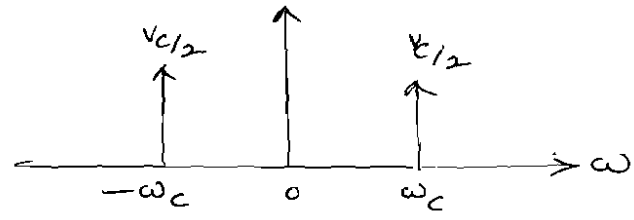

$\leftrightarrow$




$c(t) = V_c \cos\omega_c t$



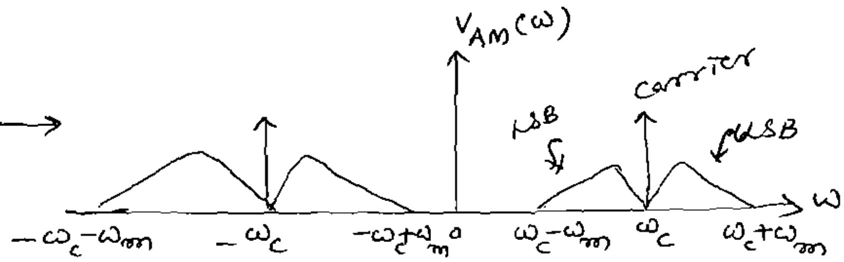

$\longleftrightarrow$




$v_{AM}(t)$

$= V_c\left[1 + K_a m(t)\right]\cos\omega_c t$

$\longleftrightarrow$




## Power relations

$$m(t) = V_{m_1}\cos\omega_{m_1}(t) + V_{m_2}\cos\omega_{m_2}t$$

$$c(t) = V_c\cos\omega_c t$$

$$v_{AM}(t) = V_c\cos\omega_c t + \frac{\mu_1 V_c}{2}\left[\cos(\omega_c+\omega_{m_1})t + \cos(\omega_c-\omega_{m_1})t\right]$$
$$+ \frac{\mu_2 V_c}{2}\left[\cos(\omega_c+\omega_{m_2})t + \cos(\omega_c-\omega_{m_2})t\right]$$

Total Power transmitted

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

$$= \frac{(V_c/\sqrt{2})^2}{R} + \frac{\left(\frac{\mu_1 V_c}{2}/\sqrt{2}\right)^2}{R} + \frac{\left(\frac{\mu_2 V_c}{2}/\sqrt{2}\right)^2}{R} + \frac{\left(\frac{\mu_1 V_c}{2}/\sqrt{2}\right)^2}{R} + \frac{\left(\frac{\mu_2 V_2}{2}/\sqrt{2}\right)^2}{R}$$

$$= \frac{V_c^2}{2R} + 2\left[\left(\frac{\mu_1 V_c}{2\sqrt{2}}\right)^2/R + \left(\frac{\mu_2 V_c}{2\sqrt{2}}\right)^2/R\right]$$

$$= \frac{V_c^2}{2R} + \frac{V_c^2}{4R}\left[\mu_1^2 + \mu_2^2\right] \qquad \mu^2 = \mu_1^2 + \mu_2^2$$

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

Ily for $\mu$ can can be calculated for any no of components.

$$\mu = \mu_1^2 + \mu_2^2 + \cdots$$

$$P_t = P_c\left[1 + \frac{\mu^2}{2}\right]$$

Efficiency :

$$\%\,\eta = \frac{\text{Power in sidebands}}{\text{total power transmitted}}$$

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

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

if $\mu = 1$

$$= \frac{1^2}{2+1^2} = \frac{1}{3} = .33 \cong 33\,\%.$$

Generation of AM waves

→ The techniques for generating Amplitude modulated waves can be classified into 2 categories.

(i) Low level modulation
(ii) High level modulation

→ In low level modulation model, the generation of AM waves takes place in the initial stage of amplification i.e at a low power level. The generated AM signal is then amplified using no of amplifier stages.

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

→ The following two techniques are well suited for low power modulation purposes.

(i) Square law modulator
(ii) Switching modulator

The above two techniques are non-linear modulators

→ a simple diode or transistor or FET can be used as non-linear modulator by restricting the operation over the non-linear region of it's characteristics.

→ The i/p to the diode is kept so small, then the diode operation in a non-linear region of it's V-I characteristics.

→ This method is useful for only small signal amplitude modulation.

## Square law modulator

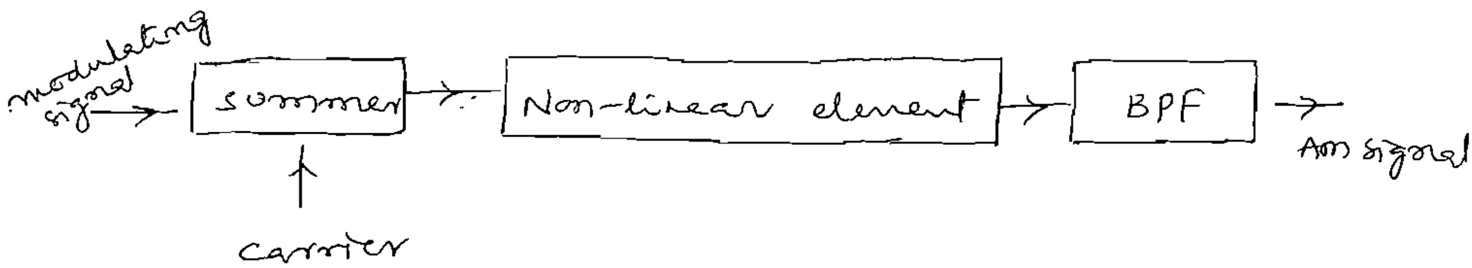
→ It contains 3 elements

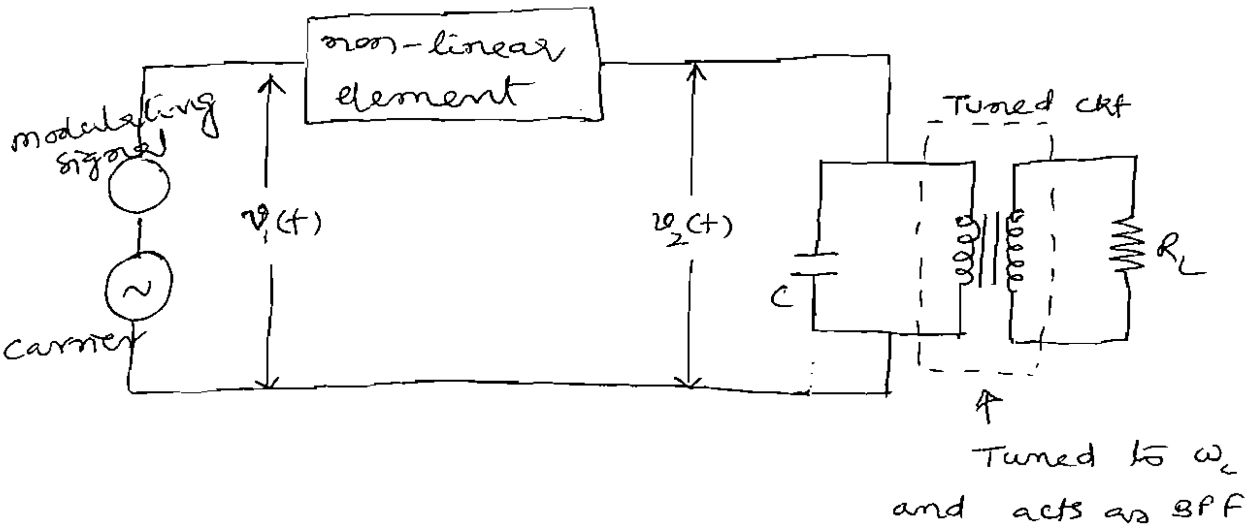
(i) summer : for summing carrier and modulating signal

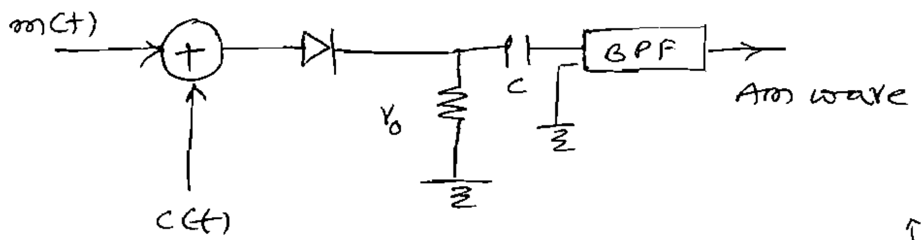
(ii) Non-linear element : for implementing square law modulator

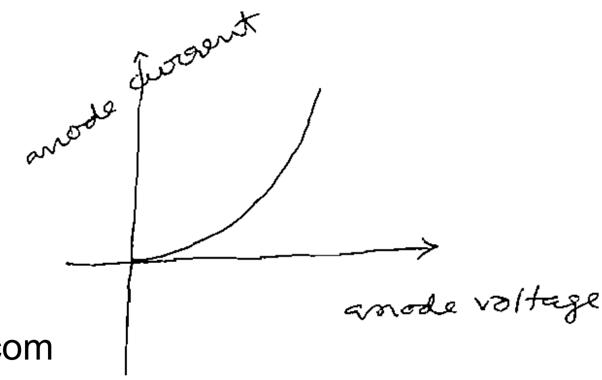
(iii) Bandpass filter (BPF) : for extracting the desired modulation products

Block diagram:



→ The output and input relationship of non-linear devices is as follows

$$v_0 = a_1 v_i + a_2 v_i^2 + a_3 v_i^3 + \cdots$$

if $v_i$ is small, first two terms are significant and remaining terms can be neglected.

$$\therefore \boxed{v_0 = a_1 v_i + a_2 v_i^2}$$

→ When a non-linear element like diode is suitably biased and operated in a restricted portion of it's characteristic curve i.e if the signal applied is to the diode is relatively weak, then we find the transfer characteristic of the diode load resistor condition can be represented closely by a square law

$$v_2(t) = a_1 v_1(t) + a_2 v_2^2(t) \quad \rightarrow ①$$

where $a_1$ and $a_2$ are constants

and i/p voltage $v_1(t) = m(t) + c(t)$

$$= m(t) + V_c \cos\omega_c t \quad \rightarrow ②$$

substituting eqn ② in ①, gives

$$v_2(t) = a_1 [m(t) + V_c \cos\omega_c t] + a_2 [m(t) + V_c \cos\omega_c t]^2$$

$$= a_1 m(t) + a_1 V_c \cos\omega_c t + a_2 m^2(t) + a_2 V_c^2 \cos^2\omega_c t + 2 a_2 m(t) V_c \cos\omega_c t$$

$$= a_1 m(t) + a_2 m^2(t) + a_2 V_c^2 \left[\frac{1+\cos 2\omega_c t}{2}\right] + a_1 V_c \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos\omega_c t$$

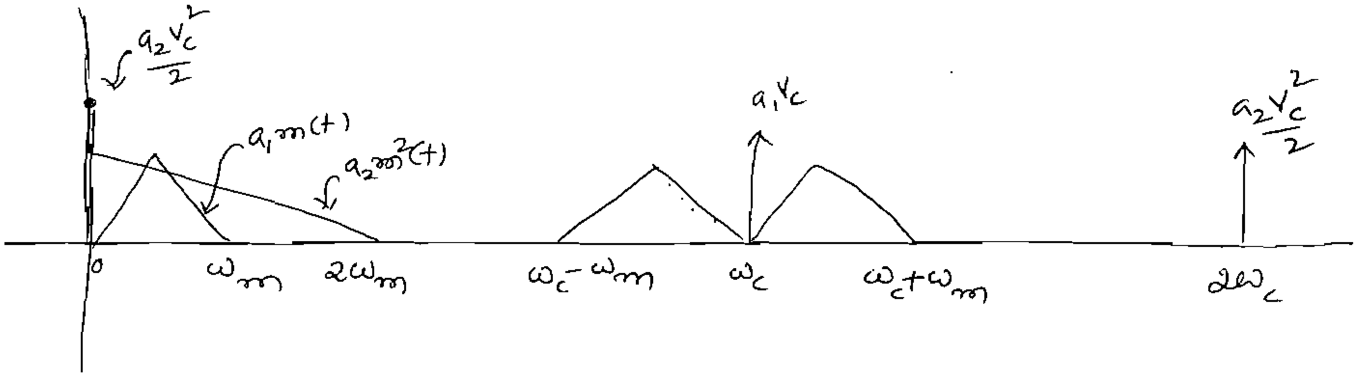
$$= a_1 V_c \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos\omega_c t \; +$$

$$a_1 m(t) + a_2 m^2(t) + \frac{a_2 V_c^2}{2} + \frac{a_2 V_c^2}{2} \cos 2\omega_c t \quad \rightarrow (3)$$

AM wave with amplitude sensitivity $K_a = \frac{2a_2}{a_1}$

unwanted components and these components can be removed by appropriate (bandpass) filtering

(remember from convolution property that $m^2(t)$ has the twice the BW of $m(t)$)




<u>conditions:</u>

$\omega_c - \omega_m >> 2\omega_m$

&

$\omega_c >> 3\omega_m$

→ The eqn (3) gives the resulting voltage developed across the primary winding of the transformer. The first term of eqn (3) is the desired AM wave and $K_a = \frac{2a_2}{a_1}$ is the amplitude sensitivity. ∴ The remaining unwanted terms present in the above equation can be eliminated by appropriate filtering method

specifications of filter:

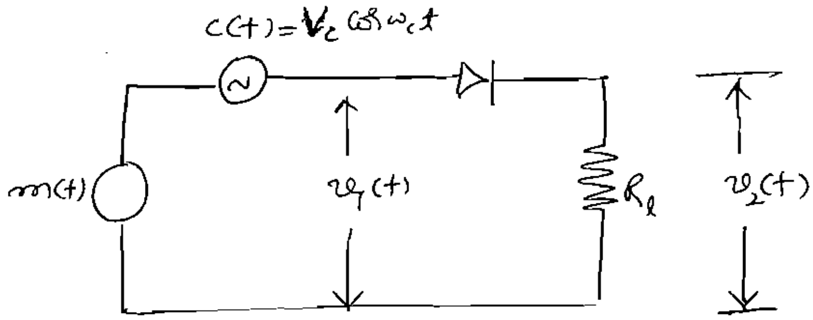
midband frequency : $\omega_c$

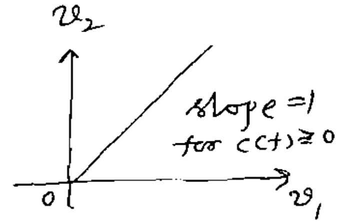
BW : $2\omega_m$

$\omega_c > 3\omega_m$

## Switching Modulator

The switching modulator is shown below.




→ Here, the diode is assumed to act as ideal switch i.e it presents zero impedance when it is forward biased and infinite impedance when it is reverse biased.

→ The diode switch is controlled by carrier wave c(t) and assume that the carrier voltage c(t) applied to the diode is large in amplitude.

→ The i/p voltage

$$v_1(t) = m(t) + c(t)$$
$$= m(t) + V_c\cos\omega_c t \rightarrow ①$$

& $m(t) << V_c(t)$

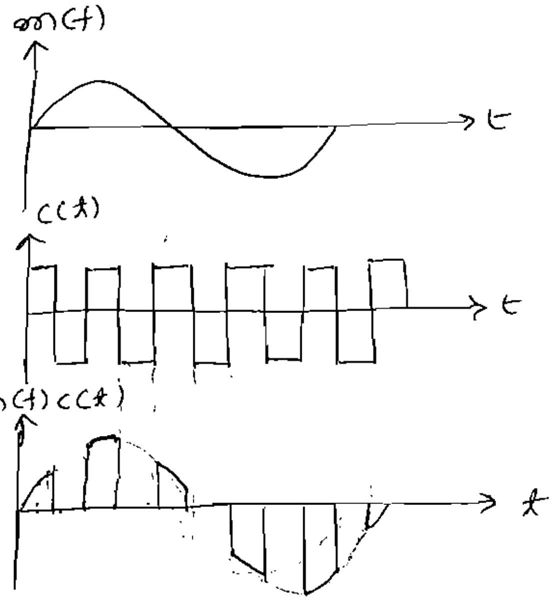
∴ the resulting load voltage

$$v_2(t) = v_1(t) \qquad \text{if } c(t) > 0$$
$$0 \qquad \text{if } c(t) < 0 \rightarrow ②$$

∴ we may thus approximate the transfer characteristic curve of the diode load resistor combination by a piecewise linear characteristic

→ i.e the load voltage periodically varies between the values $v_1(t)$ and zero at a rate equal to the carrier frequency $\omega_c$.

→ Thus, a modulating signal which is weak compared with the carrier wave, we can replace the non-linear behavior of a diode by an approximately equivalent linear time varying operation.




→ Eqn ② can be written as

$$v_2(t) \cong v_1(t) \cdot g_p(t)$$

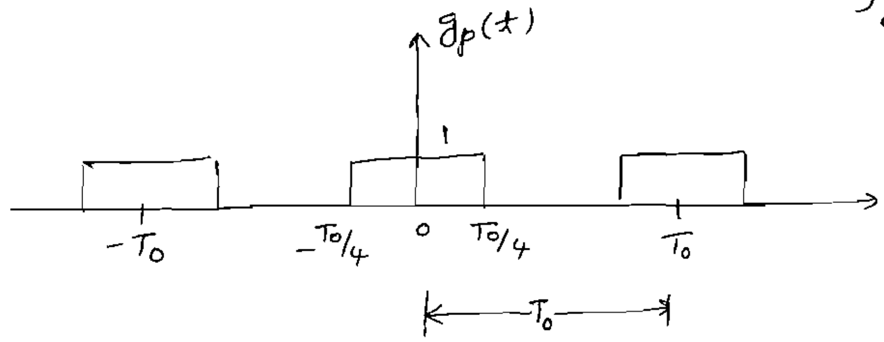
$$\cong [m(t) + V_c \cos\omega_c t] \cdot g_p(t)$$

$$\cong [m(t) + c(t)]\, g_p(t)$$

where $g_p(t)$ is a periodic pulse train

duty cycle = one half of time period

$T_0 = \frac{1}{f_c}$




→ using fourier series, $g_p(t)$ can be written as

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

$$= \frac{1}{2} + \frac{2}{\pi}\cos\omega_c t + \text{odd harmonic components} \quad \rightarrow ④$$

substituting eqn ④ in eqn ③, we get

$$v_2(t) = [V_c \cos\omega_c t + m(t)]\left\{\frac{1}{2} + \frac{2}{\pi}\cos\omega_c t + \text{odd components}\right\}$$

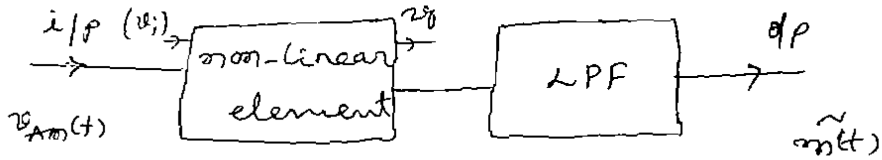
$$= \frac{V_c}{2}\cos\omega_c t + \frac{1}{2}m(t) + V_c\frac{2}{\pi}\cos^2\omega_c t + \frac{2\times V_c\times}{2\times V_c\times}\frac{2}{\pi}m(t)\cos\omega_c t + \text{some unwanted component}$$

$$= \underbrace{\frac{V_c}{2}\left[1 + \frac{4}{\pi V_c}m(t)\right]\cos\omega_c t}_{\text{AM wave}} + \underbrace{\frac{1}{2}m(t) + \frac{2V_c}{\pi}\cos^2\omega_c t +}_{\text{unwanted components}}$$

$K_a = \frac{4}{\pi V_c}$ amplitude sensitivity

→ all unwanted components can be removed using appropriate filtering.

# Square law detector




$$v_o = a_1 v_i + a_2 v_i^2$$

here $\quad v_i = V_c\,[1 + K_a m(t)]\cos\omega_c t$

$$\therefore v_o = a_1\{V_c[1+K_a m(t)]\cos\omega_c t\} + a_2\{V_c[1+K_a m(t)]\cos\omega_c t\}^2$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + a_2[V_c\cos\omega_c t + V_c K_a m(t)\cos\omega_c t]^2$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + a_2 V_c^2\cos^2\omega_c t + a_2 V_c^2 K_a^2 m^2(t)\cos^2\omega_c t + 2a_2 V_c\cos\omega_c t\, V_c K_a m(t)\cos\omega_c t$$

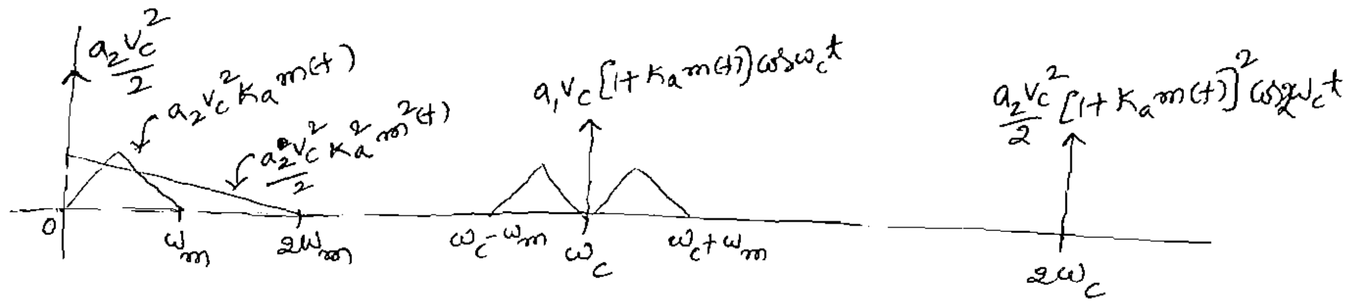
$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + [a_2 V_c^2 + a_2 V_c^2 K_a^2 m^2(t) + 2a_2 V_c^2 K_a m(t)]\cos^2\omega_c t$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + \frac{a_2 V_c^2}{2}[1+K_a^2 m^2(t) + 2K_a m(t)][1+\cos 2\omega_c t]$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + \frac{a_2 V_c^2}{2}[1+K_a m(t)]^2[1+\cos 2\omega_c t]$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + \frac{a_2 V_c^2}{2}[1+K_a m(t)]^2\cdot 1 + \frac{a_2 V_c^2}{2}[1+K_a m(t)]^2\cos 2\omega_c t$$

$$= a_1 V_c[1+K_a m(t)]\cos\omega_c t + \frac{a_2 V_c^2}{2} + a_2 V_c^2 K_a m(t) + \frac{a_2 V_c^2}{2}K_a^2 m^2(t) + \frac{a_2 V_c^2}{2}[1+K_a m(t)]^2\cos 2\omega_c t$$




→ The d.c component and all frequency components whose frequency is more than $\omega_m$ can be removed by using LPF.

→ Still the term $\frac{a_2V_c^2}{2}K_a^2m^2(t)$ is interfering with $a_2V_c^2K_am(t)$.

The ratio between wanted and unwanted terms is given by

$$\frac{\text{wanted}}{\text{unwanted}} : \frac{a_2V_c^2K_am(t)}{\frac{a_2V_c^2}{2}K_a^2m^2(t)}$$

$$: \frac{2}{K_am(t)} = \frac{2}{\mu}$$

→ for this ratio to be large $K_am(t) << 1$

i.e $\mu << 1$

i.e the modulation index must be selected so small to avoid distortion in case of square law method.

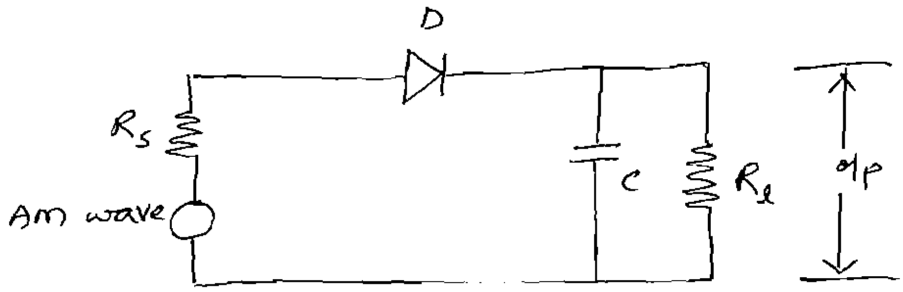
→ we know that, for better efficiency the modulation index $\mu$ should be as high as possible ($<=1$)

∴ Square law method is generally not preferred for detection purpose.

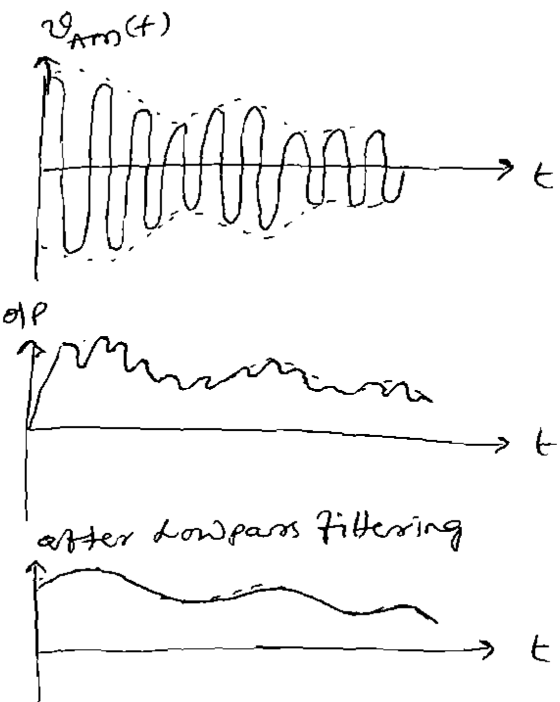
## Envelope Detector (or) diode detector

→ An envelope detector is a simple and highly effective device that is well suited for the demodulation of narrow band AM wave (i.e the carrier frequency is large compared with the modulating signal frequency BW) for which the % modulation is less than 100%.. In envelope detector, the o/p follows the envelope of the modulated signal, hence the name.

→ The envelope detector circuit is shown below.




<u>operation:</u>




→ during the +ve half cycle of the i/p AM wave, the diode is forward biased and the capacitor C charges up rapidly to the peak value of the i/p signal.

→ when the i/p signal falls below this value, the diode becomes reverse biased and the capacitor c discharges slowly the load resistor $R_l$.

The discharging process continues until the next +ve half cycle.

→ When the i/p signal becomes greater than the voltage across capacitor, the diode conducts again and the process is repeated.

→ assume diode is ideal.

offers zero impedance during forward biased time and infinite impedance during reverse biased time.

→ we also assume that the AM wave applied to the envelope detector is supplied by a voltage source of Internal impedance $R_s$.

→ The charging time constant $R_sC$ must be small compared with carrier period $\frac{1}{f_c}$

$$R_sC \ll \frac{1}{f_c}$$

→ Discharging time constant $R_lC$ must be very high enough to ensure that the capacitor discharges slowly through the load resistor $R_l$ between +ve peaks of the carrier wave. at the same moment, it should not be so long that the capacitor voltage will not discharge at the maximum rate of change of the modulating signal.

$$\frac{1}{f_c} \ll R_lC \ll \frac{1}{\omega_m}$$

→ The capacitor voltage or detector o/p is very nearly to the envelope of the AM wave.

→ The o/p usually has a small ripples at the carrier frequency rate, it can easily be removed by low pass filtering.

Problems:

P1 : A commercial AM station is broadcasting with an average transmitted carrier power of 10 KW. The modulation index is set at 0.707 for a sinusoidal message signal. Find the transmission power, efficiency and power in the sidebands of the transmitted signal

Soln:

(i) Total power transmitted

$$P_t = P_c\left[1+\frac{\mu^2}{2}\right]$$

$$= 10\times10^3\left[1+\frac{(0.707)^2}{2}\right] = 12.5\ \text{KW}$$

(ii) % efficiency

$$\%\ \eta = \frac{P_{SB}}{P_t}\times100$$

$$= \frac{\frac{\mu^2 \cancel{P_c}}{2}}{\cancel{P_c}\left[1+\frac{\mu^2}{2}\right]} = \frac{\frac{(0.707)^2}{2}}{\left[1+\frac{(0.707)^2}{2}\right]} = 20\%.$$

(iii) Side band power $= \frac{\mu^2 P_c}{2}$

$$= \frac{(0.707)^2}{2}\times10\times10^3 = 2.5\ \text{KW}$$

P2 A complex modulating wave consisting of a sine wave of amplitude 3V and frequency 1 KHz plus a cosine wave of amplitude of 5V and frequency 3 KHz is amplitude modulated by a sine carrier of 500KHz and 50V peak. Determine the average power transmitted when a wave is fed to a 50 Ω load.

Sol^n :

$$v_m(t) = 3 \sin(2\pi \times 1 \times 10^3 \times t) + 5 \cos(2\pi \times 3 \times 10^3 \times t)$$

$$v_c(t) = 50 \sin(2\pi \times 500 \times 10^3 \times t)$$

ave power transmitted

$$P_t = P_c\left[1 + \frac{\mu^2}{2}\right] = \frac{V_c^2}{2R}\left[1 + \frac{\mu^2}{2}\right]$$

$V_c = 50\,V$

$R = 50\,\Omega$

$$\mu^2 = \mu_1^2 + \mu_2^2$$

$$= \frac{V_{m_1}}{V_c} + \frac{V_{m_2}}{V_c}$$

$$= \frac{3}{50} + \frac{5}{50}$$

$$= \frac{50^2}{2 \times 50}\left[1 + \frac{\left(\frac{3}{50}\right)^2 + \left(\frac{5}{50}\right)^2}{2}\right]$$

$$= 25.17\ W$$

P3 The total transmitted power content of Am signal is 1000 Watts. Determine the power being transmitted at the carrier frequency and each of the sidebands when the % modulation is 100.

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

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

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

if $\mu = 1$

$$= P_c + \frac{P_c}{2}$$

$$= 1.5\,P_c$$

$$1000 = 1.5\,P_c$$

$$P_c = 666.67 \text{ Watts}$$

$$P_{SB} = P_t - P_c$$

$$= 1000 - 666.67 = 333.3 \text{ Watts}$$

$$P_{LSB} = P_{USB} = \frac{P_{SB}}{2} = \frac{333.3}{2}$$

$$= 166.67 \text{ Watts}$$

P4 An AM signal is given by

$$v_{Am}(t) = 10\cos 2\pi 10^6 t + 5\cos 2\pi 10^6 t \cos 2\pi 10^3 t + \\ + 2\cos 2\pi 10^6 t \cos 4\pi 10^3 t$$

Find the various frequency components and the corresponding modulation indices. Draw the spectrum, BW, modulated power, side band power and modulation index.

Sol^n:

$$v_{Am}(t) = 10\cos 2\pi 10^6 t + 5\cos 2\pi 10^6 t \cos 2\pi 10^3 t + 2\cos 2\pi 10^6 t \cos 4\pi 10^3 t$$

$$= 10\cos 2\pi 10^6 t\left[1 + 0.5\cos 2\pi 10^3 t + 0.2\cos 4\pi 10^3 t\right]$$

$V_c = 10V$ $\quad \mu_1 = 0.5$ $\quad \mu_2 = 0.2$

$\omega_c = 2\pi 10^6$ $\quad \omega_{m_1} = 2\pi 10^3$ $\quad \omega_{m_2} = 2\pi \times 2 \times 10^3$

$f_c = 1000\,KHz$ $\quad f_{m_1} = 1\,KHz$ $\quad f_{m_2} = 2\,KHz$

$= 1\,mHz$ $\quad = 0.001\,mHz$ $\quad = 0.002\,MHz$

frequencies

USB terms

$f_c + f_{m_1} = 1\,mHz + 0.001\,mHz = 1.001\,mHz$

$f_c + f_{m_2} = 1\,mHz + 0.002\,mHz = 1.002\,mHz$

LSB term

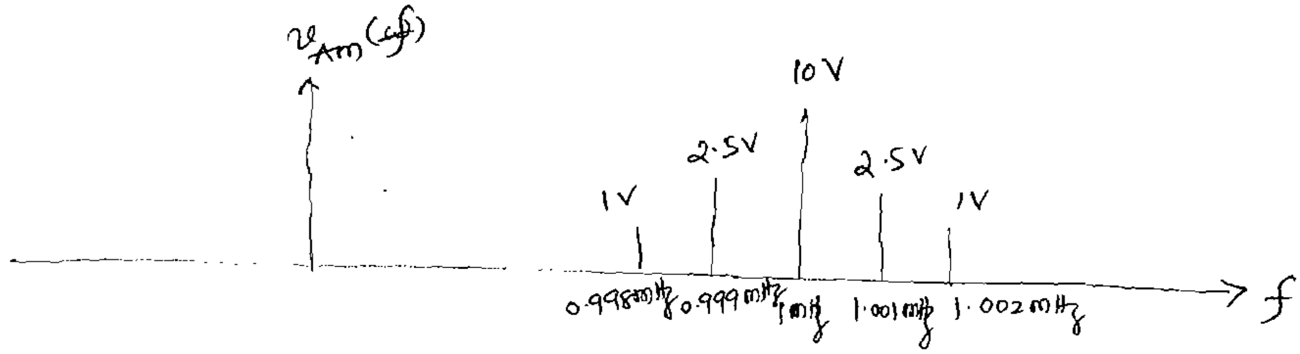
$f_c - f_{m_1} = 1\,mHz - 0.001\,mHz = 0.999\,mHz$

$f_c - f_{m_2} = 1\,mHz - 0.002\,mHz = 0.998\,mHz$

magnitudes

for $f_{m_1} \longrightarrow \dfrac{\mu_1 V_c}{2} = \dfrac{0.5 \times 10}{2} = 2.5\,V$

for $f_{m_2} \longrightarrow \dfrac{\mu_2 V_c}{2} = \dfrac{0.2 \times 10}{2} = 1\,V$




carrier power $P_c = \frac{V_c^2}{2R} = \frac{10^2}{2} = 50$ watts

where $R = 1\Omega$

$$P_t = P_c\left[1 + \frac{\mu^2}{2}\right]$$

$$= 50\left[1 + \frac{(0.29)^2}{2}\right]$$

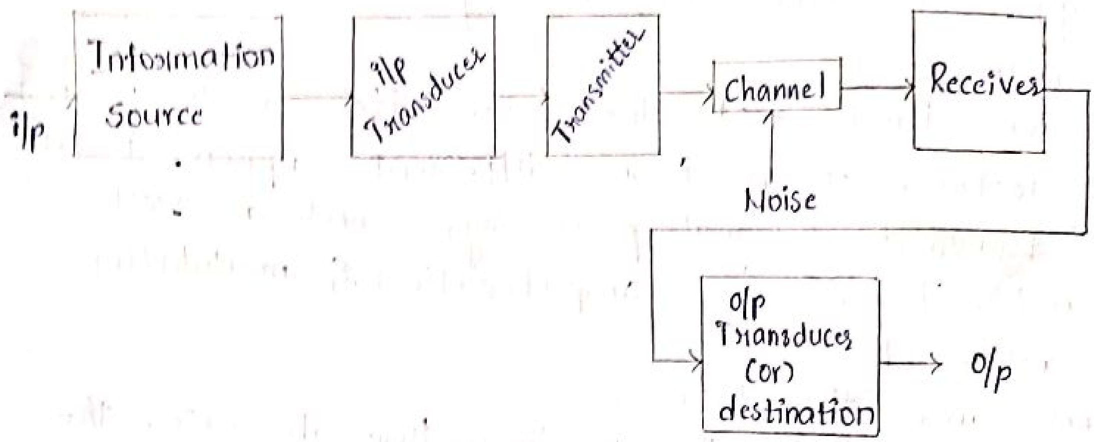
$$= 52.1 \text{ watts}$$

$$\mu^2 = \mu_1^2 + \mu_2^2$$
$$= 0.5^2 + 0.2^2$$
$$= 0.25 + 0.04$$
$$= 0.29$$

$$P_{SB} = P_t - P_c$$
$$= 52.1 - 50$$
$$= 2.1 \text{ watts}$$

# Analog Communications

Communication:
The process of establishing a connection b/w 2 points for information interchange is called as communication
× The process of exchanging information b/w any 2 points is called as communication.
The electronic equipment which are used for the purpose of communication is called as communication equipments.
when different communication equipments are connected together it forms a system called as communication system.
Elements of Communication System (or)
Block diagram of Communication system:




Block diagram of Analog Communication System

Transducer:-
It is an electronic device which converts one form of energy into another form & Vice Versa.
Ex:- Electromagnetic waves → Electric signals
Transmitter:-
The o/p of the transducer are lowlevel signals.
· Transmitter performs 3 operations.
1. Amplification
2. Modulation
3. Processing
DeModulation:-
The process of recovering the signal from the noise signal.

Information Source :-

The main source of information source is to produce the required message which has to be transmitted. The input source may be a voice signal, Music signal, Video signal (or) a data from a computer

Input Transducer :-

A transducer is a device which converts one form of energy into another form. The message from the i/p source may or maynot be in electric form. In case when the message is not in electric form an input transducer is used to convert it into a time varying electrical signal. One example of transducer is microphone which converts sound energy into electrical energy.

Transmitter :

The main function of the transmitter is to process the electrical signal from different aspects. Inside the transmitter signal processings such as restriction of audio frequencies, amplification & modulation are achieved.

Channel and the Noise :-

The channel means the medium through which the message signal travels from transmitter to the receiver. In other words we can say that the function of a channel is to provide a physical connection b/w the transmitter and receiver

During the process of transmission and reception the signal gets distorted due to the noise introduced in the system. Noise is an unwanted signal which tends to interfere with the required message signal

**Receiver :-**

The main function of the receiver is to reproduce the original message signal in electric form. The process of recovering (or) reproducing the original message signal is called as demodulation.

Demodulation is the reverse process of modulation carried out in transmitter block.

**Output Transducer :-**

Output transducer is to convert the electricalsignal coming out from the receiver section into its original form.

**Destination :-**

Destination is the final stage which produces the message signal in its original form.

For example: In radiobroadcasting, the destination is a loud speaker.

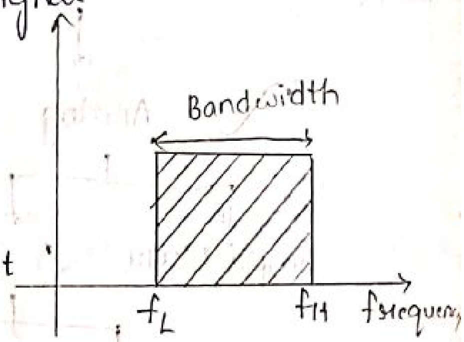
**Concept of Band width :-**

The portion of electromagnetic or spectrum occupied by a signal can be called as a band width. (or)

The frequency range over which an information signal is transmitted is called as band width.

Band width is the difference between upper and lower frequency limits of a signal.

$\times$ Band width = $f_H - f_L$




The table below shows the frequency ranges and the band width requirement of different types of signals.

| Type of signal | Frequency range | Bandwidth |
|---|---|---|
| Voice signal | 300 - 3400Hz | 3100Hz |
| Music signal | 20 - 15000 Hz | 14980Hz |
| Video signal | 0 - 5MHz | 5MHz |
| Data from Computer (if data cables are used) | 300 - 3400Hz | 3100Hz |

Modulation Process:

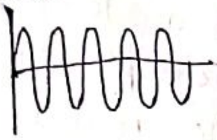
Generally, the purpose of any communication system is to deliever a message (or) information from the i/p source to the destination in a recognizable form. To do this, the transmitter modifies the information (or) message signal into a form which is suitable for the transmission over the channel.

This modification is achieved by using the process named as 'modulation'

Modulation :- ***

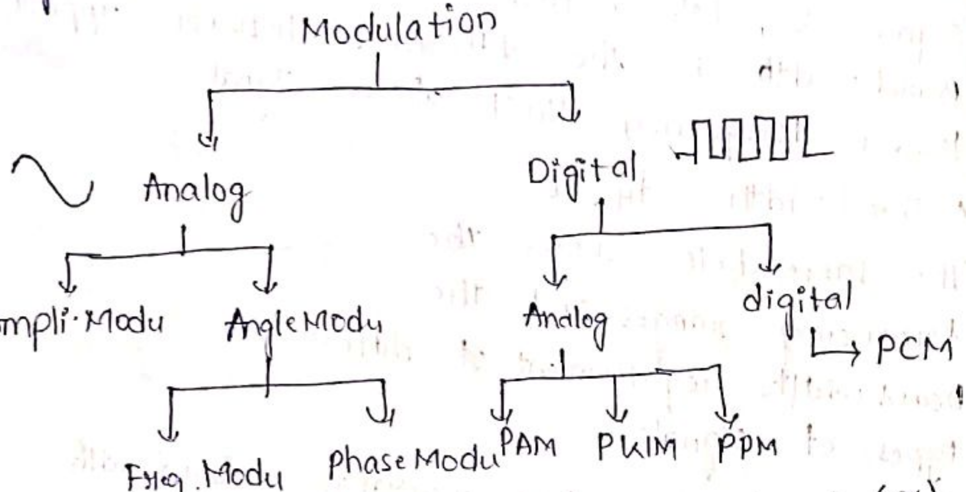
The process of changing (or) varying characteristics of a carrier signal (amplitude, frequency, phase) in accordance with the instantaneous values of a message signal is called as 'modulation'.

[Carrier signal - high frequency signal]



Types of Modulation:-

Basically, modulation is classified into 2 types depending on the type of the carrier signal.




→ input signal is known as message signal (or) information signal (or) modulating signal (or) base band signal.

Amplitude Modulation :-
The process of changing the amplitudes of a carrier signal in according to instantaneous values of the amplitudes of message signal keeping frequency and phase constant is called as amplitude modulation.

Frequency Modulation :-
The process of changing the frequency of a carrier signal according to instantaneous values of the frequencies of message signal keeping amplitude and phase constant is called as frequency modulation.

Phase Modulation :-
The process of changing the phase of a carrier signal according to instantaneous values of the phases of message signal keeping amplitude and frequency constant is called as phase modulation.

Need for modulation :-
1. To reduce the height of antenna.
when free space is selected as a transmission medium, then antennas are used for transmitting and receiving the signals.
Suppose if we transmit a 5KHz signal then the height of the antenna required to be constructed is given as $l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times5\times10^3} = 15000$ mts.
This is practically not possible to construct an antenna of such height.
If we translate a 5KHz signal into a high frequency 3MHz signal by using modulation technique. Then, the height of the antenna required to be constructed is given as $l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times3\times10^6} = 25$ mts.
This is practically possible to construct an antenna of such height.
2. To increase the range of communication
Generally, most of the message signal are of low frequency type. They get easily attenuated and effected by noise signals. So, the low frequency msg signals cannot be transmitted directly

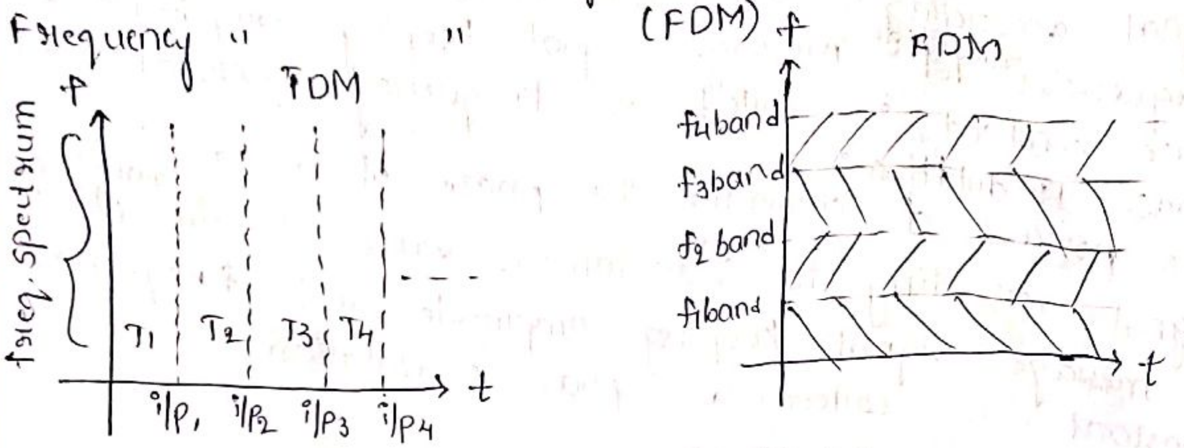
over long distances. In order to transmit over long distances, modulation is needed.

3. It allows multiplexing.

The process of transmitting two or more i/p signals through a single channel is called as multiplexing. Multiplexing is classified into 2 types.

Time division Multiplexing (TDM)

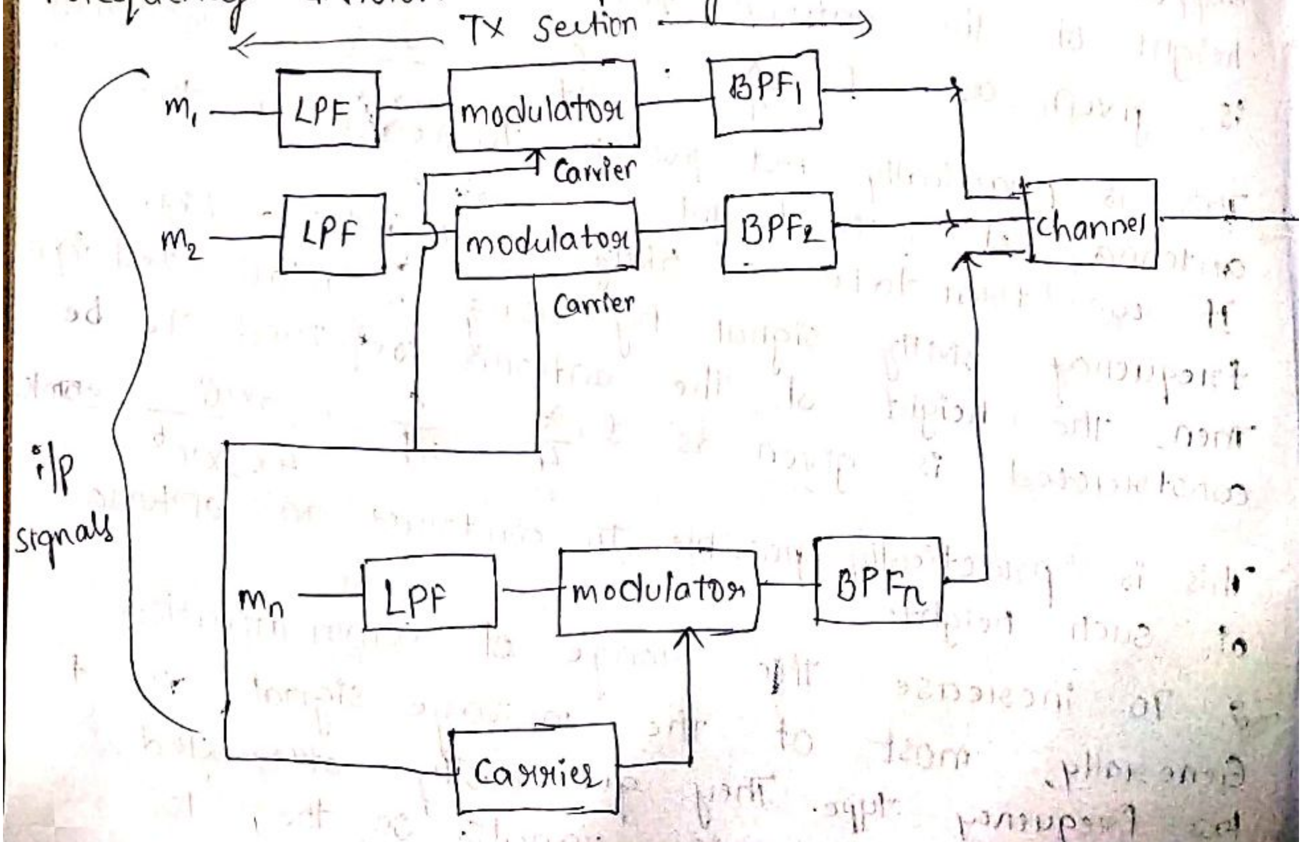
Frequency " " (FDM)




4. Reduction of noise.

Noise is the major draw back in any communication system. Although, noise cannot be eliminated completely, using modulation we can reduce the effect of noise

Frequency division Multiplexing:



$m(t) = 2\cos 5000t \qquad \omega_m = 5000$

$c(t) = 5\cos 20000t \qquad \omega_c = 20000$

$m_a = \frac{2}{5} = 0.4 \qquad 2\pi f_m = 5000$

$f_m = 795.77$

$f_{USB} = f_c + f_m \qquad 2\pi f_c = 20000$

$f_{USB} = 3978.86 \qquad f_c = 3183.09$

$f_{LSB} = f_c - f_m = 2387.32$

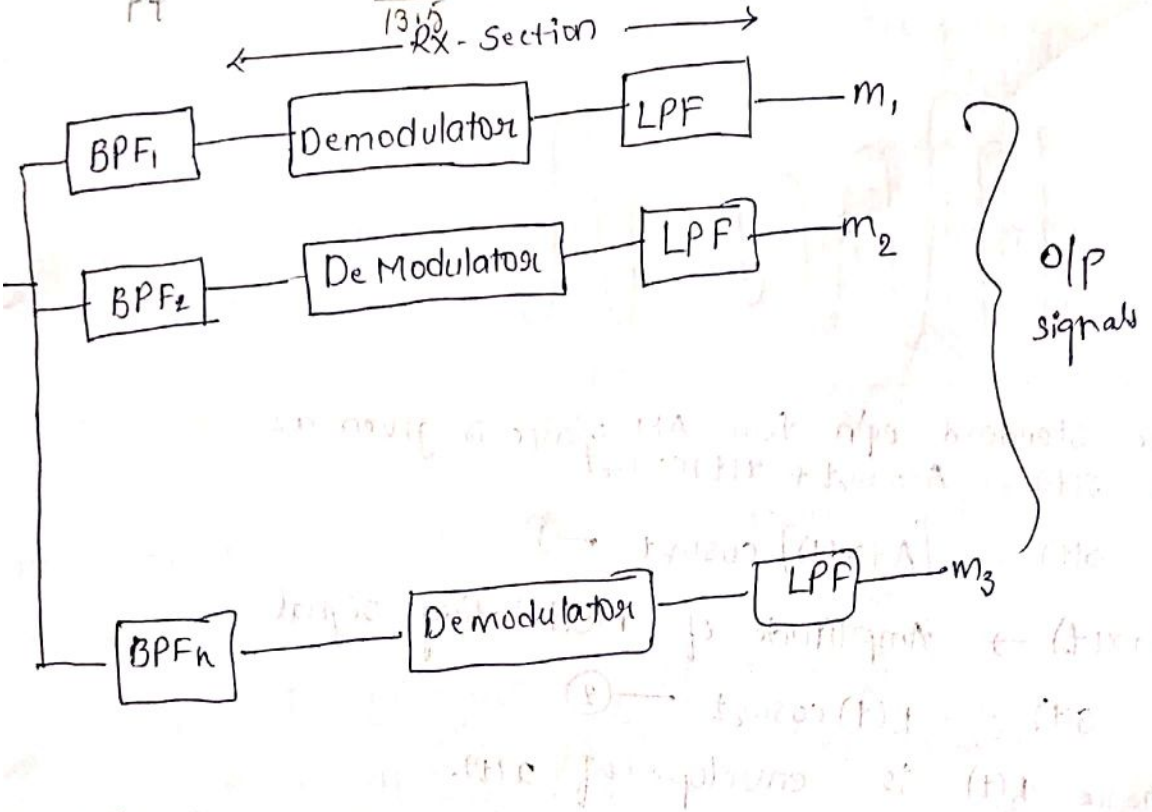
$BW = 2f_m = 1591.54$

$\frac{P_s}{P_t} = ? \qquad P_t = P_c\left[1 + \frac{m_a^2}{2}\right] \qquad P_t = P_c + P_s$

$\frac{P_t - P_c}{P_t} \qquad P_t = \frac{25}{2}\left[1 + \frac{(0.4)^2}{2}\right]$

$P_t = 13.5$

$1 - \frac{P_c}{P_t} = 1 - \frac{12.5}{13.5} = 0.074$

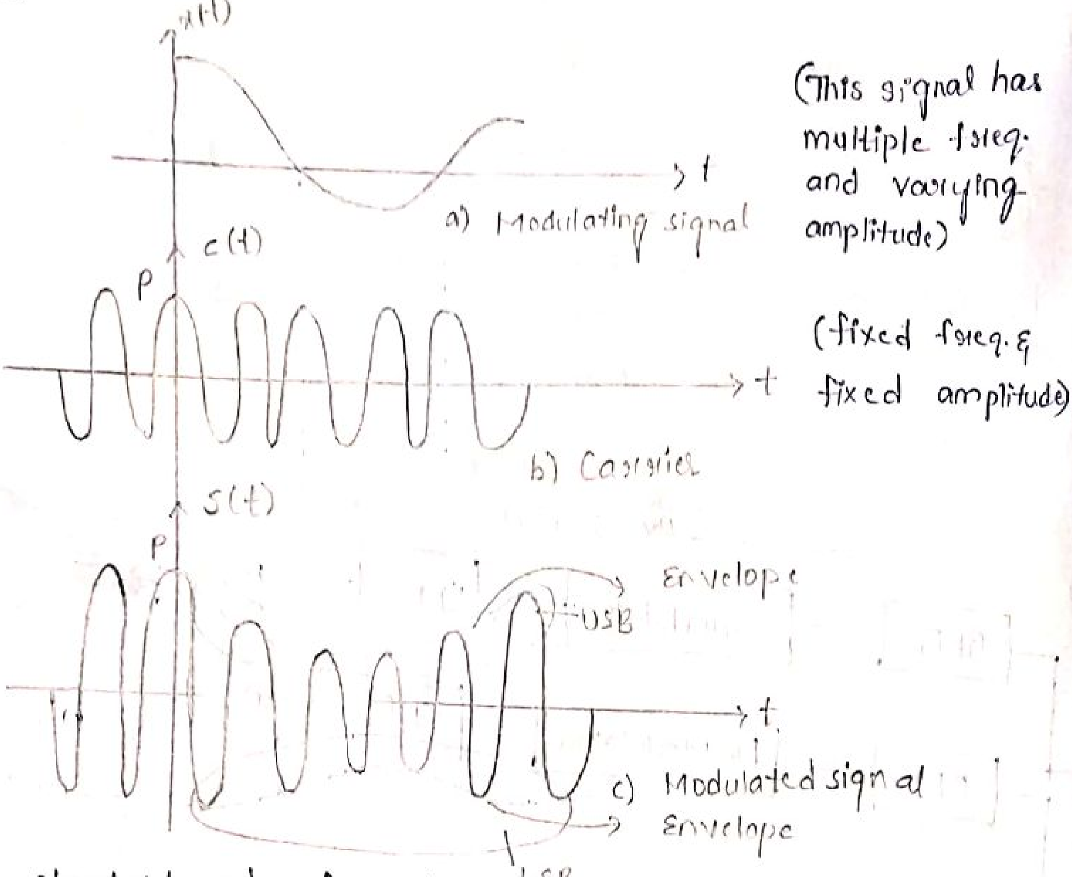


Amplitude Modulation (Time domain Analysis) :- (Multi-tone AM)

It is a process in which the amplitude of carrier signal is made proportional to the instantaneous amplitudes of message signal/modulating signal.

→ Let $x(t)$ be the i/p signal (or) modulating signal to be transmitted.

Let us consider a high frequency carrier signal $c(t) = A\cos\omega_c t$ where $A$ = max. amplitude of carrier signal and $\omega_c$ represents angular frequency of carrier signal.




The standard eq'n for AM wave is given as

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

$s(t) = [A + x(t)]\cos\omega_c t$ —①

$A + x(t)$ → Amplitude of modulating signal

$s(t) = E(t)\cos\omega_c t$ —②

where $E(t)$ is envelope of $x(t)$.

According to AM, the amplitude of carrier signal is made proportional to " " msg " as shown in fig (c).

eq 1 & eq 2 represents the time domain behaviour of AM wave.

Frequency domain Analysis of AM wave:

If $x(t)$ is the i/p signal (or) modulating signal and $c(t) = A\cos\omega_c t$ is a high frequency carrier signal then the standard equation for AM wave is given as

$s(t) = A\cos\omega_c t + x(t)\cos\omega_c t$ —(1)

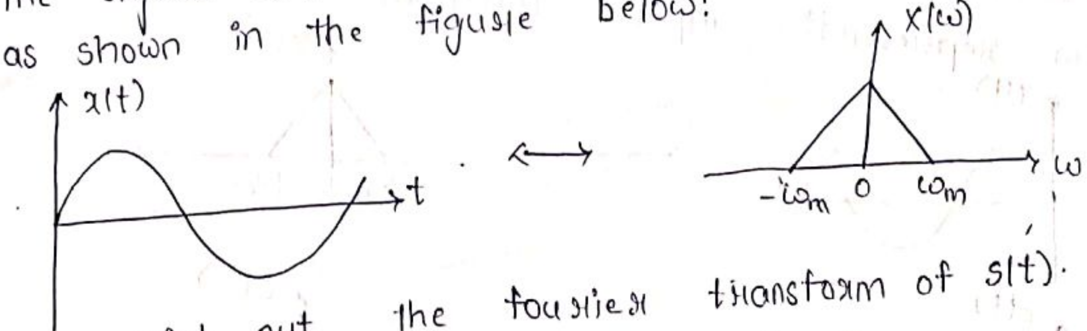
This eq. represents the time domain behaviour of AM wave.

If we want to know the frequency representation (or) the frequencies present in AM wave, we need to find out the frequency spectrum (or) frequency domain analysis of FM wave. For that purpose, we need to find out fourier transform of $s(t)$.

Let $X(\omega)$ is a fourier transform of $x(t)$ and $c(\omega)$ is the fourier transform of $c(t)$ and $S(\omega)$ is the FT of $s(t)$ respectively.

Let the modulating signal $x(t)$ be band limited to the interval $-\omega_m < \omega < \omega_m$. This indicates that modulating signal doesnot have any frequency component outside the interval $(-\omega_m, \omega_m)$.

The signal $x(t)$ and its fourier transform is as shown in the figure below:




Now, point out the fourier transform of $s(t)$.

$$FT[s(t)] = FT[A\cos\omega_c t + x(t).\cos\omega_c t] \quad -(2)$$

$$= FT[A\cos\omega_c t] + FT[x(t)\cos\omega_c t]$$

$$= FT\left[A\left(\frac{e^{j\omega_c t} + e^{-j\omega_c t}}{2}\right)\right] + FT[x(t)\cos\omega_c t]$$

Consider, $FT[A\cos\omega_c t] = \frac{A}{2} FT\left[e^{j\omega_c t} + e^{-j\omega_c t}\right]$

$$= \frac{A}{2}\left[2\pi\delta(\omega-\omega_c) + 2\pi\delta(\omega+\omega_c)\right]$$

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$$FT[A\cos\omega_c t] = A\pi[\delta(\omega-\omega_c)+\delta(\omega+\omega_c)] \quad \text{③}$$

Consider $FT[x(t)\cos\omega_c t] = FT\left[x(t)\left(\frac{e^{j\omega_c t}+e^{-j\omega_c t}}{2}\right)\right]$

$$= \frac{1}{2}\left\{FT[x(t)e^{j\omega_c t}] + FT(x(t)e^{-j\omega_c t})\right\} \quad \text{④}$$

From the frequency shifting theorem of fourier transform, we know that

$$x(t) \longleftrightarrow X(\omega)$$

$$x(t)e^{j\omega_c t} \longleftrightarrow X(\omega-\omega_c)$$

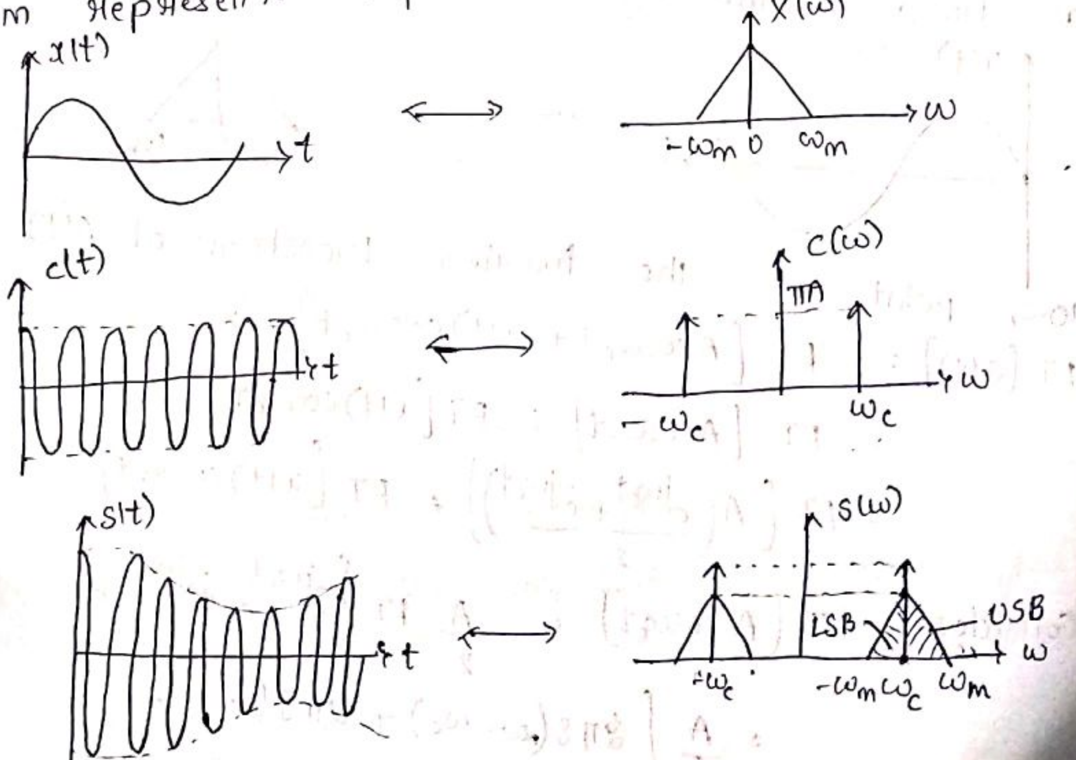
$$x(t)e^{-j\omega_c t} \longleftrightarrow X(\omega+\omega_c)$$

eq ④ $\Rightarrow$ $FT[x(t)\cos\omega_c t] = \frac{1}{2}[X(\omega-\omega_c)+X(\omega+\omega_c)]$ ⑤

From eq's 2, 3, 5 we can write

$$\boxed{S(\omega) = A\pi[\delta(\omega-\omega_c)+\delta(\omega+\omega_c)] + \frac{1}{2}[X(\omega-\omega_c)+X(\omega+\omega_c)]}$$

The above eq. represents the frequency domain of AM wave.
It represents 2 components. The first component represents the spectrum of original base band signal shifted in positive direction as well as negative direction by an amount of $\omega_c$. The 2nd term represents impulse functions.



The band frequencies which lies above the carrier signal frequency is called upper side band else lower side band.

From the fig. band width modulated signal is

band width = $\omega_c + \omega_m - (\omega_c - \omega_m)$

$BW = 2\omega_m$

$\therefore$ It is represented as twice of the maximum frequency of modulating signal.

Modulation Index (Amplitude Sensitivity):-

Modulation Index is basically the ratio by which the modulated signal varies from an unmodulated carrier signal (or)

Modulation Index is simply the ratio of maximum amplitude of modulating signal and the maximum amplitude of carrier signal. It is denoted by 'ma'

It is given by $\boxed{m_a = \frac{|x(t)|_{max}}{A}}$

→ It can also be expressed in terms of percentage called percentage of modulation.

$m_a \times 100 \rightarrow \frac{|x(t)|_{max}}{A} \times 100$

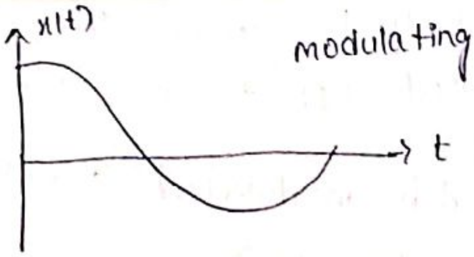
→ It is also known as depth of modulation (or) Modulation factor.

* $m_a \leq 1$, if we want to completely recover our msg signal, from the envelope of AM wave.

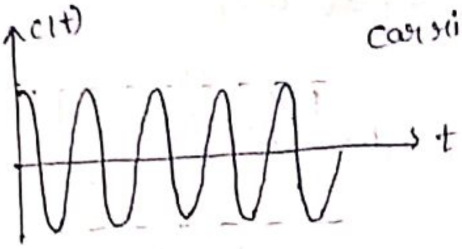
* $m_a > 1$, Envelope distortion occurs.

* In order to get distorted o/p.

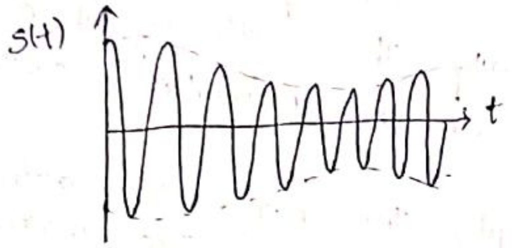
$m_a < 1$ (or) $x(t)_{max} < A_c$ (or) $m.I < 100\%$



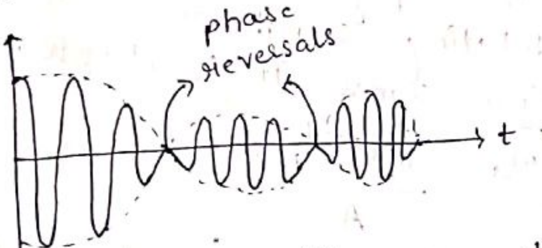

modulating signal




carrier sgl.




AM wave for ma<1 (under modulation)




AM wave for ma > 1

over modulation

*** Power Content in a multitone amplitude Modulation

Multitone modulation is a type of modulation in which the modulating signal consists of multiple frequencies in it

Let us consider a carrier signal $A\cos\omega_c t$ is modulated by a multitone modulating signal which is given as $x(t) = V_1\cos\omega_1 t + V_2\cos\omega_2 t + V_3\cos\omega_3 t$ —①

we know that the standard equation for an AM wave is represented by s(t).

$$s(t) = A\cos\omega_c t + x(t)\cos\omega_c t \quad \text{—②}$$

Sub. eq① in eq②

$$s(t) = A\cos\omega_c t + (V_1\cos\omega_1 t + V_2\cos\omega_2 t + V_3\cos\omega_3 t)\cos\omega_c t$$

$$= A\cos\omega_c t\left[1 + \frac{V_1}{A}\cos\omega_1 t + \frac{V_2}{A}\cos\omega_2 t + \frac{V_3}{A}\cos\omega_3 t\right]$$

wkT, $\dfrac{\text{The maximum amplitude of modulating signal}}{\text{Maximum amplitude of carrier signal}}$

$= M_a$

Here $\frac{V_1}{A} = m_1$ ; $\frac{V_2}{A} = m_2$ ; $\frac{V_3}{A} = m_3$

$\therefore s(t) = A\cos\omega_c t\left[1 + m_1\cos\omega_1 t + m_2\cos\omega_2 t + m_3\cos\omega_3 t\right]$

$s(t) = A\cos\omega_c t + m_1 A\cos\omega_c t\cos\omega_1 t + m_2 A\cos\omega_2 t\cos\omega_c t +$

$m_3 A\cos\omega_3 t\cos\omega_c t$ —(3)

Let us calculate the power content in an AM wave.
The power content in an AM wave can be calculated as $P_t = P_c + P_S$ —(4) where as $P_t$ = total power
$P_c$ : carrier power
$P_S$ : sidebands power

Let us calculate the carrier power.
The power carried out by carrier signal is nothing but the mean square value of $A\cos\omega_c t$

$$P_C = \overline{[A^2\cos^2\omega_c t]}^2$$

$$= \frac{1}{2\pi}\int_0^{2\pi} A^2\cos^2\omega_c t\, dt$$

$$= \frac{A^2}{2\pi}\int_0^{2\pi}\left(\frac{1+\cos 2\omega_c t}{2}\right)dt$$

$$= \frac{A^2}{4\pi}\left(t + \frac{\sin 2\omega_c t}{2\omega_c}\right)_0^{2\pi}$$

$$= \frac{A^2}{4\pi}(2\pi) = \frac{A^2}{2}$$

$$\therefore P_C = \frac{A^2}{2} \quad \text{—(5)}$$

Generally, in AM transmission, a band pass filter (or) a tuned ckt with cut off frequency $\omega_c$, is used to reject higher frequency component i.e., the 2nd integral term in above is zero.
Now, let us calculate the side band power.
The power carried out by side bands can be calculated by using the equation.

$$P_S = \frac{1}{2}\overline{[x(t)]}^2$$

$$= \frac{1}{2}\left[\overline{(V_1\cos\omega_1 t)}^2 + \overline{(V_2\cos\omega_2 t)}^2 + \overline{(V_3\cos\omega_3 t)}^2\right]$$

$$= \frac{1}{2}\left[\overline{(m_1 A\cos\omega_1 t)}^2 + \overline{(m_2 A\cos\omega_2 t)}^2 + \overline{(m_3 A\cos\omega_3 t)}^2\right]$$

$$P_s(t) = \frac{1}{2}\left[\frac{m_1^2A^2}{2} + \frac{(m_2A)^2}{2} + \frac{(m_3A)^2}{2}\right]$$

$$P_s(t) = \frac{A^2}{4}\left[m_1^2 + m_2^2 + m_3^2\right] \quad \text{--- (6)}$$

Sub. eq 5, 6 in eq (4)

$$P_t = P_c + P_s$$

$$= \frac{A^2}{2} + \frac{A^2}{4}(m_1^2 + m_2^2 + m_3^2)$$

$$P_t = \frac{A^2}{2}\left\{1 + \frac{1}{2}(m_1^2 + m_2^2 + m_3^2)\right\}$$

$$\boxed{P_t = P_c\left\{1 + \frac{m_1^2 + m_2^2 + m_3^2}{2}\right\}}$$

The above expression can be extended upto 'n' frequency components

$$P_t = P_c\left\{1 + \frac{m_1^2 + m_2^2 + m_3^2 + \cdots + m_n^2}{2}\right\}$$

Total modulation Index (or) net modulation index:-

We know that, the total power transmitted (or) carried out by an multitone AM wave is

$$P_t = P_c\left[1 + \frac{m_1^2 + m_2^2 + m_3^2 + \cdots + m_n^2}{2}\right]$$

The above equation can also be expressed as

$P_t = P_c\left[1 + \frac{m_t^2}{2}\right]$, where $m_t^2 = m_1^2 + m_2^2 + \cdots + m_n^2$ (or)

$m_t = \sqrt{m_1^2 + m_2^2 + m_3^2 + \cdots + m_n^2}$.

$m_t$ = total modulation index (or) net modulation index

Single tone Amplitude Modulation :- (Time domain)

Single tone amplitude modulation is a type of modulation in which the modulating signal is a fixed frequency signal (single frequency).

Let us consider a carrier signal $c(t) = A\cos\omega_c t$ is modulated by a single frequency modulating signal $x(t)$ ~~which~~ is

$x(t) = V_m \cos \omega_m t$ — ①

WKT, the standard equation for an AM wave is

$s(t) = A \cos \omega_c t + \overset{x(t)}{V_m} \cos \omega_c t$ — ②

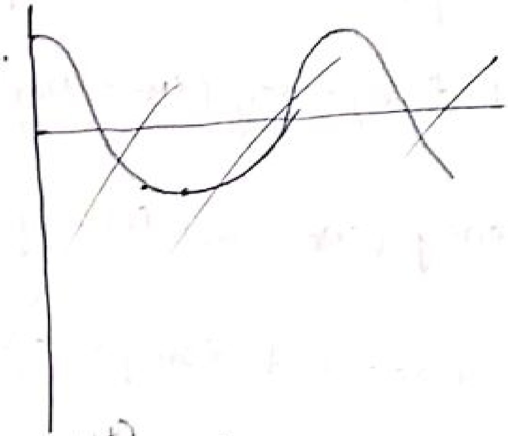
substitute eq ① in eq ②

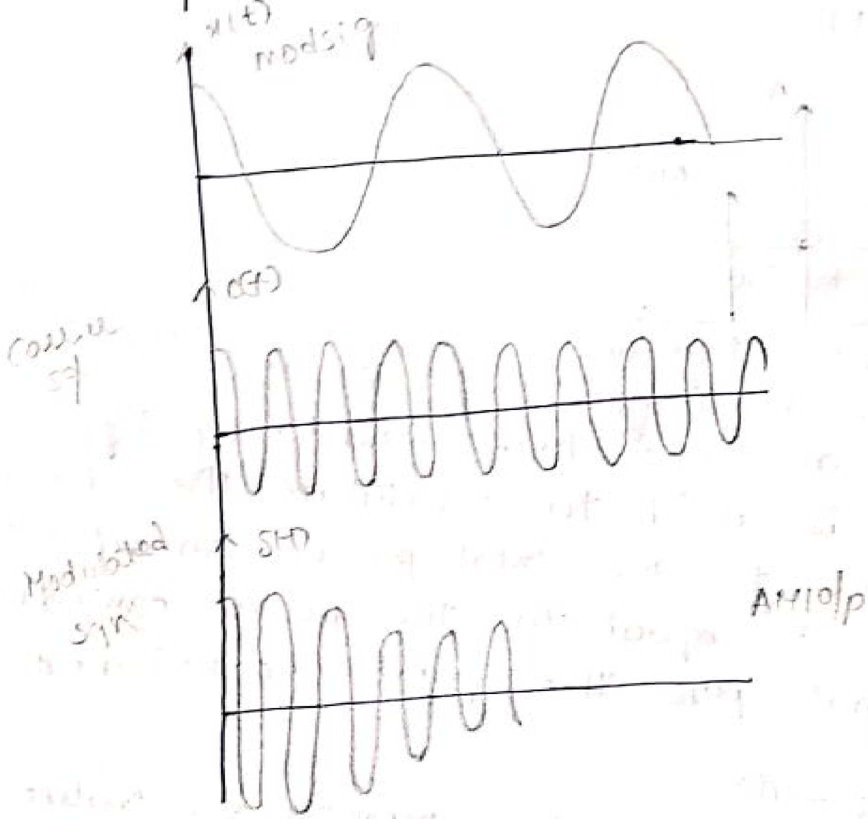
$s(t) = A \cos \omega_c t + V_m \cos \omega_m t \cos \omega_c t$

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

here $\frac{V_m}{A} = m_a$

$\therefore \boxed{s(t) = A \cos \omega_c t \left[ 1 + m_a \cos \omega_m t \right]}$





Frequency domain representation

For a single tone AM,

WKT $s(t) = A\cos\omega_c t\,[1 + m_a\cos\omega_m t]$

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

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

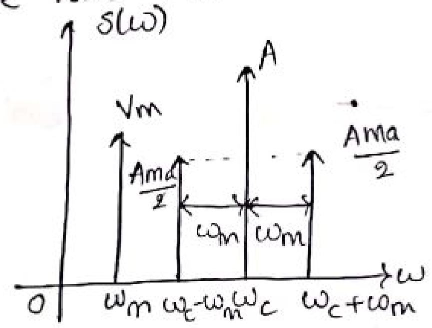
$$s(t) = A\cos\omega_c t + \frac{Am_a}{2}\left[\cos(\omega_c t + \omega_m t) + \cos(\omega_c t - \omega_m t)\right]$$

① ② ③

The above eq. reveals 3 frequency components.

1. Carrier frequency component '$\omega_c$' having an amplitude of 'A'.
2. Represents upper side band frequency $(\omega_c + \omega_m)$ having an amplitude $\frac{Am_a}{2}$
3. Lower side band frequency $(\omega_c - \omega_m)$ having an amplitude $\frac{Am_a}{2}$

The fig. below shows the onesided frequency spectrum of single tone AM




Power Content in a Single tone AM :-

Power Content in a Single tone modulating signal

Let us consider a single tone modulate the carrier $x(t) = V_m\cos\omega_m t$ is used to modulate the carrier signal $c(t) = A\cos\omega_c t$. The total power carried out by an AM wave is equal to the power carried out by carrier signal plus the power carried out by side bands.

$$P_t = P_c + P_s \quad \text{—①}$$

Here $P_c$ is nothing but the meansquare value of $A\cos\omega_c t$

$$P_c = \left[\overline{A\cos\omega_c t}\right]^2$$

$$P_c = \frac{A^2}{2} \quad \text{—②}$$

Similarly, the side band power $P_s$ is given by

$$P_s = \frac{1}{2}\overline{x(t)}$$

$$= \frac{1}{2}\overline{[V_m \cos\omega_m t]^2}$$

$$= \frac{1}{2}\frac{V_m^2}{2}$$

$$P_s = \frac{V_m^2}{4} \quad \text{--- (3)}$$

$\therefore$ The total power transmitted in AM wave is

$$P_t = \frac{A^2}{2} + \frac{V_m^2}{4}$$

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

$$\boxed{P_t = P_c\left[1 + \frac{1}{2}m_a^2\right]}$$

Current Calculation in a Single tone AM wave :-

In AM it is generally more convient to measure the AM transmitter current than the power. In this case, the modulation index may be calculated from the values of unmodulated and modulated currents in the AM transmitter.

WKT, the total power transmitted in a singletone AM wave is

$$P_t = P_c\left(1 + \frac{m_a^2}{2}\right) \quad \text{--- (1)}$$

Let us assume $I_t$ is the total transmitted current (or) modulated current and $I_c$ is the carrier current (or) unmodulated current.

From eq (1)

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

$$\frac{I_t^2 \cdot \cancel{R}}{I_c^2 \cdot \cancel{R}} = 1 + \frac{m_a^2}{2}$$

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

$$\frac{I_t}{I_c} = \sqrt{1 + \frac{m_a^2}{2}}$$

$$I_t = I_c\sqrt{1 + \frac{m_a^2}{2}}$$

1. Derive the expression for modulation index in terms of $P_t$ and $P_c$.
Derive the expression for modulation index in terms of $I_t$ and $I_c$.
2. A modulating signal $10\sin(2\pi\times10^3 t)$ is used to modulate a carrier signal $20\sin(2\pi\times10^4 t)$. Determine the modulation index, % of $m_a$, frequencies of side band components and their amplitudes. what will the bandwidth of modulated signal.
3. A 400 watts carrier is modulated to a depth of 75 %. Find the total power in the amplitude modulated wave
4. An AM broadcast radio transfer radiates 10K watts of power if % modulation % is 60. Calculate the carrier power.

1. WKT $P_t = P_c\left[1+\frac{m_a^2}{2}\right]$ ; $I_t = I_c\sqrt{\left(1+\frac{m_a^2}{2}\right)}$

$$\frac{P_t}{P_c} = 1+\frac{m_a^2}{2} \qquad \frac{I_t}{I_c} = \sqrt{\frac{m_a^2}{2}+1}$$

$$2\left(\frac{P_t}{P_c}-1\right) = m_a^2 \qquad \left(\frac{I_t}{I_c}\right)^2-1 = \frac{m_a^2}{2} \qquad 2\left(\frac{I_t}{I_c}-1\right)$$

$$m_a = \sqrt{2\left(\frac{P_t-P_c}{P_c}\right)} \qquad m_a = \sqrt{2\left(\frac{I_t-I_c}{I_c}\right)}$$

$$m_a = \sqrt{2\left(\frac{I_t}{I_c}\right)^2-2}$$

2. Given,

$x(t) = 10\sin(2\pi\times10^3 t)$ ; $c(t) = 20\sin(2\pi\times10^4) t$

$$m_a = \frac{|x(t)|_{max}}{|c(t)|_{max}} = \frac{10}{20} = 0.5$$

$\% m_a = 50\%$

$f_L = 10^3$ $f_H = 10^4$

$BW = f_H - f_L \Rightarrow 10^4 - 10^3 \Rightarrow 10$

Upper Side band frequency $= \omega_c+\omega_m = f_c+f_m$
$= 10^4+10^3$

$$\text{Amplitude} = \frac{A m_a}{2} = \frac{20\times(0.5)}{2} = 5 = 11000$$

Lower side band frequency $= \omega_c - \omega_m = 10^4 - 10^3 = 9000\,Hz$

Amplitude $= \frac{A m_a}{2} = \frac{\overset{10}{\cancel{20}} \times 0.5}{\cancel{2}} = 5$

Band width $= 2f_m = 2 \times 10^3 = 2000\,Hz.$

3) Given,

$P_c = 400\,W$

$m_a = 75\% = 0.75$

$P_t = ?$

$$P_t = P_c\left[1 + \frac{m_a^2}{2}\right]$$

$$P_t = 400\left[1 + \frac{(0.75)^2}{2}\right]$$

$$P_t = 512.5\,W$$

4) $m_a = 60\% = 0.6$

$P_c = ?$

$P_t = 10000$

$$P_t = P_c\left[1 + \frac{m_a^2}{2}\right]$$

$$10000 = P_c\left[1 + \frac{(\overset{0.6}{\cancel{60}})^2}{2}\right]$$

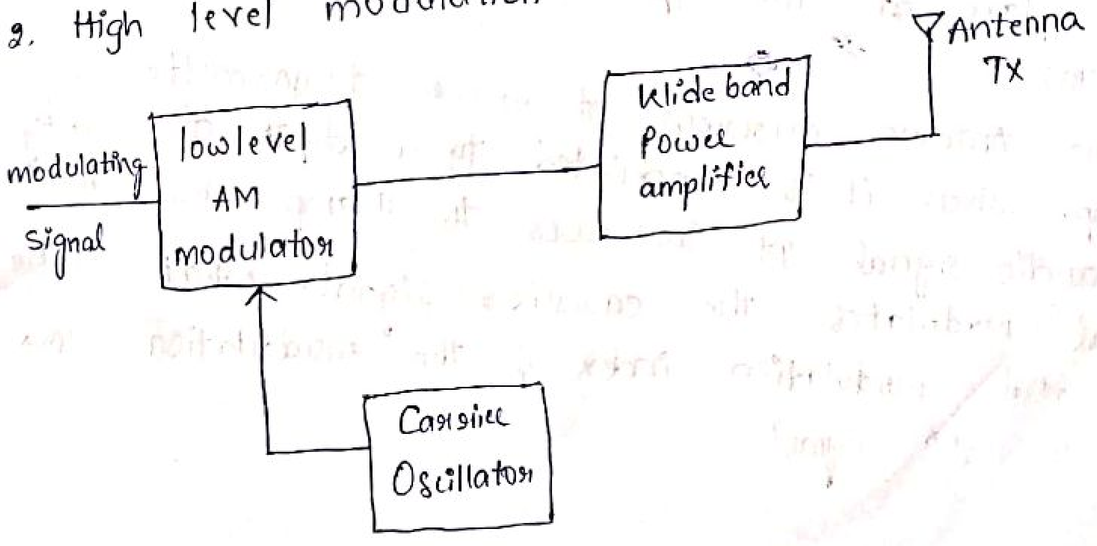
$P_c = \cancel{555\,W}\ 8474.5\,W = 8.47\,kW$

Generation of Amplitude Modulation:

The circuit which is used to generate AM wave is called as AM generator.

Generally, Amplitude modulation can be achieved by using two types of techniques:

1. Low level modulation
2. High level modulation



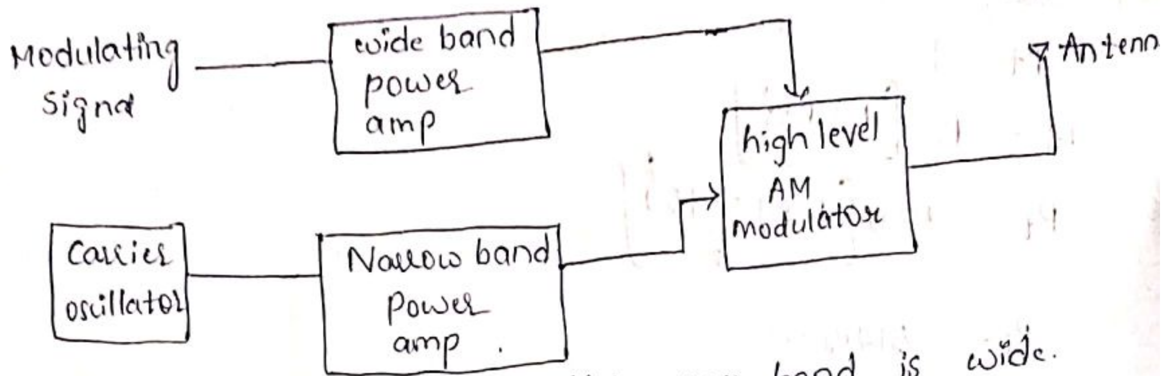
Carrier Oscillator is used to generate Carrier Signal.

High level Modulation :-

Some of the examples of low level modulation :-

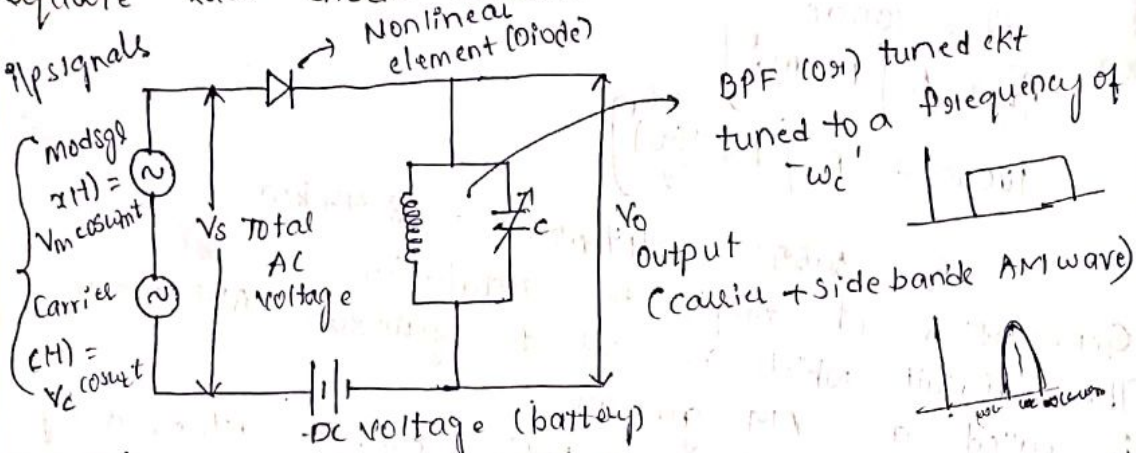
1. Square law diode modulation
2. Switching Modulation

High level Modulation :-




In wide band power amplifier pass band is wide.
In narrow band power amplifier pass band is narrow.

Square Law diode Modulation :-




Principle :-
when two signals of different frequencies are passed through a non linear element, then AM modulation takes place at the o/p.

Problems :-

5) The Antenna current of an AM transmitter is 10 Amp. when it is modulated to a depth of 30% by an audio signal. It increases to 11 amp when another signal modulates the carrier signal. what will be the total modulation index & the modulation index due to 2nd signal?

$I_{t1} = 10A$

$M_1 = m_a = 0.3$ ; $m_t = ?$ when $I_{t2} = 11A$

$M_2 = ?$

wKT $I_t = I_c \sqrt{1 + \frac{m_t^2}{2}}$

$10 = I_c \sqrt{1 + \frac{(0.3)^2}{2}}$

$10 = I_c (1.022)$

$I_c = 9.784\,A$

$I_{t2} = I_c \sqrt{1 + \frac{m_t^2}{2}}$

$11 = 9.784 \sqrt{1 + \frac{m_t^2}{2}}$

$1 + \frac{m_t^2}{2} = 1.263$

$\frac{m_t^2}{2} = 0.263$

$m_t^2 = 0.527$

$m_t = 0.725$

wKT $m_t^2 = m_1^2 + m_2^2$

$(0.725)^2 = (0.3)^2 + m_2^2$

$m_2^2 = 0.435$

$m_2 = 0.65 = 65\%$

6) The antenna current of an AM transmitter is modulated to a depth of 40% by an audio wave is 11 amp. It increases to 12 amp as a result of simultaneous modulation by another audio wave. what is the modulation index due to the 2nd wave

$I_{t1} = 11$ amp ; $I_{t2} = 12A$

$m_1 = 0.4$ ; $m_2 = ?$

$I_{t1} = I_c \sqrt{1 + 0.08}$

$11 = I_c (1.039)$

$I_c = 10.58$

$I_{t2} = I_c \sqrt{1 + \frac{m_t^2}{2}}$

$\frac{12}{10.58} = \sqrt{1 + \frac{m_t^2}{2}}$

$1.134 = \sqrt{1 + \frac{m_t^2}{2}}$

$$\frac{m_t^2}{2} = 0.286$$

$$m_t = 0.756$$

wkT $m_t^2 = m_1^2 + m_2^2$

$$(0.756)^2 = 0.16 + m_2^2$$

$$0.411 = m_2^2$$

$$m_2 = 0.641 = 64\%$$

7) A carrier wave is represented by the expression $V_c(t) = 10\sin\omega t$ then draw the wave form of an AM wave for $m = 0.5$.

Given that, $V_c(t) = 10\sin\omega t$

$M = 0.5$

$A_c = 10V$

Inorder to draw AM output we need to find out $E_{max}$ & $E_{min}$.

wkT the modulation index $m_a = \frac{A_m}{A_c}$

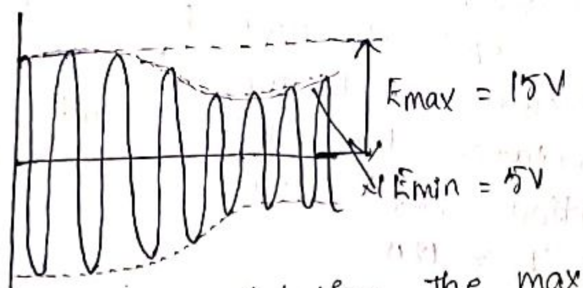
$$0.5 = \frac{A_m}{10}$$

$$A_m = 5V$$

$$E_{max} = A_c + A_m$$

$$= 5 + 10 = 15V$$

$$E_{min} = A_c - A_m = 10 - 5 = 5V$$




8) P.T in amplitude modulation the maximum avg. power transmitted by an antenna is 1.5 times the carrier power?

Given that,

$$P_t = P_c\sqrt{1 + \frac{m_a^2}{2}} \quad \text{—①}$$

$M_a = 1$ ∵ The maximum value, without introducing any distortion in AM o/p

$m_a > 1$ = o/p gets distorted.

substitute Ma = 1 in above equation.

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

$$P_t = P_c(1.5)$$

Square law diode Modulator:

The fig. shows the ckt diagram of square law diode modulation in which modulating signal x(t) and carrier signal c(t) are applied across the diode.

Let us consider $x(t) = V_m \cos\omega_m t$ and $c(t) = V_c \cos\omega_c t$ where $\omega_m$ is modulating frequency and $\omega_c$ is carrier frequency.

From the fig. we can observe that the total AC voltage which is applied across a non linear element (diode) is $V_s$.

$$V_s = V_m \cos\omega_m t + V_c \cos\omega_c t \quad -(1)$$

wkt the non linear relationship b/w voltage and current for a diode can be expressed as

$$i_o = a + bV_s + cV_s^2 \quad -(2)$$

i: o/p current

$V_s$: total i/p voltage applied across the diode

Substitute eq (1) in eq (2)

$$i = a + b\left[V_m \cos\omega_m t + V_c \cos\omega_c t\right] + c\left[V_m \cos\omega_m t + V_c \cos\omega_c t\right]^2$$

$$i = a + b\left[V_m \cos\omega_m t + V_c \cos\omega_c t\right] + c\left[V_m^2 \cos^2\omega_m t + V_c^2 \cos^2\omega_c t + 2V_m V_c \cos\omega_m t \cos\omega_c t\right]$$

$$i = a + b\left[V_m \cos\omega_m t + V_c \cos\omega_c t\right] + c\left[V_m^2\left(\frac{1+\cos 2\omega_m t}{2}\right) + V_c^2\left(\frac{1+\cos 2\omega_c t}{2}\right) + 2V_m V_c \cos\omega_m t \cos\omega_c t\right]$$

$$i = a + bV_m \cos\omega_m t + bV_c \cos\omega_c t + \frac{cV_m^2}{2} + \frac{cV_m^2 \cos 2\omega_m t}{2} + \frac{cV_c^2}{2} + \frac{cV_c^2 \cos 2\omega_c t}{2}$$

$$+ c\left[V_m V_c \cos(\omega_m + \omega_c)t + \cos(\omega_m - \omega_c)t\right]$$

$$i = a + bV_m \cos\omega_m t + bV_c \cos\omega_c t + \frac{cV_m^2}{2} + \frac{cV_m^2 \cos 2\omega_m t}{2} + \frac{cV_c^2}{2}$$

$$+ \frac{cV_c^2 \cos 2\omega_c t}{2} + cV_m V_c \cos(\omega_m + \omega_c)t + cV_m V_c \cos(\omega_m - \omega_c)t$$

$$i = \underbrace{a + \frac{cV_m^2}{2} + \frac{cV_c^2}{2}}_{1} + \underbrace{bV_m \cos\omega_m t}_{2} + \underbrace{bV_c \cos\omega_c t}_{3} + \underbrace{\frac{cV_m^2 \cos 2\omega_m t}{2}}_{4}$$

$$+ \underbrace{\frac{cV_c^2 \cos 2\omega_c t}{2}}_{4} + \underbrace{cV_m V_c \left\{\cos(\omega_m + \omega_c)t + \cos(\omega_m - \omega_c)t\right\}}_{5}$$

The above equation reveals 5 frequency components. They are

1. DC components of the signal
2. modulating signal frequency $\omega_m$.
3. Carrier signal frequency $\omega_c$ ✓
4. harmonics of modulating signal and carrier signal having frequencies $2\omega_m$ and $2\omega_c$.
5. represents upper side band and lower side band frequencies $(\omega_c+\omega_m)$, $(\omega_c-\omega_m)$ ✓

Now, all these frequency components are applied to a tuned circuit which is set to a centre frequency of '$\omega_c$'. From all these frequency components only the carrier frequency and the sideband frequencies are allowed by the tuned circuit and all other remaining frequency components are rejected by the tuned circuit.

Therefore, the required expression for the o/p current is $i = bV_c\cos\omega_c t + cV_mV_c\left[\cos(\omega_m+\omega_c)t + \cos(\omega_m-\omega_c)t\right]$.

$$i = bV_c\cos\omega_c t + c2V_mV_c\cos\omega_m t\cos\omega_c t$$

$$i = bV_c\cos\omega_c t\left[1 + \frac{c2V_m}{b}\cos\omega_m t\right]$$

$$i = bV_c\cos\omega_c t\left[1 + m_a\cos\omega_m t\right]$$

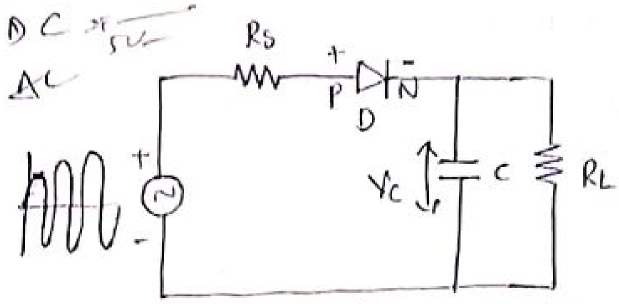
where $m_a = \dfrac{2cV_m}{b}$ represents modulation index.

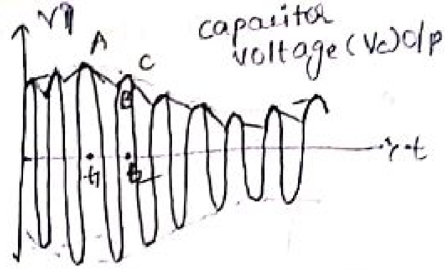
The above expression represents carrier frequency and two side band frequencies which is nothing but amplitude modulated wave. The ckt acts as a AM generator.

AM detection (or) Envelope detection (or) AM demodulation:-

The figure shows the circuit diagram of envelope detector.

we know that, the envelope of AM wave follows the amplitude variations of the modulating signal. This circuit tries to detect the envelope of the AM wave Thats why it is called as Envelope Detector




Envelope detection is a very simple and inexpensive as it contains diode and a few resistors, capacitor

Circuit Operation :-

For positive half cycle of AM wave.

Diode is ON - conducting - capacitor charge through $R_S$.

For negative half cycle of AM wave

Diode is OFF - stops conducting - capacitor discharges through $R_L$.

* $R_S C << \frac{1}{f_c}$

$R_L C >> \frac{1}{f_c}$

Efficiency of amplitude modulation :-

The efficiency of a AM wave is the ratio of power carried out by the sidebands to the total power carried out by an AM wave

power carried out by side bands

$$\eta = \frac{\text{Power carried out by side bands}}{\text{total power carried out by an AM wave}}$$

$$= \frac{\frac{V_m^2}{4}}{P_c\left[1+\frac{m_a^2}{2}\right]}$$

wkT $m_a = \frac{V_m}{A_c}$

$V_m = m_a A_c$

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

$$= \frac{\frac{m_a^2 \cancel{A_c^2}}{\cancel{4}\, 2}}{\left(\frac{\cancel{A_c^2}}{\cancel{2}}\right)\left[1+\frac{m_a^2}{2}\right]}$$

$$= \frac{\frac{m_a^2}{\cancel{2}}}{\frac{2+m_a^2}{\cancel{2}}} = \frac{m_a^2}{m_a^2+2}$$

$$\eta = \frac{m_a^2}{m_a^2+2}$$

For 100% modulation, for $m_a = 1$,

$$\eta = \frac{1}{1+2} = \frac{1}{3} = 0.33$$

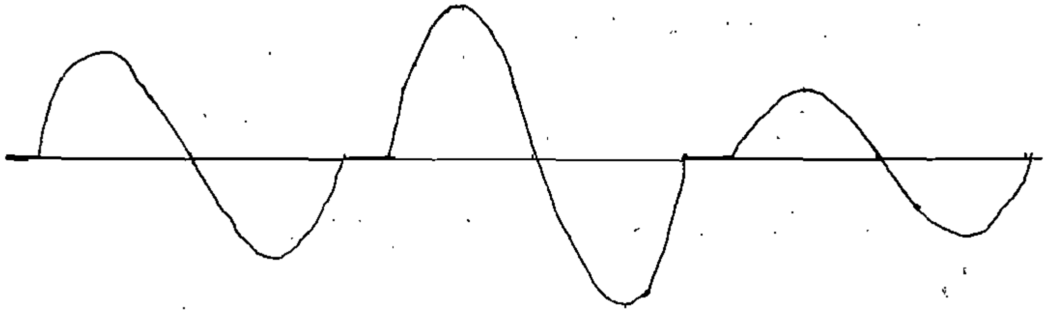
$$\eta\% = 33.3\%$$

# DSB-Modulation

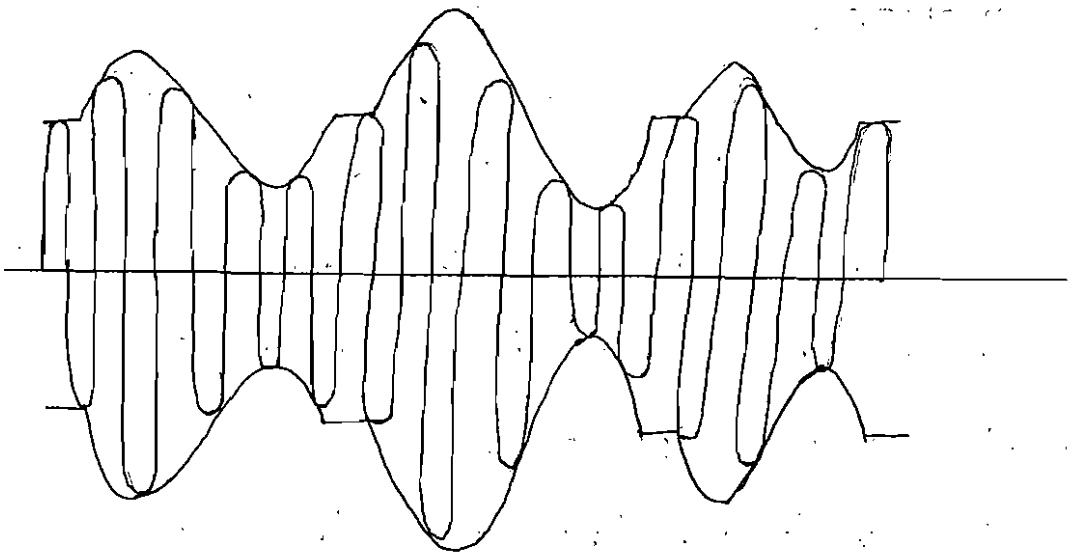
## Introduction :-

When the carrier is amplitude modulated by a single sine wave the resulting signal consists of three frequencies i.e... Original carrier and two side bands. This system is known as Double side-band full carrier (DSBFC).

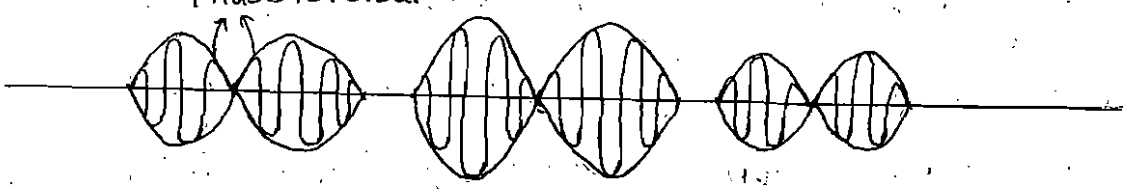
Generally we know that the carrier signal does not convey any information. The real information is conveyed by two side bands. But two third of power is wasted to transmit the carrier signal. So we can supress the carrier and remaining signal contains only lower and upper side bands. Then the signal is called Double side band supressed carrier (DSBSC). Now the power wasted for carrier can be put in to the side bands, to transmit for longer distances.



(a) modulating signal



(b) AM wave




(c) Supressed carrier wave.

Time domain description :-

When the carrier is supressed during the modulation process, the resulting is only the algebraic sum of upper and lower side bands. The equation of DSBSC AM signal is given as

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

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

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

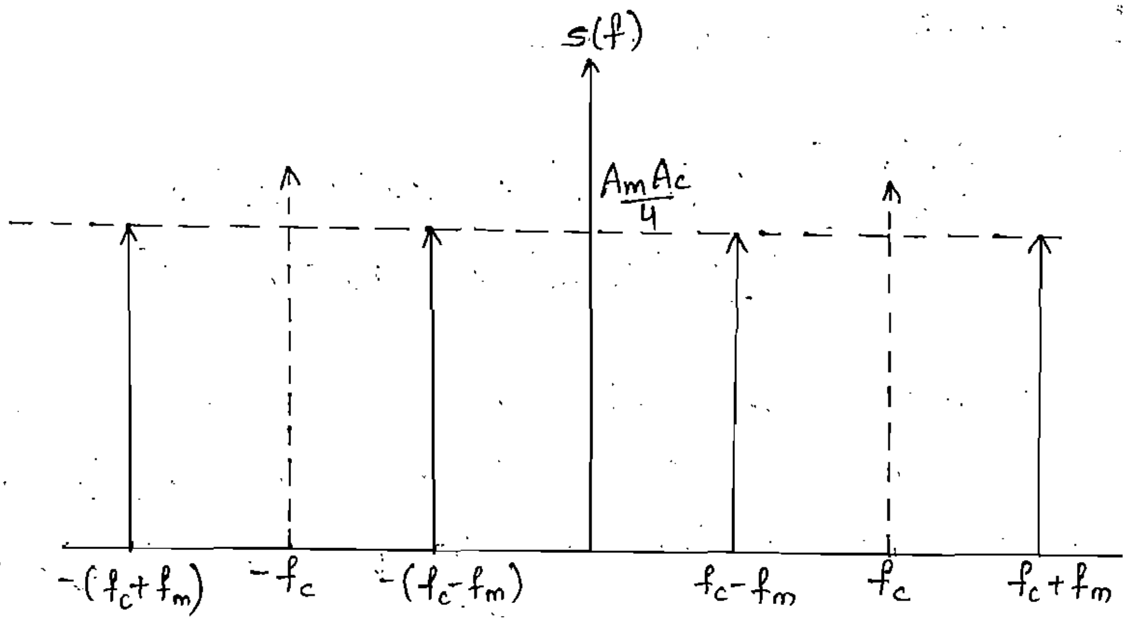
$f_c + f_m$ is upper side band ; $f_c - f_m$ is lower side band.

**Frequency domain description :=**

In DSBSC, as the carrier is supressed it is denoted by dotted line. The spectrum space occupied by DSB is $2f_m$ or $2W$, which is same as that for a conventional AM signal.

From time domain description we have

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




Spectrum of DSBSC modulated wave

**Band width :=**

The spectrum space occupied by DSB-SC is $2f_m$, so its band width is $2f_m$.

## Single tone modulation :-

If the modulating signal contains only one frequency then it is called single tone modulation.

The sinusoidal modulating signal is given by

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

There fore, DSBSC modulated wave is given by

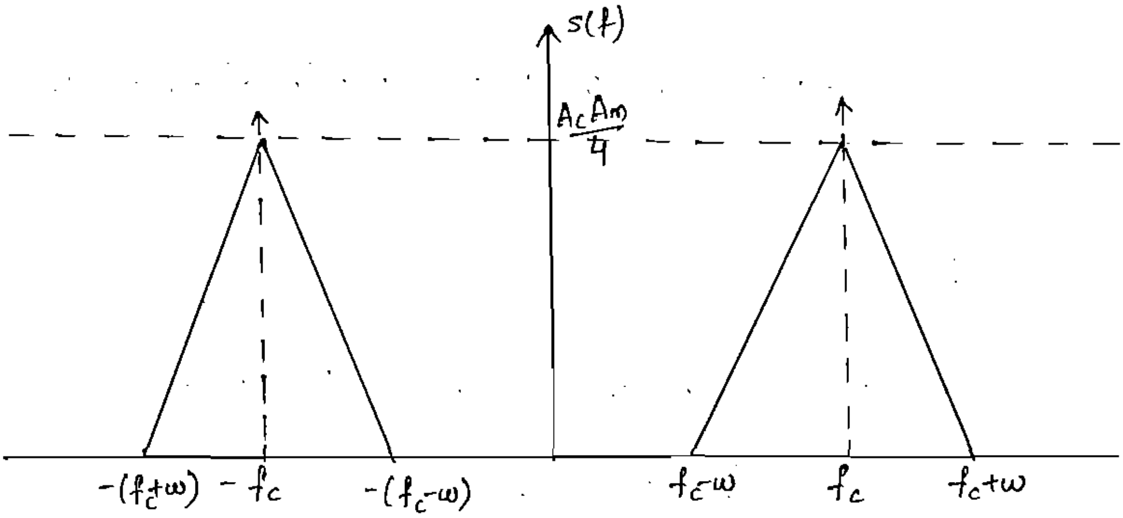
$$s(t) = A_c A_m \cos 2\pi f_c t \cos 2\pi f_m t$$

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

By applying fourier transform

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

We can observe that the spectrum of DSBSC modulated wave for the case of a sinusoidal modulating wave consists of delta functions located at $f_c \pm f_m$ & $-f_c \pm f_m$




spectrum of single tone AM-SC

Power :=

As the carrier is absent the total power is only due to side bands.

$$P_{DSBFC} = P_c + P_{sc} = P_c + P_c \frac{\mu^2}{2}$$

Carrier is supressed so

$$P_{DSBSC} = P_c \frac{\mu^2}{2}$$

$$\% \text{ power saving in DSBSC} = \frac{P_{DSBFC} - P_{DSBSC}}{P_{DSBFC}} \times 100$$

$$= \frac{P_c\left(1+\frac{\mu^2}{2}\right) - P_c \frac{\mu^2}{2}}{P_c\left(1+\frac{\mu^2}{2}\right)} \times 100$$

$$= \frac{2}{2+\mu^2} \times 100.$$

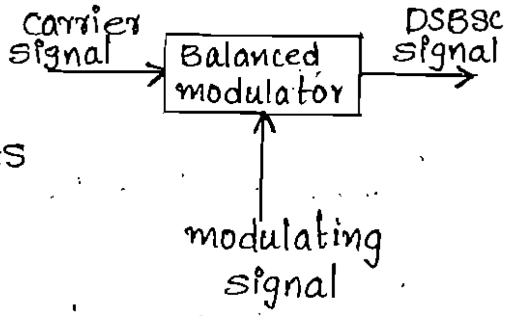
if $\mu = 1$ then % power saving in DSBSC is 66.6%.

Generation of DSBSC wave :=

For generating a DSBSC signal we have seen that m(t) and c(t) should get multiplied with each other. So this type of can be obtained by using a product modulator or balanced modulator.

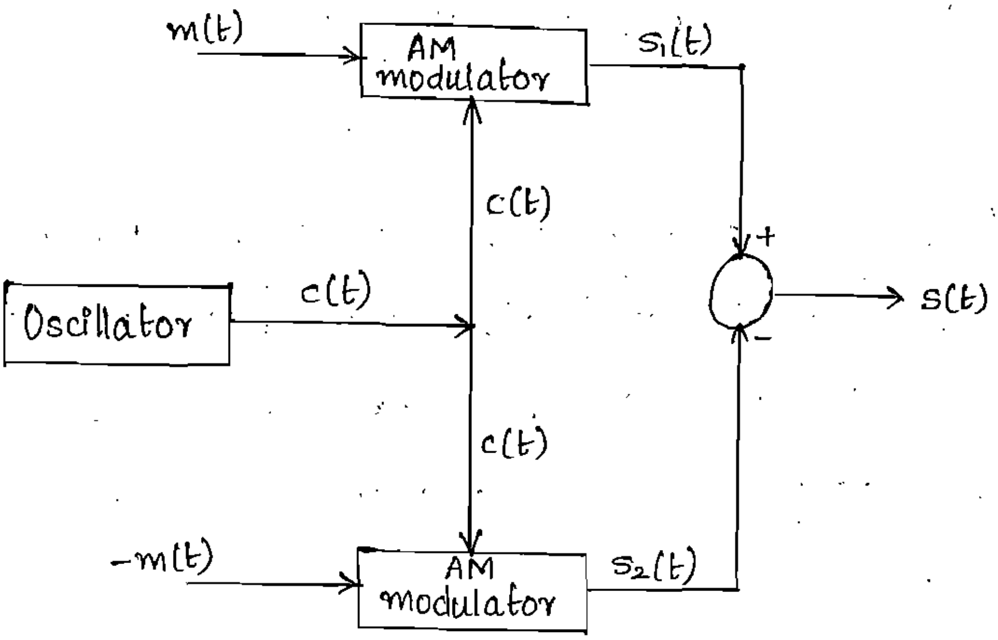
## Balanced Modulator :=

The balanced modulator is used to supress the carrier from the AM signal. The inputs are carrier and modulating signal. The outputs are upper and lower side bands.




## Principle used in balanced modulator :=

When two signals at different frequencies are passed through a non linear resistance, the AM signal is generated with supressed carrier. The non linear devices such as diode, JFET and transistor are used in balanced modulator.



The balanced modulator circuit consists of two standard amplitude modulators which are arranged in balanced configuration, to supress the carrier. The two modulators are identical except for the sign reversal of the modulating wave applied to the input of one of them.

The outputs of modulators can be given as

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

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

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

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

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

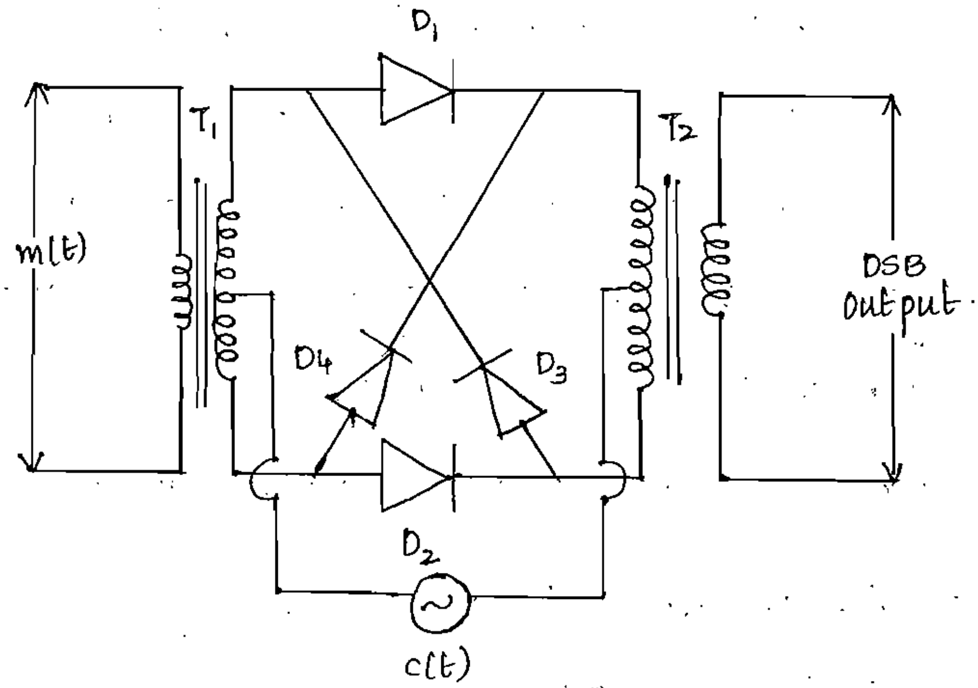
$$= 2 k_a m(t) \, c(t).$$

The balanced modulator output is equal to the product of modulating and carrier.

There are two types of balanced modulators.

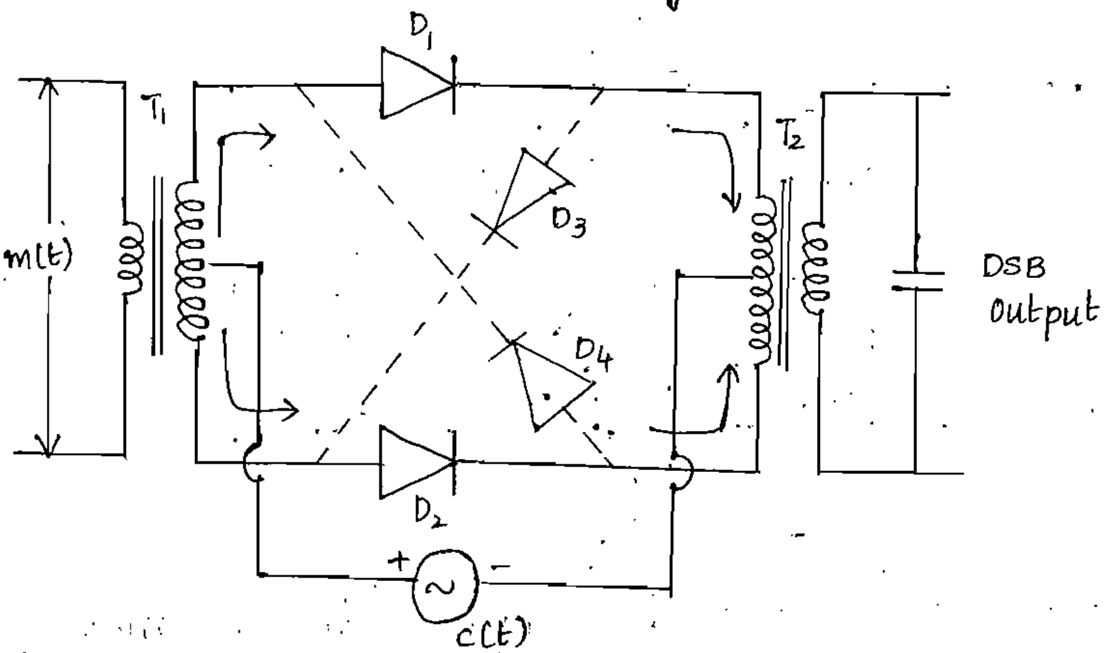
* Diode or Ring or Lattice modulator.
* JFET modulator.

## Diode Modulator :-




Diode modulator consists of an input transformer $T_1$ and an output transformer $T_2$ and four diodes connected in a bridge circuit. In this the carrier signal is applied to center taps of input and output transformers and modulating signal is applied to input transformer $T_1$. The output appears across secondary of the transformer. The diodes connected in the bridge acts like switches and are controlled by carrier signal because it is higher in frequency and amplitude than the modulating signal.

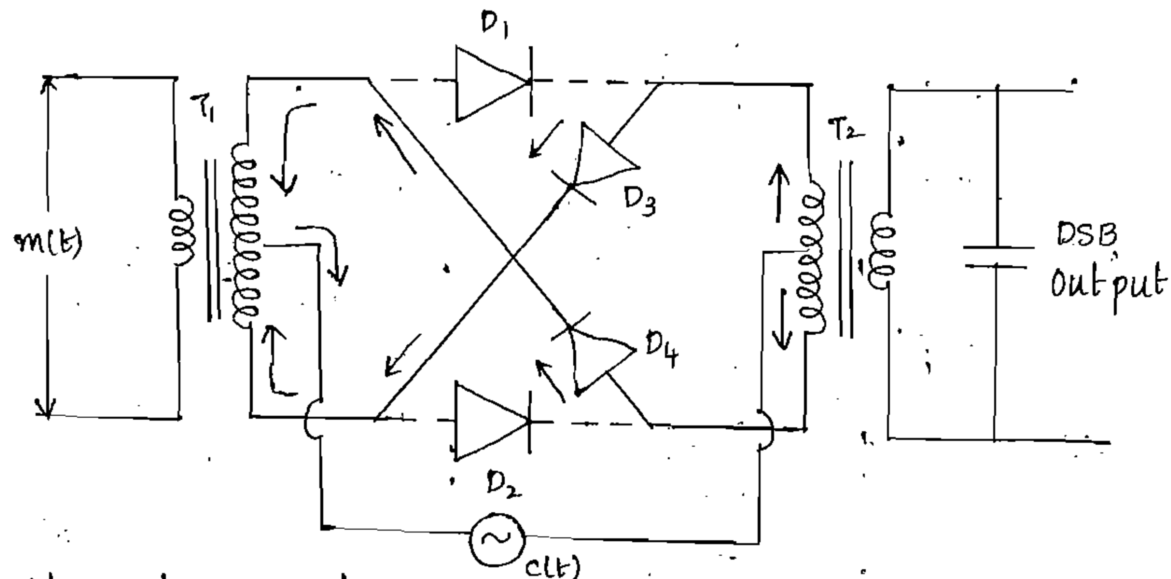
Positive half cycle of carrier signal:-




In this we assume that the modulating input is zero. In positive half cycle of carrier signal, the diodes $D_1$ and $D_2$ are forward biased and $D_3$ and $D_4$ are reverse biased. The current divides equally in upper and lower portions of primary winding of $T_2$. The current produce magnetic field which is equal and opposite in upper and lower part of the winding. Hence they cancel each other and produce no output at secondary of $T_2$. Thus the carrier is supressed.

Negative half of carrier signal:-

In negative half of carrier signal the diodes $D_1$ and $D_2$ are reverse biased and $D_3$ and $D_4$ are forward biased.




Similar to positive half cycle, the magnetic fields in primary winding of $T_2$ are equal and opposite canceling each other. There fore output at $T_2$ secondary is zero.

The supression of carrier in both the half cycles depends on diode characteristics and exactness of center tap of transformer to give exactly equal upper and lower currents and magnetic fields.

With modulating signal:-

Before we assumed that the modulating input is zero. But now consider a low frequency sine wave is applied to primary of $T_1$ as modulating signal. This will appears across $T_1$ secondary. The diodes $D_1$ and $D_2$ are forward biased and they connect secondary of $T_1$ to primary ~~of $T_2$. As a result the modulating~~ signal is applied to primary of $T_2$. In negative half cycle, the diodes $D_3$ and $D_4$ are forward biased. athey connect secondary of $T_1$ to primary of $T_2$ with reverse connections which results in 180° phase shift in modulated signal.

This balanced modulator works perfectly when the carrier signal is square wave

The expression can be given as

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

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

$$= \left(\frac{4}{\pi} \cos 2\pi f_c t - \frac{4}{3\pi} \cos(3 \times 2\pi f_c t) + \cdots\cdots\right) m(t)$$

$$= \frac{4}{\pi} \cos 2\pi f_c t \; m(t) - \frac{4}{3\pi} \cos(3 \times 2\pi f_c t) m(t) + \cdots\cdots$$

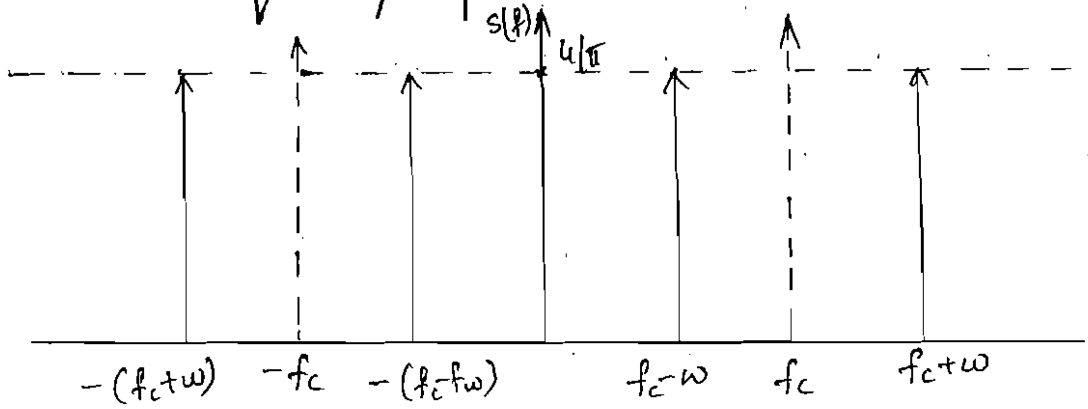
Apply fourier transforms

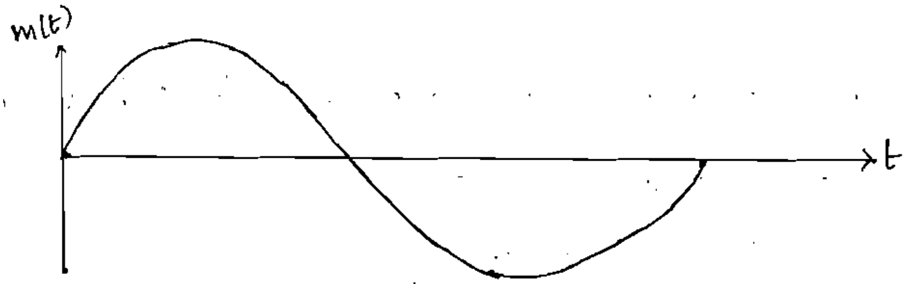
$$s(f) = \frac{4}{\pi}\left[M(f-f_c) + M(f+f_c)\right] - \frac{4}{3\pi}\left[M(f-3f_c) + M(f+3f_c)\right] + \cdots$$

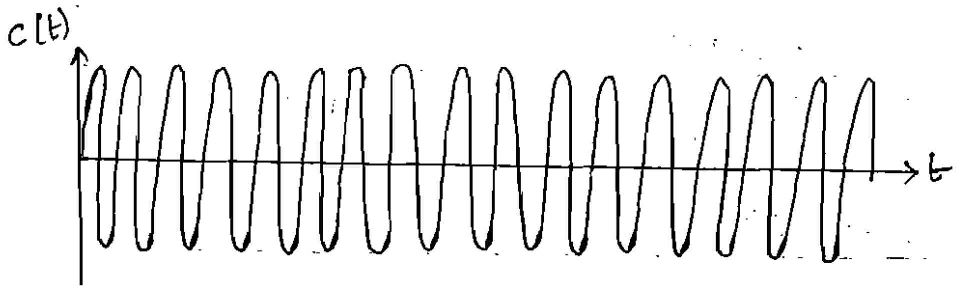
Eliminate high frequency terms then we get

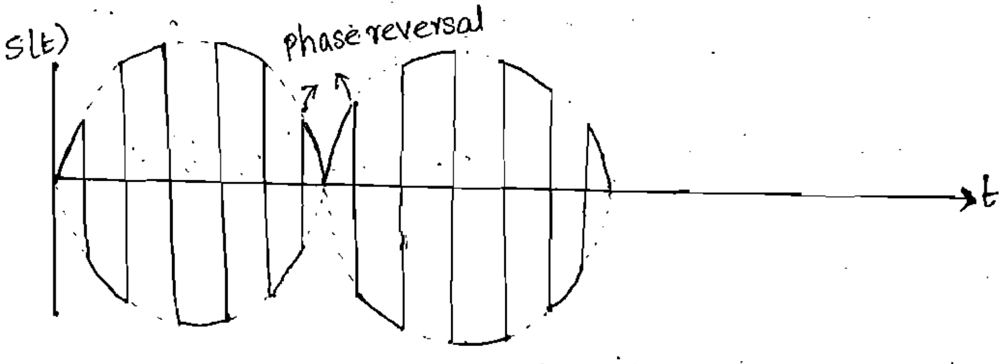
$$s(f) = \frac{4}{\pi}\left[M(f-f_c) + M(f+f_c)\right].$$

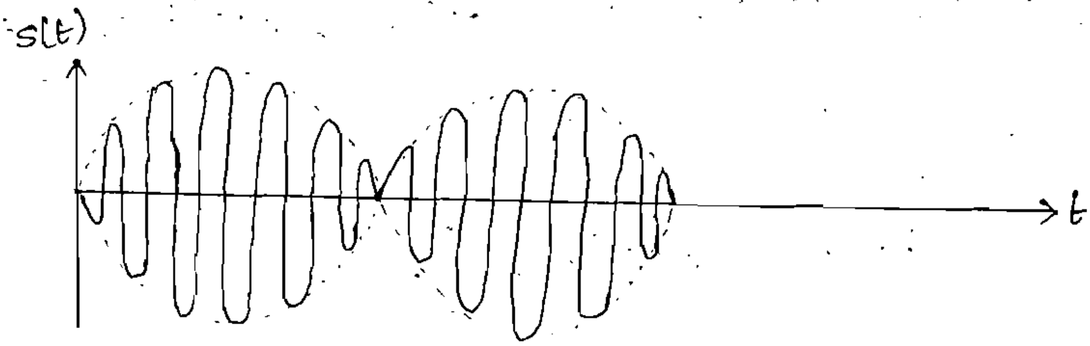
Then the frequency spectrum is



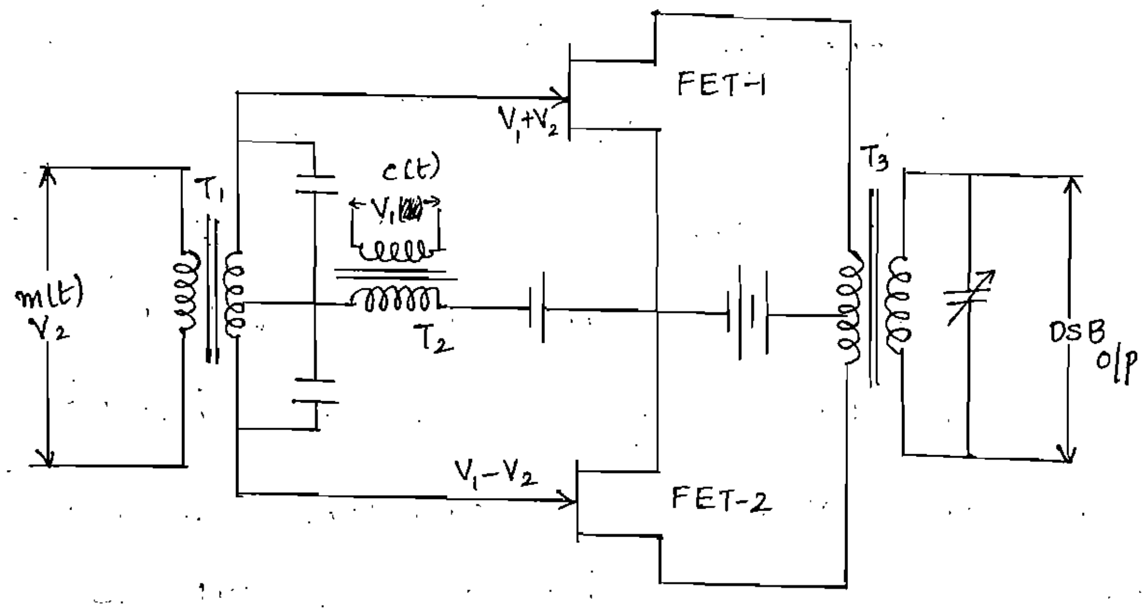



JFET balanced Modulator :-




This modulator consists of input transformers $T_1$ and $T_2$, an output transformer $T_3$ and two FETs. Here the carrier signal is applied to centre taps of input $T_1$ and output $T_3$ through transformer $T_2$. The modulating signal is applied to input transformer $T_1$. The carrier voltage is applied to two ~~taq~~ gates in phase where as modulating voltage appears 180° out of phase, at the gates, since they are at the opposite ends of center tapped transformer.

In the absence of modulating signal, the FET currents due to carrier signal are equal in amplitude but opposite in direction and cancel each each other resulting in no output of secondary of Transformer. Thus carrier is supressed.

When modulating signal is applied, the FET currents due are due to carrier as well as the modulating signal. The FET currents due to carrier are equal and opposite and cancel eachother. As modulating signal is applied 180° out of phase at the gate, the FET currents are equal but not opposite. Hence they do not cancel each other. Therefore the out put is two side bands with supressed carrier. The supression of carrier depends on symmetry of circuit components.

Mathematical Expression for balanced modulator:-

The transfer characteristics of FET ($I_D$, $V_{GS}$) can be given as $I_D = I_0 + aV_{gs} + bV_{gs}^2 + \cdots$

Here $I_0$ is maximum drain current when $V_{GS} = 0$

a and b are constants.

From this $I_{D_1} = I_0 + aV_{gs_1} + bV_{gs_1}^2$

$$I_{D_2} = I_0 + aV_{gs_2} + bV_{gs_2}^2$$

As the drain currents are in opposite direction the current through primary of coil of $T_3$ can be given as $I_P = I_{D_1} - I_{D_2}$

$I_p = a(V_{gs_1} - V_{gs_2}) + b(V_{gs_1}^2 - V_{gs_2}^2)$

Here $V_{gs_1} = V_1 + V_2$ and $V_{gs_2} = V_1 - V_2$

$I_p = a(2V_2) + b(4V_1V_2)$.

We know that $V_1 = c(t) = A_c \cos\omega_c t$

$V_2 = m(t) = A_m \cos\omega_m t$

$I_p = 2aA_m \cos\omega_m t + 4bA_cA_m \cos\omega_c t \cos\omega_m t$.

$= 2aA_m \cos\omega_m t + 2bA_cA_m[\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$

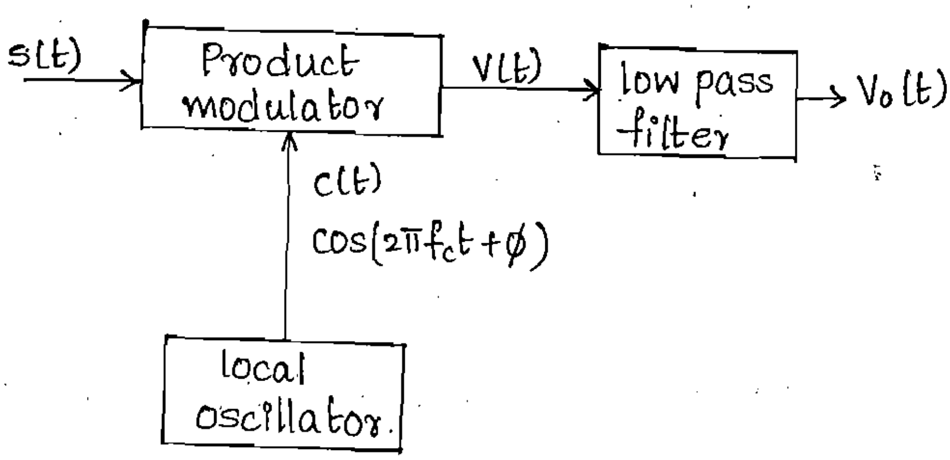
$2aA_m \cos\omega_m t$ is message signal.

$\omega_c - \omega_m$ is lower side band

$\omega_c + \omega_m$ is upper side band.

From the above expression we can say that the carrier is supressed.

Coherent detection of DSBSC modulated wave:=



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

For faithful recovery of message signal m(t) the output of local oscillator should be exactly coherent or synchronized in both frequency and phase with carrier wave c(t) which is used to generate s(t). This method of demodulation is called coherent or synchronous detection.

The output of product modulator is given as

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

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

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

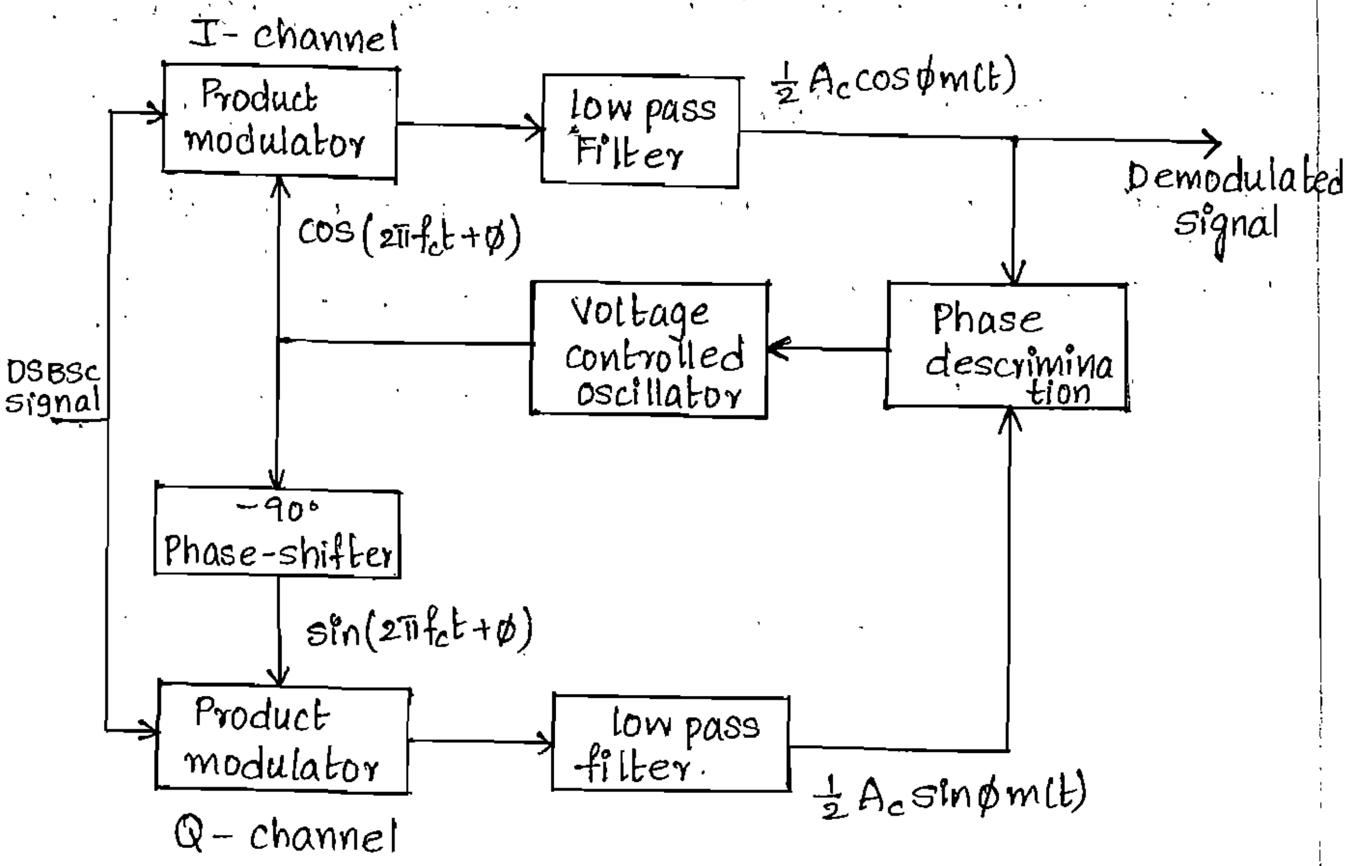
The output consists of two terms. They are scaled version of message signal and unwanted term. The unwanted term is removed by passing through low pass filter.

The output is $V_o(t) = \frac{1}{2} A_c \cos\phi\, m(t)$.

When phase error $\phi$ is constant, the demodulated signal $V_o(t)$ is proportional to $m(t)$. It is maximum when $\phi = 0$ and minimum when $\phi = \pm\pi/2$. The zero demodulated signal which occurs when $\phi = \pm\pi/2$ is called quadrature null effect.

Because of phase error $\phi$ the detector output is attenuated by a factor $\cos\phi$. If $\cos\phi$ is constant the detector output gives undistorted modulating signal. But practically $\cos\phi$ is not constant. So to maintain the local oscillator in perfect synchronism we use costas loop.

Costas loop :=



The coastas loop is nothing but the negative feedback system designed to ~~main~~ maintain the local-oscillator synchronous with carrier wave.

It consists of two coherent product modulators supplied with same input signal. The local oscillator signal supplied to product modulators are 90° out of phase. The frequency of local oscillator is adjusted to be same as carrier frequency $f_c$. The product modulator in the upper path is called "in phase coherent detector or I-channel" where as, which is in lower path is called "quadrature phase detector or Q-channel". The outputs of these two channels are given to phase descriminator which consists of a multiplier followed by a low pass filter, produces a dc control signal proportional to phase error $\phi$. This dc control signal is used to correct the phase error in local oscillator.

# SSB Modulation

## Introduction:=

When the carrier is amplitude modulated by a single sine wave, the resulting signal consists of three frequencies i.e. Original carrier and two side bands ($f_c \pm f_m$). In normal AM system both the side bands and carrier are transmitted. This is known as DSBFC. But we know that the carrier signal doesnot convey any information. so we can supress the carrier and we get DSBSC.

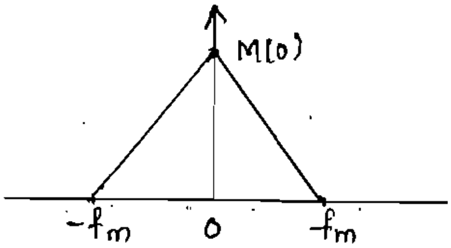
As the two side bands carry same information there is no need to transmit both the side bands to convey information. One side band may be supressed. The resulting signal is single side band known as single side band supressed carrier (SSB).

Definition := "A modulation process in which the modulated signal contains no carrier component and has only one side band is called single side band modulation or SSB modulation"

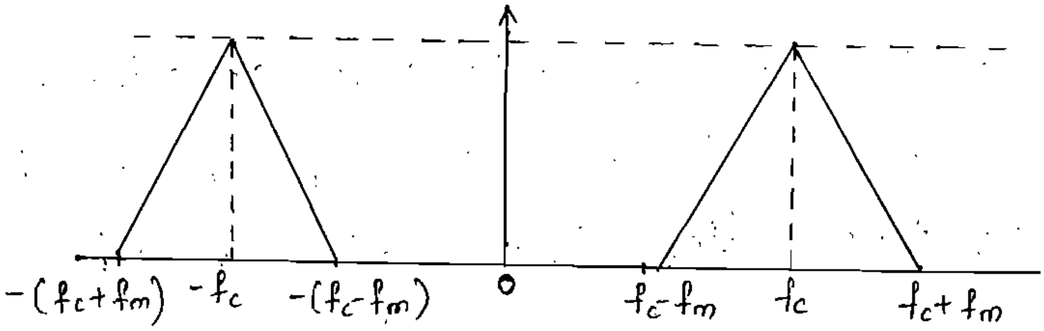
So the power which is wasted to transmit the carrier signal and a side band, can be put in to a single side band for stronger signals over longer distances.

## Frequency domain description :-

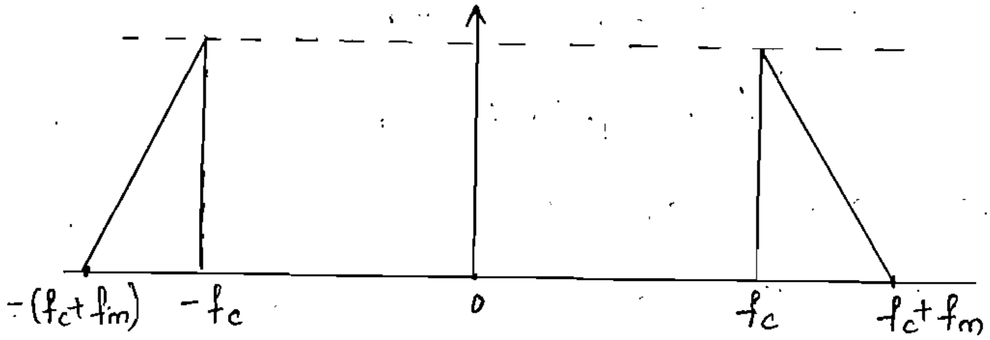
The frequency domain display of single side band modulated wave depends on the side band which is transmitted. The spectrum of DSBSC modulated wave can be obtained by multiplying m(t) and c(t).



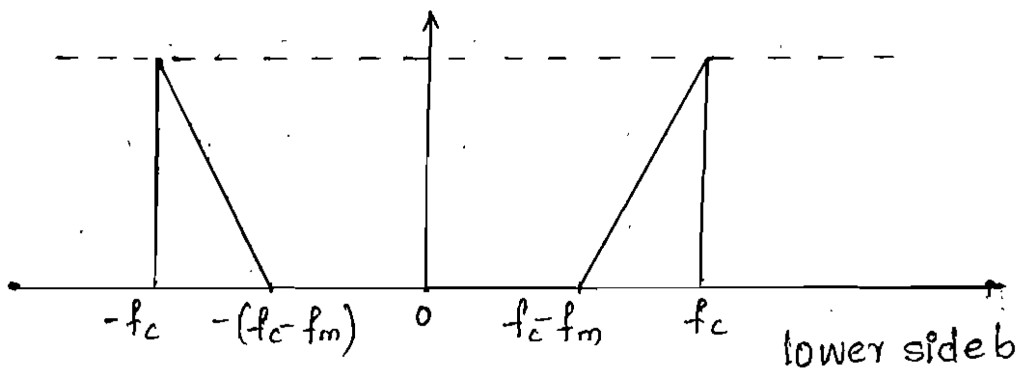

Spectrum of message signal




Spectrum of DSBSC wave




Spectrum of SSB with upper sideband




lower sideband.

Advantages of SSB :=

* The spectrum space occupied by SSB signal is $f_m$ which is only half that of AM and DSB. This reduction in frequency or band width allows more signals to transmit in same frequency range.
* Due to supression of carrier and one side band, power is saved.
* SSB signal has less band width than an AM or DSB so there will be less noise on it.
* Fading does not occur in SSB.

Dis advantages:=

* The generation and reception of SSB signal is a complex process.
* SSB is not used for transmission of high frequency signals such as music signals.

Power relations in SSB wave:=

% power saving w.r.to DSBFC.

We know that total power transmitted in AM is

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

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

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

If the carrier and one side band is supressed, then the total power transmitted in SSB-SC is

$$P_{SSB} = P_{LSB} = P_{USB}$$

$$P_{SSB} = \frac{\mu^2 A_c^2}{8R} = P_c\left(\frac{\mu^2}{4}\right).$$

Power saved due to transmitting only one side band component is given by

$$\%\text{ power saving} = \frac{P_t - P_{SSB}}{P_t} \times 100$$

$$= \frac{P_c\left(1+\frac{\mu^2}{2}\right) - \frac{\mu^2}{4} P_c}{P_c\left(1+\frac{\mu^2}{2}\right)} \times 100$$

$$= \frac{1+\frac{\mu^2}{2} - \frac{\mu^2}{4}}{1+\frac{\mu^2}{2}} \times 100$$

$$= \frac{4+\mu^2}{4+2\mu^2} \times 100.$$

if $\mu=1$ then

$$\%\text{ power saving} = \frac{5}{6} \times 100 = 83.33\%.$$

Thus 83.33% power is saved due to supression of carrier wave and one side band.

% power saving w.r.to DSB-SC system :-

We know that the total power transmitted in DS DSB-SC is

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

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

total power transmitted in SSB-SC is

$$P_{SSB} = P_{LSB} = P_{USB}$$

$$P_{SSB} = \frac{A_c^2 \mu^2}{8R} = P_c \frac{\mu^2}{4}$$

$$\% \text{ power saving} = \frac{P_{DSBSC} - P_{SSB}}{P_{DSBSC}} \times 100$$

$$= \frac{P_c \frac{\mu^2}{2} - P_c \frac{\mu^2}{4}}{P_c \frac{\mu^2}{2}} \times 100$$

$$= \frac{\mu^2}{2} \times 100$$

if $\mu = 1$ then

$$\% \text{ power saving} = \frac{1}{2} \times 100 = 50\%$$

Thus 50% of power is saved due to supression of one side band from DSB-SC signal.

Time domain description :=

The SSB signal is generated by passing a DSBSC modulated wave through a band pass filter having a transfer function $H_u(f)$.

We know that $S_{DSBSC}(t) = A_c m(t) \cos(2\pi f_c t)$.

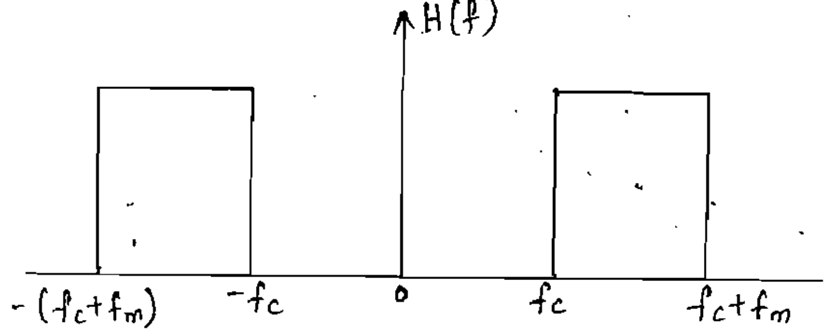
The low pass complex envelope of DSBSC modulated wave is expressed as

$$\tilde{S}_{DSBSC}(t) = A_c m(t).$$

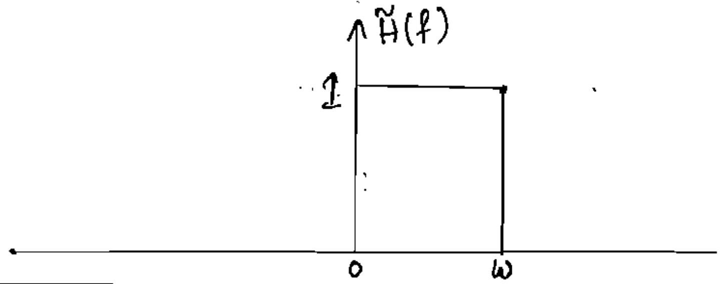
Let $s_u(t)$ be the SSB wave which has only upper-side band; and $\tilde{s}_u(t)$ be the complex envelope of $s_u(t)$.

$$s_u(t) = Re[\tilde{s}_u(t) e^{j2\pi f_c t}]$$

The frequency response of ideal band pass filter is.




Replace transfer function $H_u(f)$ by an equivalent low-pass filter transfer function $\tilde{H}_u(f)$.



Where $\tilde{H}_u(f) = \frac{1}{2}(1 + sgn(f)) \quad 0 < f < \omega$

$= 0 \quad$ otherwise.

the spectrum of complex envelope of DSBSC modulated wave $\tilde{s}_{DSBSC}(t)$ is

$$\tilde{S}_{DSBSC}(f) = A_c M(f).$$

Now $\tilde{S}_u(f) = \tilde{H}(f)\, \tilde{S}_{DSBSC}(f)$

$$= \frac{1}{2}(1 + sgn(f))\, A_c M(f).$$

$$= \frac{A_c}{2}\left[M(f) + M(f)\, sgn(f)\right].$$

from the definition of Hilbert transform

We have $\hat{m}(t) = -j\, sgn(f)\, M(f)$.

$$\therefore \tilde{S}_u(f) = \frac{A_c}{2}\left[M(f) + j\hat{m}(t)\right]$$

$$\tilde{S}_u(t) = \frac{A_c}{2}\left[m(t) + j\hat{m}(t)\right].$$

We have $s_u(t) = Re\left[\tilde{s}_u(t)\, e^{j2\pi f_c t}\right]$.

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

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

This is time domain representation of a wave containing only an upper side band.

The time domain representation of SSB modulated wave containing only lower side band is given by

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

$\frac{A_c}{2}$ = scaling factor ; $m(t)$ = inphase component,

$\hat{m}(t)$ = out of phase component.

Single tone modulation :=

If the modulating signal contains only one frequency, then it is single tone modulation.

Consider the sinusoidal modulating signal

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

By changing phase of input signal by 90° without changing its amplitude we get hilbert transform $\hat{m}(t)$.

$$\hat{m}(t) = A_m \cos(2\pi f_m t - 90°)$$

$$= A_m \sin 2\pi f_m t.$$

We know that

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

Substitute $m(t)$ and $\hat{m}(t)$ in $S_u(t)$

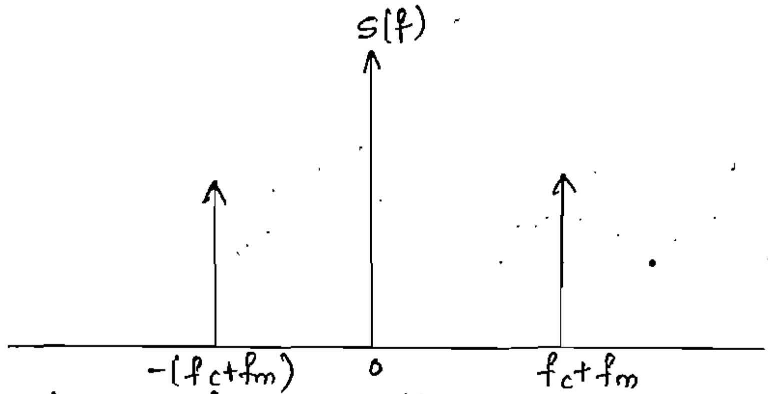
$$S_u(t) = \frac{A_c}{2}\left[A_m \cos 2\pi f_m t \cos 2\pi f_c t - A_m \sin 2\pi f_m t \sin 2\pi f_c t\right].$$

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

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

This is SSB wave obtained by transmitting only upper side band.

Now



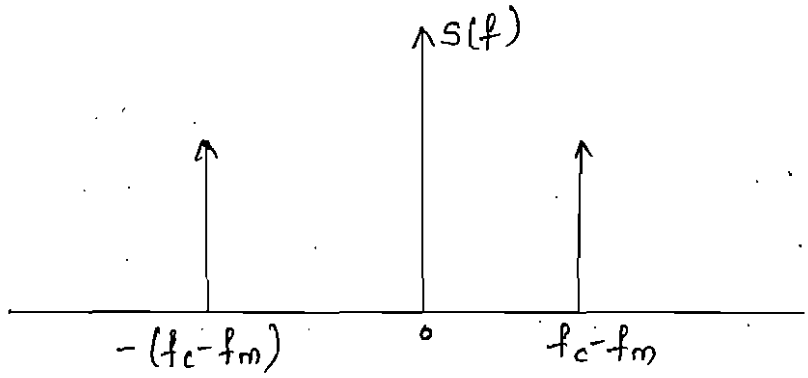

Spectrum of SSB with USB

Now for lower side band we have

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

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

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



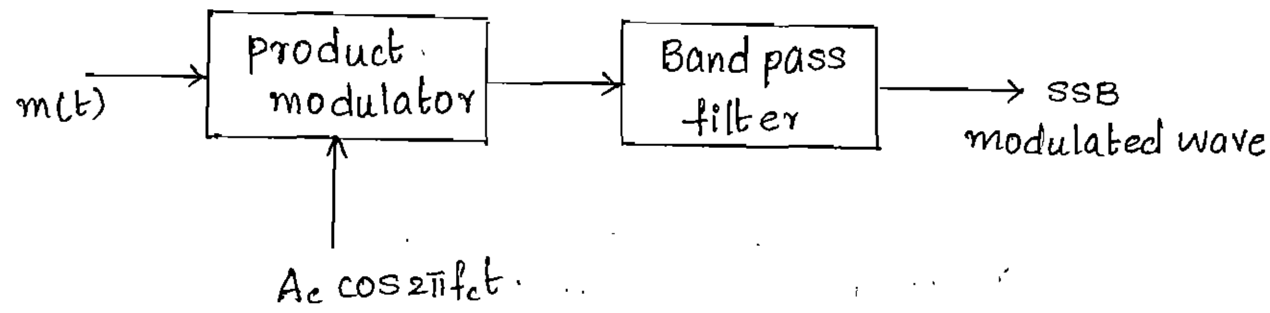

Spectrum of SSB with LSB

## Generation of SSB signals :=

There are 3 different methods for generation of SSB signal.

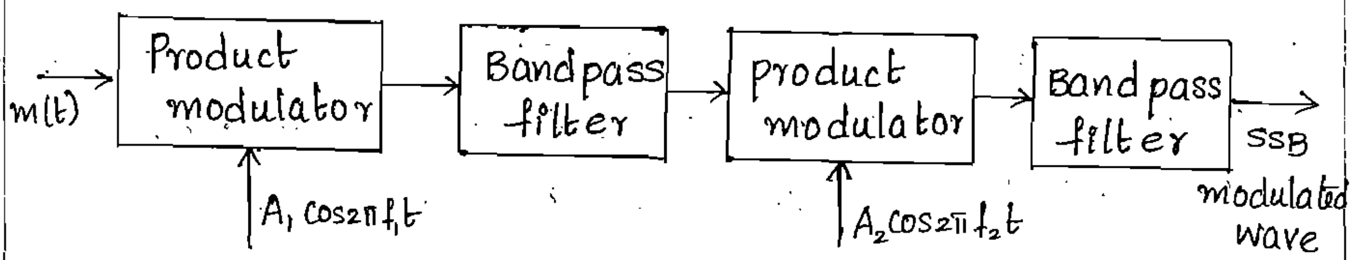
* Filter or frequency descrimination method.
* Phase shift method.
* Weavers or third method.

Filter or frequency descrimination method :=

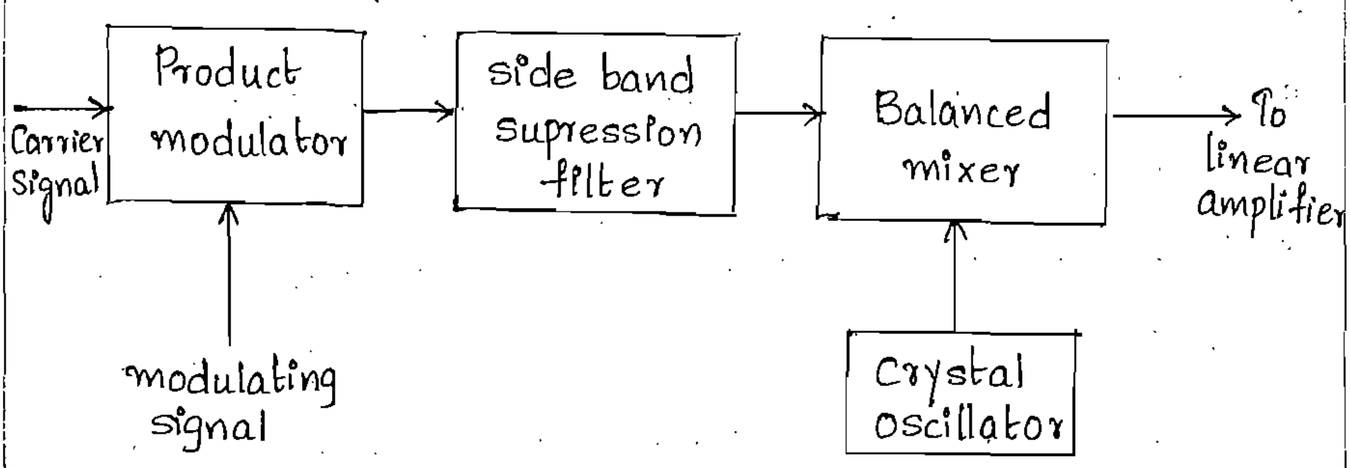



The filter is used to remove unwanted side band. The filter must have flat pass band and extremely high attenuation out side the pass band. In order to have above response the quality factor (Q) of the tuned circuits must be very high. Q-factor increases as the difference between modulating frequency $f_m$ and carrier frequency $f_c$ increases. Carrier frequency is usually same as transmitter frequency.

To generate an SSB modulated wave occupying a frequency that is higher than message signal, it is very difficult to design a filter, that pass desired side band and reject the other, using single stage modulation. In double stage modulation the SSB wave at the first filter is used as modulating wave for second product modulator. It produce a DSBSC wave with a spectrum that is symmetrically spaced about second carrier frequency $f_2$. The frequency seperation between side bands of DSBSC modulated wave is effectively twice the first carrier frequency $f_1$. ∴ It is easy to remove unwanted side band.




Block diagram of two stage SSB modulator.




Filter method of side band supression

For transmitting higher frequency, the Q value should be very high which can not be achieved practically. So initial modulation is carried out. The balanced modulator supress the carrier and filter supress the side band. The frequency of SSB is very low when compared to transmitter frequency. This frequency is boosted up to transmitter frequency by balanced mixer and crystal oscillator. This process is called "up conversion". The side band signal is then amplified by linear amplifier.

Advantages:-

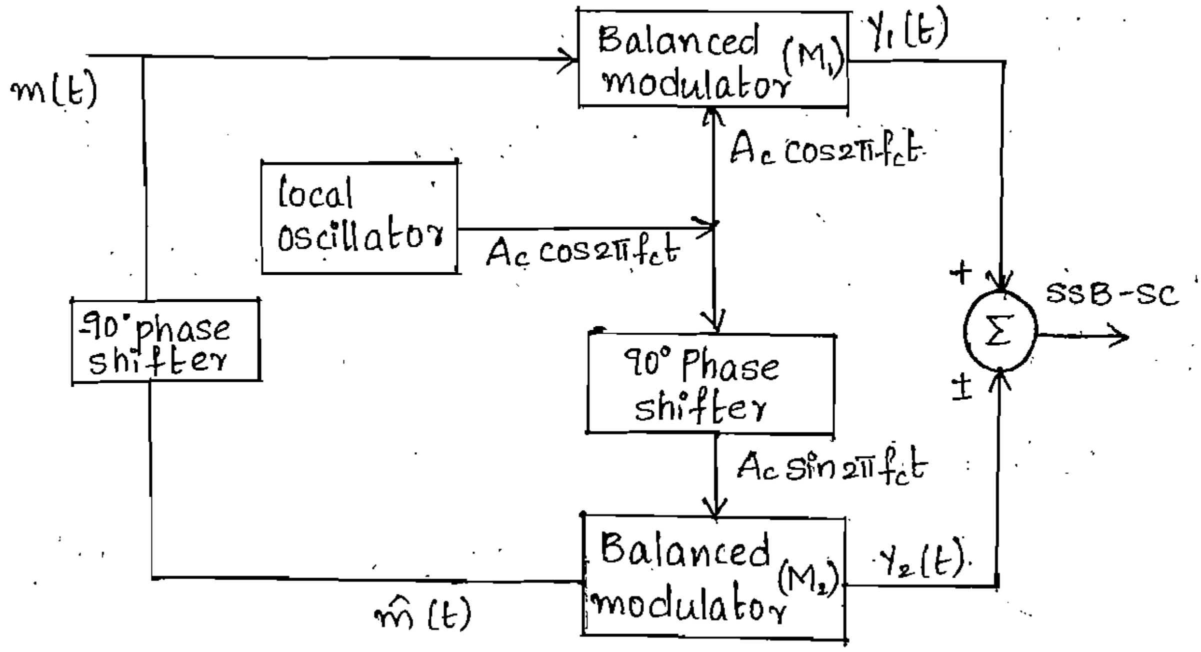
* The filter method gives side band supression up to 50 dB which is quite adequate.
* The side band filters also helps to attenuate carrier if present in output of balanced modulator.
* Band width is sufficiently flat and wide.

Dis advantages:-

* They are bulky.
* At lower audio frequencies expensive filters are required.
* As modulation takes place at lower carrier frequency repeated mixing is required in conjunction with extremely stable oscillators to generate SSB at high radio frequency.

Phase descrimination method :-

In this method we use two balanced modulators and two phase networks to obtain an SSB wave




The modulating signal m(t) and carrier signal are applied to balanced modulator ($M_1$) produce DSBSC wave that translates the spectrum of m(t) symmetrically spaced about the carrier frequency $f_c$. The modulating signal m(t) and carrier signal are phase shifted by 90° is applied to ($M_2$) which produce DSBSC wave with side bands of same amplitude but different phase spectra. Then we get $\hat{m}(t)$ which is hilbert transform of m(t).

The outputs of balanced modulator are identical in amplitude spectra, but phase spectra may be vector addition or subtraction.

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

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

The output of balanced modulator $M_1$ is $y_1(t)$

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

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

The input of balanced modulator $M_2$ is $\hat{m}(t)$.

The output is $y_2(t)$.

$$y_2(t) = A_c A_m \sin 2\pi f_c t \cdot \sin 2\pi f_m t$$

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

Expression for lower side band is

$$y_1(t) + y_2(t) = A_c A_m \cos(2\pi(f_c - f_m)t).$$

Expression for upper side band is

$$y_1(t) - y_2(t) = A_c A_m \cos 2\pi(f_c + f_m)t.$$

Advantages:-

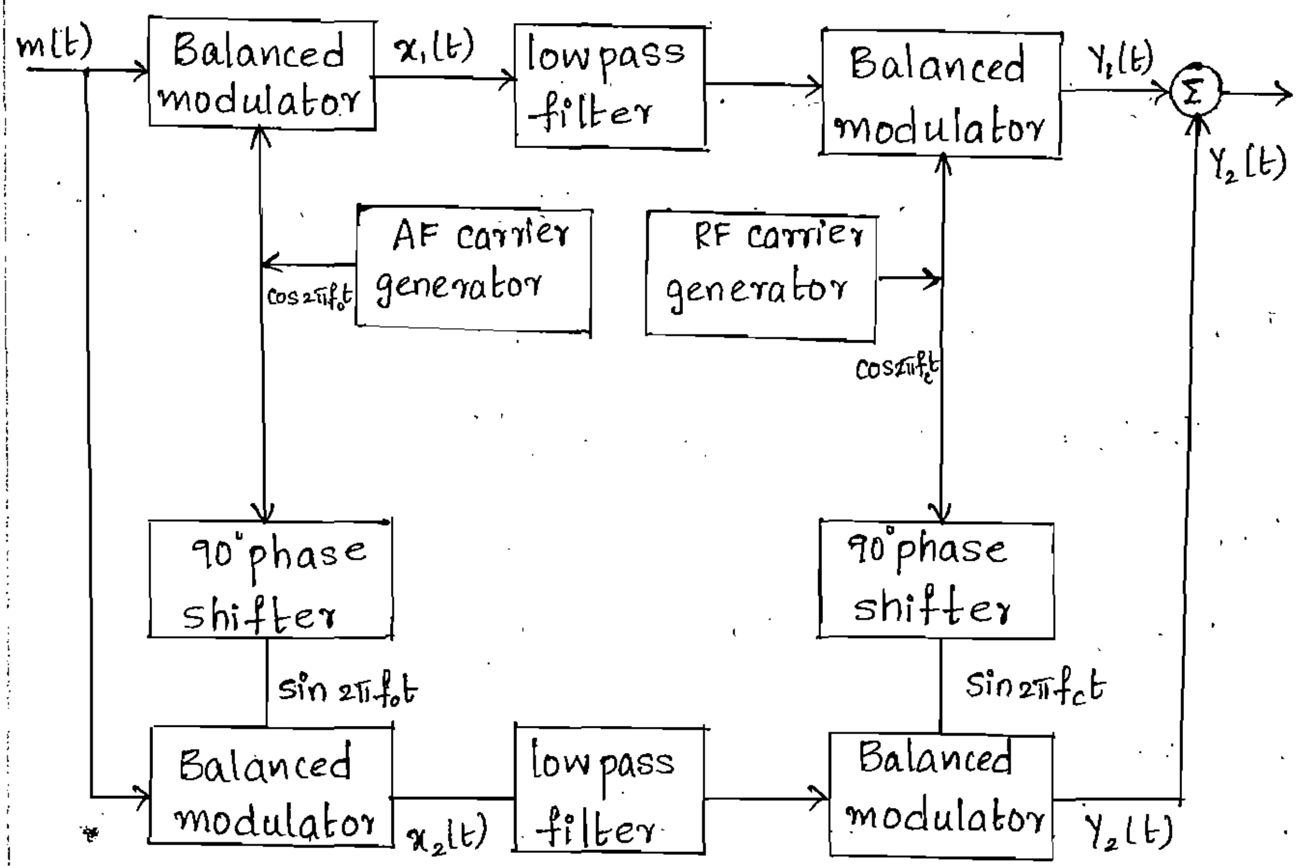
* One side band along with carrier are supressed.
* There is no need of very sharp cut off filter.
* Choice of addition or subtraction of modulator out put gives upper or lower side band.

Disadvantages:-

* The circuit is complex and expensive.
* The phase shifter should produce exact phase shift of 90°. In practise it is very difficult to design.

Weavers or third method :=

Third method of generating SSB-SC wave was invented by D.K. Weaver in 1950. This method eliminates the limitations of using sharp band pass filter. This method consists of four balanced modulators and two lowpass filters.




The input of low pass filter -1 is

$$x_1(t) = \cos(2\pi f_m t) \cdot \cos(2\pi f_0 t).$$

$$= \frac{1}{2}\left(\cos 2\pi(f_0+f_m)t + \cos 2\pi(f_0-f_m)t\right).$$

As it is lowpass filter it allows only low frequency i.e... $\frac{1}{2}\cos 2\pi(f_0-f_m)t$. This is multiplied with $\cos 2\pi f_c t$ to yield $Y_1(t)$.

$$Y_1(t) = \frac{1}{2}\cos 2\pi(f_0-f_m)t \cdot \cos 2\pi f_c t.$$

$$y_1(t) = \frac{1}{4}\left(\cos 2\pi(f_c+f_0-f_m)t + \cos 2\pi(f_c-f_0+f_m)t\right).$$

Similarly, the input of lowpass filter -2 is

$$x_2(t) = \cos 2\pi f_m t \ \sin 2\pi f_0 t$$

$$= \frac{1}{2}\left(\sin 2\pi(f_0+f_m)t + \sin 2\pi(f_0-f_m)t\right).$$

The out put is $\frac{1}{2}\sin 2\pi(f_0-f_m)t$. This is multiplied by $\sin 2\pi f_c t$ to get $y_2(t)$.

$$\therefore y_2(t) = \frac{1}{2}\sin\left(2\pi(f_0-f_m)t\right).\sin 2\pi f_c t.$$

$$= \frac{1}{4}\left(\cos 2\pi(f_c-f_0+f_m)t - \cos 2\pi(f_c+f_0-f_m)t\right).$$

Expression for lower side band.

$$y_1 - y_2 = \frac{1}{2}\cos 2\pi\left(f_0+f_c-f_m\right)t.$$

Expression for upper side band.

$$y_1 + y_2 = \frac{1}{2}\cos 2\pi(f_c-f_0+f_m)t.$$

Advantages :-

* It does not require a side band supression filter nor a wide band audio phase shift network.
* Critical parts or adjustments are not required.
* low audio frequencies can be easily transmitted.
* Side bands also be switched easily.

Disadvantages :-

* The system is more complex than other two.
* DC coupling may be needed to avoid loss of signal components close to audio carrier frequency.

Comparison between SSB generation methods.

| Sl no | Parameter | Filter method | Phase shift method | Third method. |
|---|---|---|---|---|
| 1 | Method used | Filter is used to remove the unwanted side band. | Phase shifting technique is used to remove unwanted side band. | Similar to phase shift method. But carrier is phase shifted by 90° |
| 2 | 90° phase shift | not required | Requires complex phase shift network. | Phase shift network is simple RC ckt |
| 3 | Possible frequency range of SSB | not possible to generate SSB at any frequency | possible to generate SSB at any frequency | possible to generate at any frequency |
| 4 | Need for up-conversion | required | not required | not required |
| 5 | Complexity | less | medium | high. |
| 6 | Design aspects | Q of tuned ckt, filter type, it size, weight and upper frequency limit | Design of 90° phase shifter for entire modulating frequency range | symmetry of balanced modulator. |
| 7 | Bulkiness | Yes | No | No |
| 8 | Switching ability | Not possible with existing circuit Extra filter and switching network is necessary | Easily possible | Easily possible. But extra crystal is required. |

## Demodulation of SSB waves:-

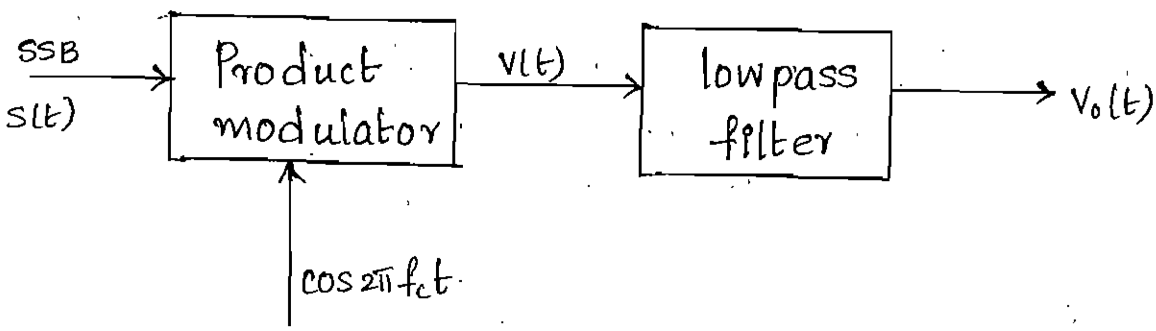
There are two types of demodulation techniques

* Coherent detection method
* Envelope detector method.

## Coherent detection method:-

We can recover the message signal m(t) by coherent detection method by following procedure

* Consider SSB wave either $s_u(t)$ or $s_L(t)$.
* Apply it to product modulator.
* Apply locally generated carrier $\cos 2\pi f_c t$ as second input to product modulator
* Apply output to lowpass filter. The output of low pass filter is recovered signal m(t).




The output of product modulator is v(t) which is product of SSB and carrier wave.

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

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

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

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

$$= \frac{A_c}{2} m(t) \left( \frac{1+\cos 4\pi f_c t}{2} \right) \pm \frac{A_c}{2} \hat{m}(t) \left( \frac{\sin 4\pi f_c t}{2} \right).$$

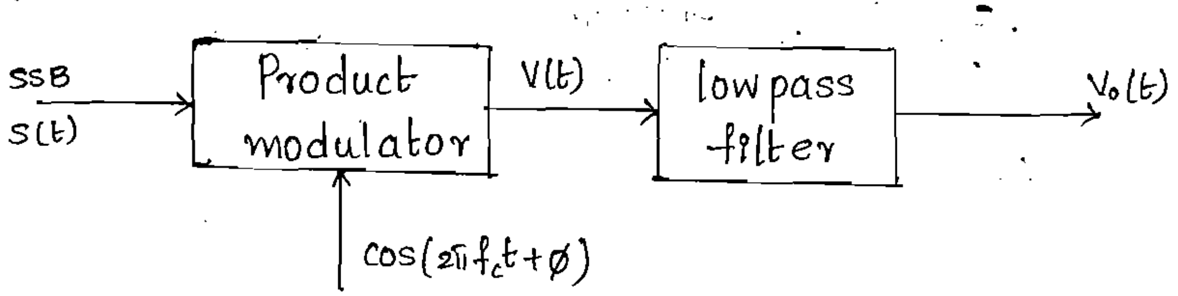
$$= \underbrace{\frac{A_c m(t)}{4}}_{\text{wanted signal}} + \underbrace{\frac{A_c}{4} \left( m(t) \cos 4\pi f_c t \pm \hat{m}(t) \sin 4\pi f_c t \right)}_{\text{unwanted signal}}.$$

If $v(t)$ is passed through low pass filter then we get

$$V_0(t) = \frac{A_c m(t)}{4}.$$

This is message signal with scaling factor $A_c/4$.

But in practise there in phase error $\phi$ in locally generated carrier wave.




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

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

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

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

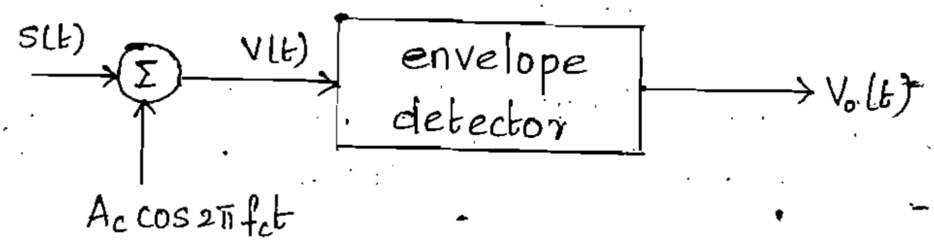
$$= \frac{A_c}{4}\left[m(t)\cos\phi \pm \hat{m}(t)\sin\phi\right] + \frac{A_c}{4}\left(m(t)\cos(4\pi f_c t+\phi) \pm \hat{m}(t)\sin(4\pi f_c t+\phi)\right)$$

$\therefore$ the desired signal

$$V_o(t) = \frac{A_c}{4}\left[m(t)\cos\phi \pm \hat{m}(t)\sin\phi\right].$$

## Envelope detector method :=

In this method the SSB signal is demodulated by adding local carrier to it at the reciever and then passing the resultant wave through an envelope detector.




Consider SSB-SC with upper side band as

$$s(t) = \left(m(t)\cos 2\pi f_c t - \hat{m}(t)\sin 2\pi f_c t\right)$$

The local carrier is given by

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

The resulting signal is given by

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

$$= \left(m(t) \cos 2\pi f_c t - \hat{m}(t) \sin 2\pi f_c t\right) + A_c \cos 2\pi f_c t$$

$$= \cos 2\pi f_c t \left(m(t) + A_c\right) - \hat{m}(t) \sin 2\pi f_c t.$$

The output of envelope detector is given by

$$V_o(t) = \sqrt{(\text{inphase component})^2 + (\text{Out of phase component})^2}$$

$$= \sqrt{(A_c + m(t))^2 + (\hat{m}(t))^2}$$

$$= \sqrt{A_c^2 + m^2(t) + 2A_c m(t) + \hat{m}^2(t)}$$

Assuming that $m^2(t) << A_c \Rightarrow \hat{m}^2(t) << A_c$

$$V_o(t) = \sqrt{A_c^2 + 2A_c m(t)}$$

$$= \sqrt{A_c^2\left(1 + \frac{2m(t)}{A_c}\right)}$$

$$= A_c \sqrt{1 + \frac{2m(t)}{A_c}}$$

Since $|m(t)| << A_c$

$$V_o(t) = A_c\left(1 + \frac{2m(t)}{2A_c}\right) \qquad \left[\because (1+x)^{1/2} = 1 + \tfrac{1}{2}x\right]$$

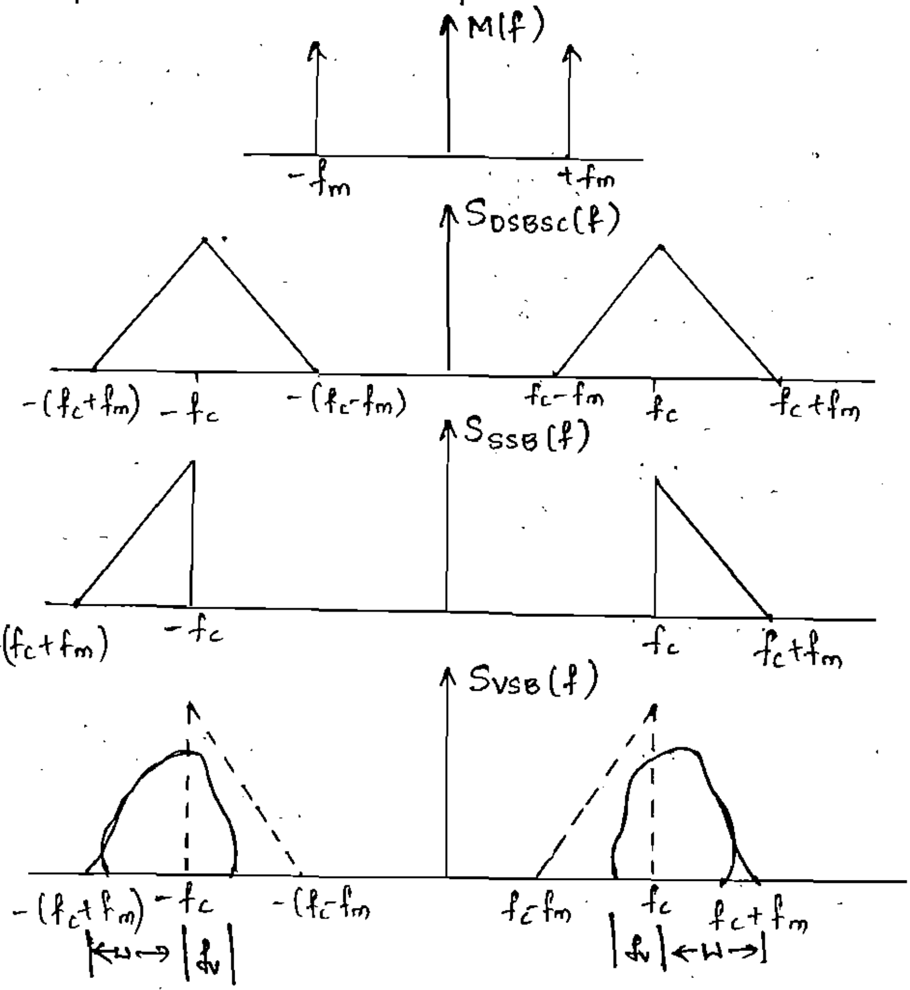
$$V_o(t) = A_c + m(t).$$

In this method local carrier frequency should match with original carrier. If not distortion occurs.

Vestigial side band modulation:=

When the message signal contains extremely low frequencies SSB modulation is not appropriate. Because the upper and lower side bands meet at carrier frequency and it becomes difficult to isolate one ~~the~~ side band. To over come this (VSB) Vestigidal side band modulation is used. In this modulation one side band is passed completely along with a trace or vestige of other side band. This technique is mostly used in television transmission.

Frequency domain description:=

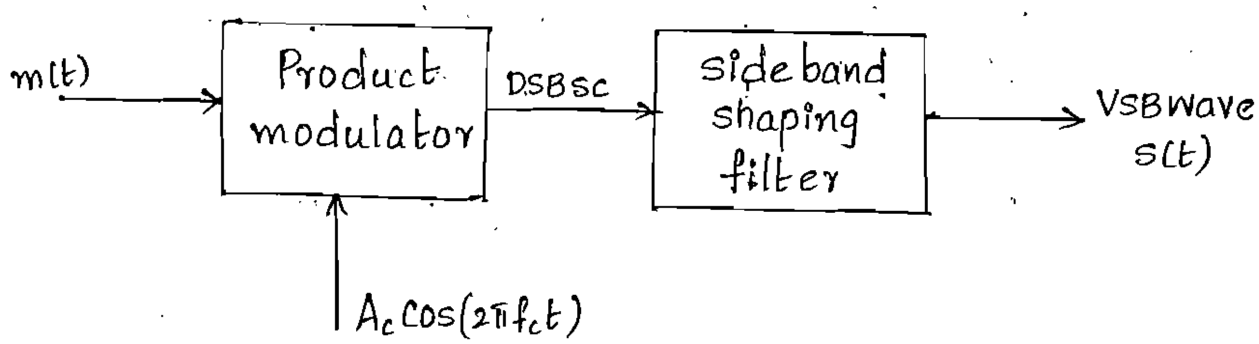


Here the lower side band is modified into vestigial side band. The transmitted vestige of LSB compensate for the removed part of USB. The transmission band width required by VSB modulated wave is given by band width $= W + f_v$.

Where $W$ is band width of message signal and $f_v$ is band width of VSB. The VSB requires band width almost equal to SSB transmission. It retains the excellent low frequency base band characteristics of double side band modulation.

Generation of VSB modulated wave:-

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




The relation between the transfer function $H(f)$ of the filter and spectrum $S(f)$ of VSB modulated wave $s(t)$ defined by

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

Demodulation of VSB modulated wave:=

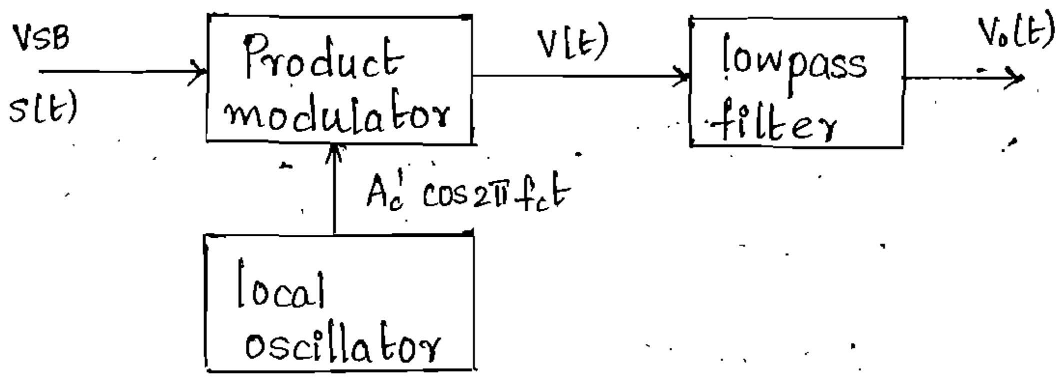
There are two methods of demodulation

* Coherent detection
* Envelope detection.

Coherent detection:=

In this method the VSB wave s(t) is passed through a coherent detector and determining the necessary condition for detector output to provide undistorted message signal m(t). In this s(t) is multiplied with locally generated wave $A_c' \cos 2\pi f_c t$. which is synchronous with carrier wave in both frequency and phase.

$$\therefore \; V(t) = s(t) \cdot A_c' \cos 2\pi f_c t$$




$$V(t) = s(t)\, A_c' \cos 2\pi f_c t$$

converting it in to frequency domain

$$V(f) = \frac{A_c'}{2}\left( S(f-f_c) + S(f+f_c) \right)$$

We have $s(f) = \frac{A_c}{2}\left(M(f-f_c) + M(f+f_c)\right) H(f)$

Substitute $s(f)$ in $v(f)$.

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

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

By passing it through a low pass filter

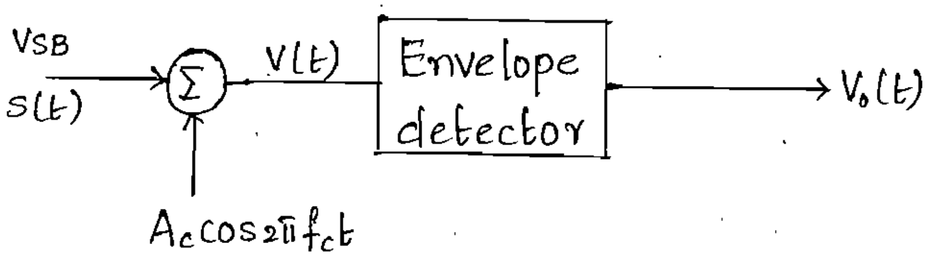
$$V_0(f) = \frac{A_c A_c'}{2} M(f)\Big[H(f-f_c) + H(f+f_c)\Big]$$

Condition for getting original message signal with a scaling factor and with out any distortion is given as $H(f-f_c) + H(f+f_c) = 2H(f)$.

So $V_0(f) = \frac{A_c A_c'}{2} M(f) \cdot 2H(f)$.

$V_0(f) = A_c A_c' M(f) H(f)$.

Envelope detection method :=



To make demodulation of VSB wave an envelope detector is used at the receiving end and it is necessary to transmit a carrier along with the modulated wave. The scaled expression of VSB wave by factor $k_a$ with carrier component $A_c \cos 2\pi f_c t$ can be given by

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

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

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

$$V_0(t) = \sqrt{(\text{inphase component})^2 + (\text{out of phase component})^2}$$

$$= \sqrt{\left(A_c \left(1 + \frac{k_a}{2} m(t)\right)\right)^2 + \left(\frac{A_c}{2} k_a m_Q(t)\right)^2}$$

$$= \sqrt{A_c^2 \left(1 + \frac{k_a}{2} m(t)\right)^2 + A_c^2 \left(\frac{k_a m_Q(t)}{2}\right)^2}$$

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

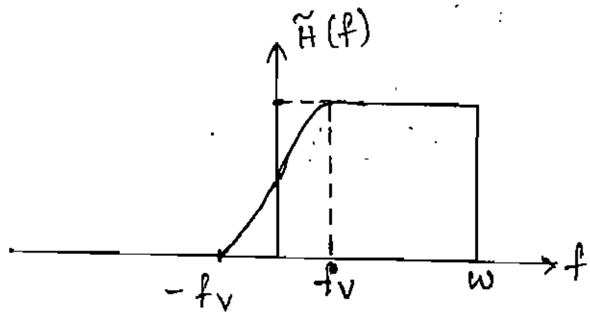
The detector output is distorted by quadrature component $m_Q(t)$. To reduce the distortion we can decrease the modulation index $k_a$ or we can decrease the quadrature phase component $m_Q(t)$.

## Time domain description :=

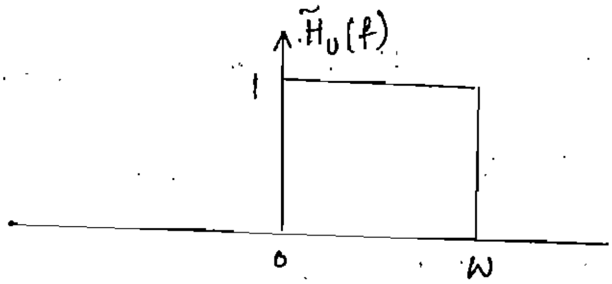
let s(t) be VSB modulated wave containing a vestige of lower side band, and it is an output of side band shaping filter in response to DSBSC modulated wave

If H(f) is transfer function of side band shaping filter then $\tilde{H}(f)$ can be given as

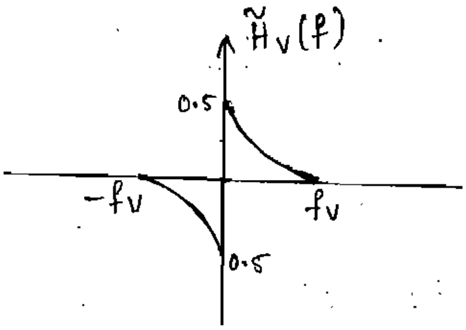
$$\tilde{H}(f) = \tilde{H}_U(f) - \tilde{H}_V(f).$$



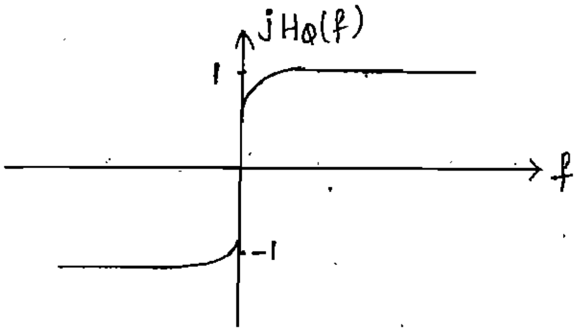

Response of side band shaping filter




first component. $\tilde{H}_U(f)$




second component $\tilde{H}_V(f)$




Response of $jH_Q(f)$

Where

(i) $\tilde{H}_v(f)$ is for complex low pass filter which is equal to the band pass filter designed to reject lower side band

(ii) $\tilde{H}_v(f)$ is for both generation of vestige of lower side band and for removal of some portion in upper side band.

We know that $\tilde{H}_v(f) = \frac{1}{2}(1+\text{sgn}(f)) \quad 0<f<\omega$

$= 0 \quad$ elsewhere.

$\therefore \tilde{H}(f) = \frac{1}{2}(1+\text{sgn}(f)) - \tilde{H}_v(f) \quad f_v < f < \omega$

$= 0 \quad$ else where.

As signum function and $\tilde{H}_v(f)$ are odd functions, they contain imaginary values inverse fourier transforms There exist a new expression

$$H_Q(f) = \frac{1}{j}\left(\text{sgn}(f) - 2\tilde{H}_v(f)\right).$$

$\tilde{H}(f)$ can be written as

$\tilde{H}(f) = \frac{1}{2}(1 + jH_Q(f)) \quad f_v < f < \omega$

$= 0 \quad$ else where.

The VSB modulated wave $s(t)$ is given as

$$s(t) = \text{Re}\left(\tilde{s}(t)\, e^{j2\pi f_c t}\right).$$

$$\tilde{S}(f) = \tilde{H}(f)\, \tilde{S}_{DSBSC}(f)$$

$$\tilde{S}_{DSBSC}(f) = A_c M(f)$$

$$\tilde{S}(f) = \frac{1}{2} A_c \left(1 + jH_Q(f)\right) M(f).$$

Converting it in to time domain

$$\tilde{s}(t) = \frac{1}{2} A_c \left(m(t) + j m_Q(t)\right)$$

$$s(t) = Re\left(\tilde{s}(t) e^{j2\pi f_c t}\right).$$

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

$$= Re\left(\frac{A_c}{2} m(t) \cos 2\pi f_c t + j \frac{A_c}{2} m(t) \sin 2\pi f_c t + j \frac{A_c}{2} m_Q(t) \cos 2\pi f_c t - \frac{A_c}{2} m_Q(t) \sin 2\pi f_c t\right).$$

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

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

This is expression for USB with a vestige of LSB. Similarly the expression for LSB with a vestige of USB is given as

$$s(t) = \frac{A_c}{2}\left(m(t) \cos 2\pi f_c t + m_Q(t) \sin 2\pi f_c t\right).$$

## Advantages of VSB :=

* Low frequencies, near $f_c$ are transmitted with out any attenuation.
* Band width is reduced compared to DSB.
* VSB signals are relatively easy to generate.
* VSB inherts the advantages of DSB and SSB but avoid their disadvantages at a small cost.

Comparison of amplitude modulation techniques:=

| Sl no | Parameter | Standard AM | SSB | DSBSC | VSB |
|---|---|---|---|---|---|
| 1 | Power | High | less | medium | less than DSBSC but greater than SSB |
| 2 | Bandwidth | $2f_m$ | $f_m$ | $2f_m$ | $f_m < BW < 2f_m$ |
| 3 | Carrier Supression | NO | Yes | yes | NO |
| 4 | Receiver Complexity | simple | complex | complex | simple. |
| 5 | Modulation type | Non linear | linear | linear | linear |
| 6 | Sideband supression | NO | One sided Completely | NO | One sideband supressed partly |
| 7 | Transmission efficiency | minimum | maximum | moderate | moderate |
| 8 | Application | Radio communication | Point to point communication Preffered for long distances | Point to point communication | Television broad casting |

# 2- DSB & SSB Modulation :-

Introduction to DSB-SC Modulation :
The standard o/p expression for an AM wave is

$S(t) = A\cos\omega_c t + x(t)\cos\omega_c t$

(or)

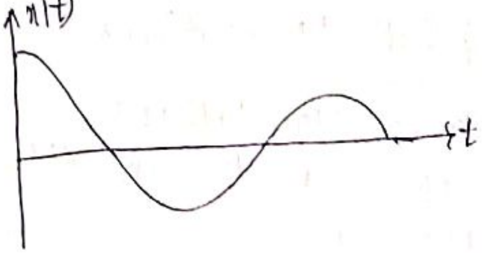
$S(t) = \underbrace{A\cos\omega_c t}_{\text{Carrier freq}} + \frac{Am_a}{2}\cos\underbrace{(\omega_c+\omega_m)t}_{\text{USB freq}} + \frac{Am_a}{2}\cos\underbrace{(\omega_c-\omega_m)t}_{\text{LSB frequency}}$ } DSB-FC.

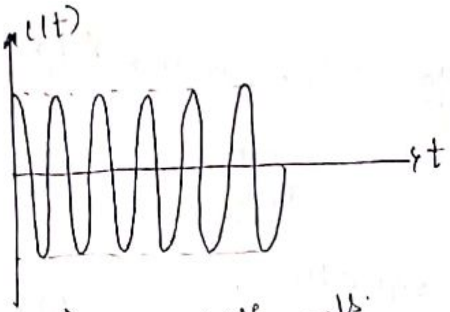
From the above equation, we can observe that the carrier frequency component in AM wave doesnot carry any information. From the power calculations of AM wave, it has been observed that 67% of total power is required for transmitting the carrier signal which doesnot contain any information. Hence, if this carrier is supressed, only the side bands remains and in this way a saving of $\frac{2}{3}$rd power may be achieved at 100% modulation This type of carrier supression doesnot affect the modulating signal in any way

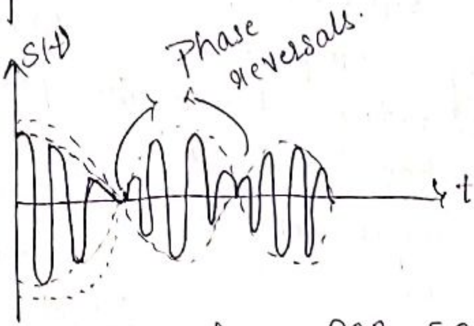
The resultant signal obtained by supressing the carrier from the modulated wave is called DSB-SC system (or) DSB-SC wave

DSB-SC :-
DSB-SC is a method of transmission where only the two sidebands are transmitted by supressing the carrier signal. (or)
The conventional AM wave in which the carrier is suppressed is called as DSB-SC modulation.

Time domain representation of Multitone DSB-SC wave :-
Let us consider a modulating signal $x(t)$ modulates a high frequency carrier signal. $c(t) = A\cos\omega_c t$
Let $x(t)$ be band limited to the interval $(-\omega_m, \omega_m)$
This means it doesnot have any frequency component outside the range $(-\omega_m, \omega_m)$




The standard equation for DSB-SC modulated wave is $s(t) = x(t) \cdot c(t) = x(t) A \cos\omega_c t$

Frequency domain description of DSB-SC wave :-

From the frequency shifting theorem of Fourier transform

wKT $\quad x(t) \longleftrightarrow X(\omega)$

$$x(t)\, e^{j\omega_c t} \longleftrightarrow X(\omega - \omega_c)$$

$$x(t)\, e^{-j\omega_c t} \longleftrightarrow X(\omega + \omega_c)$$

Now, $\quad s(t) = x(t) A\cos\omega_c t$

Apply FT on above eq.

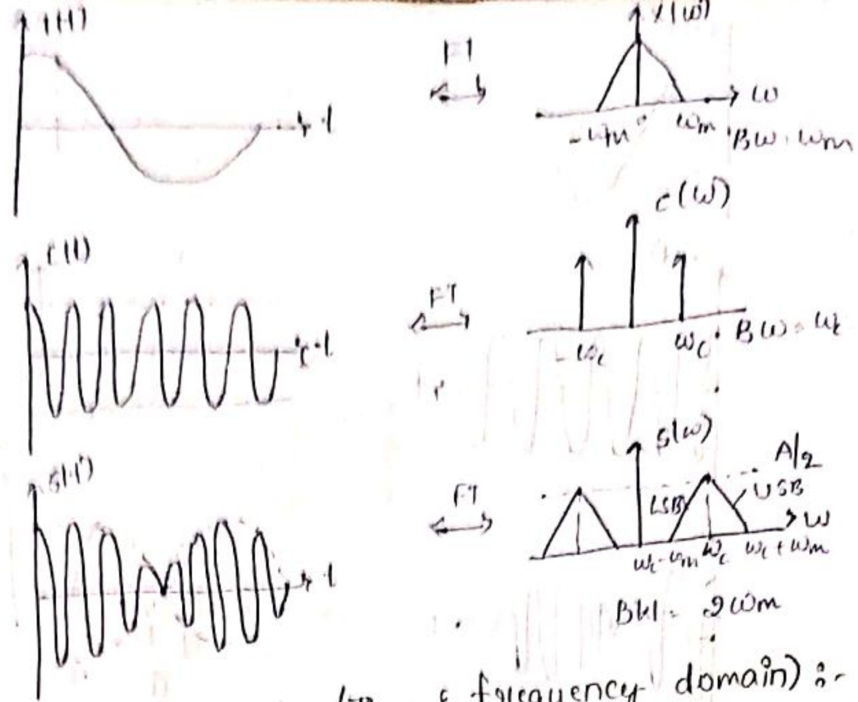
$$S(\omega) = FT\left[x(t) A \cos\omega_c t\right]$$

$$= A\, FT\left[x(t)\left(\frac{e^{j\omega_c t} + e^{-j\omega_c t}}{2}\right)\right]$$

$$= \frac{A}{2} FT\left[x(t) e^{j\omega_c t}\right] + FT\left[x(t) e^{-j\omega_c t}\right]$$

$$S(\omega) = \frac{A}{2}\left[X(\omega - \omega_c) + X(\omega + \omega_c)\right]$$

→ Band width of modulating signal is $\omega_m$




**Single tone DSB-SC Modulation (Time & frequency domain) :-**

If the modulating signal consists of fixed frequency/single, then it is called as single tone modulation.

Let us consider a modulating signal $x(t) = A_m \cos \omega_m t$ modulates a high frequency carrier signal $c(t) = A_c \cos \omega_c t$. WKT According to the standard eq'n of DSB-SC

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

$$= A_m \cos \omega_m t \cdot A_c \cos \omega_c t$$

$$= A_m A_c \cos \omega_m t \cos \omega_c t$$

$$= A_m A_c \frac{1}{2}\left[\cos(\omega_m + \omega_c)t + \cos(\omega_m - \omega_c)t\right]$$

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

WKT

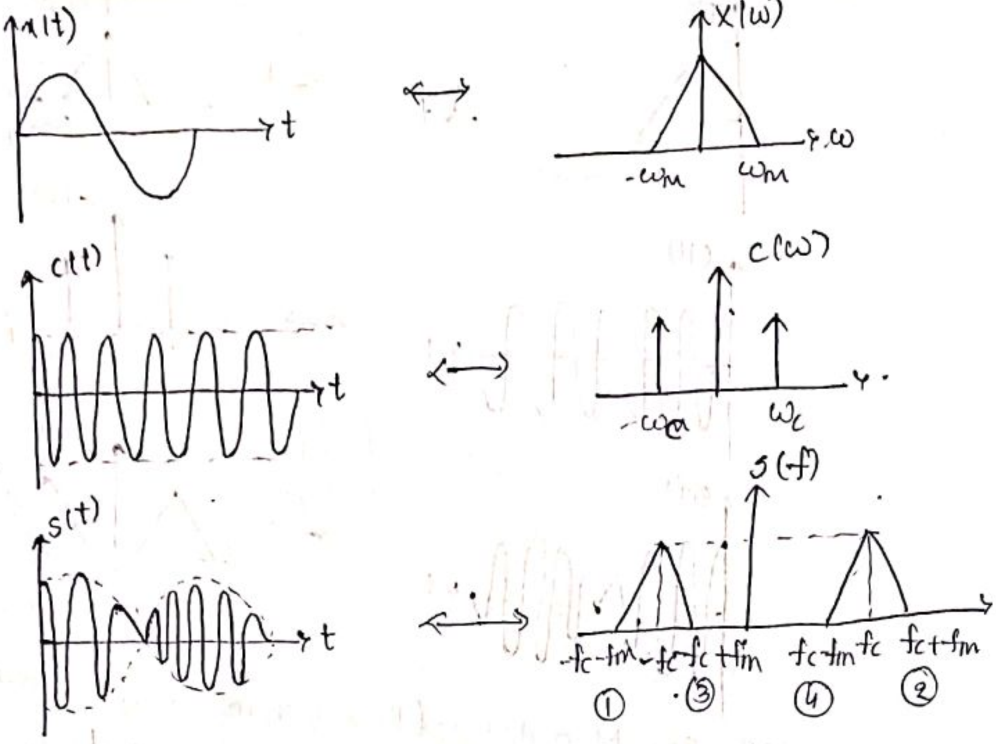
$$FT[\cos 2\pi f_0 t] = \frac{1}{2}\left[\delta(f - f_0) + \delta(f + f_0)\right]$$

Now Apply the FT on eq (1) in order to get frequency description of single tone DSB-SC.

$$S(f) = \frac{A_m A_c}{2 \cdot 2}\left[\delta[f - (f_m + f_c)] + \delta(f + f_m + f_c) + \delta(f - (f_c - f_m)) + \delta(f + f_c - f_m)\right]$$

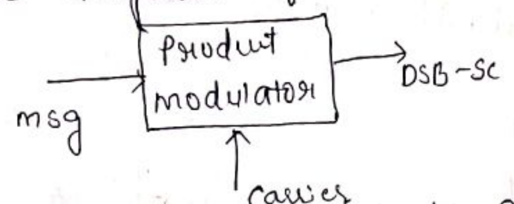
$$= \frac{A_m A_c}{4}\left[\underset{(1)}{\delta(f - f_c - f_m)} + \underset{(2)}{\delta(f + f_m + f_c)} + \underset{(3)}{\delta(f - f_c + f_m)} + \underset{(4)}{\delta(f + f_c - f_m)}\right] \quad \text{—(2)}$$

The above eq. reveals four frequency components and they are upper and lower sidebands ~~of~~ on either side of $\pm f_c$




BW = 2fm

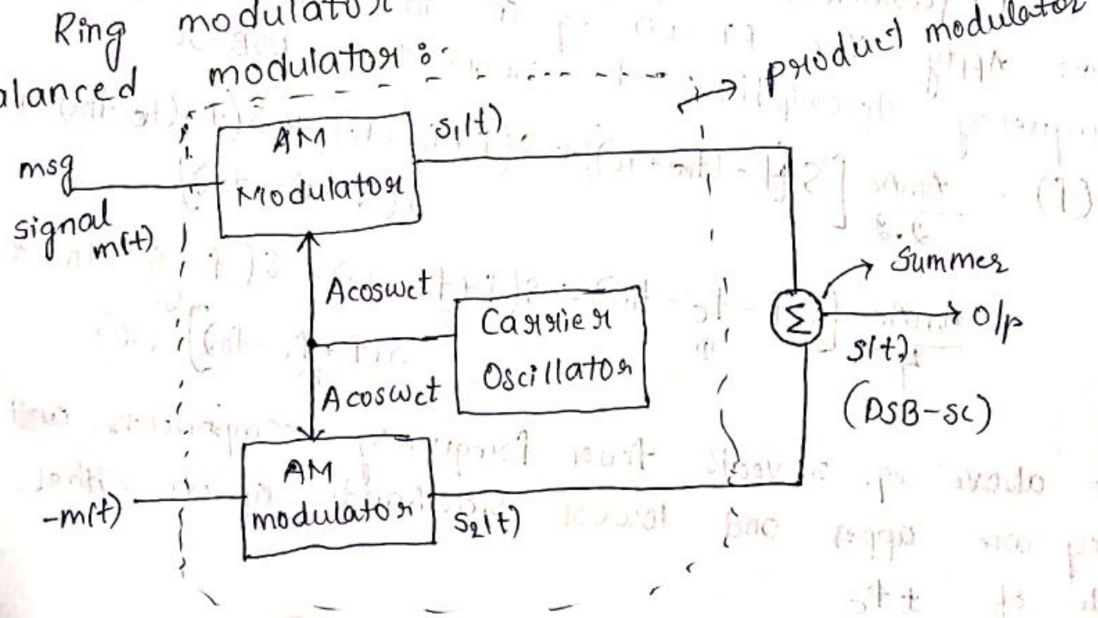
Generation of DSB-SC :
A DSB-SC wave can be generated by simply multiplying message signal and carrier signal.
Symbolic diagram of DSB-SC generation




The ckt which is used to generate a DSB-SC wave by multiplying message and carrier signals is called as product modulator.
Product modulators are of 2 types. They are

1. Balanced modulator
2. Ring modulator

Balanced modulator :



$s_1(t) = A\cos w_c t \,[1 + m_a m(t)]$

$s_2(t) = A\cos w_c t \,(1 - m_a m(t))$

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

$= \cancel{A\cos w_c t} + A m_a m(t)\cos w_c t - \cancel{A\cos w_c t} + A m_a m(t)\cos w_c t$

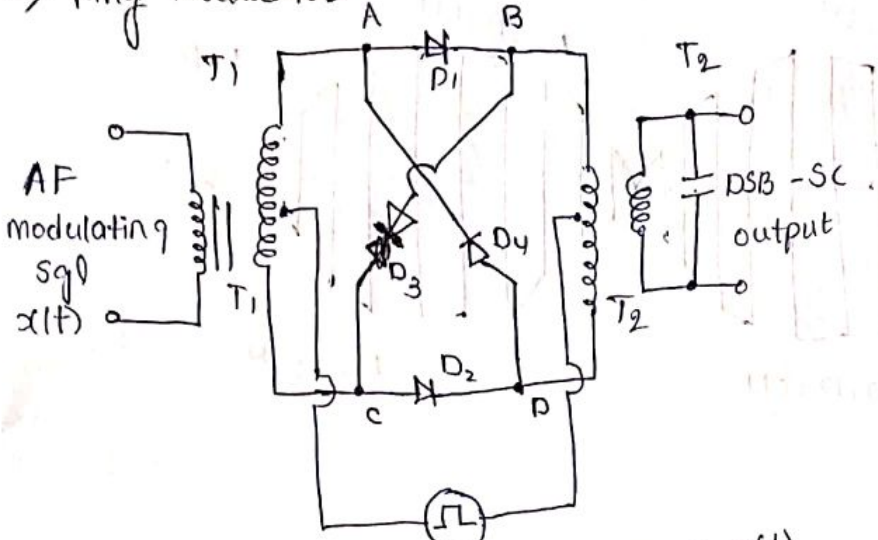
$s(t) = 2A m_a m(t)\cos w_c t$

(or)

$s(t) = 2m_a \underbrace{c(t)\, m(t)}_{\text{DSB-SC wave}}$

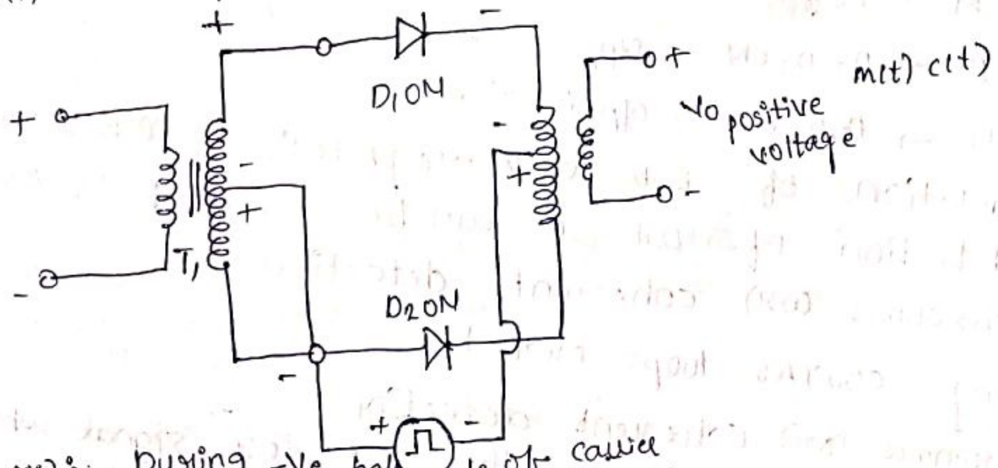
Mid. exp.

→ Ring Modulator

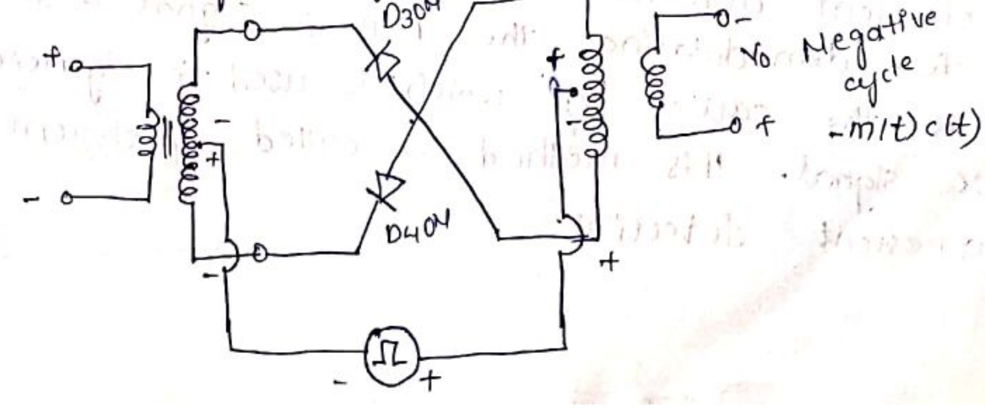


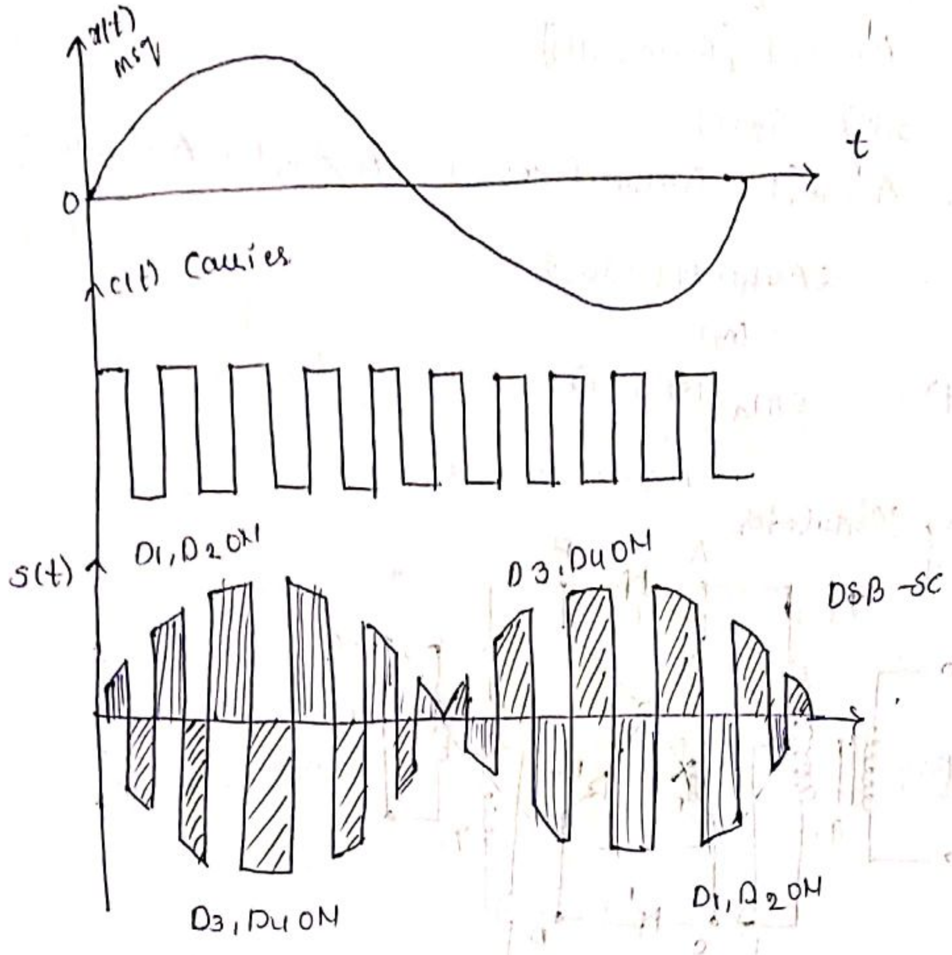

A diode ring modulator

Case (i): During +ve half cycle of carrier




Case (ii): During -ve half cycle of carrier






**

| M | C | | | |
|---|---|---|---|---|
| P | P | → $D_1, D_2$ ON | - o/p voltage is +Ve | → $x(t)\,c(t)$ |
| P | N | → $D_3, D_4$ ON | - o/p voltage is -Ve | → $-x(t)\,c(t)$ |
| N | P | → $D_3, D_4$ ON | - o/p is +Ve | |
| N | N | → $D_1, D_2$ ON | - o/p is -Ve | |

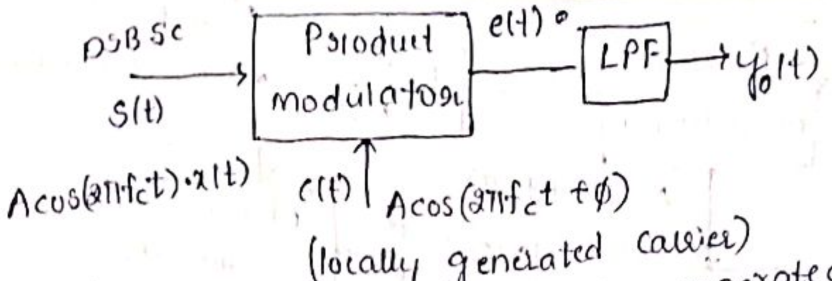
Demodulation of DSB-SC wave / Detection of DSB-SC wave :-

The detection of DSB-SC can be achieved in 2 ways.

1. synchronous (or) coherent detection
2. using COSTAS loop method.

Synchronous (or) Coherent detection :-

In coherent detection, the carrier signal which is used for demodulating the DSB-SC signal is exactly same as the carrier sgl which is used for generating DSB-SC signal. This method is called synchronous (or) coherent detection.




The DSB-SC signal s(t) and a locally generated carrier sgl are applied as an inputs to a product modulator. The o/p of product modulator is $e(t) = s(t)\cdot c(t)$

$$e(t) = A\cos(2\pi f_c t)\,x(t)\cdot A\cos(2\pi f_c t + \phi)$$

where $\phi$ is phase difference b/w the carrier used at the transmitter and the receiver.

$$e(t) = A^2 x(t)\cos(2\pi f_c t)\cos(2\pi f_c t + \phi)$$

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

$$e(t) = \underset{①}{\frac{A^2}{2}x(t)\cos\phi} + \underset{②}{\frac{A^2}{2}x(t)\cos(4\pi f_c t + \phi)}$$

Now the signal ① is allowed to pass to a LPF so 2nd term is attenuated by LPF.

$$y_0(t) = \frac{A^2}{2}x(t)\cos\phi$$

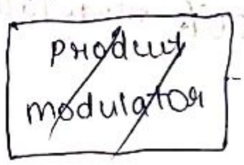
here $y_0(t)$ → demodulated signal is maximum when the phase difference $\phi = 0$

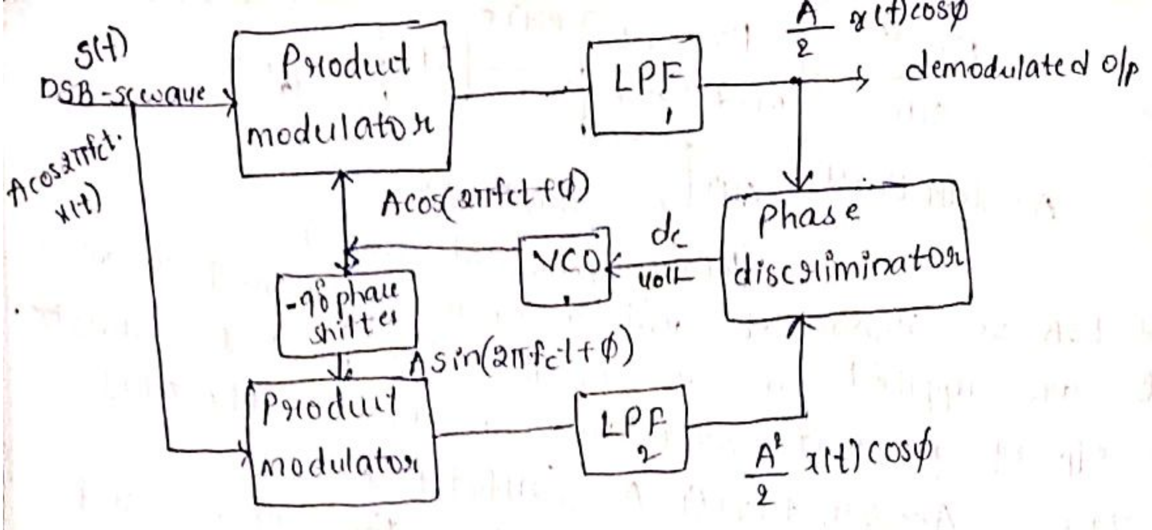
$|||^{ly}$, when $\phi = \pm \pi/2$, the demodulated o/p is zero. This effect is called Quadrature Null effect.

COSTAS Loop Method:-

The COSTAS loop method is used for demodulation of DSB-SC wave and for the correction of face difference b/w the carrier used at transmitter section and for the locally generated signal used at the receiver section.

The COSTAS Loop Method uses two product modulators with the common i/p s(t) i.e., DSB-SC wave as shown in fig. below.






VCO - Carrier generator
VCO - voltage controlled oscillator
As shown in fig, The second i/p of product modulator is taken from VCO with -90° phase shift to one of the product modulator. The o/p's of product modulators are then applied to lowpass filter ckt to allow low frequency signals. Therefore, the o/p of LPF 1 is $\frac{A^2}{2} x(t)\cos\phi$ and the o/p of LPF 2 is $\frac{A^2 x(t)}{2}\sin\phi$. Now, The o/p's of lowpass filters are applied as an inputs to phase discriminator which is used to findout the phase difference b/w these 2 signals. Depending on the phase difference b/w these 2 signals it will generate a DC voltage which is used to control the oscillations of voltage controlled oscillator by this way COSTAS loop corrects the phase difference b/w the carrier and locally generated sgl which results in error free o/p.

Power saving in DSB-SC:-

WKT, the total power transmitted by an antenna is given by $P_t = P_c\left[1+\frac{m_a^2}{2}\right]$.

For 100% modulation,
i.e., for $m_a = 1$, $P_t = P_c\left(\frac{3}{2}\right)$, $P_t = 1.5P_c$

The percentage of power saving in DSB-SC is given by

$\frac{P_c}{1.5P_c} \times 100 = 66.66\%$

i.e., 67% of power is saved for 100% modulation.

For 50% modulation, $m_a = \frac{1}{2}$.

$P_t = P_c\left[1+\frac{1}{8}\right] = 1.125P_c$

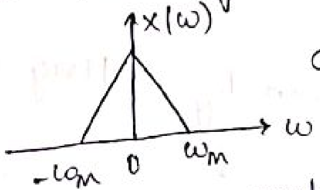
$\frac{P_c}{1.125 P_c} \times 100 = 88.88\%$

## SSB - SC Modulation :-

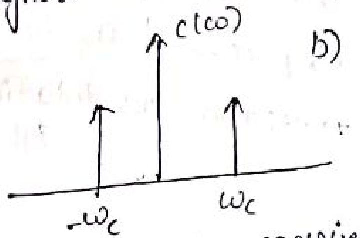
If we consider the fact that, in DSB-SC, the two side bands carries same information i.e., the Basic information is transmitted twice in DSB-SC. So, if any one of the two side bands is suppressed in DSB-SC, the resultant signal consists of single side band with the suppressed carrier is called as SSB-SC wave and the modulation is called as SSB-SC modulation.

Frequency (or) spectrum representation of SSB-SC wave

Let us consider a modulating signal $x(t)$ having the frequency range $-\omega_m$ to $\omega_m$ modulates a high frequency carrier signal $\cos\omega_c t$. Then the resulting spectrum of DSB-SC and SSB-SC are shown in the fig. below.

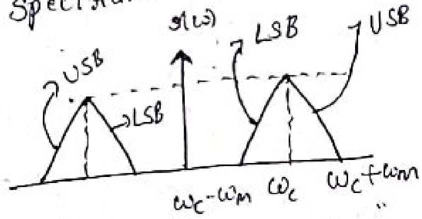



spectrum of modulating signal



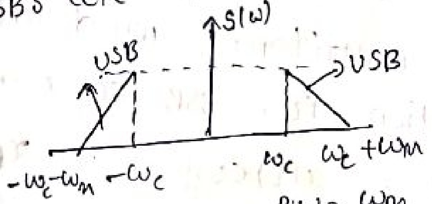

Spectrum of carrier sgl

Spectrum of DSB-SC



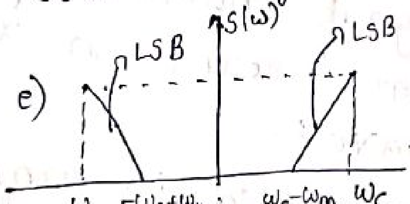

$BW = 2\omega_m$

Spectrum of SSB-SC only when USB's are transmitted.




$BW = \omega_m$

Spectrum of SSB-SC only when LSB's are Txed.




$BW = \omega_m$

Time domain description of SSB-SC wave :-

Time domain description of single tone modulating signal

Let us consider a single tone modulating signal $x(t) = \cos\omega_m t$ modulates a high frequency carrier sgl $c(t) = \cos\omega_c t$. The time domain discription of

SSB-SC signal can be obtained with the help of fig(d), fig(e) in frequency spectrum representation.

From fig(d), it is clear that this spectrum represents the time domain description of the signal $\cos(\omega_c+\omega_m)t$

$\cos(\omega_c+\omega_m)t = \cos\omega_c t \cos\omega_m t - \sin\omega_c t \sin\omega_m t$

Similarly, from fig(e), it is clear that this spectrum of the signal $\cos(\omega_c-\omega_m)t$ corresponds to a time domain

$\cos(\omega_c-\omega_m)t = \cos\omega_c t \cos\omega_m t + \sin\omega_c t \sin\omega_m t$

From the above two expressions, the standard expression for SSB-SC wave can be expressed as

$$\boxed{s(t)_{SSB} = \cos\omega_c t \cos\omega_m t \pm \sin\omega_c t \sin\omega_m t.}$$ single tone modulating sgl

Here '+' sign represents for lower side bands and '−' sign represents for upper side bands.

Here the signal $\sin\omega_c t$ can be obtained by shifting the phase of $\cos\omega_c t$ by $-90°$ i.e.,

$\sin\omega_c t = \cos(\omega_c t - \pi/2)$.

111ly, the signal $\sin\omega_m t$ can be obtained by shifting the phase of $\cos\omega_m t$ by $-90°$ i.e.,

$\sin\omega_m t = \cos(\omega_m t - \pi/2)$

For multitone modulating signal, the standard equation for SSB-SC wave can be expressed as

$$\boxed{s(t) = x(t)\cos\omega_c t + x_h(t)\sin\omega_c t}$$ Multi tone modulating sgl

$x(t) \rightarrow$ Multitone msg sgl and

$x_h(t)$ is the resultant signal obtained by shifting every frequency component present in $x(t)$.

Generation of SSB-SC wave:

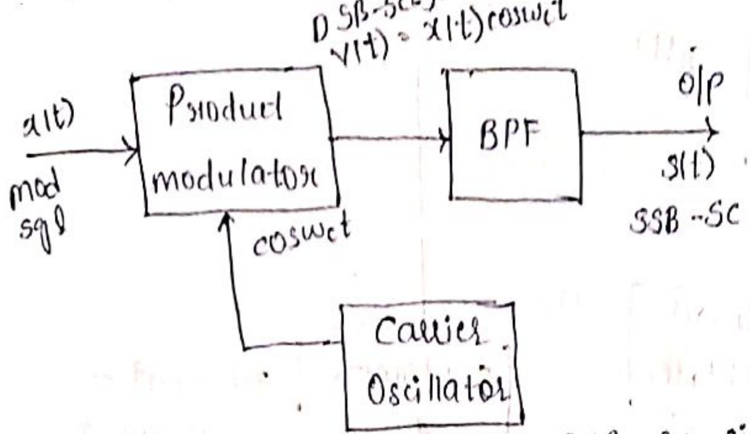
SSB-SC wave can be generated by using two methods.

They are

i) Frequency discrimination method

ii) Phase discrimination method

Frequency (or) filter discrimination method :-

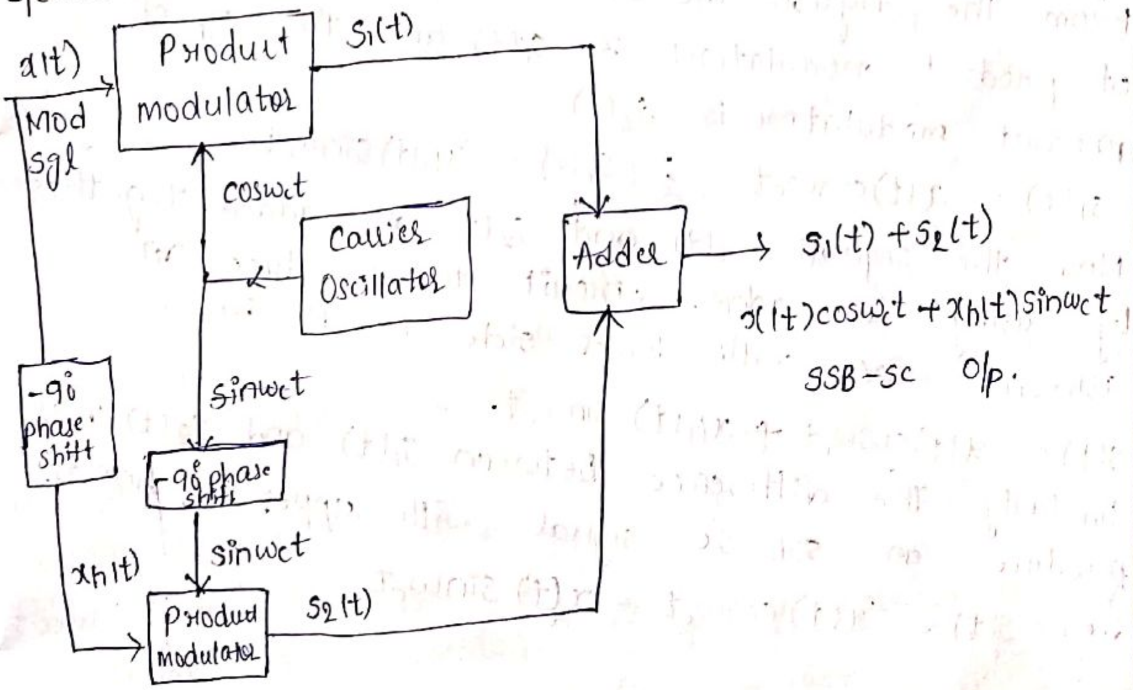



In this method, first a DSB-SC signal is generated by applying message and carrier signal as an inputs to one product modulator.

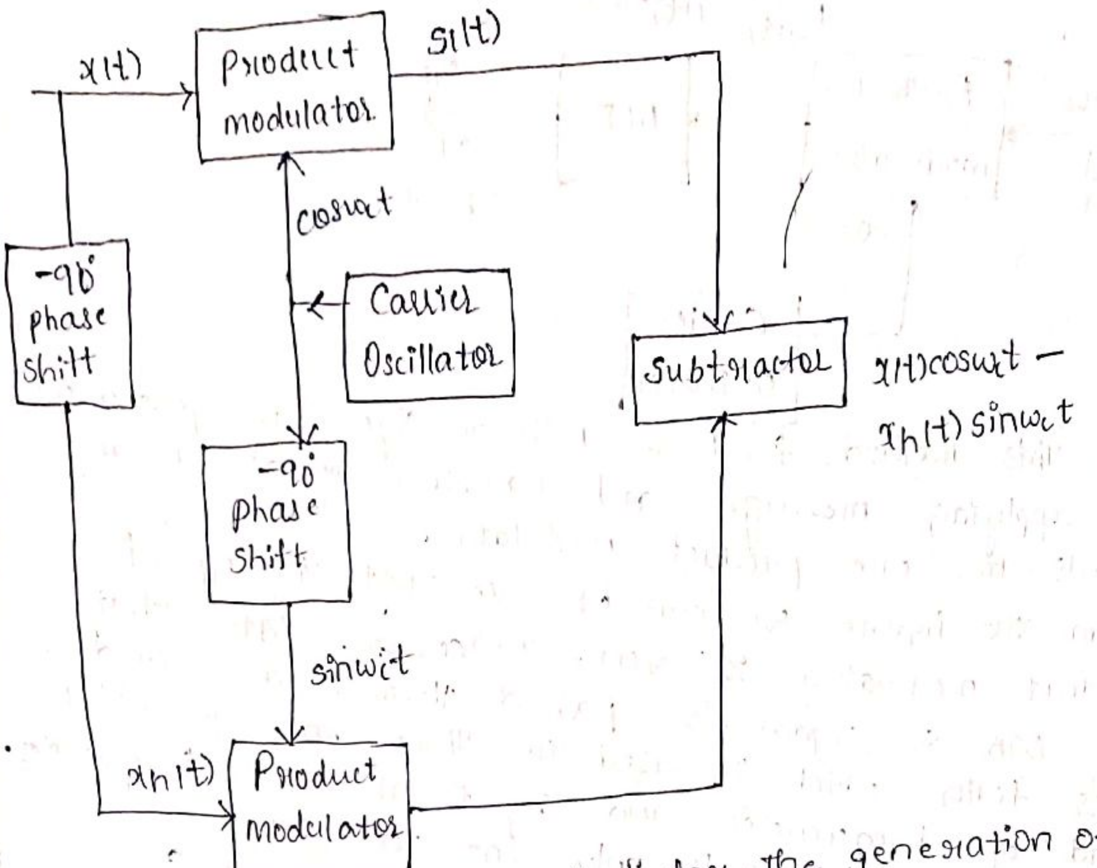
From the figure, we can observe that the o/p of product modulator is $v(t) = x(t)\cos\omega_c t$. After that the DSB-SC signal is passed through a band pass filter which is used to allow only particular band of frequencies and rejects all other frequencies. so, out of two sands only one side band is allowed by a band pass filter resulting in a SSB-SC signal generated at the o/p.

Phase discrimination Method :-

Generation of SSB-SC sgl with lower side bands:



For upper side bands :-




The phase discrimination method for the generation of SSB-SC uses two product modulators and two phase shifting circuits for the carrier and modulating signals respectively.

From the figure, we can observe that the output of product modulator1 is $s_1(t)$ and the o/p of product modulator is $s_2(t)$.

$s_1(t) = x(t)\cos\omega_c t$ ; $s_2(t) = x_h(t)\sin\omega_c t$

Now the signals $s_1(t)$ and $s_2(t)$ are added together by using an adder circuit to produce an SSB-SC wave with lower side bands. i.e.,

$s(t) = x(t)\cos\omega_c t + x_h(t)\sin\omega_c t$.

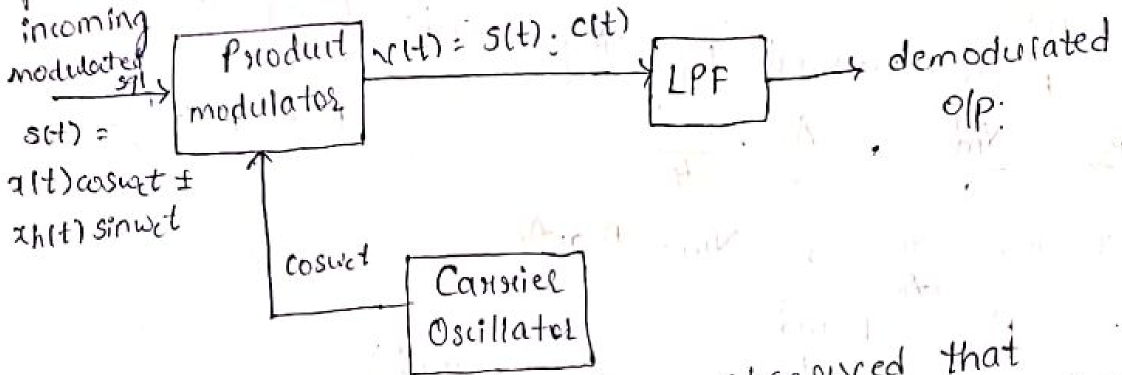
Similarly, the difference between $s_1(t)$ and $s_2(t)$ will produce an SSB-SC signal with upper side bands.

i.e., $s(t) = x(t)\cos\omega_c t - x_h(t)\sin\omega_c t$

DeModulation of SSB-SC:-

Synchronous (or) Coherent detection method:-

In synchronous detection method, the demodulated signal can be obtained by simply multiplying the incoming modulated signal s(t) with the locally generated carrier signal.




From the figure, it has been observed that s(t), the incoming modulated signal and $\cos\omega_c t$, locally generated carrier sgl are applied as an inputs to one product modulator. The o/p of product modulator is v(t). i.e., $v(t) = s(t)\cos\omega_c t$

$$v(t) = [x(t)\cos\omega_c t \pm x_h(t)\sin\omega_c t]\cos\omega_c t$$

$$v(t) = x(t)\cos^2\omega_c t \pm x_h(t)\sin\omega_c t\cos\omega_c t$$

$$v(t) = \frac{x(t)[1+\cos 2\omega_c t]}{2} \pm \frac{x_h(t)}{2}[\sin(\omega_c t+\omega_c t) + \sin(\omega_c t-\omega_c t)]$$

$$v(t) = \underbrace{\frac{x(t)}{2}}_{1} + \underbrace{\frac{x(t)\cos 2\omega_c t}{2}}_{2} \pm \underbrace{\frac{x_h(t)}{2}\sin(2\omega_c t)}_{3}$$

when this signal v(t) is passed through a low pass filter, only the ~~first term~~ is allowed by low pass filter and remaining terms are rejected by it. So, the o/p of LPF is nothing but the scaled version of our message sgl x(t)

Power saving in SSB-SC:-

* Calculate the percentage of power saving in SSB-SC for 100% modulation and 50% modulation.

Percentage of power saving in SSB-SC =

$$\frac{\text{carrier power} + \text{single side band power}}{\text{total power}} \quad \text{--- (1)}$$

WKT, the total power transmitted by an antenna is

$$P_t = P_c\left[1 + \frac{m_a^2}{2}\right]$$

The power carried by both the side bands is

$$P_{\substack{\text{dual}\\\text{side}\\\text{bands}}} = P_S = \frac{V_m^2}{4}$$

The power carried by single side band is

$$P_{SS} = \frac{V_m^2}{8} = \frac{A_c^2}{2} \cdot \frac{m_a^2}{4} \quad , \text{here } P_c = \frac{A_c^2}{2}$$

WKT $m_a = \frac{V_m}{A_c} \Rightarrow V_m = m_a \cdot A_c$

$$\therefore \quad P_{SS} = \frac{P_c m_a^2}{4}$$

The % of our power $= \frac{P_c + P_c\left(\frac{m_a^2}{4}\right)}{P_c\left[1 + \frac{m_a^2}{2}\right]} \times 100$

$$= \frac{\cancel{P_c}\left[1 + \frac{m_a^2}{4}\right]}{\cancel{P_c}\left[1 + \frac{m_a^2}{2}\right]} \times 100$$

For 100% modulation i.e., $m_a = 1$

$$= \frac{(1 + 1/4)}{(1 + 1/2)} \times 100$$

$$= \left[\frac{1 + 0.125}{1 + 0.5}\right] \quad \frac{\cancel{1.125}}{\cancel{1.5}} \, 0.83 \times 100 = 83.3\,\%$$

For 50% modulation $\therefore m_a = 0.5$

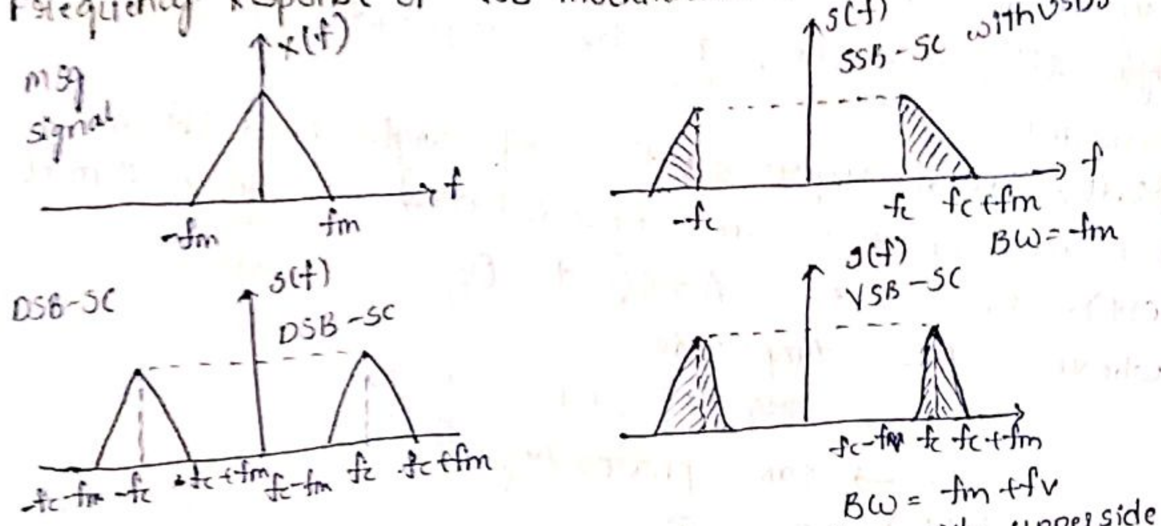
$$= \frac{\left[1 + \frac{(0.5)^2}{4}\right]}{1 + \frac{(0.5)^2}{2}} = 94.4\,\%$$

Vestigial Side band Modulation:

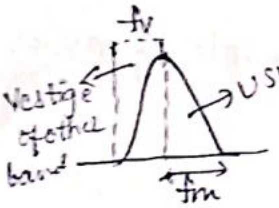
The stringent frequency response requirements on the designing of side band filter (band pass filter) in SSB-SC modulation can be relaxed by allowing some part of the other side band (vestige) to appear at the o/p of modulated ckt.

This increases the band width of the system slightly but the design of side band filter is simplified greatly. This type of modulation technique is called as VSB-SC modulation. This is the compromise b/w DSB-SC and SSB-SC modulation techniques.

Frequency Response of VSB modulation:-

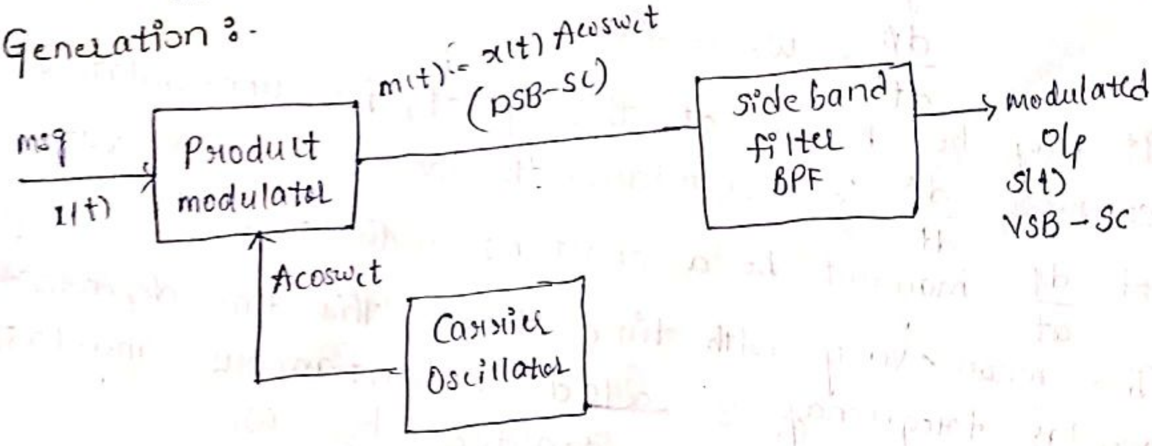



VSB modulation with upperside bands along with vestige of LSB.




$BW = f_m + f_v$

Generation :-



UNIT-3

# Angle Modulation

→ In amplitude modulation, we had seen the effect of slowly varying the amplitude of sinusoidal carrier wave in accordance with the baseband or information carrying or message signal i.e the carrier is amplitude modulated by the message signal m(t), hence the information content of m(t) is carried by the amplitude variations of the carrier.

→ There is another way of modulating a sinusoidal carrier wave namely "Angle modulation", in which the angle of the carrier wave is varied according to the baseband signal. The reason is any sinusoidal signal is described by 3 variables amplitude, frequency and phase.

→ Therefore, there exists a possibility of carrying the same information either by varying the frequency or phase of the carrier.

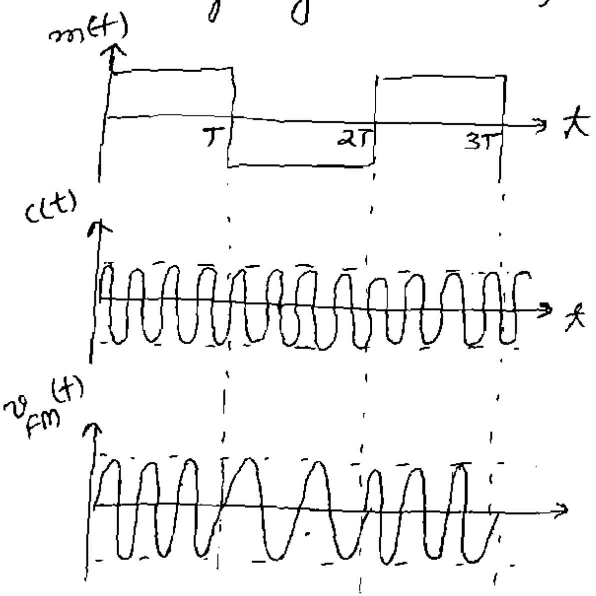



Fig (a)

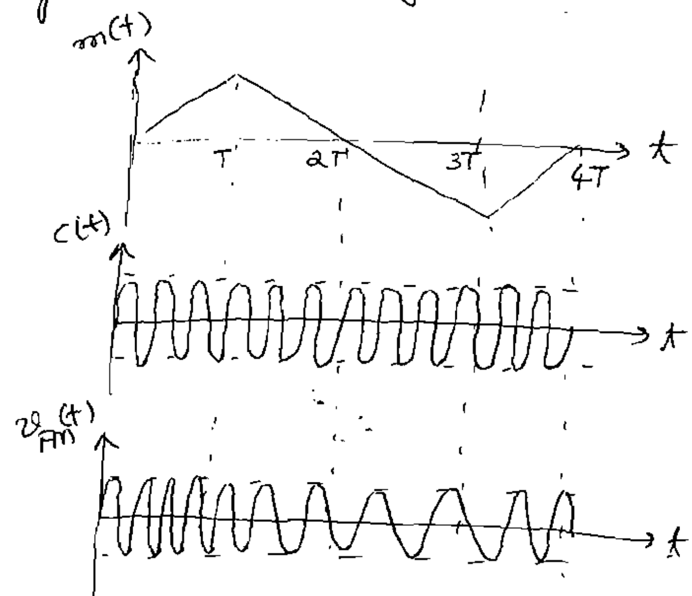



Fig (b)

from Fig (a)

→ maintaining same frequency $\omega_0$ between 0 and T as the amplitude of m(t) is constant in that interval

→ maintaining another frequency $\omega_1$ between T and 2T as the amplitude of m(t) is different in this interval

→ again the frequency is $\omega_0$ between 2T and 3T

from Fig (b):

→ we allowed a gradual variation in frequency as the amplitude is continuously varying in message m(t) at a uniform rate. Therefore, the frequency is different at every point

→ m(t) in Fig (b) can't be expressed by a simple sinusoidal expression, because, according it's definition, sinusoidal signal is a train of constant amplitude, frequency and phase.

→ Therefore, the new one should be defined as a generalized function whose amplitude, frequency and phase may vary with time.

→ The generalized sinusoidal signal is given by

$$f(t) = A \cos(\omega_c t + \phi) \rightarrow ①$$

where A - amplitude

$\omega_c$ - angular frequency

$f_c$ - $\frac{\omega_c}{2\pi}$ frequency

$\phi$ - phase

$$= A \cos \theta(t) \rightarrow ②$$

I case:

$\theta(t)$ is varied linearly with time

$$\theta_i(t) = \omega_c t + \phi + k_p m(t)$$

where $k_p$ is constand called as phase deviation

The new carrier after modulation is given by

$$v_{PM}(t) = V_c \cos[\omega_c t + \phi + k_p m(t)]$$

represents the phase modulated carrier

→ the instantaneous frequency $\omega_i$ for a phase modulated carrier is given by

$$\omega_i = \frac{d\theta}{dt} = \frac{d}{dt}[\omega_c t + \phi + k_p m(t)]$$

$$= \omega_c + K_p \frac{d}{dt} m(t)$$

i.e the instantaneous frequency $\omega_i$ for a phase modulated carrier signal varies linearly with derivative of the modulating signal.

II case:

direct modulation of instantaneous frequency

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

$$\theta_i(t) = \int \omega_i \, dt$$

$$= \int [\omega_c + K_f m(t)] \, dt$$

$$= \omega_c t + K_f \int m(t) \, dt + \underset{\substack{\uparrow \\ \text{integration constant}}}{\phi}$$

comparing ① and ②, we have

$$\theta(t) = \omega_c t + \phi$$

differentiating both sides w.r.to t

$$\frac{d\theta(t)}{dt} = \omega_c + 0$$

$$\omega_c = \frac{d\theta}{dt} \rightarrow ③$$

→ normally, the angular frequency is constant and is given by derivative of $\theta(t)$.

→ $\frac{d\theta}{dt}$ represents the instantaneous frequency $\omega_i$ which may or may not be constant.

$$\omega_i = \frac{d\theta}{dt}$$

$$\theta = \int \omega_i \, dt$$

→ from equation Ⓐ, it is clear that, information in $m(t)$ can be sent by varying the angle of a carrier.

→ Therefore, the techniques of changing the angle of a carrier in the same manner according to the modulating / message signal $m(t)$ are called angle modulation.

$$\theta(t) = \underbrace{\omega_c t}_{\text{I term}} + \underbrace{\phi}_{\text{II term}}$$

→ by changing the I term or II term, we can change the value of angle $\theta(t)$.

where $\beta = \frac{\Delta\omega}{\omega_m}$ is called modulation index of the FM signal

it is the ratio of the frequency deviation to the modulating signal frequency.

$$\theta_i(t) = \omega_c t + \beta \sin\omega_m t$$

$\beta$ also represents the phase deviation of the FM signal. i.e maximum departure of the angle $\theta_i(t)$ from the angle $\theta(t) = \omega_c t$ of the unmodulated carrier.

$\beta$ is measured in radians.

→ finally the FM signal is given by

$$v_{FM}(t) = A_c \cos[\omega_c t + \beta \sin\omega_m t]$$

## (i) Single tone frequency Modulation :

→ Let the sinusoidal modulating signal be

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

Let the high frequency carrier signal be

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

the instantaneous frequency of FM signal is given by

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

$$= \omega_c + K_f A_m \cos\omega_m t$$

$$= \omega_c + \Delta\omega \cos\omega_m t$$

where $\Delta\omega$ is called frequency deviation

→ $\Delta\omega$ represents the maximum departure of the instantaneous frequency of FM signal from the carrier frequency $\omega_c$.

→ $\Delta\omega = K_f A_m$

i.e $\Delta\omega \propto A_m$

proportional to amplitude of the modulating signal but not depending on it's frequency

→ the instantaneous angle of FM signal is given by

$$\theta_i(t) = \int \omega_i \, dt$$

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

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

$$= \omega_c t + \beta \sin\omega_m t$$

The new modulated carrier is given by

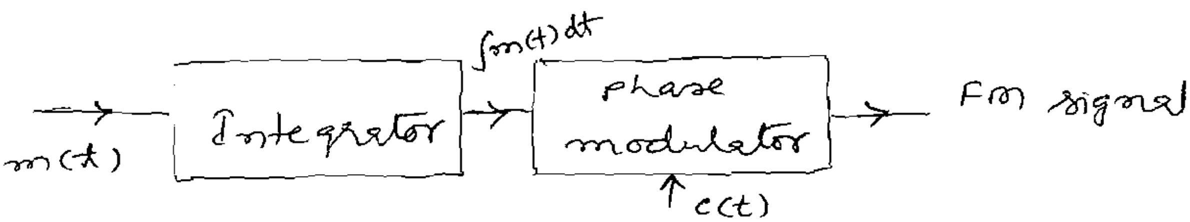
$$v_{FM}(t) = V_c \cos[\omega_c t + k_f \int m(t)\,dt + \phi]$$

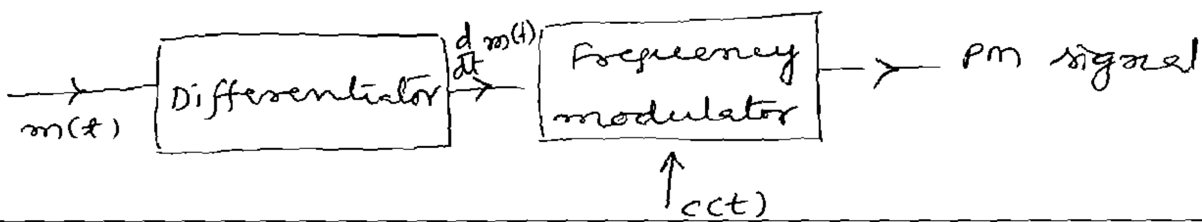
→ PM signal $v_{PM}(t) = V_c \cos[\omega_c t + \phi + k_p m(t)]$ → I

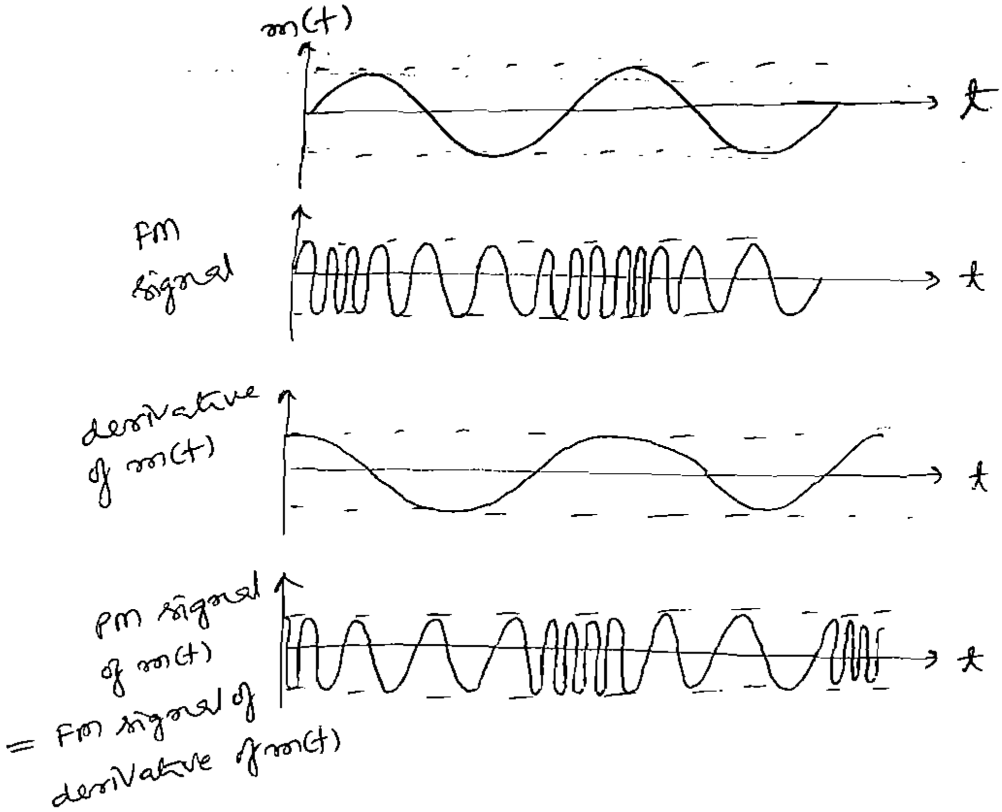
→ FM signal $v_{FM}(t) = V_c \cos[\omega_c t + \phi + k_f \int m(t)\,dt]$ → II

Basic differences between PM and FM

| PM | FM |
|---|---|
| → angle is varied linearly with the modulating signal | → angle is varied linearly with the integral of the modulating signal. |
| → differentiate the modulating signal m(t) first and allow it to frequency modulation of the carrier, we obtain phase modulated signal | → Integrate the modulating signal m(t) first and allow it to phase modulation of the carrier, we obtain frequency modulated signal. |






→ from eqn (II), it is clear that, FM signal is a non linear function of the modulating signal m(t), which makes frequency modulation a non-linear modulation process.

∴ the spectrum of a FM signal is not related in a manner to that of the modulating signal. The analysis will be more difficult than amplitude modulation due to this non-linear behavior.

→ There are several methods of FM

(i) Single tone FM – NBFM

(ii) Single tone FM – WBFM

(iii) Multi tone FM –

# Narrow Band Frequency Modulation (NBFM)

$$v_{FM}(t) = A_c \cos[\omega_c t + \beta \sin \omega_m t]$$

$$= A_c \cos(\omega_c t) \cos(\beta \sin \omega_m t) - A_c \sin(\omega_c t) \sin(\beta \sin \omega_m t)$$

if $\beta$ is very small then,

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

$\cos(\beta \sin \omega_m t) \cong 1$

$\sin(\beta \sin \omega_m t) \cong \beta \sin \omega_m t$

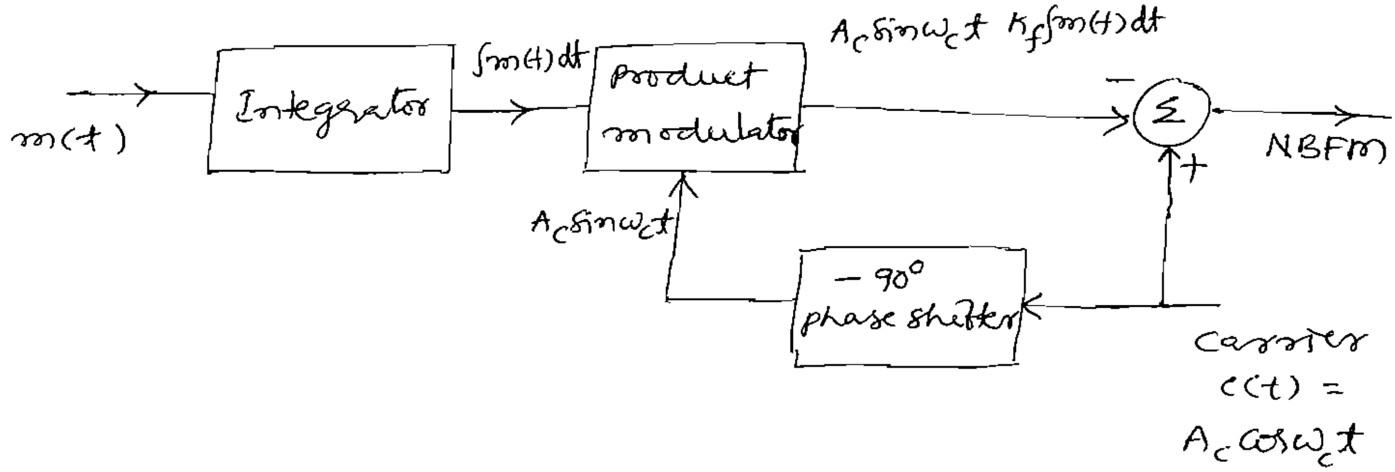
for small values of $\theta$

$\cos\theta = 1$ $\quad$ $\cos\theta = 1 - \frac{\theta^2}{2!} + \cdots$

$\sin\theta = \theta$ $\quad$ $\sin\theta = \theta - \frac{\theta^3}{3!} + \cdots$

$$= A_c \cos \omega_c t \, (1) - A_c \sin \omega_c t \, \beta \sin \omega_m t$$

$$= A_c \cos \omega_c t - \beta A_c \sin \omega_c t \sin \omega_m t \longrightarrow (A)$$




→ This arrangement produces some distortion. In FM, the amplitude should be constant and $\theta_i(t)$ should be sinusoidal with same frequency $\omega_m$, but the output from the above method, it is clear that the envelope is containing a residual amplitude modulation and that varies with time

→ The second problem is that, the instantaneous angle $\theta(t)$ is containing harmonic distortion in the form of third and higher order harmonics of the modulating frequency $\omega_m$

→ But, by selecting $\beta << 0.3$ radians, the above problems can be rectified.

$$v_{FM}(t) = A_c \cos\omega_c t - \beta A_c \sin\omega_c t \sin\omega_m t \; \frac{\times 2}{\times 2}$$

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

$$= A_c \cos\omega_c t - \frac{\beta A_c}{2}\cos(\omega_c - \omega_m)t + \beta\frac{A_c}{2}\cos(\omega_c+\omega_m)t \rightarrow ①$$

↳ standard form of NBFM — $v_{NBFM}(t)$.

comparison with $v_{AM}(t)$

$$v_{AM}(t) = V_c \cos\omega_c t - \frac{\mu V_c}{2}\cos(\omega_c - \omega_m)t + \frac{\mu V_c}{2}\cos(\omega_c+\omega_m)t \rightarrow ②$$

$\mu$ and $\beta$ are modulation indices

$V_c$ and $A_c$ are amplitudes of carrier

→ hence eqns ① and ② are taking the same form except the sign of lower sideband

→ This form of frequency modulation is called Narrow Band Frequency modulation (NBFM) as the bandwidth is $2\omega_m$ which is same as bandwidth of AM signal.

Wideband frequency modulation (WBFM)

→ The basic equation of frequency modulation is given by

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

This FM signal is a non-periodic unless the carrier frequency $\omega_c$ is an integral multiple of the modulating signal frequency $\omega_m$.

→ Assume, the carrier frequency $\omega_c >>$ BW of FM signal, rewriting the above equation as given below

$$s(t) = Re\{ A_c e^{j(\omega_c t + \beta \sin\omega_m t)}$$

$$= Re\{ A_c e^{j\omega_c t + j\beta \sin\omega_m t} \}$$

$$= Re\{ A_c \, e^{j\omega_c t} . e^{j\beta \sin\omega_m t} \}$$

$$= Re\{ \tilde{s}(t) \; e^{j\omega_c t} \}$$

where $\tilde{s}(t) = A_c e^{j\beta \sin\omega_m t}$ is the complex envelope of the signal $e^{j\omega_c t}$.

But, this complex envelope is a periodic function of time with fundamental frequency equal to modulating frequency $\omega_m$.

→ As we know that, any periodic signal can be represented as Fourier series.

∴ The complex Fourier series representation of $\tilde{s}(t)$ is as follows.

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} C_n e^{jn\omega_m t}$$

where $$C_n = \frac{1}{T=\frac{1}{f_m}} \int_{-\frac{1}{2}f_m}^{+\frac{1}{2}f_m} \tilde{s}(t)\, e^{-jn\omega_m t}\, dt$$

$$= f_m \int_{-\frac{1}{2}f_m}^{+\frac{1}{2}f_m} A_c e^{j\beta \sin \omega_m t} e^{-jn\omega_m t}\, dt$$

$$= f_m A_c \int_{-\frac{1}{2}f_m}^{\frac{1}{2}f_m} e^{j\beta \sin \omega_m t - jn\omega_m t}\, dt$$

<u>limits</u>

l.l $\to \omega_m(-\frac{1}{2}f_m)$

$\cancel{2}\pi \cancel{f_m} \times -\frac{1}{\cancel{2}\cancel{f_m}} = -\pi$

u.l $\to \omega_m(+\frac{1}{2}f_m) = \pi$

$\cancel{2}\pi \cancel{f_m} (\frac{1}{\cancel{2}\cancel{f_m}})$

put $x = \omega_m t \Rightarrow x = 2\pi f_m t$

$$\frac{dx}{dt} = \omega_m \qquad f_m = \frac{x}{2\pi t}$$

$$-\tfrac{1}{2} f_m = \frac{x}{-4\pi t}$$

$$dx = \omega_m\, dt$$

$$+\tfrac{1}{2} f_m = \frac{x}{4\pi t}$$

$$= 2\pi f_m\, dt$$

$$dt = \frac{dx}{2\pi f_m}$$

$$= \cancel{f_m} A_c \int_{-\pi}^{\pi} e^{j\beta \sin x - jnx} \frac{dx}{2\pi \cancel{f_m}}$$

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

→ except scaling factor $A_c$, it is recognized as the $n^{th}$ order Bessel function of the first kind and argument $\beta$. ∴ it is denoted by $J_n(\beta)$.

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

$$C_n = A_c J_n(\beta)$$

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} A_c J_n(\beta)\, e^{jn\omega_m t} \quad \text{and}$$

$$s(t) = Re\left\{ \sum_{n=-\infty}^{\infty} A_c J_n(\beta)\, e^{jn\omega_m t}\, e^{j\omega_c t} \right\}$$

$$= A_c\, Re\left\{ \sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{j(\omega_c + n\omega_m)t} \right\}$$

after interchanging order of summation and evaluating the real part in the RHS of above equation gives

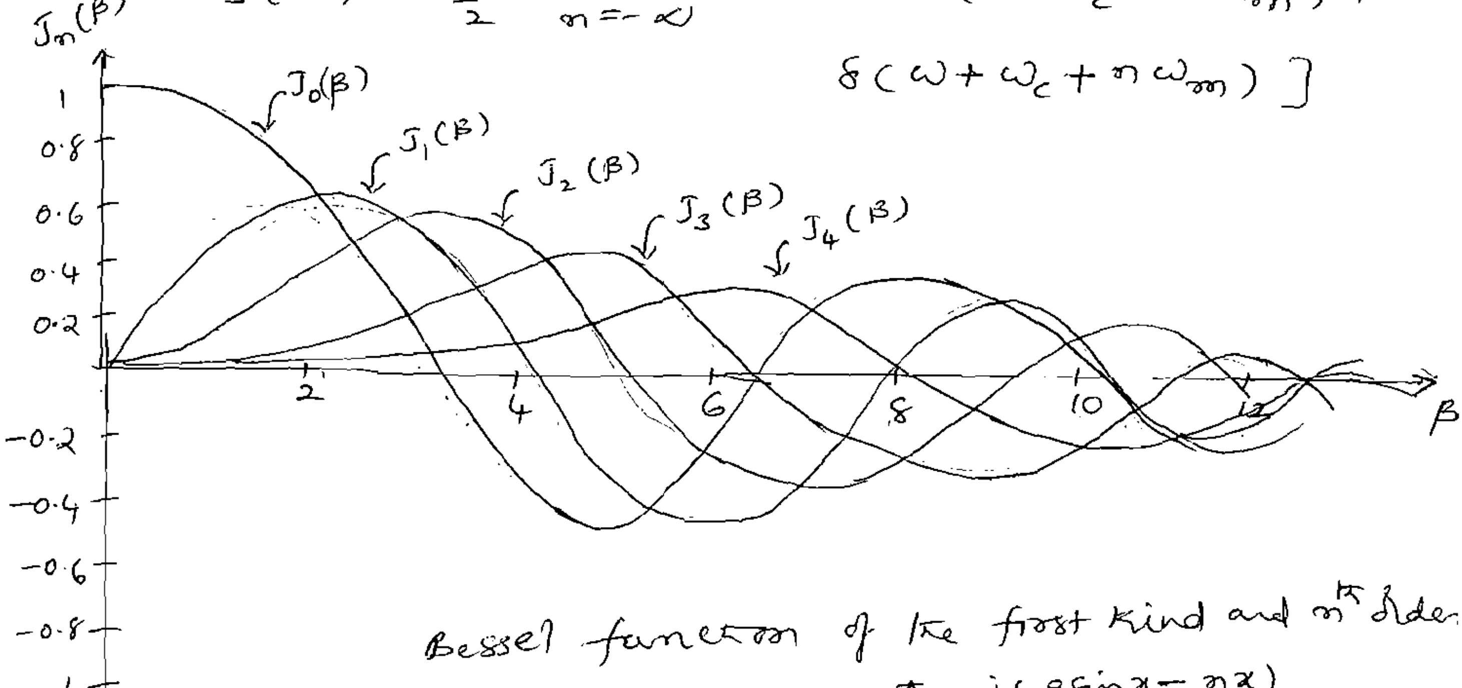
$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos(\omega_c + n\omega_m)t$$

↳ standard form of WBFM $v_{WBFM}(t)$.

→ This is the required form of Fourier series representation of FM signal for single tone modulation for an arbitrary value of $\beta$.

→ The spectrum of $s(t)$ can be obtained by taking the Fourier transform of on both sides

$$S(\omega) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta) \left[ \delta(\omega - \omega_c - n\omega_m) + \delta(\omega + \omega_c + n\omega_m) \right]$$




Bessel function of the first kind and $n^{th}$ order

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

Properties of Bessel for $J_n(\beta)$

(i) $J_n(\beta) = J_{-n}(\beta)$ → for $n$ even

$J_n(\beta) = -J_{-n}(\beta)$ → for $n$ odd

$$J_n(\beta) = (-1)^n J_{-n}(\beta)$$

→ for all values of $n$

(ii) for small values of $\beta$

$J_0(\beta) = 1$

$J_1(\beta) = \beta/2$

$J_n(\beta) = 0$ for $n \geq 2$

(iii) $$\sum_{n=-\infty}^{\infty} J_n^2(\beta) = 1$$

→ The WBFM is given by (for $\beta$ is large value)

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos(\omega_c + n\omega_m)t \rightarrow ①$$

$$= A_c \{ J_0(\beta) \cos\omega_c t + J_1(\beta) \cos(\omega_c + \omega_m)t +$$
$$J_{-1}(\beta) \cos(\omega_c - \omega_m)t + J_2(\beta) \cos(\omega_c + 2\omega_m)t +$$
$$J_{-2}(\beta) \cos(\omega_c - \omega_m)t + - - - - \}$$

from property (i) of $J_n(\beta)$

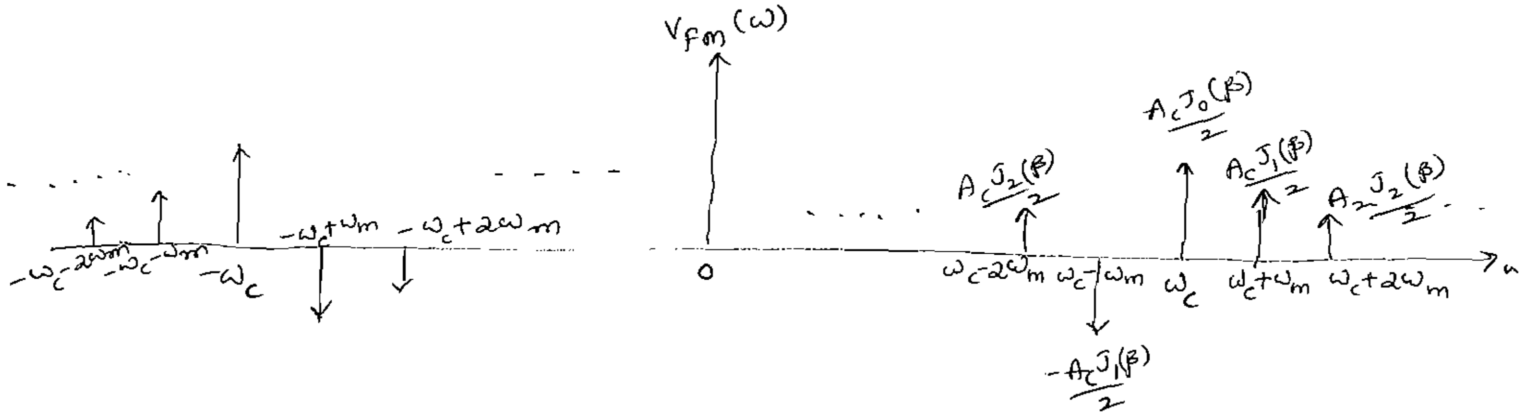
$J_1(\beta) = -J_{-1}(\beta)$

$J_2(\beta) = J_{-2}(\beta)$ and so on

$$= A_c \{ J_0(\beta)\cos\omega_c t + J_1(\beta) \cos(\omega_c + \omega_m)t -$$
$$J_1(\beta) \cos(\omega_c - \omega_m)t + A_c J_2(\beta) \cos(\omega_c + 2\omega_m)$$
$$+ J_2(\beta) \cos(\omega_c + 2\omega_m)t + - - - \}$$

$$= A_c J_0(\beta)\cos\omega_c t + A_c J_1(\beta)\{\cos(\omega_c+\omega_m)t - \cos(\omega_c-\omega_m)t\}$$
$$+ A_c J_2(\beta)\{\cos(\omega_c+2\omega_m)t + \cos(\omega_c-2\omega_m)t\}$$
$$+ \cdots \quad \rightarrow (2)$$

→ Thus, the frequency modulated signal contains carrier component and an infinite number of side frequency components at $\omega_c \pm \omega_m$, $\omega_c \pm 2\omega_m$ ---- an so on as shown below




$$J_n(\beta) = \left(\frac{\beta}{2}\right)^n \left[\frac{1}{n!} - \frac{(\beta/2)^2}{1!(n+1)!} + \frac{(\beta/2)^4}{2!(n+2)!} - \cdots\right] \rightarrow (3)$$

→ From eqns (1), (2) & (3), it is clear that

(1) FM has infinite number of sidebands as well as carrier and all they are separated from carrier by $\pm\omega_m$, $\pm 2\omega_m$, $\pm 3\omega_m$ ----. But, in AM there are only 3 components.

(2) The modulation index determines the how many sidebands components have significant amplitude i.e for large value of $\beta$, more number of significant sidebands are present. if $\beta$ is small, less no. of sidebands will exist.

(3) The sidebands are at equal distance from $\omega_c$ have equal amplitudes so that the sideband distribution is symmetrical about the carrier frequency.

(4) Theoritically, infinite sidebands are produced and the amplitude of each sideband is decided by the Bessel function.

(5) The presence of infinite no. of sidebands makes the bandwidth of FM infinite. However, the sidebands with negligible amplitudes are ignored, then the BW of FM becomes finite

(6) for small values of $\beta$ i.e $\beta << 1$ only the amplitudes of $J_0(\beta)$ and $J_1(\beta)$ are significant and other terms can be neglected.
This is equivalent to NBFM

(7) The amplitude of FM remains unchanged and hence the power of FM is same as that of the unmodulated carrier

(8) The total power of FM signal depends on the power of the unmodulated carrier, whereas in AM, the total power depends on the modulation index

(9) In AM, the increased modulation index increases the sideband power
In FM, the total power remains constant with increased modulation index and only the BW is increased.

## Multitone Modulation

→ modulation can be carried out with more than one message signal is called multitone modulation.

Let us consider the message signal as

$$m(t) = A_{m_1} \cos\omega_{m_1} t + A_{m_2} \cos\omega_{m_2} t$$

Let the carrier signal be

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

The frequency of carrier is changed according to instantaneous values of message signal

$$\therefore \quad \omega_i = \omega_c + K_f\, m(t)$$
$$= \omega_c + K_f (A_{m_1} \cos\omega_{m_1} t + A_{m_2} \cos\omega_{m_2} t)$$

The frequency deviation is maximum when $\cos\omega_{m_1} t = \pm 1$ and $\cos\omega_{m_2} t = \pm 1$

The frequency deviation is proportional to the amplitude of the modulating signal, which is

$$\Delta\omega_1 = K_f A_{m_1} \quad \& \quad \Delta\omega_2 = K_f A_{m_2}$$

$$\omega_i = \omega_c + \Delta\omega_1 \cos\omega_m t + \Delta\omega_2 \cos\omega_{m_2} t$$

The instantaneous phase is given by

$$\theta_i = \int \omega_i \, dt$$
$$= \int [\omega_c + \Delta\omega_1 \cos\omega_{m_1} t + \Delta\omega_2 \cos\omega_{m_2} t]\, dt$$
$$= \omega_c t + \Delta\omega_1 \frac{\sin\omega_{m_1} t}{\omega_{m_1}} + \Delta\omega_2 \frac{\sin\omega_{m_2} t}{\omega_{m_2}}$$
$$= \omega_c t + \beta_1 \sin\omega_{m_1} t + \beta_2 \sin\omega_{m_2} t$$

where $\beta_1 = \frac{\Delta\omega_1}{\omega_{m_1}}$ & $\beta = \frac{\Delta\omega_2}{\omega_{m_2}}$

→ The frequency modulated signal is given by

$$v_{FM}(t) = A_c \cos\theta_i t$$

$$= A_c \cos[\omega_c t + \beta_1 \sin\omega_{m_1} t + \beta_2 \sin\omega_{m_2} t]$$

$$= A_c \cos(\omega_c t + (\alpha_1 + \alpha_2))$$

where $\alpha_1 = \beta_1 \sin\omega_{m_1} t$

$\alpha_2 = \beta_2 \sin\omega_{m_2} t$

$$= A_c [\cos\omega_c t \cos(\alpha_1 + \alpha_2) - \sin\omega_c t \sin(\alpha_1 + \alpha_2)]$$

$$= A_c[\cos\omega_c t(\cos\alpha_1 \cos\alpha_2 - \sin\alpha_1 \sin\alpha_2) -$$

$$\sin\omega_c t(\sin\alpha_1 \cos\alpha_2 + \cos\alpha_1 \sin\alpha_2)] \longrightarrow (1)$$

→ Bessel functions can be used to solve eqn (1)

$$v_{FM}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\alpha_1) J_n(\alpha_2) \cos(\omega_c t \pm n\omega_{m_1} t + n\omega_{m_2} t)$$

Observations:

(i) a carrier frequency component with an amplitude $J_0(\alpha_2)\, J_0(\alpha_2)\, A_c$

(ii) A set of sidebands having amplitude $J_n(\alpha_1)\, J_n(\alpha_2)$ and frequencies $(\omega_{m_1} + n\omega_{m_2})$ where $n = 1, 2, 3 - - -$

## Transmission BW of FM

→ The no. of sidebands which are having significant amplitudes 'n' produced in FM waves can be obtained from the plot of Bessel fn $J_n(\beta)$

→ for $\beta < n$, the value of $J_n(\beta)$ are negligible particularly when $\beta >> 1$, ∴ the significant sidebands produced in WBFM may be considered to be an integer approximately equal to $\beta$.

$$\text{i.e } n \cong \beta \quad \text{if} \quad \beta >> 1$$

→ The USB frequencies are separated by $\omega_m$ and form a frequency span $n\omega_m$, also it is similar by LSBs.

→ ∴ transmission BW of FM signal is

$$BW = 2n\omega_m \quad \text{rad/sec}$$

where $n$ = no. of sidebands

$$= 2\beta\omega_m$$

$$= 2\frac{\Delta\omega}{\cancel{\omega_m}}\cancel{\omega_m}$$

$$= 2\Delta\omega \quad \text{rad/sec}$$

approximately BW of FM is twice the frequency deviation for $\beta >> 1$

but, for smaller values of $\beta$, it may be greater than $2\Delta\omega$

→ this is given by approximate rule for transmission BW of FM signal

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

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

$$= 2\Delta\omega\left(1+\frac{1}{\beta}\right)$$

This empirical relation is known as "Carson's rule".

## Power Content in FM

consider the signal

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

Frequency modulated signal (FM) is given by

$$v_{FM}(t) = A_c \cos\left[\omega_c t + K_f \int m(t)\,dt\right]$$

for $m(t) = A_m \cos\omega_m t$

$$= A_c \cos\left(\omega_c t + \beta \sin\omega_m t\right)$$

where $\beta = \frac{\Delta\omega}{\omega_m} = \frac{K_f A_m}{\omega_m}$

→ ∴ The envelope of the FM signal is constant by observing the above expression.

→ The power of the signal is calculated from it's amplitude

→ In FM, amplitude is constant which is also equal to the carrier amplitude, only frequency of the carrier is varied according to modulating signal.

∴ Power of the FM signal is

$$P = \frac{A_c^2}{2} \quad \text{per unit resistor}$$

power is independent of $\beta$

$$v_{FM}(t) = A_c J_0(\beta)\cos\omega_c t +$$
$$A_c J_1(\beta)\cos(\omega_c+\omega_m)t - A_c J_1(\beta)\cos(\omega_c-\omega_m)t +$$
$$A_c J_2(\beta)\cos(\omega_c+2\omega_m)t - A_c J_2(\beta)\cos(\omega_c-2\omega_m)t + \cdots$$

$$\text{Power} = \left(\frac{A_c J_0(\beta)}{\sqrt{2}}\right)^2 + \left(\frac{A_c J_1(\beta)}{\sqrt{2}}\right)^2 + \left(\frac{A_1 J_1(\beta)}{2}\right)^2 +$$
$$\left(\frac{A_c J_2(\beta)}{\sqrt{2}}\right)^2 + \left(\frac{A_c J_2(\beta)}{\sqrt{2}}\right)^2 + \cdots$$

$$= \frac{A_c^2}{2} J_0^2(\beta) + \frac{A_c^2}{2} J_1(\beta) + \frac{A_c^2}{2} J_1(\beta) +$$
$$\frac{A_c^2}{2} J_2^2(\beta) + \frac{A_c^2}{2} J_2^2(\beta) + \cdots$$

$$= \frac{A_c^2}{2}\left[J_0^2(\beta) + 2J_1^2(\beta) + 2J_2^2(\beta) + \cdots\right]$$

$$= \frac{A_c^2}{2}[1] \qquad \sum_{n=-\infty}^{\infty} J_n^2(\beta) = 1$$

$$= \frac{A_c^2}{2} \text{ per unit resistor}$$

$$= \text{carrier power}$$

→ if modulation index $\beta$ increases, then BW of FM signal increases but power transmitted by FM is not changed.

→ As modulation index $\beta$ increases, sidebands increases, thereby most of the power located in sidebands. Thus, the efficiency is improved in FM system.

Problem : 1

Determine the instantaneous frequency in Hz of each of the following signal

(a) $10 \cos(200\pi t + \pi/3)$ (b) $10 \cos(20\pi t + \pi t^2)$

(c) $\cos 200\pi t \cos(5 \sin 2\pi t) + \sin 200\pi t \sin(5 \sin 2\pi t)$

Sol$^n$ : (a) $\theta(t) = 200\pi t + \pi/3$

$$\omega_i = \frac{d\theta_i}{dt} = \frac{d}{dt}(200\pi t + \pi/3)$$

$$= 200\pi = 2\pi(100) = 2\pi f$$

$$f = 100 \text{ Hz}$$

(b) $\theta(t) = 20\pi t + \pi t^2$

$$\omega_i = \frac{d\theta_i}{dt} = \frac{d}{dt}(20\pi t + \pi t^2)$$

$$= 20\pi + 2\pi t = 2\pi(10 + t)$$

$$= 2\pi f$$

$$f = (10 + t) \text{ Hz}$$

(c) $\theta_i(t) = \cos(200\pi t - 5 \sin 2\pi t)$

$$\omega_i = \frac{d\,\theta_i(t)}{dt} = \frac{d}{dt}(200\pi t - 5 \sin 2\pi t)$$

$$= 200\pi - 5 \cos 2\pi t\,(2\pi)$$

$$= 2\pi(100 - 5 \cos 2\pi t)$$

$$f = 100 - 5 \cos 2\pi t$$

Problem : 2

consider the angle modulated signal

$s(t) = 10 \cos[10^8 \pi t + 5 \sin 2\pi 10^3 t]$. Find maximum phase deviation and frequency deviation

Sol$^n$ :

$$s(t) = 10 \cos(10^8 \pi t + 5 \sin 2\pi 10^3 t)$$

$$v_{FM}(t) = A_c \cos(\omega_c t + \beta \sin \omega_m t)$$

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

$$\therefore \theta(t) = \omega_c t = 10^8 \pi t$$

$$\theta_i(t) = \omega_c t + \beta \sin \omega_m t$$

$$10^8 \pi t + 5 \sin 2\pi 10^3 t$$

$\theta_i(t)$ is deviated from $\theta(t)$ by 5 radians

$$\omega_i = \frac{d\theta_i}{dt} = \frac{d}{dt}\left(10^8 \pi t + 5 \sin 2\pi 10^3 t\right)$$

$$= 10^8 \pi + 5 \cos 2\pi 10^3 t \; (2\pi 10^3)$$

$$\omega_c = 10^8 \pi$$

$$\omega_{i\,(max)} = 10^8 \pi + 5(2\pi 10^3)$$

instantantaneous frequency deviated by $5 2\pi 10^3$

**Problem: 3**

An angle modulated signal is described by

$$s(t) = 10 \cos(2\pi 10^6 t + 0.1 \sin 10^3 \pi t)$$

Find $m(t)$

(a) considering $s(t)$ as a PM signal with $K_p = 10$

(b) considering $s(t)$ as a FM signal with $K_f = 10$

**Sol$^n$**:

$$s_{PM}(t) = A_c \cos(\omega_c t + K_p m(t))$$

$$= 10 \cos(2\pi 10^6 + 0.1 \sin 10^3 \pi t)$$

$$= 10 \cos(2\pi 10^6 t + 10 \times 0.01 \sin 10^3 \pi t)$$

$$\therefore m(t) = 0.01 \sin 10^3 \pi t$$

$$s_{FM}(t) = A_c \cos(\omega_c t + K_f \int m(t)\, dt)$$

$$= 10 \cos(2\pi 10^6 t + 0.1 \sin 10^3 \pi t)$$

$$K_f \int m(t)\, dt = 0.1 \sin 10^3 \pi t$$

put $m(t) = A_m \cos \omega_m t$

$$K_f \int A_m \cos\omega_m t \, dt = 0.1 \sin 10^3 \pi t$$

$$K_f A_m \frac{\sin\omega_m t}{\omega_m} = 0.1 \sin 10^3 \pi t$$

$$\frac{K_f A_m}{\omega_m} = 0.1$$

$$\frac{10 A_m}{10^3 \pi} = 0.1 \Rightarrow A_m = 10\pi$$

$$\therefore m(t) = 10\pi \cos 10^3 \pi t$$

**Problem: 4**

Given an angle modulated signal as

$$s(t) = 10 \cos(2\pi 10^8 t + 200 \cos 2\pi 10^3 t)$$

what is it's BW

**Sol$^n$:** $BW = 2\Delta\omega = 2\beta\omega_m$

$$\omega_i = \frac{d\theta_i}{dt} = \frac{d}{dt}(2\pi 10^8 t + 200 \cos 2\pi 10^3 t)$$

$$= 2\pi 10^8 + 200 \sin 2\pi 10^3 t \,(2\pi 10^3)$$

$$= \omega_c + \Delta\omega \sin 2\pi 10^3 t$$

$$\Delta\omega = 200\,(2\pi 10^3)$$
$$= 4\pi 10^5$$

$$\beta = \frac{\Delta\omega}{\omega_m} = \frac{4\pi 10^5}{2\pi 10^3} = 200$$

(i) From Carson's rule

$$BW = 2\Delta\omega\left(1 + \frac{1}{\beta}\right)$$
$$= 2\omega_m(1+\beta)$$
$$= 2(2\pi 10^3)(1+200)$$
$$= 8.04\pi 10^5 \text{ rad/s}$$
$$= 2\pi(402)10^3 \text{ rad/sec}$$
$$= 402 \text{ KHz}$$

(ii) from approximate meth

since $\beta >> 1$

$$BW = 2\Delta\omega$$
$$= 2(4\pi 10^5)$$
$$= 2\pi \times 400 \times 10^3 \text{ rad/s}$$
$$= 400 \text{ KHz}$$

# 3- Angle Modulation

Def:-
It is defined as a modulation process in which the total phase angle of a carrier sgl is varied in accordance with the instantaneous values of modulating sgl $x(t)$ while keeping the amplitude of carrier constant

Mathematical representation of angle modulation:-
Let us consider an unmodulated carrier signal $c(t)$. i.e., $c(t) = A\cos(\omega_c t + \theta_0)$

where $A \rightarrow$ Amplitude
$\omega_c \rightarrow$ freq. of carrier sgl.
$\theta_0 \rightarrow$ same phase angle.

Let $(\omega_c t + \theta_0) = \phi$ —①

$\therefore \quad c(t) = A\cos\phi$ —②

where $\phi = (\omega_c t + \phi_0)$ represents total phase angle of carrier.
Differentiating on b.s of above equation

$$\frac{d\phi}{dt} = \omega_c \quad —③$$

It may be noted at this point, for unmodulated carrier $\frac{d\phi}{dt}$ is a constant. However this derivative of $\frac{d\phi}{dt}$ may not be a constant with time
This mean vary with time. Hence this time dependent angular frequency is called instantaneous angular frequency and it is denoted by $\omega_i$.

$$\omega_i = \frac{d\phi}{dt}$$

$$(or) \quad \phi = \int \omega_i dt \quad —④$$

If this total phase angle $\phi$ is varied linearly with the msg signal then the carrier wave is said to be angle modulated
$c(t) = A\cos\phi$

Types of angle modulation:-
It is classified into 2 types.

Angle modulation

Phase Modulation (phase of carrier changes)

Frequency modulation (freq. of carrier changes)

Phase Modulation:-

The modulation process in which the phase of the carrier signal is changed according to instantaneous values of message signal is called as phase modulation.

Frequency Modulation:

The modulation process in which the frequency of carrier signal is changed according to instantaneous values of message signal is called as frequency modulation

Advantages of Angle modulation:

1. Noise reduction increases when compared to amplitude modulation.

2. Increases system Fidelity.

Disadvantages:

1. The transmission band width requirement more compared to amplitude modulation.

2. Complexity of circuit designing increases.

Applications:

1. Radio broad casting

2. Microwave communication

3. Satellite communication

4. point to point communication

Phase Modulation:-

Phase Modulation is that type of modulation in which the phase of the carrier signal $\phi$ varies linearly with the base band signal (or) message signal $x(t)$. This means that, in phase modulation the instantaneous phase angle is equal to an modulated carrier phase angle plus a time varying component which is proportional to $x(t)$.

$\phi_i$ = unmodulated phase angle + time varying component proportional to $x(t)$.

Mathematical Representation of phase Modulation:

Let us consider an unmodulated carrier signal

$c(t) = A\cos(\omega_c t + \phi_0)$

$c(t) = A\cos\phi$ —①

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

by neglecting $\theta_0$ we can write $\phi = \omega_c t$

According to phase modulation this phase angle $\phi$ is varied linearly with the message signal $x(t)$ to get instantaneous phase angle.

Let the instantaneous phase angle be denoted by $\phi_i$.

$\therefore \phi_i = \omega_c t + k_p x(t)$

where $K_p$ = proportionality constant, known as phase sensitivity of modulation, and that is expressed as rad/Volts.

Since the expression for carrier signal is $c(t) = A\cos\phi$ the expression for o/p phase modulator signal is

$s(t) = A\cos\phi_i$.

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

which is the required expression for phase modulated wave

Frequency Modulation :-

FM modulation is the type of modulation in which the frequency of carrier signal is varied linearly with the message signal $x(t)$. This means that, in frequency modulation the instantaneous value of angular frequency is equal to unmodulated carrier frequency plus a time varying component which is proportional to $x(t)$.

$\omega_i$ = unmodulated carrier freq. + time varying component proportional to $x(t)$.

Mathematical representation of freq. modulation:

$$\boxed{\omega_i = \omega_c + k_f\, x(t)} \quad \text{--- (1)}$$

$k_f$ = proportionality constant which is known as frequency sensitivity → units: Hz/volts

Let us consider an unmodulated carrier signal

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

$\phi = \omega_c t + \theta_0$

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

Let $\phi_i$ be the instantaneous phase angle of modulated signal. Now according to FM, the amplitude of the carrier signal remains constant and only total phase angle $\phi$ will changes. Hence the expression for freq. modulated wave will be

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

wkT $\phi_i = \int \omega_i\, dt$ --- (2)

sub. eq (1) in eq (2)

$\therefore \phi_i = \int [\omega_c + k_f\, x(t)]\, dt$

$\phi_i = \omega_c t + k_f \int x(t)\, dt$

$$\therefore \boxed{S(t) = A\cos(\omega_c t + k_f \int x(t)\, dt)}$$

Now with the phase angle of unmodulated carrier is taken at $t=0$, then the limits of integration in the above integration will be 0 to $t$

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

(For Multitone)

which is the required expression for freq. modulated wave.

Frequency Deviation ($\Delta\omega$) :-

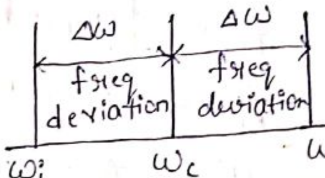
In FM, the instantaneous frequency is given by unmodulated frequency ~~and~~ plus a time varying component which is proportional to $x(t)$.

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

This instantaneous frequency '$\omega_i$' varies with respect to time according to the message signal $x(t)$. The maximum change in instantaneous frequency from the average frequency (or) carrier frequency is called as frequency deviation. and it depends on the magnitude and the sign of $k_f\, x(t)$. This means that the frequency deviation would be either +ve or -ve and it depends on the sign of $k_f\, x(t)$. However, the amount of frequency deviation in both the cases depends on maximum magnitude of $k_f\, x(t)$.

Frequency deviation is denoted by $\Delta\omega$.

$$\boxed{\Delta\omega = |k_f\, x(t)|_{max}}$$




Difference b/w phase modulation and frequency modulation.

We know that, the angle modulated wave is given by $s(t)$.

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

where $\phi_i$ is the total phase angle.

III'ly, phase modulated and frequency modulated o/p's are expressed as

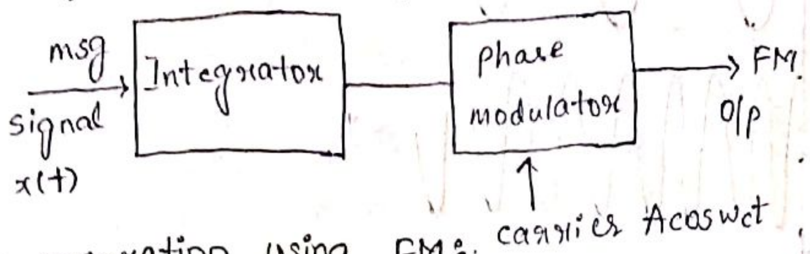
$$s(t) = A\cos[\omega_c t + k_p\, x(t)] \quad \text{- For PM}$$

$$s(t) = A\cos\left[\omega_c t + k_f\int_0^t x(t)\, dt\right] \quad \text{- For FM}$$

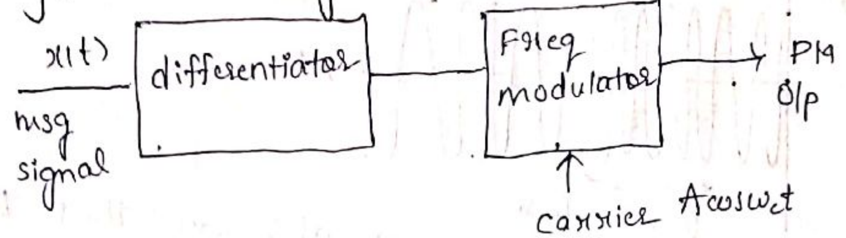
From the above equations, it may be noted that PM and FM waves are closely related to each other because in both the cases the total phase angle changes with respect to message signal.

In PM, the phase angle changes linearly with the message signal $x(t)$ whereas in FM, the phase angle varies linearly with the integral of message signal $x(t)$. So, a PM wave can be generated by using FM and an FM wave can be generated by using PM.

FM generation using PM :-




PM generation using FM :-




Modulation Index in FM :-
Modulation Index in FM is defined as the ratio of freq. deviation to the maximum frequency of message signal. It is denoted by $m_f$.

$$m_f = \frac{\text{freq. deviation}}{\text{freq. of msg sgl}}$$

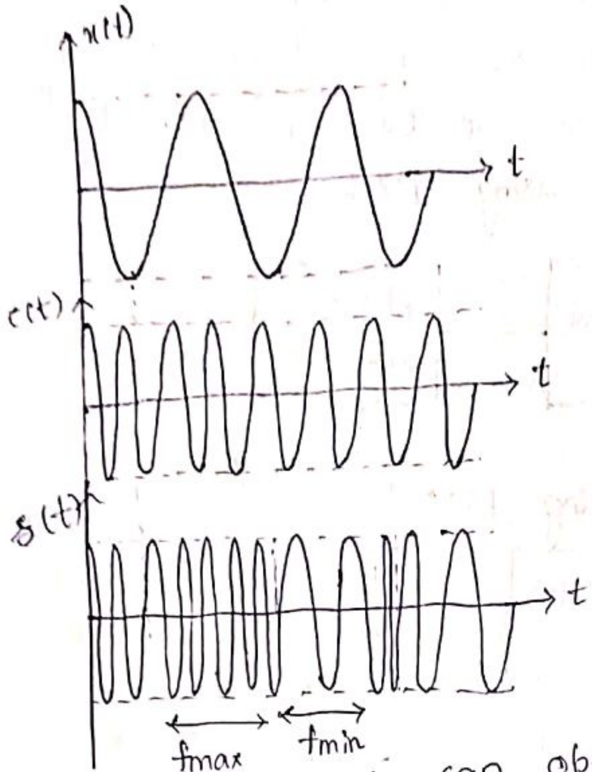
$$\boxed{m_f = \frac{\Delta\omega}{\omega_m}} \quad (\text{or}) \quad m_f = \frac{\Delta f}{f_m}$$

Percentage of modulation in FM :-
In FM, the percentage of modulation is the ratio of actual freq. deviation to the maximum allowable freq. deviation

$$\boxed{\%\text{ of mod} = \frac{\Delta f_{actual}}{\Delta f_{allowable}} \times 100}$$

* Single tone frequency modulation :-
Let us consider a carrier signal, $c(t) = A\cos\omega_c t$ and modulating signal $x(t) = V_m\cos\omega_m t$.
Now according to the definition of frequency modulation the frequency of the carrier signal is changed

according to the instantaneous value of message signal. The figure below illustrates the process of frequency modulation.




From the above figure, we can observe that the frequency of the carrier signal shifts up and down from the center frequency / carrier frequency is $f_c$. This change (or) shift in carrier frequency is called as frequency deviation. and it is denoted by $\Delta\omega$. The frequency deviation always depends on the amount of amplitude of message signal which means that louder the music, greater the frequency deviation.

Mathematical Expression:

Let us consider the standard expression for angle modulated wave $s(t) = A\cos\phi$ —① where $\phi$ is the total phase angle $= \int \omega_i \, dt$ —②

here $\omega_i = \omega_c + k_f \, x(t)$

$$= \omega_c + V_m \cos\omega_m t \; k_f$$

WKT, frequency deviation $\Delta\omega = |k_f \, x(t)|_{max}$

$$\omega_i = \omega_c + \Delta\omega \cos\omega_m t$$

substitute '$\omega_i$' in eq ②

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

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

$\therefore \frac{\Delta\omega}{\omega_m} = m_f / \beta$.

$\phi = \omega_c t + m_f \sin\omega_m t$ —③

sub. ③ in eq ①.

$\boxed{s(t) = A\cos(\omega_c t + m_f \sin\omega_m t)}$ (or) $s(t) = A\cos(\omega_c t + \beta \sin\omega_m t)$

1) A single tone FM is represented by the voltage eq$^n$ as $v(t) = 12\cos(6\times10^8 t + 5\sin 1250t)$ then find out the following:
i) Carrier freq. ii) Modulating freq. iii) Modulation index iv) maximum deviation v) what amount of power will this FM wave dissipate in 10 Ω resistor?

Given,
$v(t) = 12\cos(6\times10^8 t + 5\sin 1250t)$
The standard equation for single tone FM is
$s(t) = A\cos(\omega_c t + m_f \sin\omega_m t)$
Carrier frequency $\omega_c = 6\times10^8$ $f_c = 955$ MHz
Modulating frequency $\omega_m = 1250$ $f_m = 198.9$ Hz.
Modulation index $m_f = 5$

WKT, $m_f = \frac{\Delta\omega}{\omega_m} = \frac{\Delta f}{f_m}$.

$\Delta f = m_f \cdot f_m = 5\times198.9 = 994.5$

$P = \frac{V_{rms}^2}{R} = \frac{\left(\frac{12}{\sqrt{2}}\right)^2}{10} = \frac{72}{10} = 7.2$ W

Types of FM:- We know that, the band width of an FM signal depends on frequency deviation i.e., if frequency deviation is high, band width will be large. similarly, if frequency deviation is low, the bandwidth is small.
Since, deviation is given by $\Delta\omega = |k_f\, x(t)|_{max}$, the band width will depend upon frequency sensitivity $k_f$. Hence when $k_f$ is quite small, the bandwidth will be narrow and when $k_f$ is large, the band width will be wide. Thus, depending upon the value of frequency sensitivity, FM may be classified as
i) Narrow band FM

ii) wide band FM

Narrow band FM :-

For $k_f$ value less than 1, the FM is treated as narrow band FM. For narrow band FM, the band width is very small which means a pass band is very less in narrow band FM.

Let us consider the standard expression for multitone FM signal is

$$s(t) = A\cos\left(\omega_c t + k_f\int x(t)\,dt\right)$$

$$s(t) = A\cos\omega_c t\cos\left(k_f\int x(t)\,dt\right) - \sin\omega_c t\,\sin\left(k_f\int x(t)\,dt\right)$$

For $k_f < 1$, we can assume to realistic approximations for the simplicity of equation

$$\begin{cases} \cos\left(k_f\int x(t)\,dt\right) \simeq 1 ; & \because \cos\theta \simeq 1, \text{ for } \theta < 1, \\ \sin\left(k_f\int x(t)\,dt\right) \simeq k_f\int x(t)\,dt & \because \sin\theta \simeq \theta, \text{ for } \theta < 1 \end{cases}$$

$$\therefore s(t) = A\cos\omega_c t - A\sin\omega_c t\; k_f\int x(t)\,dt$$

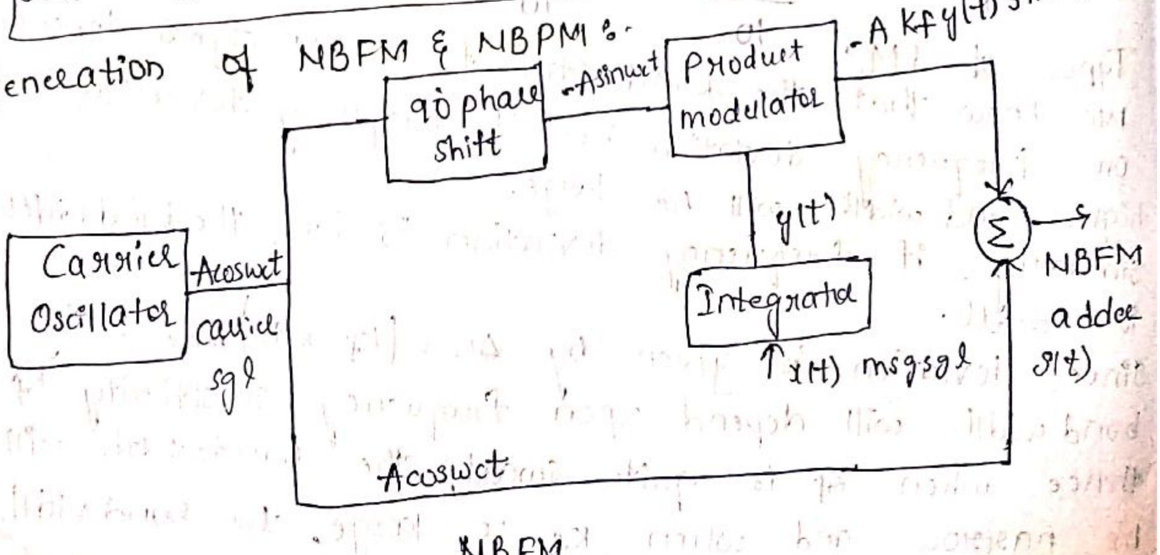
Let $\int x(t)\,dt = y(t)$

$$\boxed{s(t) = A\cos\omega_c t - A k_f\, y(t)\sin\omega_c t}$$ , which is the required standard expression for narrow band FM.

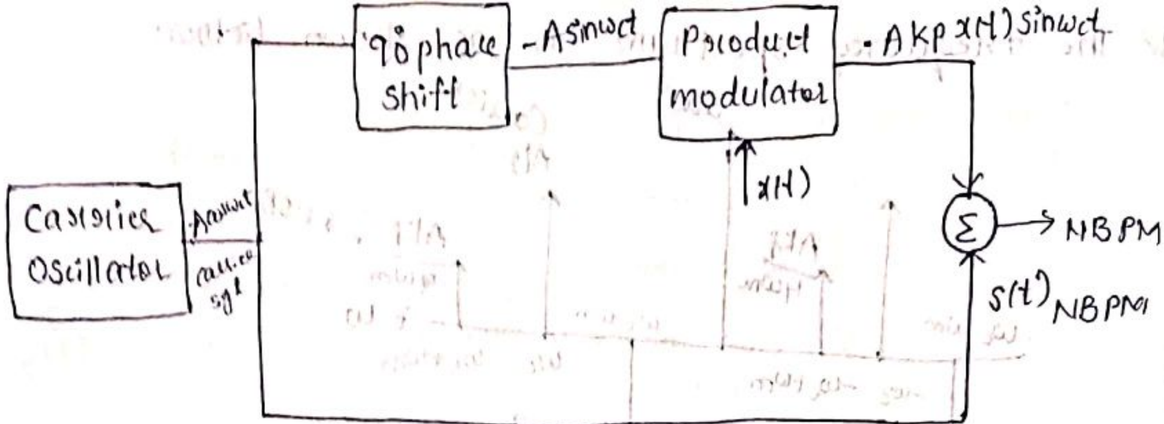
Similarly, for narrow band phase modulation the standard expression can be written as

$$\boxed{s(t) = A\cos\omega_c t - A k_p\, x(t)\sin\omega_c t}$$

Generation of NBFM & NBPM :-




NBFM




Frequency Spectrum of NBFM :-
Let us consider the standard expression for NBFM.

$s(t) = A\cos\omega_c t - A k_f y(t)\sin\omega_c t$

here $y(t) = \int x(t)\,dt$

Let $x(t) = \sin\omega_m t$

$\therefore y(t) = \dfrac{-\cos\omega_m t}{\omega_m}$

$s(t) = A\cos\omega_c t + \dfrac{Ak_f}{\omega_m}\cos\omega_m t\sin\omega_c t$

$= A\cos\omega_c t + \dfrac{Ak_f}{2\omega_m}\left[\sin(\omega_c+\omega_m)t + \sin(\omega_c-\omega_m)t\right]$

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

$y(t) = \dfrac{\sin\omega_m t}{\omega_m}$

$s(t) = A\cos\omega_c t - \dfrac{Ak_f}{\omega_m}\sin\omega_m t\sin\omega_c t$

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

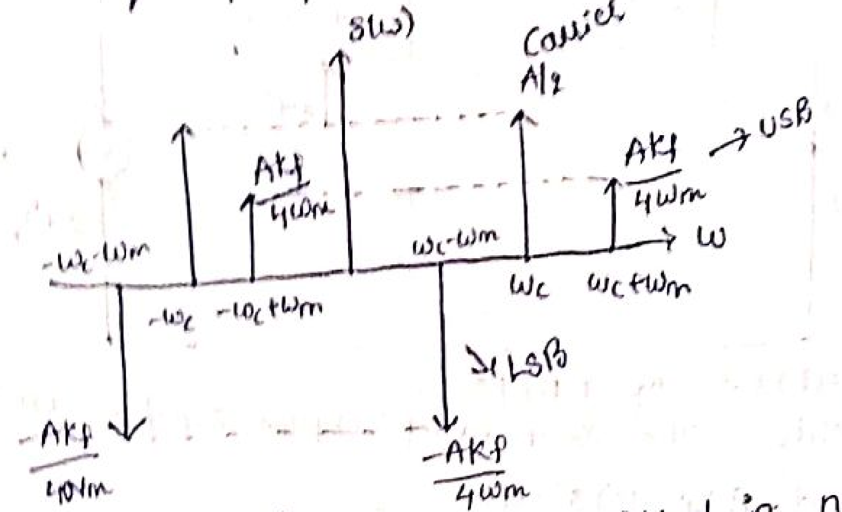
$= \underbrace{A\cos\omega_c t}_{\text{Carrier}} - \underbrace{\dfrac{Ak_f}{2\omega_m}\cos(\omega_c-\omega_m)t}_{\text{LSB}} + \underbrace{\dfrac{Ak_f}{2\omega_m}\cos(\omega_c+\omega_m)t}_{\text{USB}}$

The equation contains 3 components.
~~The first component~~
Apply F.T on b-s

$s(\omega) = \dfrac{A}{2}\left[\delta(\omega-\omega_c) + \delta(\omega+\omega_c)\right] - \dfrac{Ak_f}{4\omega_m}\left[\delta(\omega-(\omega_c-\omega_m)) + \delta(\omega+(\omega_c-\omega_m))\right]$

$+ \dfrac{Ak_f}{4\omega_m}\left[\delta(\omega-(\omega_c+\omega_m)) + \delta(\omega+(\omega_c+\omega_m))\right]$

(iv) The frequency spectrum is as shown below:




* The amount of power transmitted in narrow band FM is $P_t = P_c + P_s = P_c + P_{LSB} + P_{USB}$

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

* The band width requirement in narrow band FM is
The band width requirement of DSB-FC AM system.
$2\omega_m$ = band width requirement with AM, NBFM is
Because of its much similarity
given least preference compared to WBFM.

Wide band frequency Modulation (Spectrum analysis of WBFM using Bessels function)
For $K_f \gg 1$, (or) $\beta > 1$ the modulation is treated as a sinusoidal wave in wide band frequency Modulation. In wide band FM, the band width is very wide.
In order to find out, the frequency spectrum of WBFM first we need to know about the Bessels function and its properties.

The Bessels function is given by

$$J_n(x) = \frac{1}{2\pi}\int_{-\pi}^{\pi} e^{J(x\sin\theta - n\theta)}\, d\theta$$

Properties :-

1. $J_n(x)$ decreases as n value increases

Eg. $J_0(x) > J_1(x) > J_2(x) > \ldots$

2. $J_{(-n)}(x) = (-1)^n J_n(x)$

i.e., $J_{-n}(x) = J_n(x)$, when $n$ = even
$= -J_n(x)$, when $n$ = odd

5) $\sum_{n=-\infty}^{\infty} J_n^2(x) = 1$

Let us derive the standard expression for NBFM wave.

WKT, the standard expression for single tone FM wave is given by $s(t) = A\cos(\omega_c t + m_f \sin\omega_m t)$ —(1)

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

$$= A\,Re\left[e^{J(2\pi f_c t + \beta\sin 2\pi f_m t)}\right]$$

$$= A\,Re\left[e^{J2\pi f_c t} \cdot e^{J\beta\sin 2\pi f_m t}\right]$$

Let $e^{J\beta\sin 2\pi f_m t} = f(t)$, is a periodic sequence having time period $\frac{1}{f_m}$. So, this function can be expressed in exponential fourier series.

$$f(t) = e^{J\beta \sin 2\pi f_m t} = \sum_{n=-\infty}^{\infty} C_n e^{Jn\omega_0 t} \quad \text{—(2)} \qquad \omega_0 = 2\pi f_m$$

where $C_n = \frac{1}{T}\int_{-T/2}^{T/2} f(t)\, e^{-Jn\omega_0 t}\, dt$

$$C_n = \frac{1}{(1/f_m)}\int_{-T/2}^{T/2} e^{J\beta\sin 2\pi f_m t} \cdot e^{-Jn\omega_0 t}\, dt$$

$$= f_m \int_{-1/2f_m}^{1/2f_m} e^{J(\beta\sin 2\pi f_m t - n 2\pi f_m t)}\, dt$$

Assume $2\pi f_m t = \theta \quad \therefore \frac{d\theta}{dt} = 2\pi f_m$

$t = -1/2f_m \Rightarrow 2\pi f_m t = 2\pi f_m \cdot \frac{-1}{2f_m} = -\pi$

$t = 1/2f_m \Rightarrow \theta = \pi$

$$\therefore C_n = f_m \int_{-\pi}^{\pi} e^{J(\beta\sin\theta - n\theta)} \frac{d\theta}{2\pi f_m}$$

$$= \frac{1}{2\pi}\int_{-\pi}^{\pi} e^{J(\beta\sin\theta - n\theta)}\, d\theta$$

$$C_n = J_n(\beta) \quad \text{—(3)}$$

③ in ⓧ

$$f(t) = \sum_{n=-\infty}^{\infty} J_n\beta\, e^{Jn2\pi f_m t}$$

substitute this eq. in eq ①

$$s(t) = A\,Re\left[e^{J2\pi f_c t} \cdot \sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{Jn2\pi f_m t}\right]$$

$$= A\,Re\left[\sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{J2\pi(f_c+nf_m)t}\right]$$

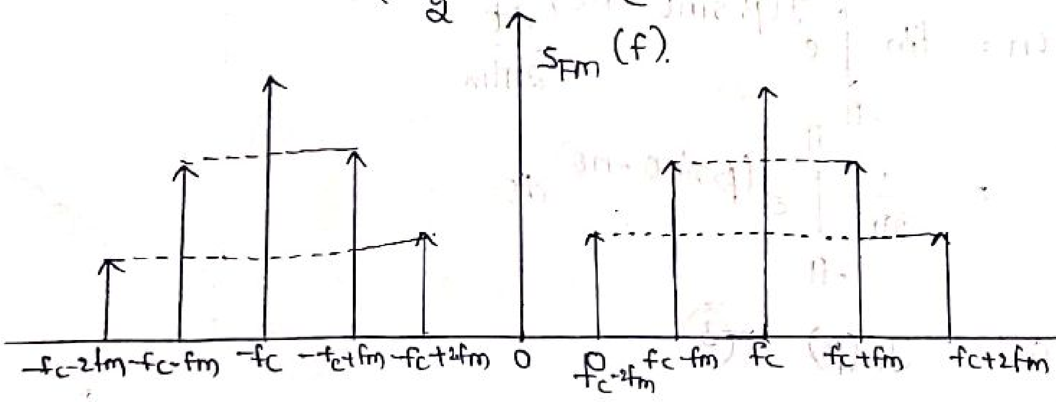
$$\boxed{s(t) = A \sum_{n=-\infty}^{\infty} J_n(\beta) \cos 2\pi(f_c+nf_m)t}$$ — KIBFM standard exprsn

Spectrum :-
By giving the values of 'n' from $[-2,2]$ the above eq becomes

$$S(t)_{FM} = A_c\, j_0(B) \cos 2\pi f_c t + j_1(B)\cos 2\pi(f_c+f_m)t$$
$$+ j_{-1}(B)\cos 2\pi(f_c-f_m)t + j_2\,B \cos 2\pi(f_c+2f_m)t$$
$$+ j_{-2}(B)\cos 2\pi(f_c-2f_m)t.$$

Taking fourier transform on both sides,

$$S(f)_{FM} = \frac{A_c}{2} j_0(B)\left[\delta(f-f_c)+\delta(f+f_c)\right] +$$
$$\frac{A_c}{2} j_1(B)\,\delta[f-(f_c+f_m)+\delta(f+(f_c+f_m))] +$$
$$\frac{A_c}{2} j_{-1}(B)\,\delta[f-(f_c-f_m)+\delta(f+f_c+f_m)] + \ldots$$
$$\ldots + \frac{A_c}{2} j_{-2}(B)\,\delta[f-(f_c+2f_m)+\delta(f+(f_c+2f_m))]$$




spectrum of FM.

Scanned by CamScanner

Carson's rule (or) Practical B.W of FM:-

Carson's rule provides a formula to calculate the band width of a single tone FM. According to this rule, the band width of an FM sgl is twice the sum of frequency deviation and the highest modulating frequency. However, it must be remembered that this rule gives the approximation values of bandwidth

$$B.W = 2(\Delta\omega + \omega_m)$$

WKT, $m_f = \frac{\Delta\omega}{\omega_m} \Rightarrow \Delta\omega = m_f \omega_m$

$$B.W = 2(m_f\omega_m + \omega_m) = 2\omega_m(m_f + 1)$$

we can analyse the above equation in two cases.

Case-(1):- For $m_f << 1$ i.e., the case for narrow band FM. we can neglect $m_f$.

$$\boxed{B.W_{NBFM} = 2\omega_m}$$

Case-(2):- For $m_f >> 1$ i.e., the case for wide band FM.

$$\boxed{B.W_{WBFM} = 2\omega_m m_f = 2\Delta\omega}$$

Ex:- Find the bandwidth of a commercial FM transmission if $\Delta\omega / \Delta f = 75KHZ$ and the modulating freq $f_m = 15KHZ$

Given, $\Delta f = 75KHZ$ ; $f_m = 15KHZ$

Bw of a FM signal is given by $2(\Delta f + f_m)$

$$B.W = 2(\Delta\omega + \omega_m) = 2(75 + 15) \times 10^3$$

$$= 2(90) \times 10^3 = 180 \times 10^3$$

$$B.W = 180KHZ.$$

Find out the B.W of a NBFM sgl which is generated by a 4KHZ audio sgl modulating a 125MHZ sgl

Given,

$f_m = 4KHZ.$

$f_c = 125MHZ$

$$B.W_{NBFM} = 2\omega_m = 2f_m = 8KHZ.$$

Generation of FM :-

It can be done in 2 ways.

1. Direct Method
2. Indirect method

Direct Method :-

In Direct Method, the FM sgl is generated by using a device called VCO (Voltage control Oscillator) in which the frequency of an oscillator is directly controlled by the msg. sgl $x(t)$. i.e., VCO produces an o/p sgl which is proportional to i/p signal voltage


fig (a)

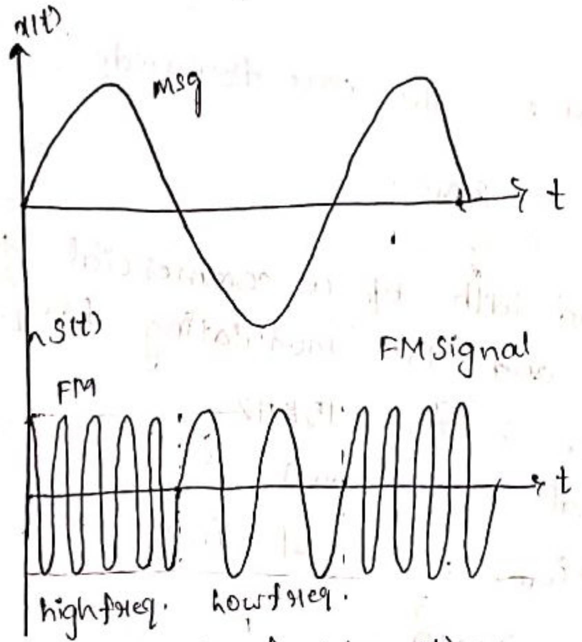
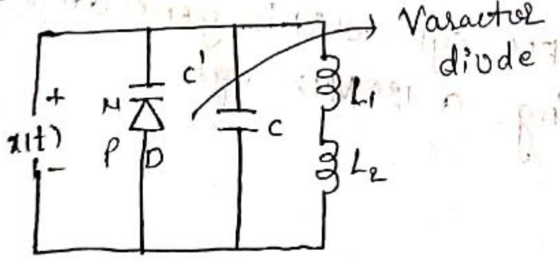



fig (b)

$f_i = f_c + K_f x(t)$

instantaneous freq

Internal structure of VCO :-




$\omega \rightarrow$ width of depletion region

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

here $C' \propto \frac{1}{\omega}$ $\because C' = \frac{\epsilon_0 A}{\omega_0 (\text{or}) d}$

Operation :-

During +ve half cycle of $x(t)$ → Diode is in F.B

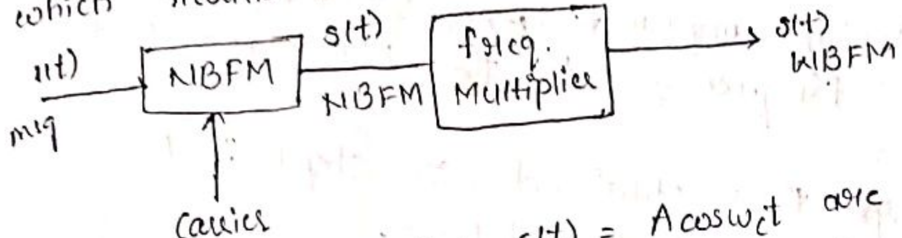
F.B ↑ ω ↓ c' ↑ $f_i$ ↓

During -ve half cycle of $x(t)$ → diode is in RB

RB ↑ ω ↑ c' ↓ $f_i$ ↑

InDirect Method :-

In this method, first a NBFM wave is generated and this NBFM wave is passed through a frequency multiplier ckt to increase the frequency deviation which results in a wide band FM sgl.

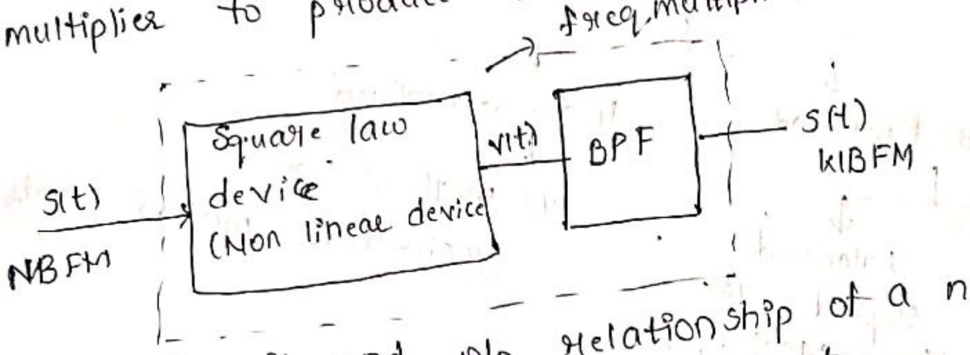



Let $x(t) = V_m\cos\omega_m t$ : $c(t) = A\cos\omega_c t$ are applied as an inputs to NBFM modulator ckt. So, the o/p of NBFM modulator is

The instantaneous freq. of the sgl is $f_i = f_c + k_f\, x(t)$

$s(t) = A\cos(\omega_c t + m_f \sin\omega_m t)$ — ①

Now this sgl is allowed to pass through a frequency multiplier to produce desired WBFM sgl.




Here, the i/p and o/p relationship of a non linear device in general form is given by

$v(t) = a_1 s(t) + a_2 s^2(t) + a_3 s^3(t) + \cdots + a_n s^n(t)$ — ②

where $a_1, a_2 \cdots a_n$ are coefficients and 'n' is the highest order of non-linearity.

By substituting eq ① in eq ② and by simplifying the frequency modulated wave having carrier frequencies $f_1, 2f_1, 3f_1 \cdots n f_1$ with frequency deviations $\Delta f_1, 2\Delta f_1, 3\Delta f_1 \cdots n\Delta f_1$ the output of frequency multiplier produces the desired WBFM having the following time domain equation

$$s(t)_{WBFM} = A\cos(n\omega_c t + n m_f \sin\omega_m t)$$

By using proper mixer circuit, the centre frequency of resulting WBFM signal will be changed from $nf_c$ to $f_c$ only.

## Demodulation of FM (or) detection of FM :

In the process of demodulation (or) detection of FM, the original message signal is recovered from the FM wave. FM detection is carried out in two steps.

1. A signal having amplitude proportional to the instantaneous frequency of the FM wave is first obtained.
2. The o/p sgl thus obtained in step 1 is then detected by using diode detector.

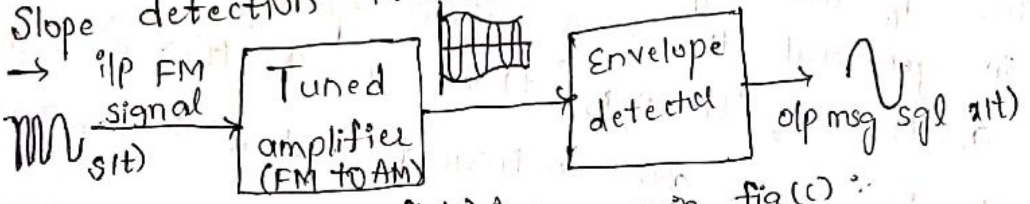
## Types of demodulation :

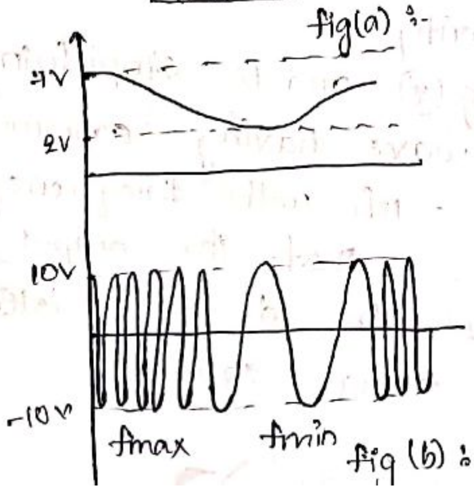
FM detection can be done in 2 ways.

1. Direct Method (freq discrimination method)
2. Indirect method (phase discrimination method)

FM detection
- Direct method
  - slope detection
  - balanced slope detection
- Indirect method
  - PLL method (or) Synchronous method

## Slope detection Method :

→ 

fig(a) :-

fig (b) :-

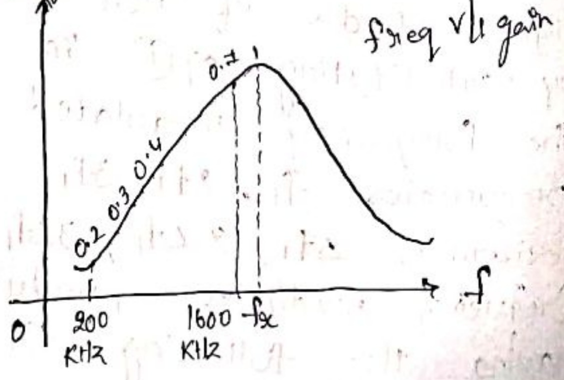
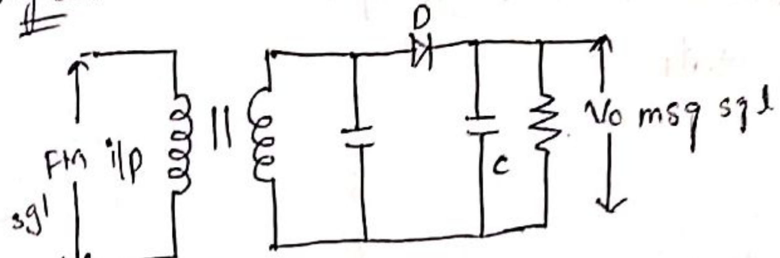


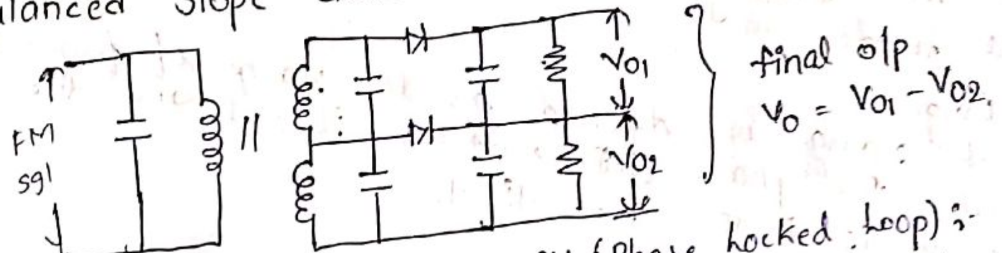
fig (c) :-

fig (d) :-




Tuned amplifier :-
As f becomes closer to fr → gain value ↑ → o/p voltage ↑.
As f moves away to fr → gain value ↓ → o/p voltage ↓
→ To increase the limited range of values, we go for balanced slope detection.

Balanced slope detection :-




Demodulation of FM using PLL (Phase Locked Loop) :-
In synchronous detection method PLL is used for removing phase error.
A PLL is a non-linear feedback system that tracks the phase of input signal and minimize the phase error of carrier at local oscillator in receiver section.
PLL is used in synchronous demodulation (or) coherent demodulation of FM wave.

fig (a) General structure of synchronous detection

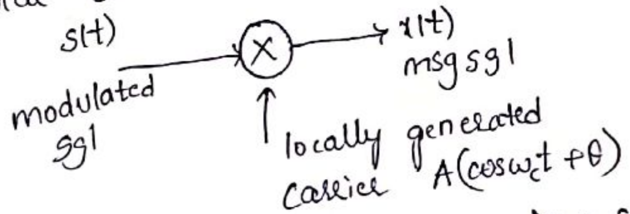
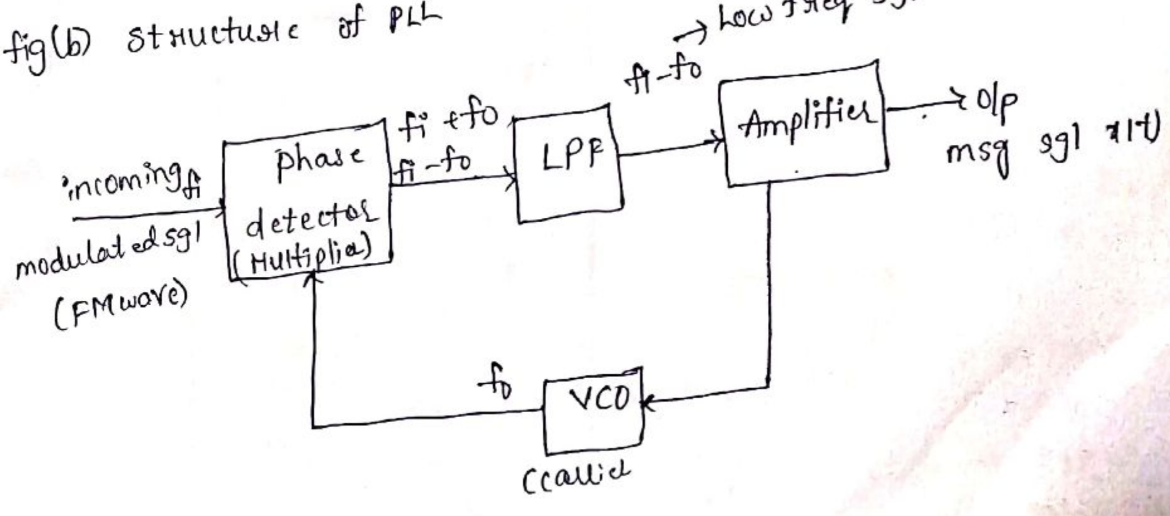



fig (b) structure of PLL



The circuit operates in 3 regions/states.

1) Free running state
2) Capture Range
3) Lock Range

Free running state :-
If no i/p sgl is applied then PLL is in free running state

Capture Range :-
Once the i/p frequency is applied the VCO frequency starts to change and PLL is said to be in capture mode

Lock Range :-
The lock range is defined as the range of frequency over which the phase difference b/w i/p sgl and the carrier sgl is zero. i.e., $f_i - f_o = 0$

# Transmitters and Recievers

## AM Transmitter

AM transmitter are classified as
* Low power level AM transmitter
* High power level AM tansmitter.

## Low-power level AM transmitter

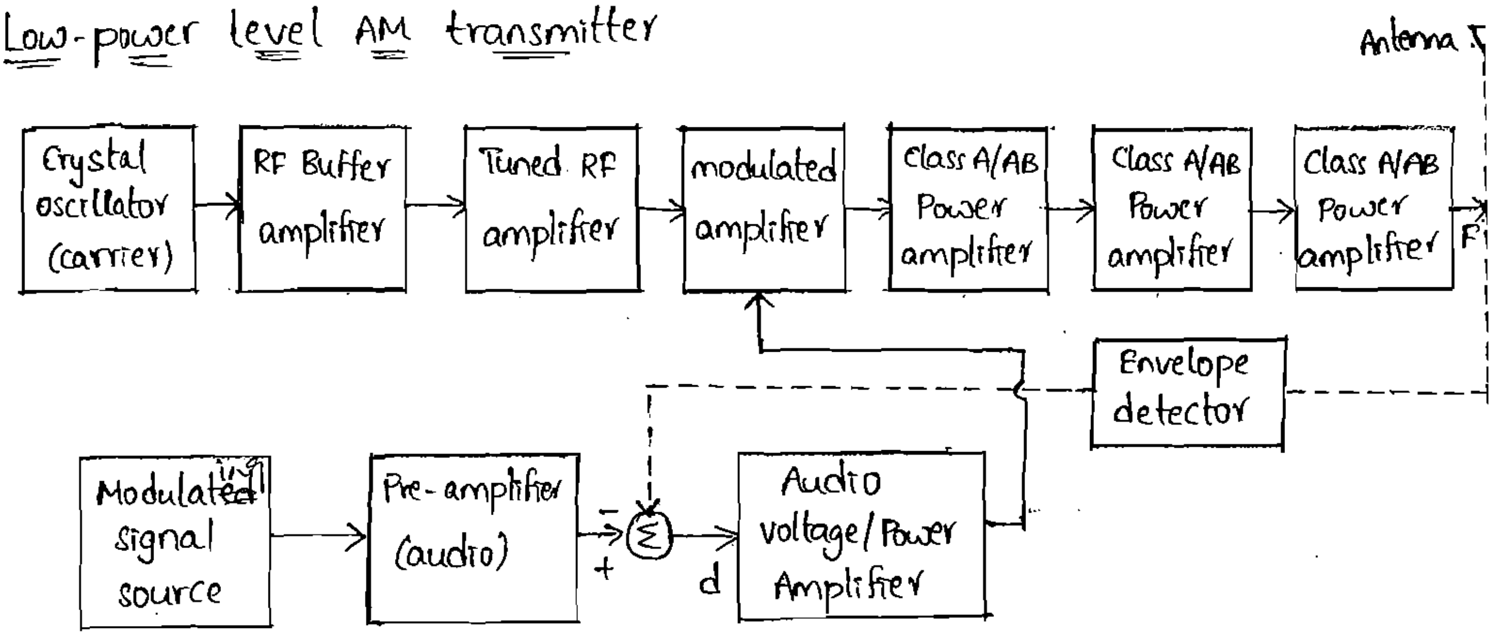



Fig : AM Transmitter with low-level

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

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

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

High power level AM transmitter

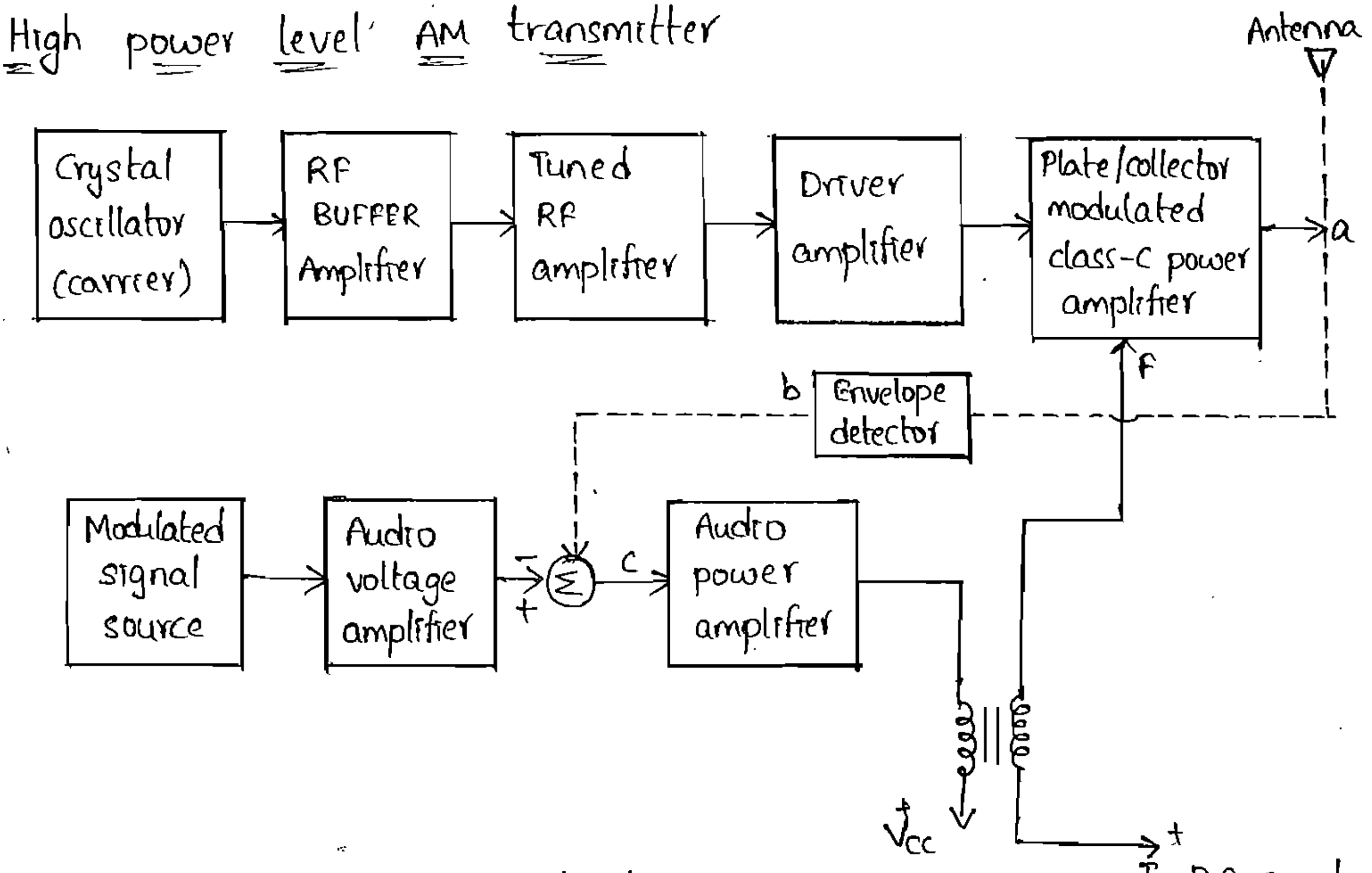



fig : High level modulator

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

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

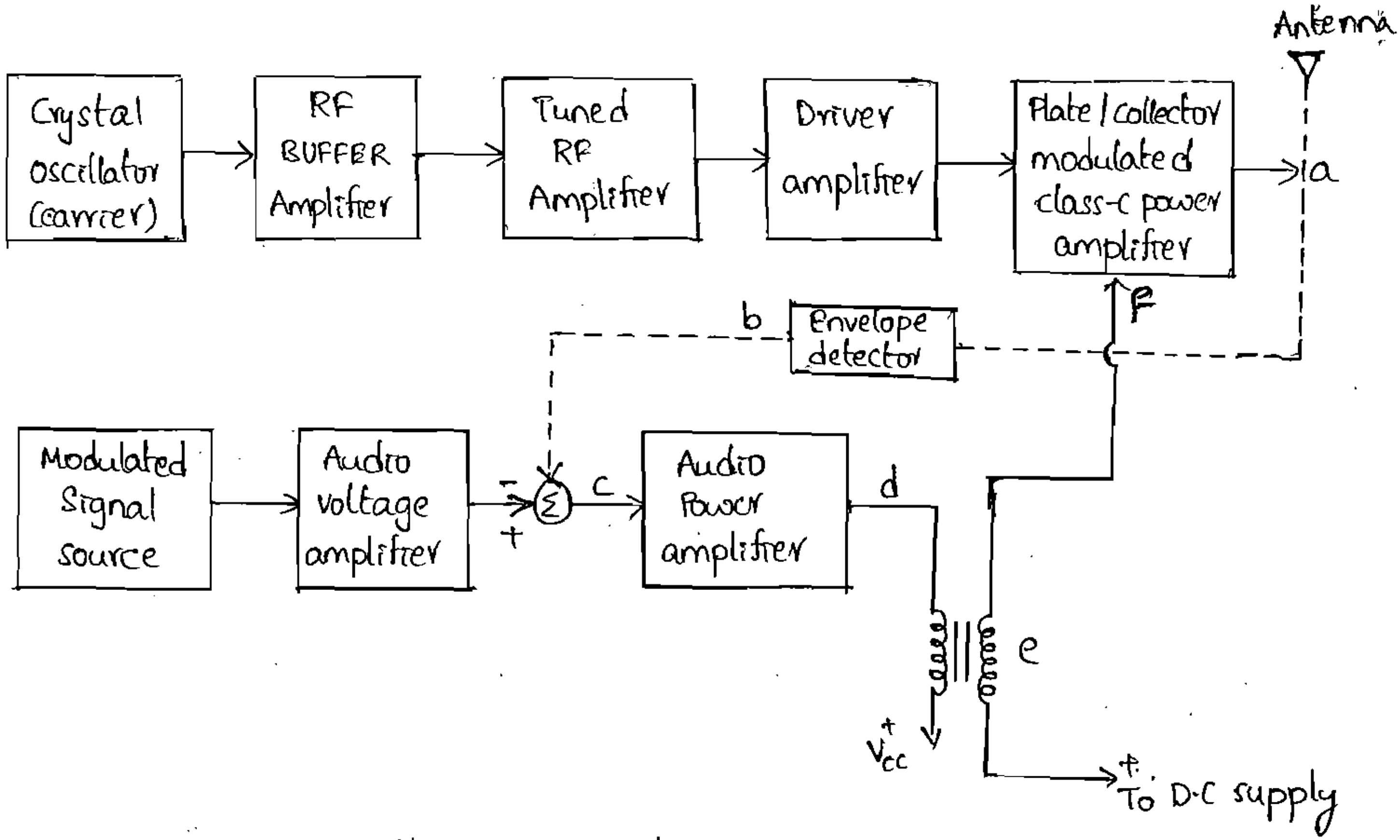
An RF oscillator and its circuit are similar to that of a low-level transmitter. The carrier signal also requires an additional power amplifier before it is given to the modulator. The final power amplifier is the actual modulator. Collector modulator class C type has a very good efficiency.

## Advantage

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

## Effect of Feedback on performance of AM transmitters

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




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

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

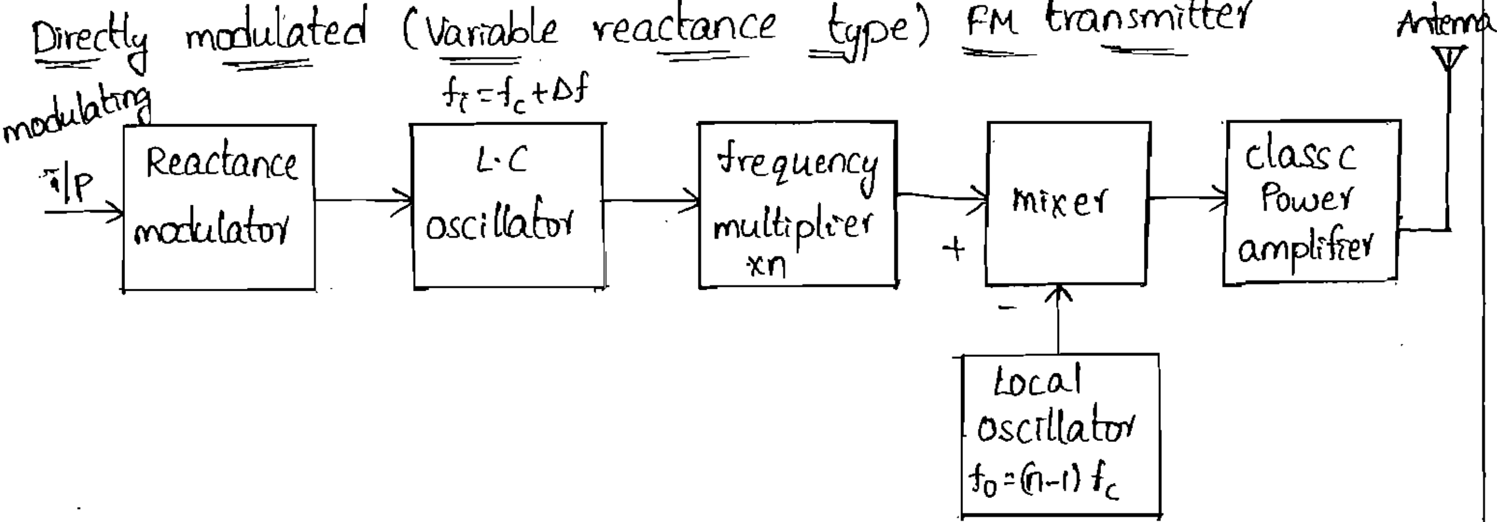
This negative feedback improves the performance of the transmitter as it reduces the distortion of the envelope of the radiated signal by making it closely resemble the message signal. It reduces the noise and power frequency also

→ FM transmitters

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

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

Directly modulated (Variable reactance type) FM transmitter



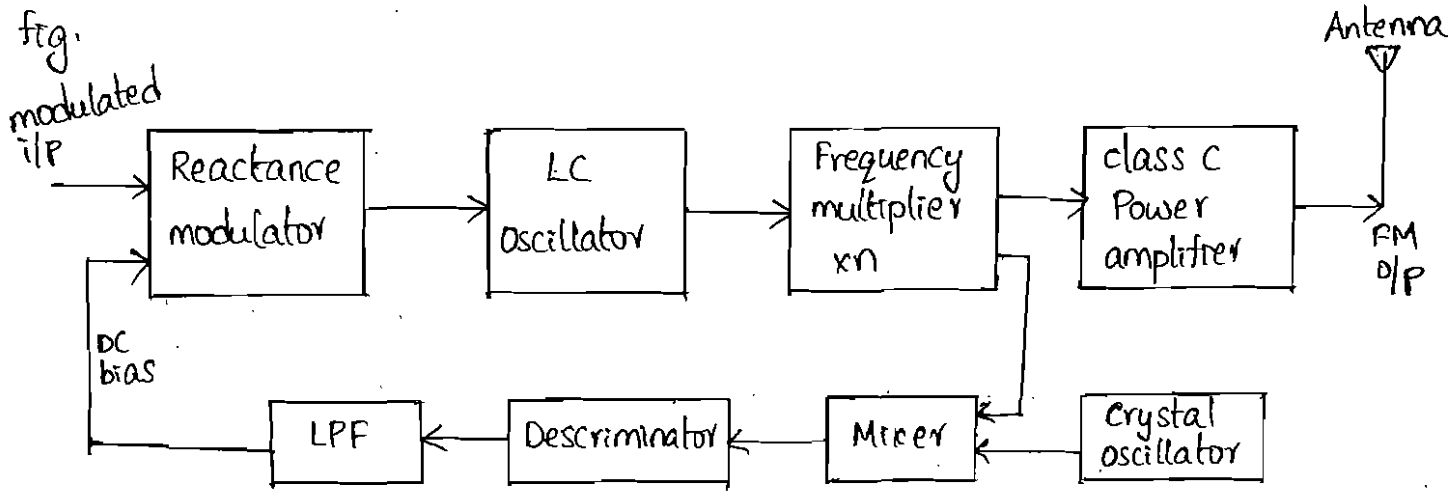

In directly modulated FM transmitters, the frequency modulation is carried out at a lower frequency and with a smaller frequency deviation. Then passing this frequency modulated wave through frequency multiplier circuit, the desired carrier frequency and desired frequency deviation is achieved.

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

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

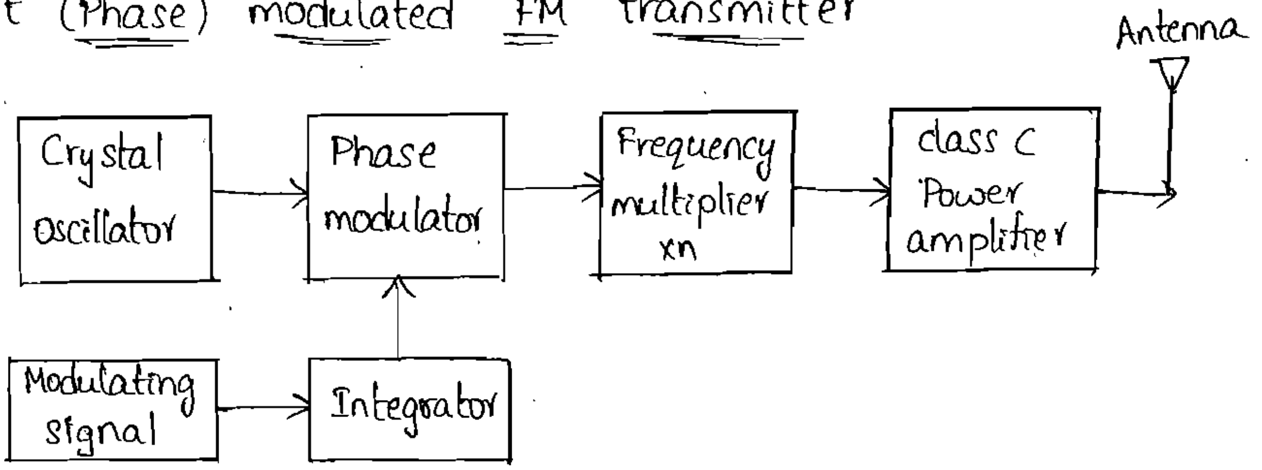
## Frequency stability using AFC

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



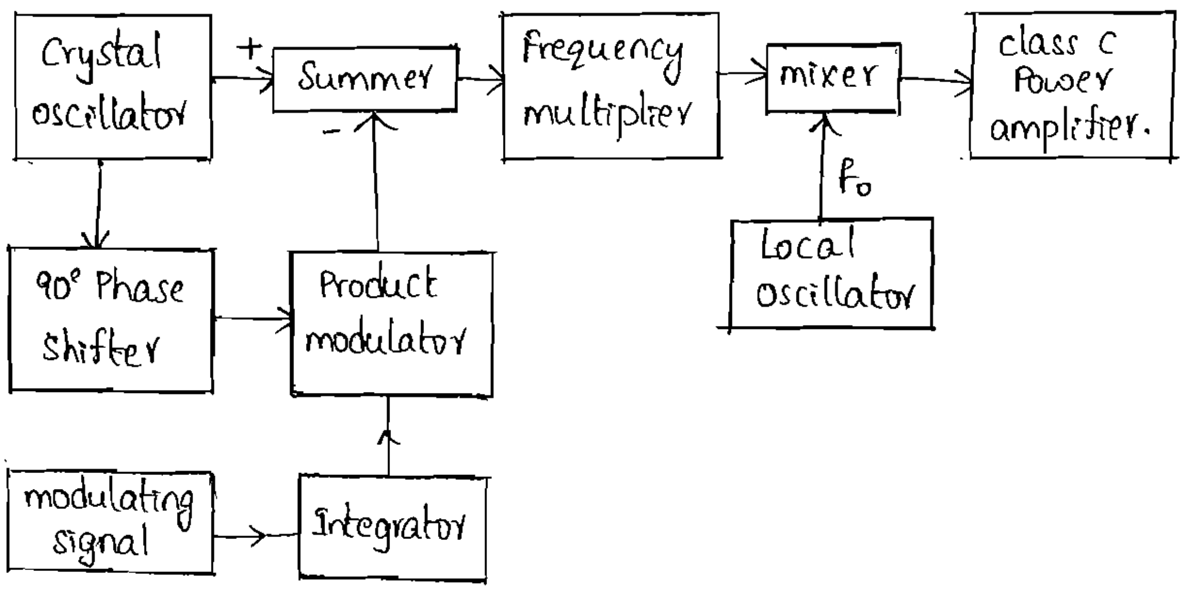

Suppose frequency of carrier increases. This higher frequency is fed to the mixer for which the other i/p frequency is from the stable crystal oscillator. A some what higher frequency will be fed to the descriminator. The descriminator will develop a positive dc voltage. The low pass filter (LPF) removes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

Indirect (Phase) modulated FM transmitter




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

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




FM transmitter: Armstrong method.

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

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

# RECEIVERS

## Introduction

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

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

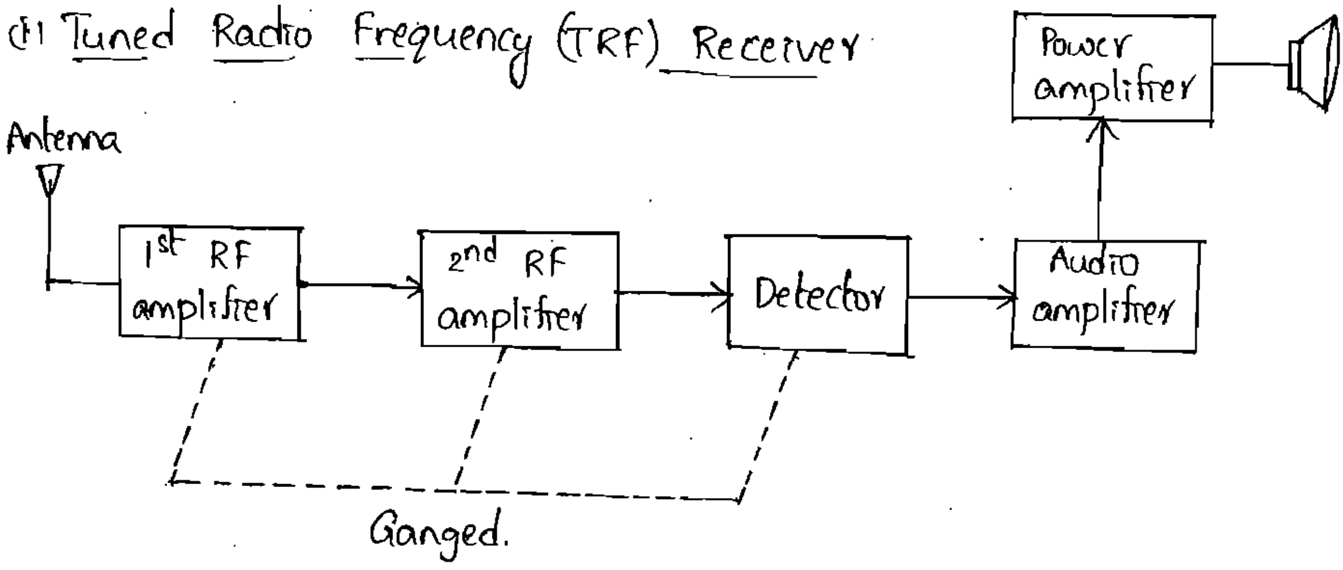
## Receiver types

1. AM Receiver    2. FM Receiver

## 1. AM Receiver

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

### (i) Tuned Radio Frequency (TRF) Receiver



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

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

Problems in TRF Receivers

1. Tracking of Tuned circuit

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

2. Instability

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

3. Variable Bandwidth

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

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

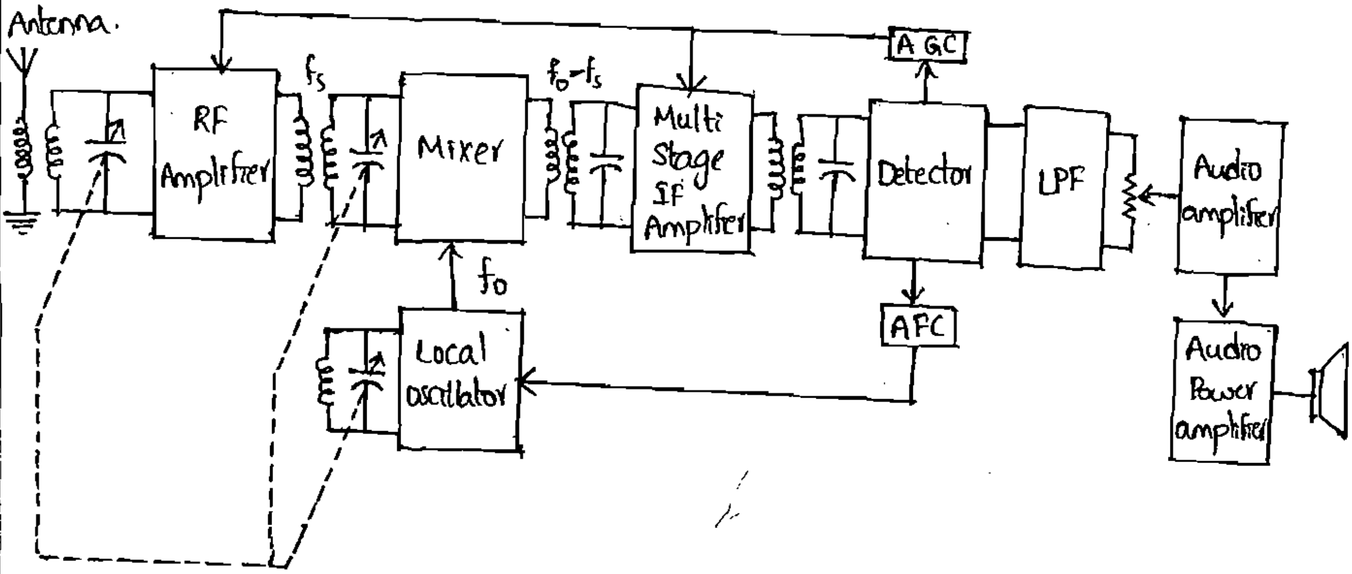
At 1640KHz Q of the coil should be 164 (1640k/10k)

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

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

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

## Superheterodyne Receivers




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

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

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

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

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

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

## Receiver characteristics

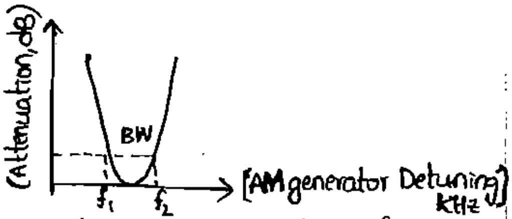
The performance of the radio receiver can be measured in terms of following receiver characteristics.

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

1. Selectivity

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

where $f_r$ is the resonant frequency




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

2. Sensitivity

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

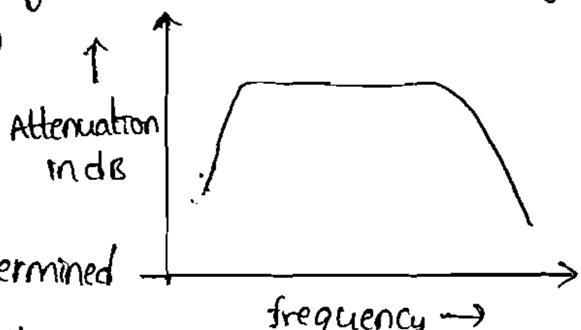
Fig. Sensitivity curve for typical receiver.


3. Fedility

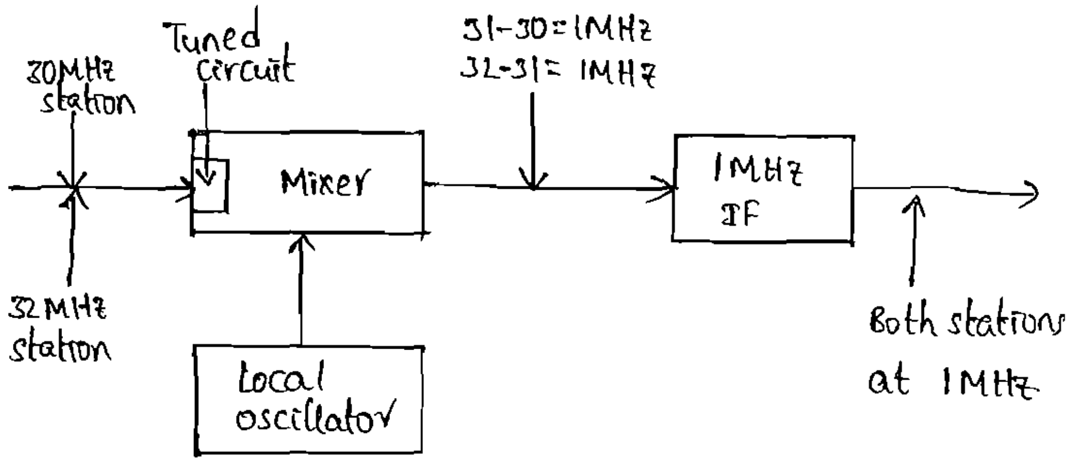
Fedility refers to the ability of the receiver to reproduce all the modulating frequencies equally. The below figure shows the typical fedility curve for radio receiver.

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

Fig: Typical Fedility curve




4. Image frequency and its Rejection



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

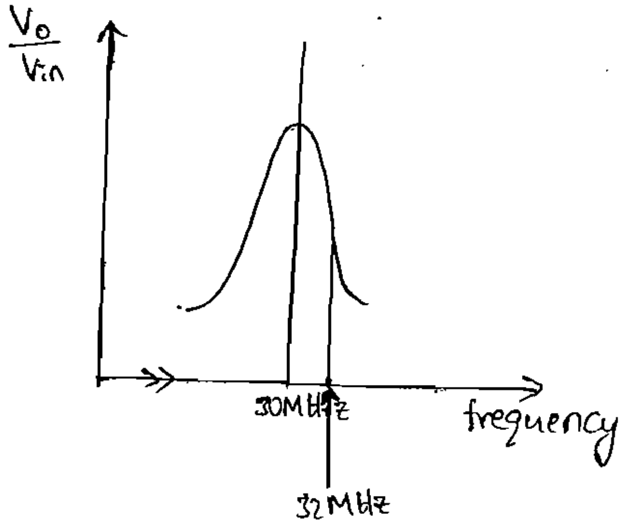



Fig: Response of the tuned.

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

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

5. Double spotting

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

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

## Receiver sections

### RF Amplifier

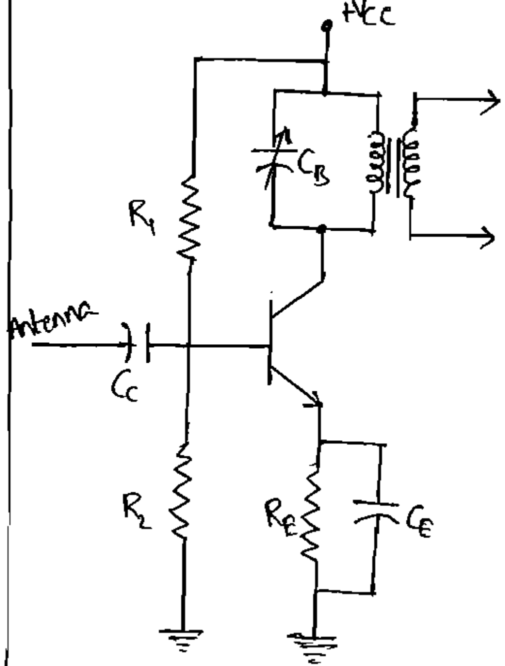



fig a.

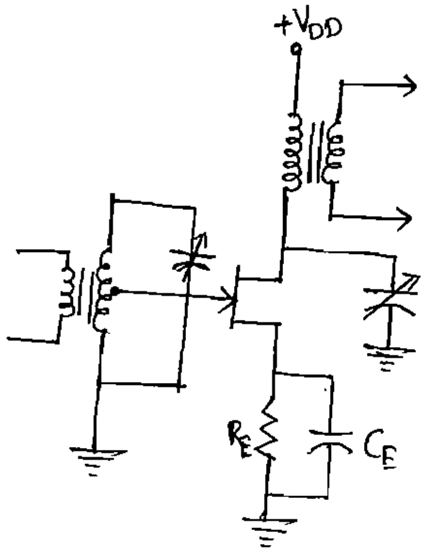



fig b.

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

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

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

The receiver having an RF amplifier stage has following advantages.

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

Mixer or frequency changer/ converter

Types
- Seperately Excited Mixer
- Self Excited Mixer.

(i) Seperately excited mixer

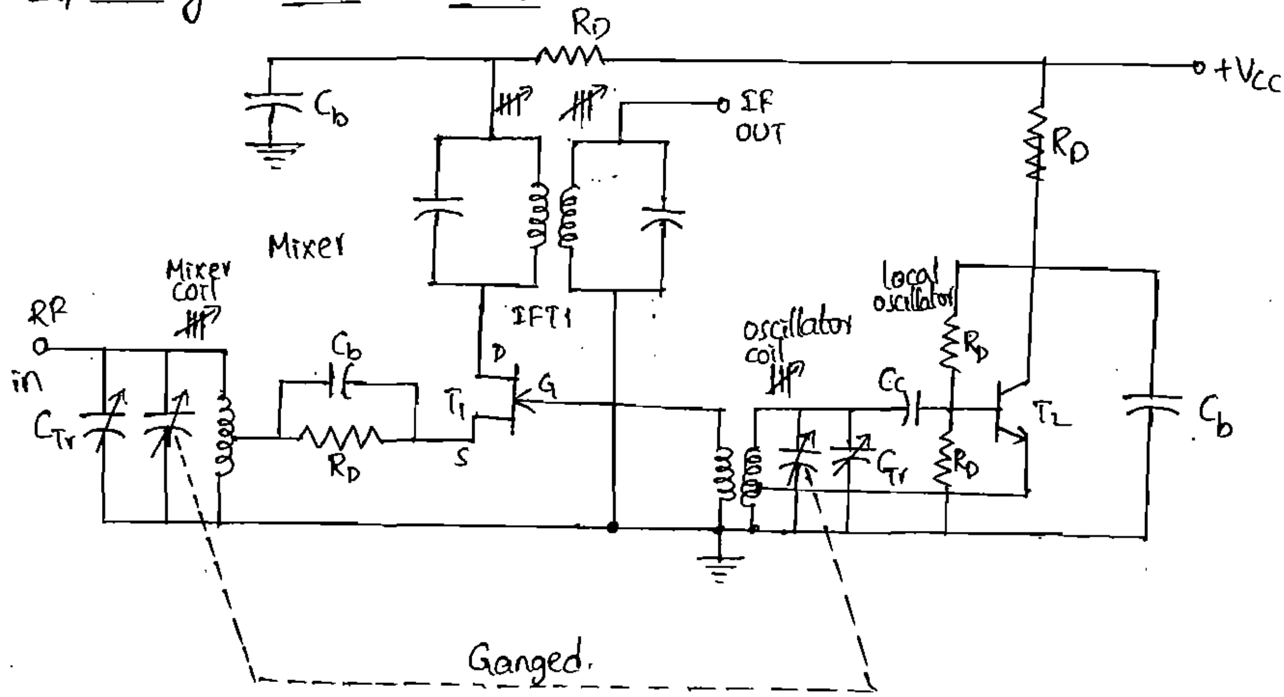



Fig: Seperately excited FET mixer.

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

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

(c) Self excited Mixer

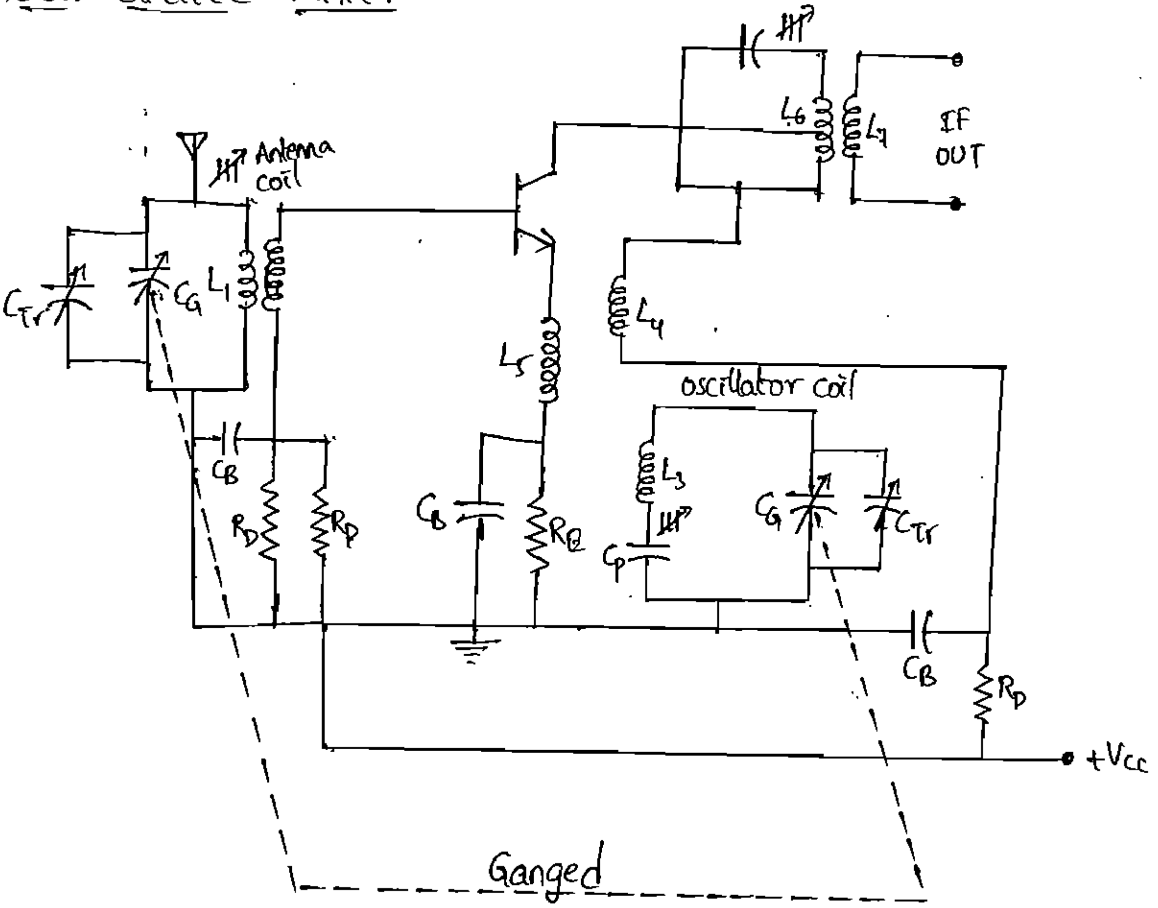



Fig: Self excited Mixer.

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

Tracking

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

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

1) Padder tracking

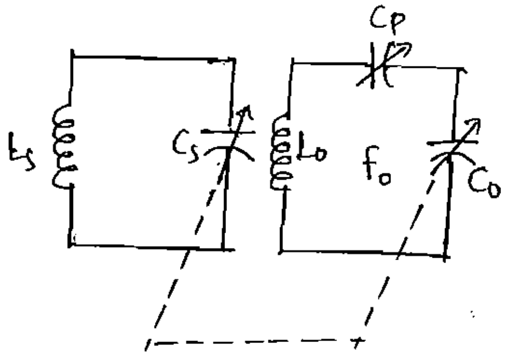



Fig: Padder tracking

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

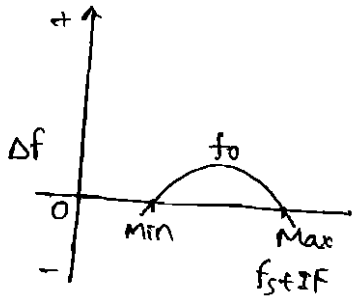



Fig: Tracking error in padder tracking.

2) Trimmer tracking

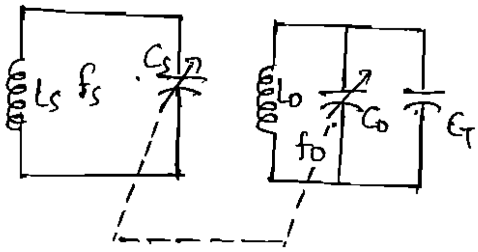



Fig: Trimmer tracking.

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

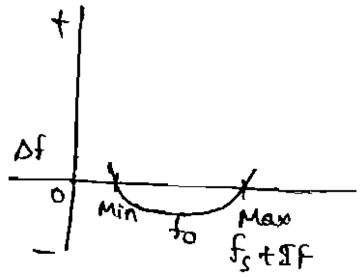



Fig: Tracking error in trimmer tracking.

3) Three-point tracking

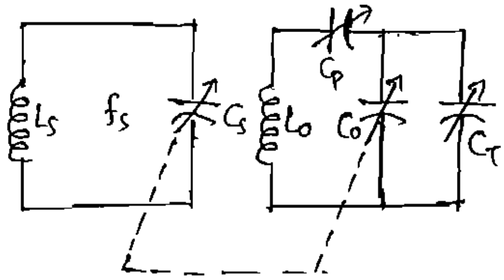



Fig: Three-point tracking.

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

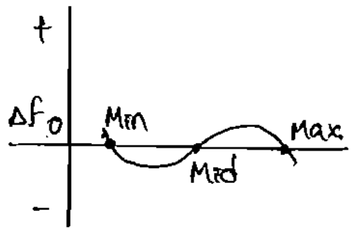



Fig. Tracking error in three point tracking.

Procedure to get values of capacitor required in above ckts

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

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

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

oscillator capacitance ratio can be given as

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

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

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

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

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

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

4. Calculate value of padder capacitor $C_p$ using ratios.

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

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

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

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

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

## Intermediate frequency amplifier

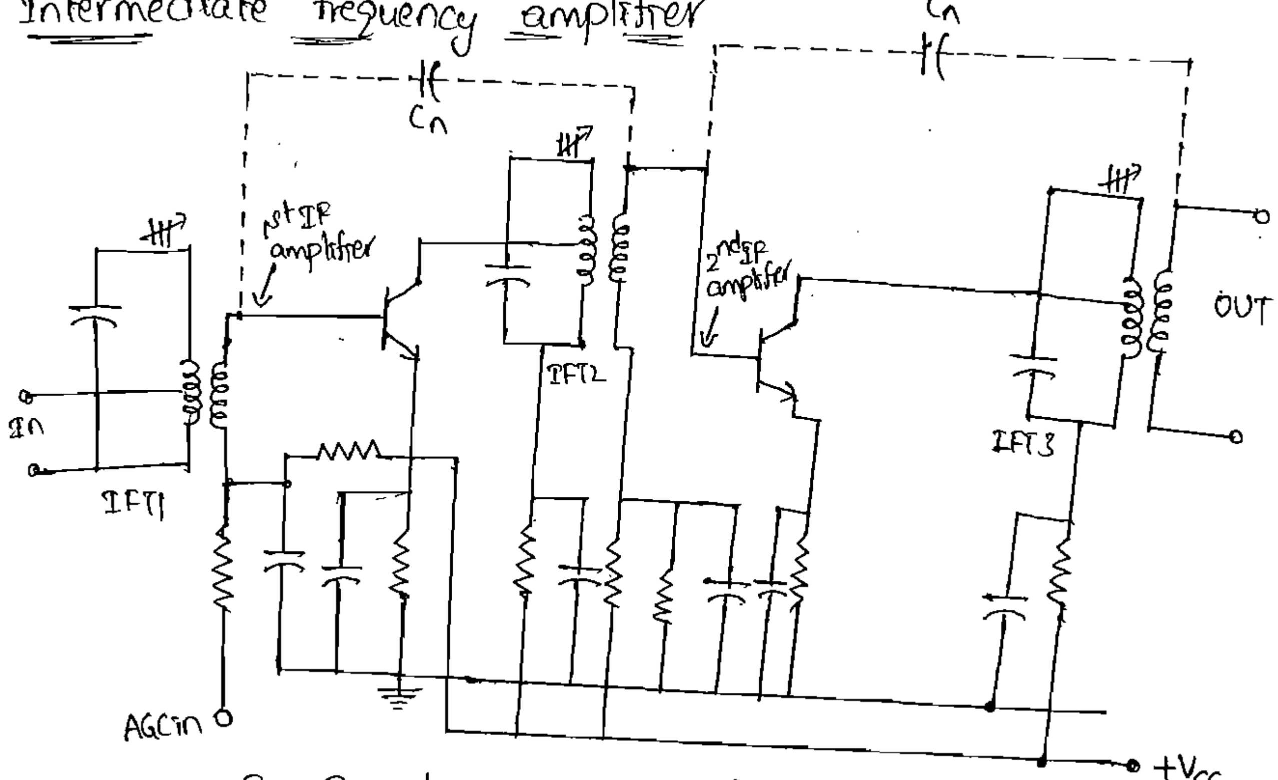



Fig: Two-stage IF amplifier.

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

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

Choice of Intermediate frequency

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

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

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

Automatic Gain Control (AGC)

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

There are two types of AGC circuits in use

→ Simple AGC

→ Delayed AGC

## Simple AGC

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

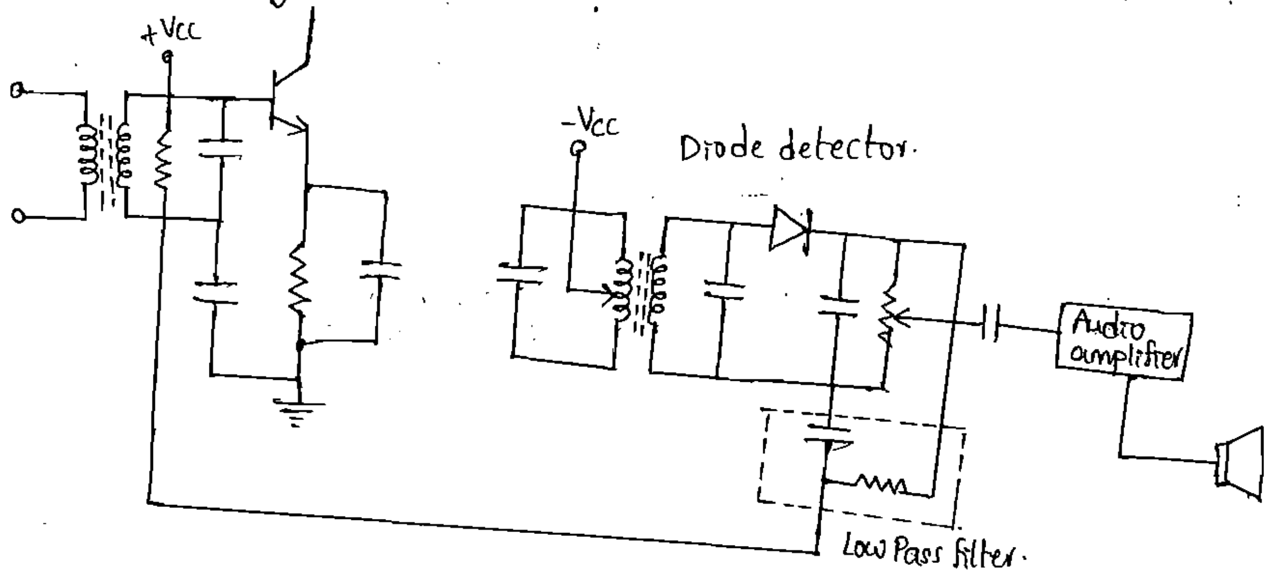



Fig: Simple automatic gain control circuit.

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

## Delayed AGC

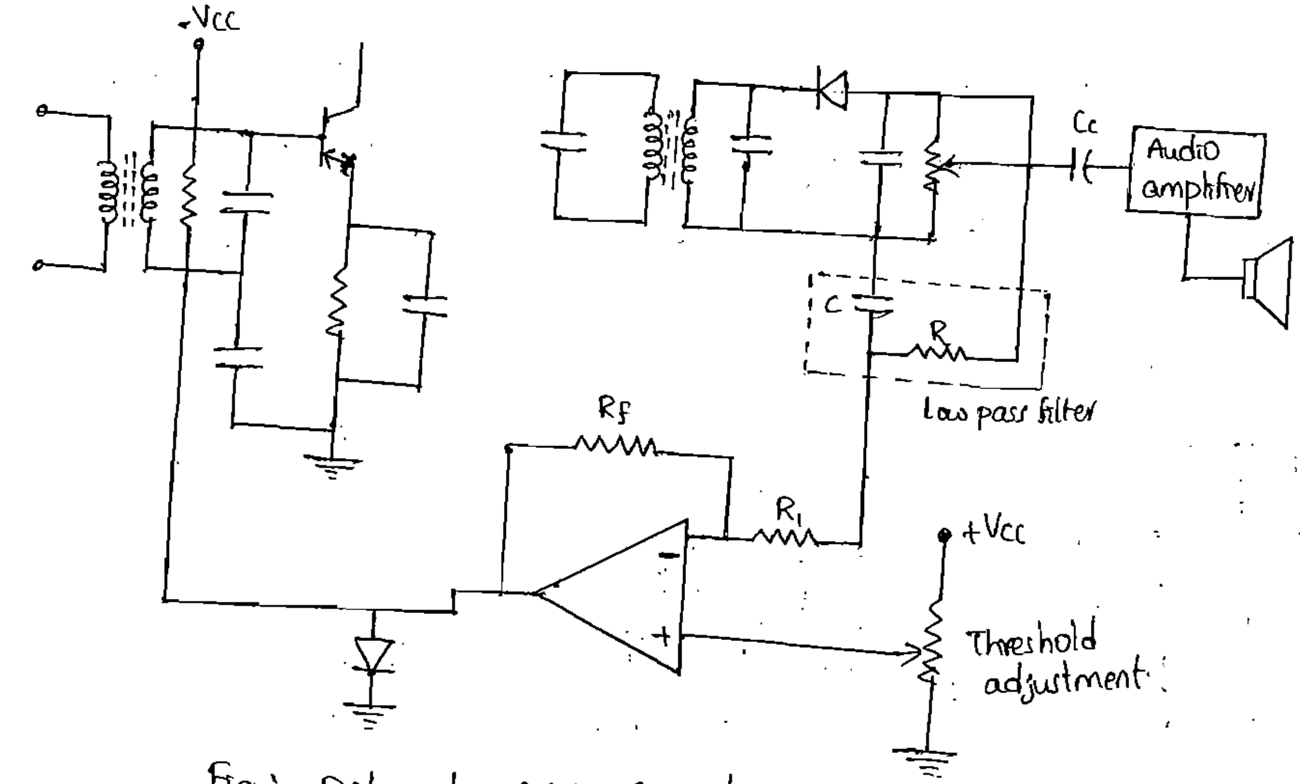
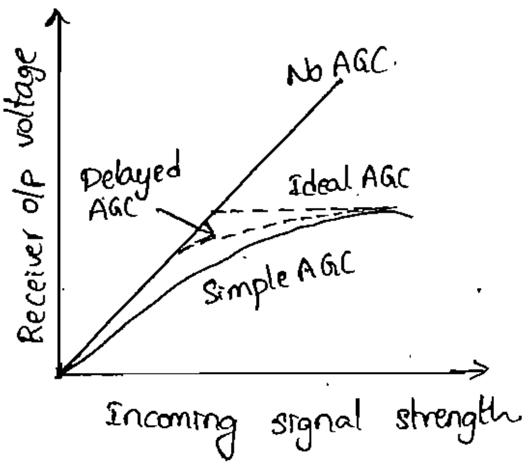



Fig: Delayed AGC Circuit.

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

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

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




## FM Receiver

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

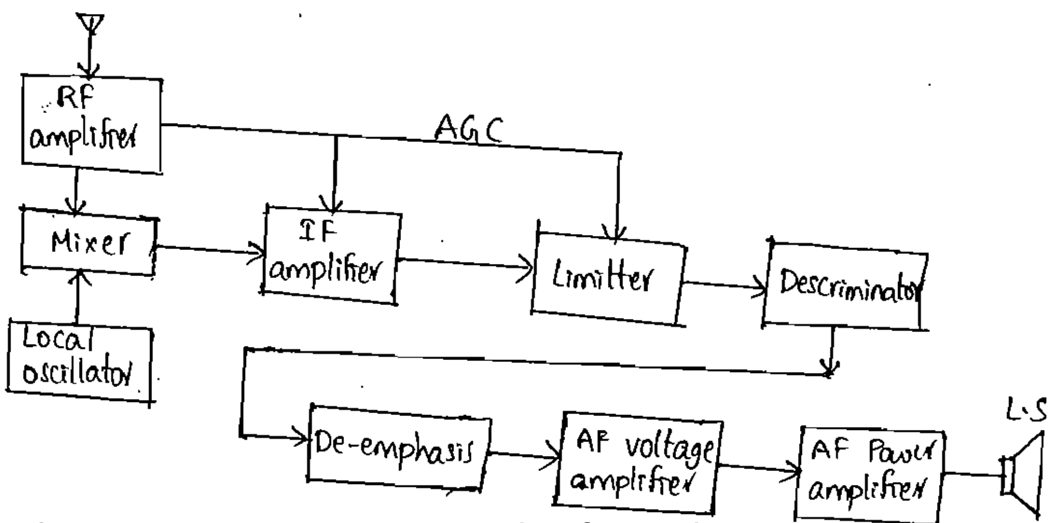
→ AM receiver operates in MW and SW bands, while FM receiver operates at much higher frequencies viz. 88MHz to 108MHz.

→ Limiter and deemphasis circuits are required only in FM receiver

→ The technique of demodulating FM signal is different from detection of AM signal.

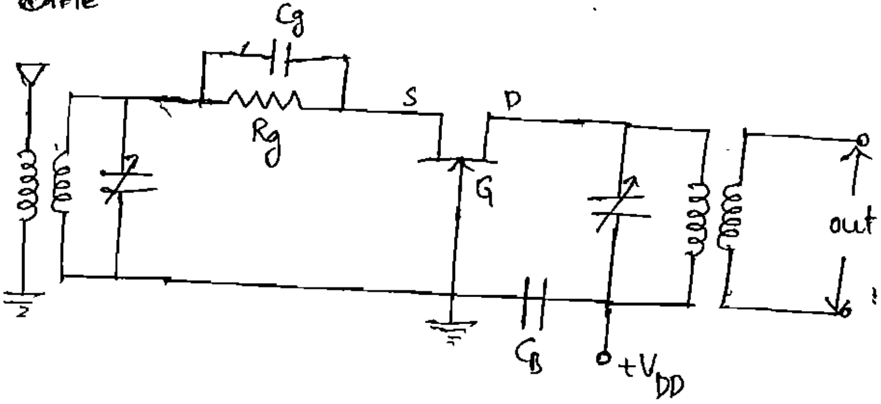
→ FM receiver uses different methods of obtaining AGC.

### Block diagram of FM Receiver



Different Stages in F.M Receivers.

## R.F Amplifier Stage




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

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

## Mixer Stage

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

Problem in FM Receiver unlike in an AM Receiver. Compared to AM Receiver, tuning range of incoming carrier frequencies for FM Receiver is small, from 88MHZ to 108MHZ i.e; about 1.25:1. Thus the tracking is comparatively easy in FM Receiver.

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

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

## I.F. Amplifier Stage

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

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

## Amplitude Limiting

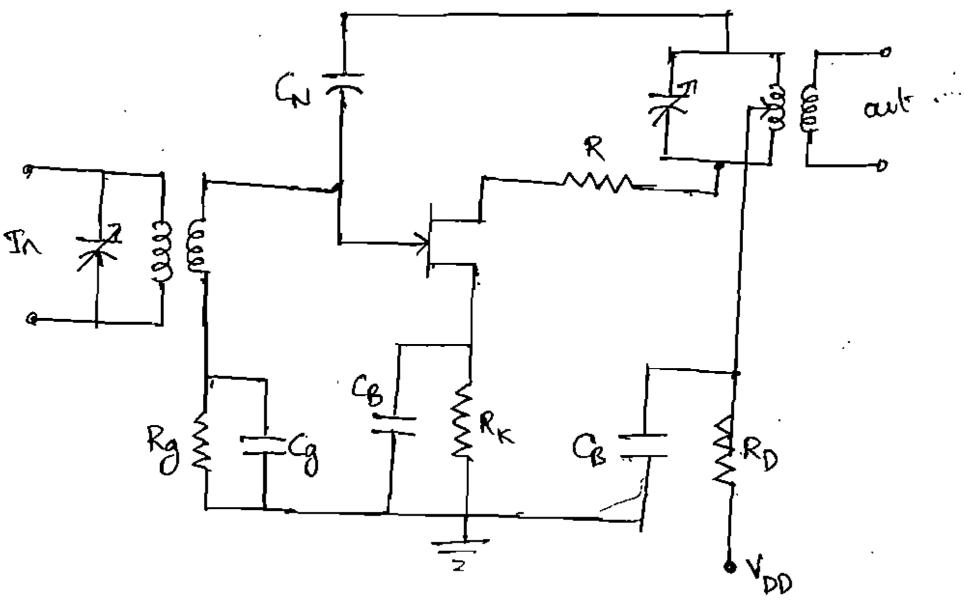



Fig: Amplitude Limitter.

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

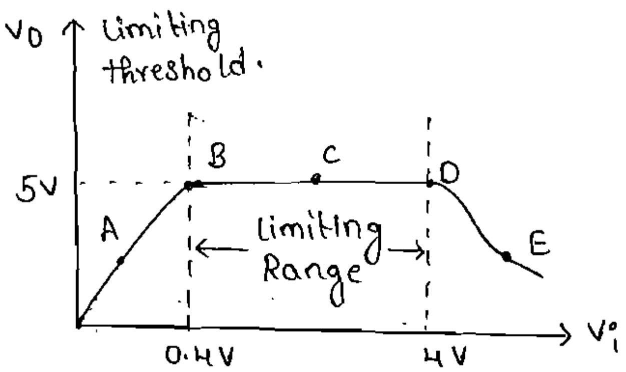



fig: Typical Limiter Response characteristics.

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

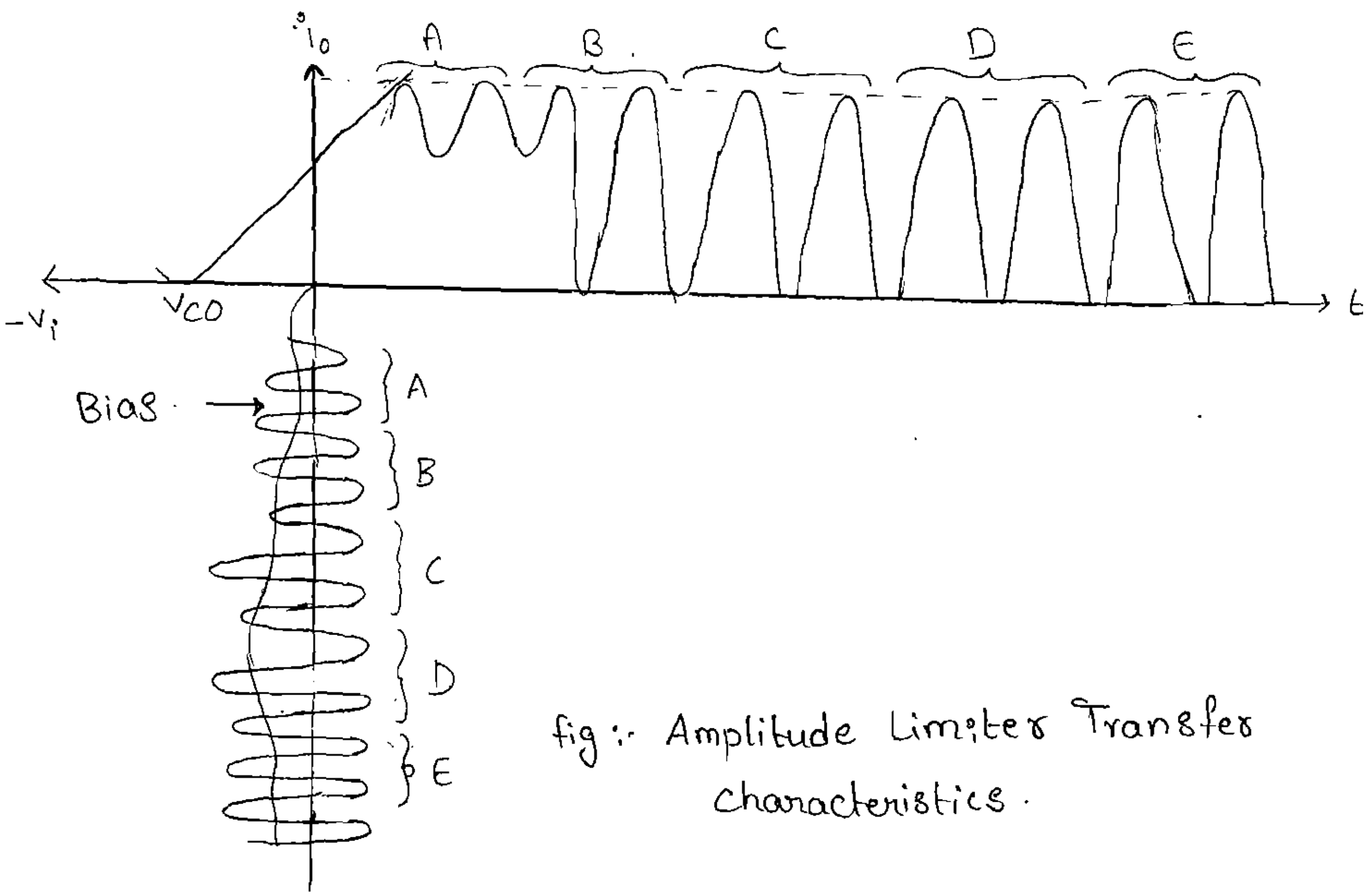



fig :- Amplitude Limiter Transfer characteristics.

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

## Introduction to noise :-

Noise is defined as any undesirable electrical energy that falls with in the passband of message signal. This gives rise to an audible noise in a system. The presence of noise degrades the performance of communication systems. In this chapter we analyze the noise in continuous wave modulation system. For such analysis we first define the receiver model. Then we analyze the noise in AM receivers namely DSBSC, SSB. Finally, we discuss the noise in FM receivers.

## Receiver model :-

The Fig. shows the receiver model in its most basic form. Modulated signal is $s(t)$ and noise is $w(t)$. Signal $w(t)$ is known as front end receiver noise. The receiver input signal is the sum of $s(t)$ and $w(t)$. The output of band-pass filter is $x(t)$. The band-width of a bandpass filter is kept just wide enough to pass the modulated signal $s(t)$ without distortion.

The demodulation process represented by the block demodulator depends on the modulation used.

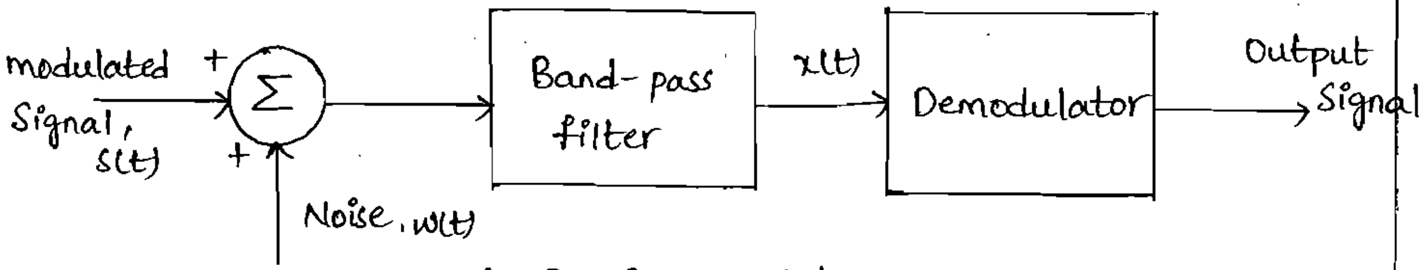



Fig. Receiver model.

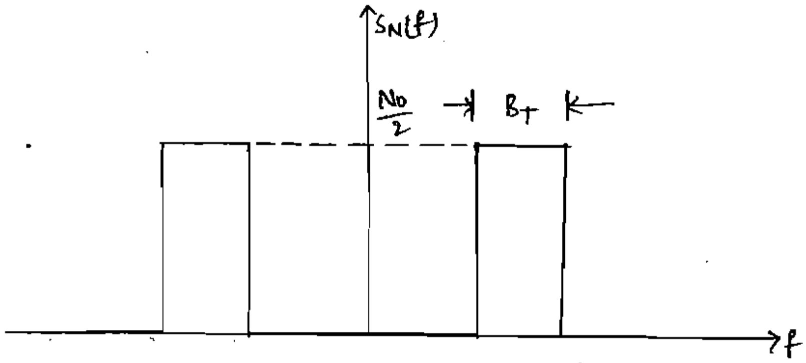



Fig. Idealized characteristic of bandpass filtered noise.

For receiver model, we may denote and define the following things.

* we denote $N_0/2$ as the power spectral density of noise w(t) for both +ve and −ve frequencies.
* '$N_0$' is the average noise power per unit bandwidth.
* Bandwidth of bandpass filter is equal to transmission bandwidth of the modulated signal and is denoted as '$B_T$'.
* Midband frequency is equal to the carrier frequency and it is denoted as $f_c$.

* Typically, the carrier frequency $f_c >> B_T$ and therefore we may consider the filtered noise n(t) as narrow band noise and it is defined in canonical form as

$$n(t) = n_I(t)\cos 2\pi f_c t - n_Q(t)\sin 2\pi f_c t.$$

* The filtered signal available for demodulation is defined by $x(t) = s(t) + n(t)$

* The average noise power is given as,

avg noise power = avg. noise power per unit bandwidth X Bandwidth

$$= N_0 B_T$$

* Input signal to noise ratio is given by

$$(SNR)_I = \frac{\text{Average power of the modulated signal, } s(t)}{\text{Average power of filtered noise, } n(t).}$$

* Output signal to noise ratio is given by

$$(SNR)_0 = \frac{\text{Average power of the demodulated msg signal}}{\text{Average power of the noise.}}$$

* Channel Signal to noise ratio is given by

$$(SNR)_c = \frac{\text{Average power of the modulated signal}}{\text{Avg. power of noise in message bandwidth}}$$

* Figure of merit $= \dfrac{(SNR)_0}{(SNR)_c}$

* Higher the value of figure of merit, better the performance of the receiver.

* The value of figure of merit also depends upon the type of modulation used.

## Noise in DSBSC Receivers :-

The below figure shows the model of a DSBSC receiver using a coherent detector. As shown in the figure, the filtered signal is applied to coherent detector $x(t)$. It is multiplied with a locally generated sinusoidal wave $\cos 2\pi f_c t$ using product modulator. The product is then filtered using low pass filter.

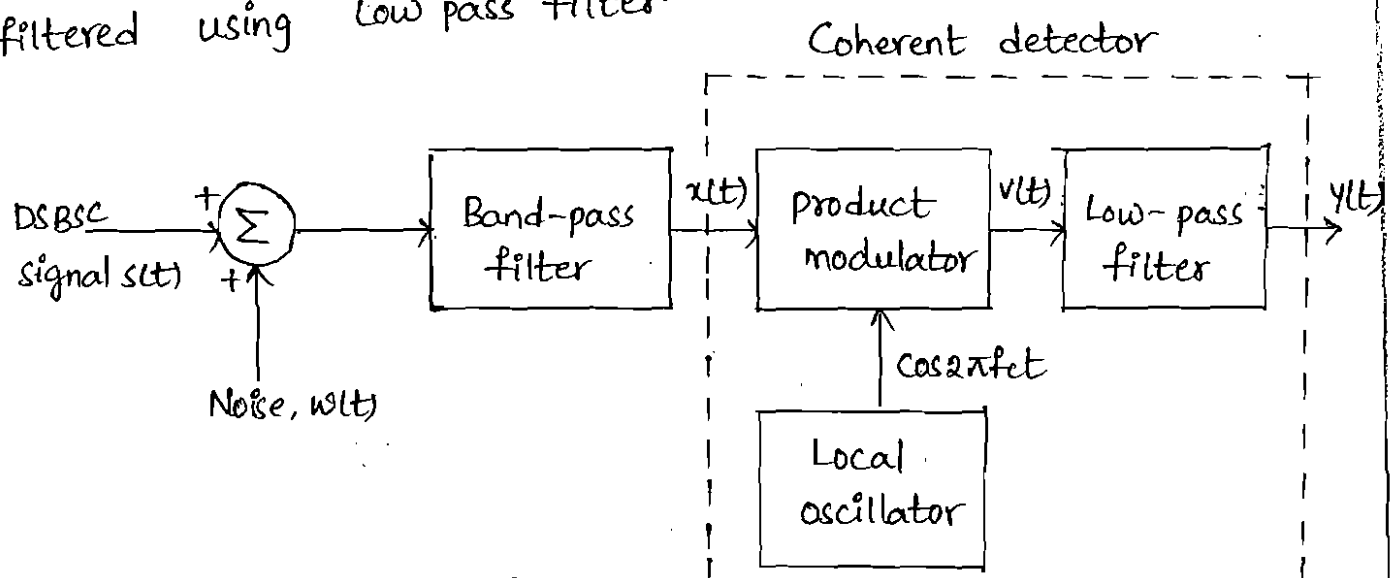



Fig. Model of DSBSC receiver

The time-domain expression of DSB-SC wave is given as,

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

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

Channel Signal-to-noise ratio :-

It is given as,

$$(SNR)_c = \frac{\text{Avg. power of modulated signal } s(t)}{\text{Avg. power of noise in msg bandwidth}}$$

$$(SNR)_c = \frac{\overline{s^2(t)}}{\overline{n_w^2(t)}}$$

Average power of modulated signal $= \overline{s^2(t)}$

$$= \overline{[m(t) A_c \cos 2\pi f_c t]^2}$$

$$= \overline{m^2(t)}\ \overline{[A_c \cos 2\pi f_c t]^2}$$

$$= A_c^2 \cdot P \cdot \left(\frac{1}{\sqrt{2}}\right)^2$$

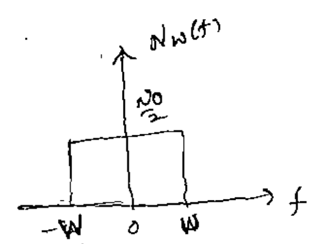
$$= \frac{A_c^2 P}{2}$$

where, 'P' represents average power of m(t).

Average power of noise in message bandwidth is given as $\overline{n_w^2(t)}$ = avg. noise power per unit bandwidth x bandwidth

$$= \frac{N_0}{2} \cdot 2W$$

$$= N_0 W$$




$$(SNR)_c = \frac{\overline{s^2(t)}}{\overline{n_w^2(t)}} = \frac{A_c^2 P}{2 N_0 W}$$

Output Signal-to-noise ratio :-

It is given as,

$$(SNR)_0 = \frac{\text{Average power of demodulated msg signal}}{\text{Average power of noise}}$$

$$(SNR)_0 = \frac{\overline{m_d^2(t)}}{\overline{n_d^2(t)}}$$

The output of band pass filter is

$$x(t) = s(t) + n(t)$$

$$x(t) = m(t) A_c \cos 2\pi f_c t + n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

This $x(t)$ is passed through product modulator. Another input to product modulator is $\cos 2\pi f_c t$. The output of product modulator is

$$v(t) = x(t) \cos 2\pi f_c t$$

$$v(t) = m(t) A_c \cos^2 2\pi f_c t + n_I(t) \cos^2 2\pi f_c t - n_Q(t) \cos 2\pi f_c t \sin 2\pi f_c t$$

$$v(t) = m(t) A_c \left(\frac{1+\cos 4\pi f_c t}{2}\right) + n_I(t) \left(\frac{1+\cos 4\pi f_c t}{2}\right) - \frac{n_Q(t)}{2} \sin 4\pi f_c t$$

$$v(t) = \frac{m(t) A_c}{2} + \frac{n_I(t)}{2} + \left(\frac{m(t) A_c}{2} + \frac{n_I(t)}{2}\right) \cos 4\pi f_c t - \frac{n_Q(t)}{2} \sin 4\pi f_c t.$$

This $v(t)$ is passed through Low-pass filter.

The output of LPF is

$$y(t) = \frac{m(t)A_c}{2} + \frac{n_I(t)}{2}$$

$$= m_d(t) + n_d(t)$$

where,

$m_d(t) = \frac{m(t)A_c}{2}$ is the desired signal component

$n_d(t) = \frac{n_I(t)}{2}$ is the noise component.

Average power of $m_d(t) = \overline{m_d^2(t)}$

$$= \overline{\left[\frac{m(t)A_c}{2}\right]^2}$$

$$= \frac{A_c^2}{4}\overline{m^2(t)} = \frac{A_c^2 P}{4}$$

Average power of $n_d(t) = \overline{n_d^2(t)}$

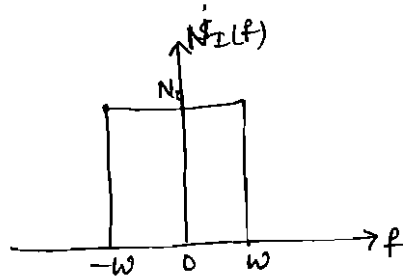
$$= \overline{\left[\frac{n_I(t)}{2}\right]^2}$$

$$= \frac{1}{4}\overline{n_I^2(t)}$$

$= \frac{1}{4} \times$ Area under PSD curve

$$= \frac{1}{4} \times N_0 \times 2W$$

$$= \frac{N_0 W}{2}$$




$$(SNR)_0 = \frac{\overline{m_d^2(t)}}{\overline{n_d^2(t)}} = \frac{A_c^2 P/4}{N_0 W/2} = \frac{A_c^2 P}{2N_0 W}$$

Figure of merit ($\gamma$) is given as,

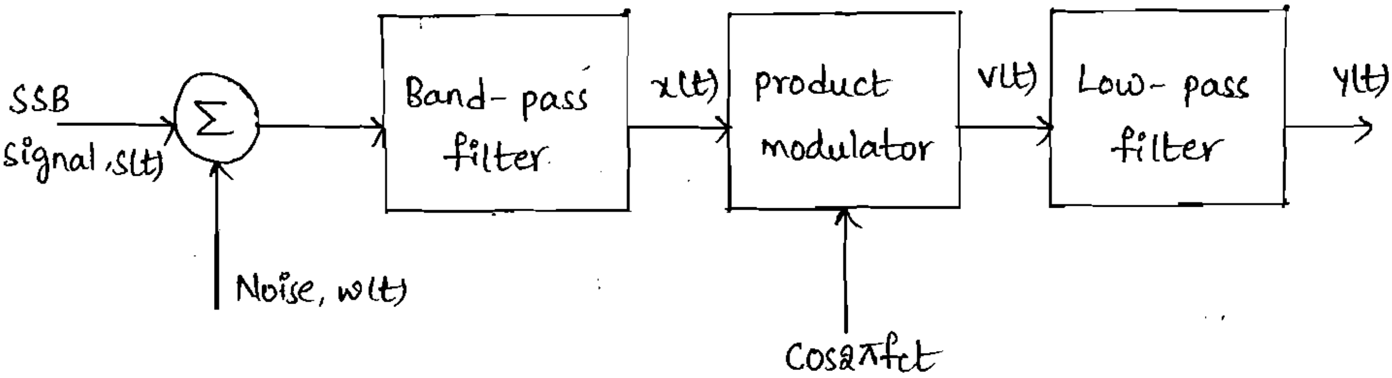
$$\gamma = \frac{(SNR)_o}{(SNR)_c}$$

$$\gamma = \frac{\frac{A_c^2 P}{2N_0W}}{\frac{A_c^2 P}{2N_0W}} = 1$$

Thus, the figure of merit of DSB-SC system is 1.

## Noise in an SSB-SC system:-

The block-diagram of the SSB-SC system is just as like DSB-SC system except for the fact that bandwidth of BPF of SSB-SC receiver is exactly half of that required for DSB-SC.




Channel Signal-to-noise ratio,

$$(SNR)_c = \frac{\text{Avg. power of modulated signal}}{\text{Avg. power of noise in msg bandwidth.}}$$

$$= \frac{\overline{s^2(t)}}{\overline{n_w^2(t)}}$$

The expression of s(t) for an SSB is given as

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

Average power of s(t) = $\overline{s^2(t)}$

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

$$\overline{s^2(t)} = \frac{A_c^2}{4} \overline{m^2(t)}\ \overline{\cos^2 2\pi f_c t} + \frac{A_c^2}{4} \overline{\hat{m}^2(t)}\ \overline{\sin^2 2\pi f_c t}$$

$$= \frac{A_c^2 P}{4}\left(\frac{1}{\sqrt{2}}\right)^2 + \frac{A_c^2 P}{4}\left(\frac{1}{\sqrt{2}}\right)^2$$

$$= \frac{A_c^2 P}{4}$$

Average power of $n_\omega(t) = \overline{n_\omega^2(t)}$

= Power Spectral density x bandwidth

= $N_0 W$

$$\therefore (SNR)_c = \frac{A_c^2 P}{4 N_0 W}$$

Output Signal-to-noise ratio :-

$$(SNR)_o = \frac{\text{Avg. power of demodulated msg signal}}{\text{Average power of noise.}}$$

$$= \frac{\overline{m_d^2(t)}}{\overline{n_d^2(t)}}$$

The output of band pass filter is

$$x(t) = s(t) + n(t).$$

Here, the type of modulation used is SSB, So the center frequency of band pass filter changes to $(f_c - W/2)$

Therefore, noise signal is expressed as

$$n(t) = n_I(t)\cos 2\pi(f_c - W/2)t - n_Q(t)\sin 2\pi(f_c - W/2)t$$

The output of product modulator is

$$v(t) = x(t)\cos 2\pi f_c t$$

$$= [s(t) + n(t)]\cos 2\pi f_c t$$

$$= s(t)\cos 2\pi f_c t + n(t)\cos 2\pi f_c t$$

$$v(t) = \left[\frac{A_c}{2}m(t)\cos 2\pi f_c t + \frac{A_c}{2}\hat{m}(t)\sin 2\pi f_c t\right]\cos 2\pi f_c t +$$
$$\left[n_I(t)\cos 2\pi(f_c - W/2)t - n_Q(t)\sin 2\pi(f_c - W/2)t\right]\cos 2\pi f_c t.$$

$$v(t) = \frac{A_c m(t)}{2}\cos^2 2\pi f_c t + \frac{A_c\hat{m}(t)}{2}\sin 2\pi f_c t\cos 2\pi f_c t +$$
$$n_I(t)\cos 2\pi(f_c - W/2)t\cos 2\pi f_c t - n_Q(t)\sin 2\pi(f_c - W/2)t\cos 2\pi f_c t$$

$$v(t) = \frac{A_c m(t)}{2}\left\{\frac{1+\cos 4\pi f_c t}{2}\right\} + \frac{A_c\hat{m}(t)}{4}\sin 4\pi f_c t +$$
$$\frac{n_I(t)}{2}\left\{\cos 2\pi(2f_c - W/2)t + \cos 2\pi(W/2)t\right\} -$$
$$\frac{n_Q(t)}{2}\left\{\sin 2\pi(2f_c - W/2)t - \sin 2\pi(W/2)t\right\}$$

$$v(t) = \frac{A_c m(t)}{4} + \frac{n_I(t)}{2}\cos\pi wt + \frac{n_Q(t)}{2}\sin\pi wt + \frac{A_c m(t)}{4}\cos 4\pi f_c t + \frac{A_c \hat{m}(t)}{4}\sin 4\pi f_c t + \frac{n_I(t)}{2}\cos 2\pi(2f_c - w/2)t - \frac{n_Q(t)}{2}\sin 2\pi(2f_c - w/2)$$

This v(t) is passed through LPF, In LPF high freq-uencies are attenuated and only low frequencies are allowed. So, the output of LPF is

$$y(t) = \frac{A_c m(t)}{4} + \frac{n_I(t)}{2}\cos\pi wt + \frac{n_Q(t)}{2}\sin\pi wt$$

$$= m_d(t) + n_d(t)$$

where,

$$m_d(t) = \frac{A_c m(t)}{4}$$

$$n_d(t) = \frac{n_I(t)}{2}\cos\pi wt + \frac{n_Q(t)}{2}\sin\pi wt$$

Average power of $m_d(t) = \overline{m_d^2(t)}$

$$= \overline{\left[\frac{A_c m(t)}{4}\right]^2}$$

$$= \frac{A_c^2}{16}\overline{m^2(t)} = \frac{A_c^2 P}{16}$$

Average power of $n_d(t) = \overline{n_d^2(t)}$

$$= \overline{\left[\frac{n_I(t)}{2}\cos\pi wt\right]^2} + \overline{\left[\frac{n_Q(t)}{2}\sin\pi wt\right]^2}$$

$$\overline{n_d^2(t)} = \frac{\overline{n_I^2(t)}}{4}\ \overline{\cos^2 \pi w t} + \frac{\overline{n_Q^2(t)}}{4}\ \overline{\sin^2 \pi w t}$$

$$\overline{n_d^2(t)} = \frac{N_0 W}{4}\left(\frac{1}{\sqrt{2}}\right)^2 + \frac{N_0 W}{4}\left(\frac{1}{\sqrt{2}}\right)^2$$

$$= \frac{N_0 W}{4}$$

$$\therefore (SNR)_0 = \frac{\overline{m_d^2(t)}}{\overline{n_d^2(t)}} = \frac{A_c^2 P/16}{N_0 w/4} = \frac{A_c^2 P}{4 N_0 W}$$

Figure of merit ($r$) is given as

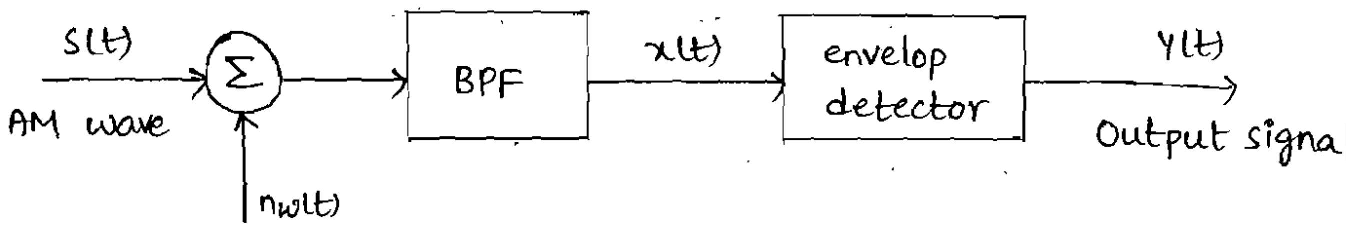
$$\text{Figure of merit} = \frac{(SNR)_0}{(SNR)_c}$$

$$= \frac{\dfrac{A_c^2 P}{4N_0 W}}{\dfrac{A_c^2 P}{4N_0 W}} = 1$$

Thus, the figure of merit of SSB-SC system is 1.

## Noise in AM system :-

The block diagram of AM receiver is as shown.




The time-domain expression for AM wave is given by

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

Channel Signal-to-noise ratio :-

$$(SNR)_c = \frac{\overline{s^2(t)}}{\overline{n_w^2(t)}}$$

Average power of $s(t) = \overline{s^2(t)}$

$$= \overline{[A_c \cos 2\pi f_c t]}^2 + \overline{[A_c k_a m(t) \cos 2\pi f_c t]}^2$$

$$= A_c^2 \left(\frac{1}{\sqrt{2}}\right)^2 + A_c^2 k_a^2 P \cdot \left(\frac{1}{\sqrt{2}}\right)^2$$

$$= \frac{A_c^2}{2}[1+k_a^2 P]$$

Average power of $n_w(t) = \overline{n_w^2(t)}$

$$= N_0 W$$

$$(SNR)_c = \frac{A_c^2(1+k_a^2 P)}{2N_0 W}$$

Output Signal-to-Noise ratio -

$$(SNR)_0 = \frac{\overline{m_d^2(t)}}{\overline{n_d^2(t)}}$$

The output of Band-pass filter is $x(t)$

$$x(t) = s(t) + n(t)$$

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

$$x(t) = \left[A_c + A_c k_a m(t) + n_I(t)\right]\cos 2\pi f_c t - n_Q(t)\sin 2\pi f_c t$$

$$e(t) = \sqrt{(\text{In phase comp.})^2 + (\text{Quadrature comp.})^2}$$

$$e(t) = \sqrt{\left[A_c + A_c k_a m(t) + n_I(t)\right]^2 + (n_Q(t))^2}$$

In this case,

$$A_c(1+k_a m(t)) >> n(t)$$

Thus,

$$A_c(1+k_a m(t)) >> n_I(t) \text{ (or) } n_Q(t)$$

$$\therefore e(t) = A_c(1+k_a m(t)) + n_I(t)$$

$$= A_c + A_c k_a m(t) + n_I(t)$$

The output of envelop detector is $y(t)$

$$y(t) = A_c k_a m(t) + n_I(t)$$

$$= m_d(t) + n_d(t)$$

Average power of $m_d(t) = \overline{m_d^2(t)}$

$$= \overline{(A_c k_a m(t))^2}$$

$$= A_c^2 k_a^2 \cdot P$$

Average power of noise is given as $\overline{n_d^2(t)}$

$$= \overline{n_I^2(t)}$$

$$= N_0(2W)$$

$$= 2N_0W$$

$$(SNR)_0 = \frac{A_c^2 k_a^2 P}{2N_0W}$$

Figure of merit ($r$) is given as

$$r = \frac{(SNR)_0}{(SNR)_c}$$

$$r = \frac{\frac{\cancel{A_c^2} k_a^2 P}{2\cancel{N_0W}}}{\frac{\cancel{A_c^2}(1+k_a^2P)}{2\cancel{N_0W}}} = \frac{k_a^2 P}{1+k_a^2 P}$$

we know that 'p' is the average power of the message signal and it is given as

$$P = \frac{1}{2} A_m^2$$

$$r = \frac{k_a^2 \frac{A_m^2}{2}}{1 + k_a^2 \frac{A_m^2}{2}} = \frac{k_a^2 A_m^2}{2 + k_a^2 A_m^2}$$

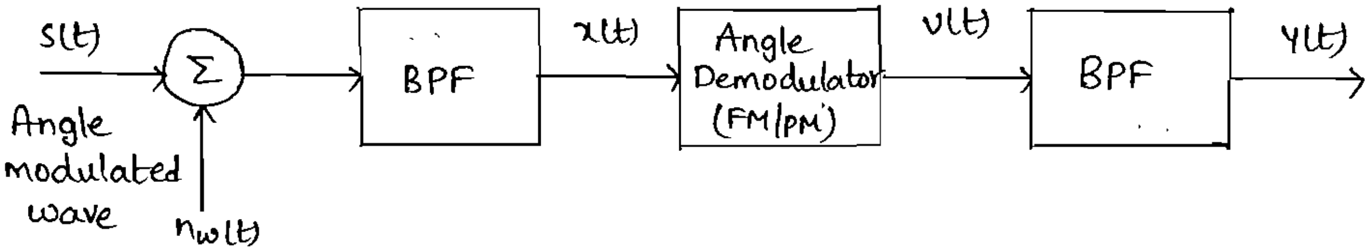
$$r = \frac{\mu^2}{2+\mu^2} \quad [\because \mu = k_a A_m]$$

For 100% modulation i.e $\mu = 1$ we get

$$r = \frac{1}{2+1} = \frac{1}{3}.$$

## Noise in Angle modulated system :-

The block diagram of an angle modulated system is as shown,




The time-domain expression for an angle modulated carrier is given by,

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

where $\phi(t)$ represents the instantaneous phase angle and is given as,

$$\phi(t) = k_p m(t) \quad \text{[For phase modulation]}$$

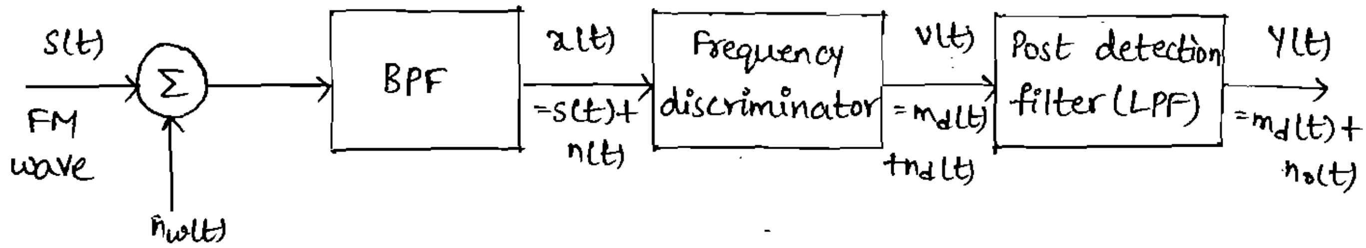
$$\phi(t) = 2\pi k_f \int_0^t m(t)\, dt \quad \text{[For frequency modulation]}$$

Here $k_p$ and $k_f$ represents the sensitivities of phase & frequency respectively. The transmission bandwidth $B_T$ in angle modulated system determined by Carson's rule is

$$B_T = 2(\Delta f + W)$$

where $W$ represents the Bandwidth of msg signal and $\Delta f$ is the peak frequency deviation.

## Noise in FM Receivers :-




The time domain expression for frequency modulated carrier is given by

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

The noise can be expressed in terms of inphase and quadrature components as,

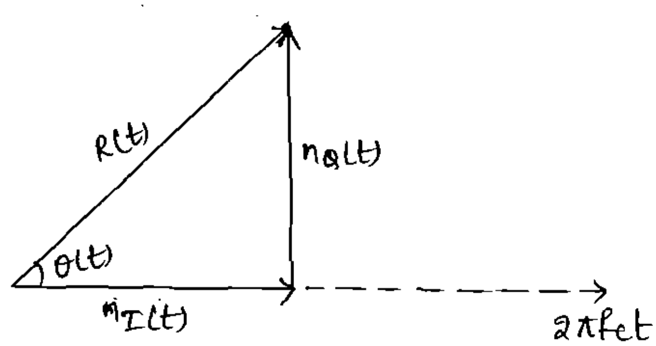
$$n(t) = n_I(t)\cos 2\pi f_c t - n_Q(t)\sin 2\pi f_c t$$

$$= R(t)\cos(2\pi f_c t + \theta(t))$$

where, $$R(t) = \sqrt{n_I^2(t) + n_Q^2(t)}$$

$$\theta(t) = \tan^{-1}\left(\frac{n_Q(t)}{n_I(t)}\right)$$

The phasor diagram of n(t) drawn by taking reference as phase of unmodulated carrier is as shown.



Channel Signal-to-noise ratio.

$$(SNR)_c = \frac{\overline{s^2(t)}}{\overline{n_w^2(t)}}$$

Average power of $s(t) = \overline{s^2(t)}$

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

$$= A_c^2 \cdot \left(\frac{1}{\sqrt{2}}\right)^2$$

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

Average power of noise $n_w(t) = \overline{n_w^2(t)}$

$$= N_0 W$$

Thus, $(SNR)_c = \dfrac{A_c^2}{2N_0W}$

Output Signal-to-noise ratio –

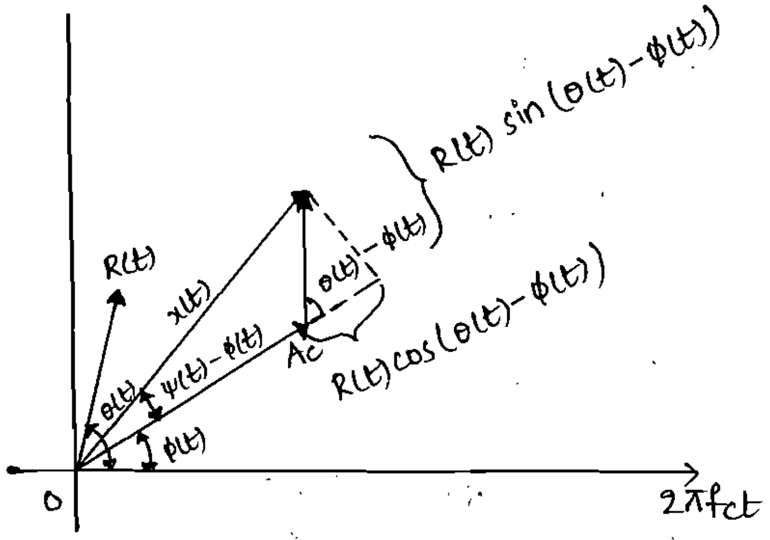
$$(SNR)_o = \frac{\text{Average power of demodulated signal}}{\text{Average power of the noise}}$$

The output of band-pass filter is

$$x(t) = s(t) + n(t)$$

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

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




The relative phase $\psi(t)$ of the resulting $x(t)$ can be obtained from the figure as,

$$\psi(t) - \phi(t) = \tan^{-1}\left\{\frac{R(t)\sin(\theta(t)-\phi(t))}{A_c + R(t)\cos(\theta(t)-\phi(t))}\right\}$$

While drawing phasor diagram, it is assumed that $\theta(t) > \phi(t)$ and $A_c > R(t)$.

We can write,

$$\tan(\psi(t)-\phi(t)) \approx \frac{R(t)\sin\theta(t)}{A_c}$$

Referring the phasor diagram of narrow band noise, we get

$$R(t)\sin\theta(t) = n_Q(t)$$

Hence,

$$\tan(\psi(t)-\phi(t)) = \frac{n_Q(t)}{A_c}$$

Since $A_c >> n_Q(t)$ we can write

$$\psi(t) - \phi(t) \approx \frac{n_Q(t)}{A_c}$$

$$\psi(t) = \phi(t) + \frac{n_Q(t)}{A_c}$$

The output of frequency discriminator is

$$v(t) = \frac{1}{2\pi} \frac{d\psi(t)}{dt}$$

$$= \frac{1}{2\pi} \frac{d}{dt}\left[\phi(t) + \frac{n_Q(t)}{A_c}\right]$$

$$v(t) = \frac{1}{2\pi} \frac{d\phi(t)}{dt} + \frac{1}{2\pi A_c} \frac{dn_Q(t)}{dt}$$

$$= \frac{1}{2\pi} \frac{d}{dt}\left[2\pi k_f \int_0^t m(t)\,dt\right] + \frac{1}{2\pi A_c} \frac{dn_Q(t)}{dt}$$

$$= k_f\, m(t) + \frac{1}{2\pi A_c} \frac{dn_Q(t)}{dt}$$

$$= m_d(t) + n_d(t)$$

where,

$$m_d(t) = k_f\, m(t)$$

$$n_d(t) = \frac{1}{2\pi A_c} \frac{dn_Q(t)}{dt}$$

Average power of $m_d(t) = \overline{m_d^2(t)}$

$$= k_f^2\, \overline{m^2(t)}$$

$$= k_f^2 \cdot P$$

Average power of $n_d(t) = \overline{n_d^2(t)}$

By applying fourier transformation, we get

$$N_d(f) = \frac{1}{\cancel{2\pi} A_c}\left[j\, \cancel{2\pi} f\; N_Q(f)\right]$$

$$= \frac{jf}{A_c} N_Q(f)$$

If $S_{nd}(f)$ and $S_{nQ}(f)$ be the power spectral densities of $N_d(f)$ and $N_Q(f)$ then relation is given as,

$$S_{nd}(f) = \frac{f^2}{A_c^2} S_{nQ}(f)$$

If the signal is passed through LPF, the value of $S_{nQ}(f)$ will be $N_0$.

$$\therefore S_{no}(f) = \frac{f^2}{A_c^2} N_0$$

The average power of noise signal in demodulated signal is given as

$$\overline{n_o^2(t)} = \int_{-W}^{W} S_{No}(f)\, df$$

$$= \int_{-W}^{W} \frac{N_0}{A_c^2} f^2\, df$$

$$= \frac{N_0}{A_c^2}\int_{-W}^{W} f^2\, df = \frac{N_0}{A_c^2}\left(\frac{f^3}{3}\right)_{-W}^{W}$$

$$= \frac{N_0}{A_c^2}\left[\frac{W^3}{3} + \frac{W^3}{3}\right] = \frac{2N_0W^3}{3A_c^2}$$

$$\therefore (SNR)_o = \frac{K_f^2 P}{\frac{2N_0W^3}{3A_c^2}} = \frac{3A_c^2 K_f^2 P}{2N_0 W^3}$$

$$\text{Figure of merit } (r) = \frac{(SNR)_o}{(SNR)_c}$$

$$r = \frac{\dfrac{3A_c^2 k_f^2 P}{2N_o W^3}}{\dfrac{A_c^2}{2N_o W}} = \frac{3k_f^2 P}{W^2}$$

$$\boxed{\therefore\ r = \frac{3k_f^2 P}{W^2}}$$

we know that $p$ is the average power of message signal and it is given as,

$$P = A_m^2/2$$

$$r = \frac{3k_f^2}{W^2} \cdot \frac{A_m^2}{2} = \frac{3}{2W^2} \Delta f^2$$

$$r = \frac{3}{2}\left(\frac{\Delta f}{W}\right)^2$$

$$\boxed{r = \frac{3}{2}\beta^2}$$ where, $\frac{\Delta f}{W} = \beta$ (modulation index)

Now Let us compare the figure of merit of FM w.r.t AM

For 100% modulation the figure of merit of AM $= \frac{1}{3}$

The figure of merit of FM $= \frac{3}{2}\beta^2$

To have less noise in FM when compared to AM we have to take

$$\frac{3}{2}\beta^2 > \frac{1}{3}$$

$$\Rightarrow \beta > \frac{\sqrt{2}}{3}$$

$$\beta > 0.47 \approx 0.5$$

The value of $\beta = 0.47$ (or) $\beta = 0.5$ actually the transition point between the narrow-band FM and wide-band FM.

If $\beta < 0.5$, the FM is considered as narrow band FM in which there is no improvement in noise when compared to AM.

## Capture Effect :-

In the frequency modulation, the signal can be affected by another frequency modulated signal whose frequency content is close to carrier frequency of the desired FM wave. The receiver may lock such an interference signal and suppress the desired FM signal when interference signal is stronger than desired signal.

When the strength of interference and desired signal are nearly equal, the receiver fluctuates back and forth between them i.e receiver locks interference signal for some time and desired signal for some time and this goes randomly. This phenomenon is Capture effect.

## Threshold effect in angle modulation System:-

The threshold effect in FM is much more pronounced than in AM. The figure of merit of FM is valid if the carrier-to-noise is high compared to unity (i.e, CNR>>1).

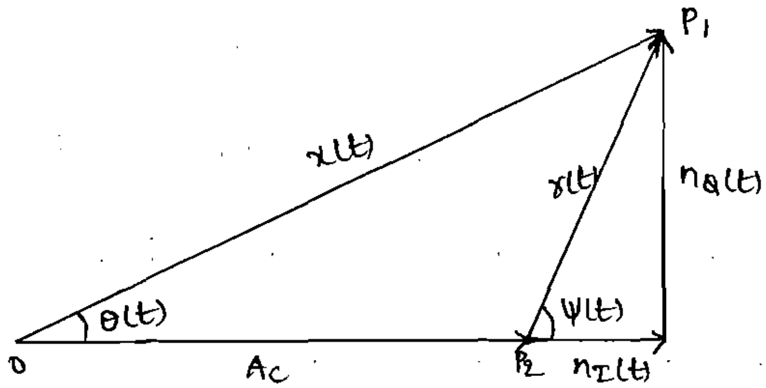
Suppose if the input noise power is increased or the carrier power is decreased, the CNR decreases consequently demodulator discriminator output becomes more and more corrupted by noise. Spikes comes out FM receiver and if CNR further decreases, continuous spikes comes out of FM receiver. The FM receiver is said to breakdown when clicks are heard. This phenomenon is called as threshold effect.

The threshold effect is defined as the minimum carrier to noise ratio that gives the output signal to noise ratio not less than the value predicted by the usual signal to noise formula assuming a small noise power.

At the frequency discriminator input is given by

$$x(t) = [A_c + n_I(t)] \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

where $n_I(t)$ and $n_Q(t)$ are in-phase and quadrature components of narrow band noise signal $n(t)$ w.r.t carrier respectively. The relationships defined by this equation are represented by phasor diagram as shown.




Let us derive the conditions for positive clicks to occur and negative clicks to occur are as follows:

Conditions for positive clicks:

$$r(t) > A_c$$

$$\psi(t) < \pi < \psi(t) + d\psi(t)$$

$$\frac{d\psi(t)}{dt} > 0$$

conditions for negative clicks:

$$r(t) > A_c$$

$$\psi(t) > -\pi > \psi(t) + d\psi(t)$$

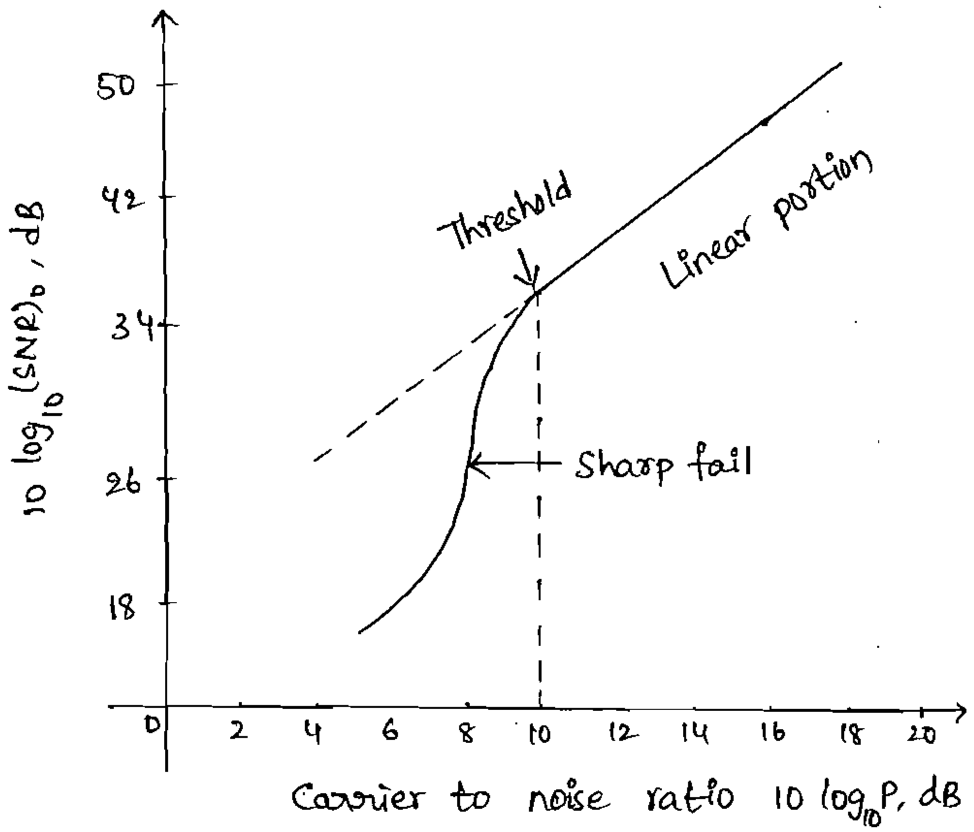
$$\frac{d\psi(t)}{dt} < 0$$

The conditions for positive clicks ensure that $\theta(t)$ changes by $2\pi$ radians and for negative clicks ensure that $\theta(t)$ changes by $-2\pi$ radians during the time increment $dt$.

Carrier-to-noise ratio is defined by

$$P = \frac{A_c^2}{2 B_T N_0}$$

The average no. of clicks per unit time is inversely proportional to P. It is seen that $(SNR)_0$ ratio is a linear function of P when P is greater than 10 dB. However, it falls sharply for lower values of P than 10 dB. This is shown below.




Threshold can be avoided by keeping $P > 20$ i.e 13 dB

$$\frac{A_c^2}{2 B_T N_0} \geq 20 \quad \Rightarrow \quad \frac{A_c^2}{2} \geq 20 B_T N_0$$

$$\gamma = \frac{3k_f^2 P}{W^2}$$

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

$$\Rightarrow P = \overline{m^2(t)} = A_m^2/2$$

$$\Rightarrow \gamma = \frac{3k_f^2 A_m^2}{2W^2}$$

The expression for frequency deviation is given by

$$\Delta f = \left|k_f\ m(t)\right|_{max} = \left|k_f\ A_m \cos(2\pi f_m t)\right|_{max}$$

$$\Rightarrow \cancel{\Delta f = k_f\, m(t)} \Rightarrow \Delta f = k_f A_m$$

$$\therefore\ \gamma = \frac{3}{2}\left[\frac{\Delta f}{W}\right]^2 \Rightarrow \boxed{\gamma = 1.5\beta^2}$$

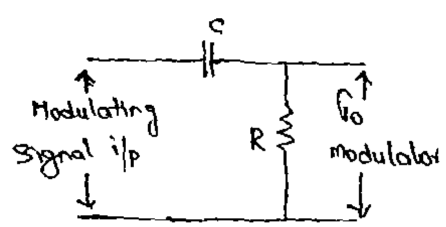
Where $\beta$ is modulation index of FM given by $\beta = \frac{\Delta f}{W}$

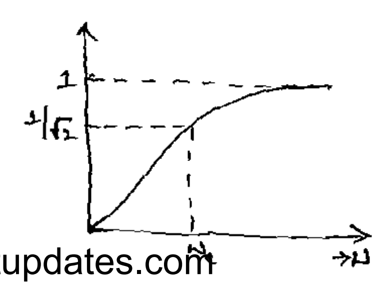
## PRE-EMPHASIS AND DE-EMPHASIS

Noise produced in electronic circuits is low in low Af range, but at higher frequencies it increases. So, for information signals with a uniform signal level, a non-uniform signal-to-noise ratio is obtained. The higher modulating signal frequencies have a lower signal-to-noise ratio than the lower frequencies. To overcome this a high frequency modulating signals are emphasized (or) boosted in amplitude in the FM transmitter before modulation. This is known as pre-emphasis. The de-emphasis circuit restores the original amplitude-frequency characteristics of the information signal. Pre-emphasis and De-emphasis give a more (or) less uniform signal-to-noise ratio over the whole Af range.

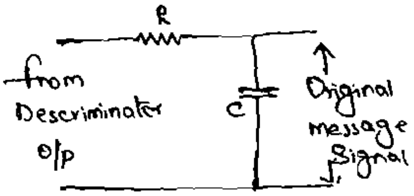
A Pre-emphasis circuit is a high pass filter i.e. ~~an~~ RC ~~circuit~~ ~~with a~~ high frequency components are boosted up at the output, because the capacitor C offers a low reactance at high frequencies.

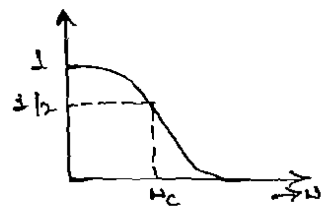
The circuit and frequency response curve of the Pre-emphasis are shown as



A de-emphasis circuit is low-pass filter, i.e high frequency components are attenuated because the capacitor short circuits such components. The circuit and frequency response curve of de-emphasis are shown as



UNIT-5

NOISE

Sources of noise:-

→ Natural

→ Manmade

→ fundamental

Classification

→ Shot noise:- It is produced due to shot effect. It produced in all the amplifying devices rather than in all the active devices.

* Shot noise is produced because of the random variations in the arrival of electrons & holes at the o/p electrode of an amplifying device. It sounds like a shower of lead shots falling on a metal sheet. The mean square shot noise current equation for diode is given as $I_n^2 = 2(I + 2I_0) q B$ (Amp$^2$)

$I \rightarrow$ direct current across the junction (Amps)

$I_0 \rightarrow$ reverse saturation current (Amps)

$q \rightarrow$ electronic charge $= 1.6 \times 10^{-19}$ Columbs

$B \rightarrow$ effective noise Bandwidth (Hz)

for the amplying devices the shot noise is inversely proportion-al to the transconductance of the device & directly proportional to the direct current.

→ Partition Noise:- It is generated when the current gets divided b/w two or more paths it is generated due to the random fluctuations in the division. Therefore the partition noise in a transistor will be higher than that in a diode.

→ Low frequency / flicker Noise:- It will appear at frequencies below a few KHz. It is sometimes called as $\frac{1}{f}$ noise. In the semiconductor devices flicker noise is generated due to the fluctuations in the carrier density. These fluctuations in the carrier density will cause the fluctuations in the conductivity of the material. This will produce a fluctuation voltage drop when a direct current flows through a device.

This fluctuating voltage is known as flicker noise voltage. The mean square value of the flicker noise voltage is proportional to the square of direct current flowing through the device.

→ <u>Thermal / Johnson / white noise</u> :- The free electrons within a conductor are always in random motion. This random motion is due to the thermal energy received by them. The distribution of this free electrons with in a conductor at a given instant of time is not uniform. It is possible that an excess no of electrons may appear at one end or the other of the conductor. The average voltage resulting from this non-uniform distribution is zero, but the avg power is not zero. As this power results from the thermal energy. It is called as the thermal noise power.

Avg thermal noise power is given by $P_n = KTB$ watts

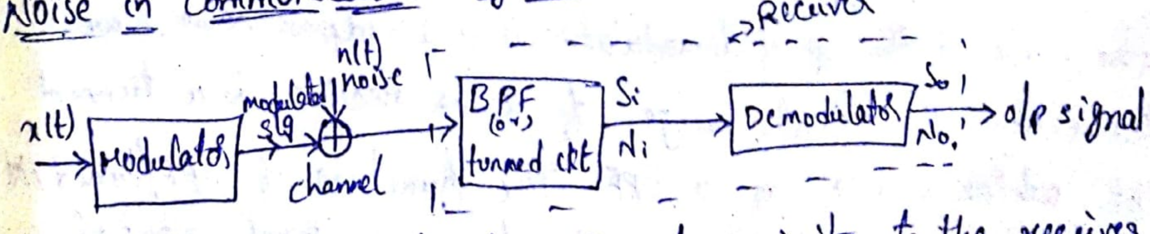
K – Boltzmann constant $1.38 \times 10^{-23}$ J/K

B – B.W of noise spectrum (Hz)

T – temp of conductor °K

→ <u>High frequency / Transit time noise</u> :- If the time taken by an electron to travel from the emitter to the collector of a transistor becomes comparable to the period of the s/g which is being amplified then the transit time effect takes places. This effect is observed at very high frequencies. Due to transit time effect some of the carriers diffused back to the emitter. This gives rise to a i/p admittance, the conductance component of which increases with frequency. The minute currents induced in input of device by the random fluctuations in o/p current, will create ~~at~~ random noise at high frequencies.

## <u>Noise in Communication System</u> :-




The message s/g travels from the transmitter to the receiver through a medium called channel. Noise is present in every communication system. The channel introduces a negative additive noise in the message s/g, and thus the msg which is received at the receiver is distorted.

Since the receiver detects both msg & noise signals. It will reproduce a msg s/g which contains noise. A noise calculation in a communication s/m is carried out ~~by~~ in the form of a parameter called figure of merit. It is noted by letter ($\gamma$).

figure of merit is defined as the ratio of o/p SNR to i/p SNR of a receiver.

$$\gamma = \frac{(SNR)_o}{(SNR)_i}$$

→ few assumptions to calculate the figure of merit for various communications systems.

1) Channel noise is always white & Gaussian :- we assume that the noise of channel n(t) is always a white noise. This means that it is uniformly distributed over the entire band of frequencies hence the PSD of channel noise will be uniform.

The total noise power may be obtained by taking the product of noise power spectrum density $N_0/2$ with the Bandwidth ~~with the bandwidth~~.

Total noise power N = white noise PSD × B.W

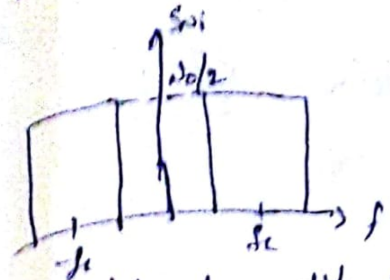
$$N = \frac{N_0}{2} \times B.W$$

Thus the noise has a Gaussian distribution.

2) Channel noise is always additive :- we assume that the disturbing effect of channel noise is always additive. This means that the effect of channel noise may be obtained by simple addition of s/g x(t) + noise n(t).

3) The noise at the i/p of demodulator is a bandpass noise :- we know that the first range of each receiver is a turned ckt which works as a BPF. The function of BPF/turned ckt is to allow only an narrow band s/g centered about $\pm f_c$ and reject all other frequencies. This means that the noise s/g lying ~~ab~~ outside this range is also rejected. And thus the B.W of noise s/g at the i/p of detector as same as that of the incoming modulated s/g.

we assumed that the channel noise is wide in nature so the PSD of white noise at i/p of demodulator is $S_{ni} = \frac{N_0}{2}$

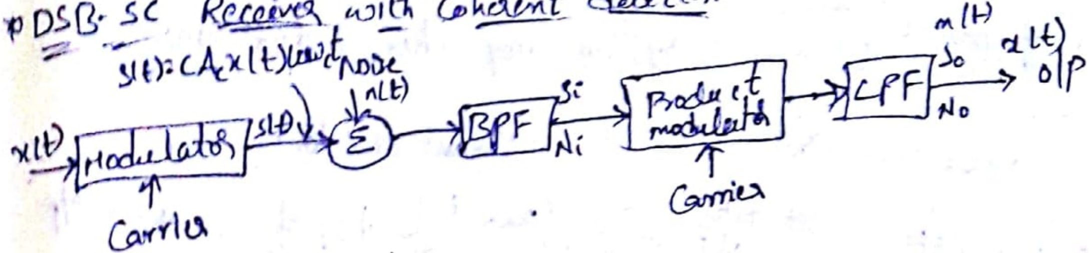



$r(t) = n_I(t)\cos\omega_c t - n_Q(t)\sin\omega_c t$

PSD of bandpass whitenoise

(i) The noise i/p power $N_i$ to calculate the figure of merit we evaluate the noise i/p power for passband / B.W of ~~i/p~~ incoming modulating signal. Total noise power $N = \frac{N_0}{2} \times 2f_m = N_0 f_m$ (for AM s/g)

* DSB-SC Receiver with Coherent detector:-




$$\gamma = \frac{(SNR)_o}{(SNR)_i} \qquad (SNR)_i = (SNR)_c$$

$(SNR)_i = \frac{\text{Avg s/g power at the receiver i/p}}{\text{Avg noise power at the receiver i/p}}$

$$\text{Avg signal power} = \frac{c^2 A_c^2 P}{2}$$

$$\text{Avg noise power} = 2 \times \frac{N_0}{2} \times 2f_m = 2N_0 f_m$$

$$(SNR)_i = \frac{c^2 A_c^2 P}{2 \times 2N_0 f_m} = \frac{c^2 A_c^2 P}{4N_0 f_m} \quad \text{— (1)}$$

After LPF i.e., receiver o/p.

$$S_o = (CA_c x(t)\cos\omega_c t + n_I(t)\cos\omega_c t - n_Q(t)\sin\omega_c t)\cos\omega_c t$$

$$= CA_c x(t)\cos^2\omega_c t + n_I(t)\cos^2\omega_c t - n_Q(t)\sin\omega_c t\cos\omega_c t$$

$$\cos^2\theta = \frac{1+\cos 2\theta}{2}$$

$$\sin 2A = 2\sin A\cos A$$

$$= CA_c x(t)\left(\frac{1+\cos 2\omega_c t}{2}\right) + n_I(t)\left(\frac{1+\cos 2\omega_c t}{2}\right) - \frac{n_Q(t)}{2}\sin 2\omega_c t$$

$$S_o = \frac{CA_c x(t)}{2} + \frac{CA_c x(t)}{2}\cos 2\omega_c t + \frac{n_I(t)}{2} + \frac{n_I(t)}{2}\cos 2\omega_c t - \frac{n_Q(t)}{2}\sin 2\omega_c t$$

After LPF

$$m(t) = \frac{CA_c x(t)}{2} + \frac{n_I(t)}{2}$$

$m(t) = \frac{CA_c}{2} x(t) + \frac{1}{2} n_I(t)$

Avg signal power $= \frac{(\frac{1}{2})^2 C^2 A_c^2 P}{2} = \frac{C^2 A_c^2 P}{8}$

Noise $(\frac{1}{2})^2\, 2 \times \frac{N_0}{2} \times 2f_m = \frac{1}{2} N_0 f_m = \frac{N_0 f_m}{2}$

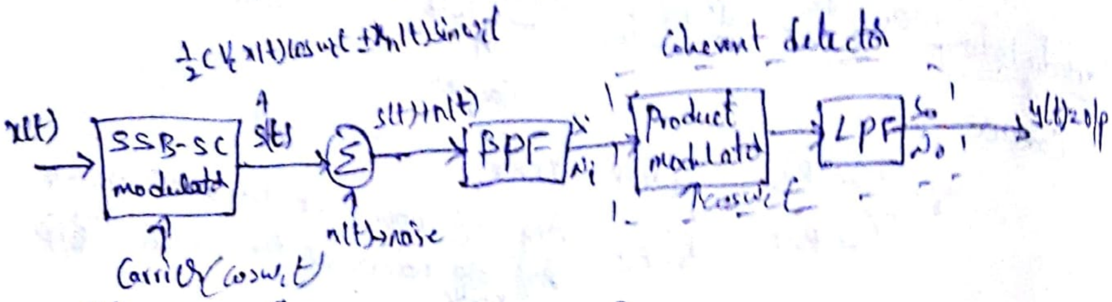
$(SNR)_o = \frac{2C^2A_c^2P}{8 N_0 f_m} = \frac{C^2A_c^2P}{4N_0 f_m}$ — ②

Sub ① & ② in $\gamma$

$\gamma = \frac{(SNR)_o}{(SNR)_i} = \frac{\frac{C^2A_c^2P}{4N_0f_m}}{\frac{C^2A_c^2P}{4N_0f_m}} = 1$

$\boxed{\gamma = 1}$

* figure of merit for SSBSC system using coherent detection:-




$s(t) = \frac{1}{2} CV_c \left[x(t)\cos w_c t \pm x_h(t)\sin w_c t\right]$

$S_i = (\frac{1}{2}CV_c)^2 \frac{P}{2} + (\frac{1}{2}CV_c)^2 \frac{P}{2}$

$= \frac{1}{4}C^2V_c^2 \frac{P}{2} + \frac{1}{4}C^2V_c^2\frac{P}{2}$

$= \frac{C^2V_c^2P}{8} + \frac{C^2V_c^2P}{8}$

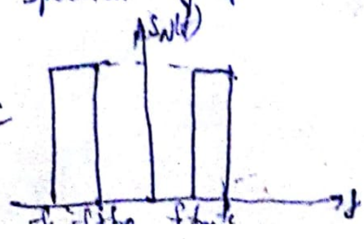
$S_i = \frac{C^2V_c^2P}{4}$

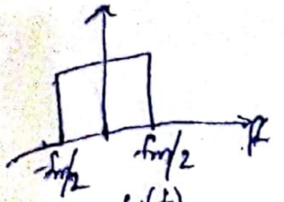
$N_i = 2 \times \frac{N_0}{2} \times f_m = N_0 f_m$

$(SNR)_i = \frac{S_i}{N_i} = \frac{C^2V_c^2P}{4N_0f_m}$ — ①

1) Shift the spectrum left $\frac{f_m}{2}$
2) Shift the spectrum right $f_m/2$
3) Divide spectrum by 4

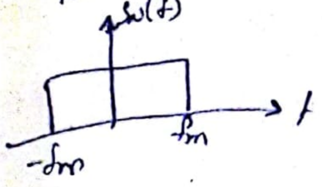
SSB-SC





$\frac{f_m}{2} - f_m = -\frac{f_m}{2}$

$-\frac{f_m}{2} + f_m = \frac{f_m}{2}$



$n(t) = n_I(t)\cos(2\pi(f_c - \frac{f_m}{2})t] + n_Q(t)\sin(2\pi(f_c - \frac{f_m}{2})t]$

$\underbrace{s(t)+n(t)}_{m(t)} = \left[\left[\left(\frac{1}{2}cV_c x(t)\cos\omega_c t + \frac{1}{2}cV_c x(t)\sin\omega_c t\right) + \left[n_I(t)\cos 2\pi(f_c - \frac{f_m}{2})t\right] + n_Q(t)\sin[2\pi(f_c - \frac{f_m}{2})t]\right]\right] \cos\omega_c t.$

$\cos^2\theta = \frac{1+\cos 2\theta}{2}$, $\sin 2A = 2\sin A\cos B$

$\cos A\cos B = \frac{1}{2}[\cos(A+B) + \cos(A-B)]$ ; $\sin A\cos B = \frac{1}{2}(\sin(A+B) + \sin(A-B))$

$m(t) = \frac{1}{2}V_c c x(t)\cos\omega_c t\cos\omega_c t + \frac{1}{2}cV_c x(t)\sin\omega_c t\cos\omega_c t + n_I(t)\cos 2\pi(f_c - \frac{f_m}{2})t\cos\omega_c t + n_Q(t)\sin(2\pi f_c - \frac{f_m}{2}t)\cos\omega_c t.$

$= \frac{1}{4}V_c c x(t)[\cos(2\omega_c t] + \cos(0)] + \frac{1}{4}cV_c x(t)[\sin(2\omega_c t) + \sin(0)] + \frac{1}{2}n_I(t)[\cos(\omega_c - \frac{\omega_m}{2} + \omega_c t) + \cos(\omega_c - \frac{\omega_m}{2} - \omega_c t)] + \frac{1}{2}n_Q(t)[\sin(\omega_c - \frac{\omega_m}{2} + \omega_c)t + \sin(\omega_c - \frac{\omega_m}{2} - \omega_c t)]$

$\Rightarrow \frac{1}{4}V_c c x(t)\cos 2\omega_c t + \frac{1}{4}V_c c x(t) + \frac{1}{4}cV_c x(t)\sin 2\omega_c t + \frac{1}{2}n_I(t)[\cos 2\omega_c - \frac{\omega_m}{2}] + \frac{1}{2}n_I(t)\cos(\frac{\omega_m}{2}) + \frac{1}{2}n_Q(t)[\sin(2\omega_c - \frac{\omega_m}{2})] + \frac{1}{2}n_Q(t)[\sin\frac{\omega_m}{2}]$

$= \frac{1}{4}V_c c x(t)\cos 2\omega_c t + \frac{1}{4}cV_c x(t) + \frac{1}{4}cV_c x(t)\sin 2\omega_c t + \frac{1}{2}n_I(t)\cos 2\omega_c t - \frac{1}{2}n_I(t)\frac{2\pi f_m}{2} + \frac{1}{2}n_I(t)\cos\left|\frac{2\pi f_m}{2}\right|t + \frac{1}{2}n_Q(t)\sin 2\omega_c t + \frac{1}{2}n_Q(t)\frac{2\pi f_m}{2} - \frac{1}{2}n_Q(t)\sin\left(\frac{2\pi f_m}{2}\right)$

$= \cancel{\frac{1}{4}V_c c x(t)\cos 2\omega_c t + \frac{1}{4}V_c x(t) + \frac{1}{4}cV_c x(t)\sin 2\omega_c t - \frac{1}{2}n_I(t)\pi f_m t + \frac{1}{2}n_I(t)\cos\pi f_m t + \frac{1}{2}n_Q(t)\sin 2\omega_c t - \frac{1}{2}n_Q(t)\pi f_m t - \frac{1}{2}n_Q(t)\sin\pi f_m t}$

$= \frac{1}{4}V_c c\,\cancel{x(t)\cos 2\omega_c t} + \frac{1}{4}cV_c x(t) + \frac{1}{4}cV_c x(t)\sin 2\omega_c t + \frac{1}{2}n_I(t)\cancel{\cos 2\omega_c t} + \cancel{\frac{1}{2}n_Q(t)} - \frac{1}{2}n_I(t)\pi f_m t + \frac{1}{2}n_I(t)\cos\pi f_m t + \frac{1}{2}n_Q(t)\sin 2\omega_c t - \frac{1}{2}Q(t)\pi f_m t - \frac{1}{2}n_Q(t)\sin\pi f_m t$

~~After~~ The above expression is passed through a Low pass filter, the filter attenuates unwanted expression & allows wanted expression. After LPF o/p is

$y(t) = \frac{1}{4}cV_c x(t) + \frac{1}{2}n_I(t)\cos(\pi f_m t) + \frac{1}{2}n_Q(t)\sin(\pi f_m t)$

This is the ~~eq~~ required o/p of LPF.

$y(t) = \frac{1}{4}(V_c\, x(t) + \frac{1}{2} n_I(t)\cos(\pi f_m t) + \frac{1}{2} n_Q(t)\sin(\pi f_m t)$

$S_0$ = Avg signal power at the o/p

$S_0 = \left(\frac{1}{4} C A_c\right)^2 \frac{P}{2} = \frac{C^2}{16} V_c^2 \frac{P}{2} = \frac{C^2 V_c^2 P}{32}$ — (2)

$N_0 = \left(\frac{1}{2}\right)^2 \frac{N_0}{4} f_m + \left(\frac{1}{2}\right)^2 \frac{N_0}{4} f_m$

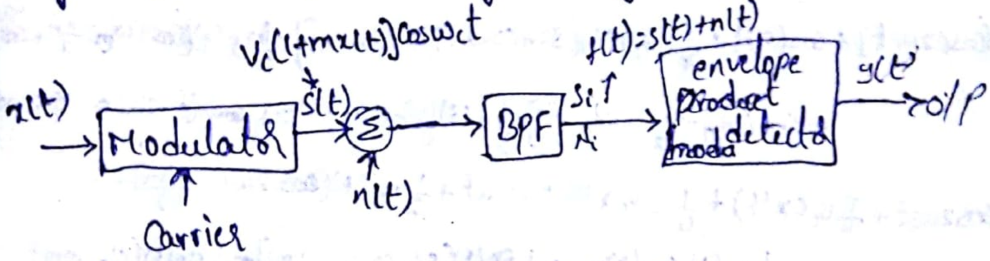
$N_0 = \frac{1}{4}\frac{N_0 f_m}{4} + \frac{1}{4}\frac{N_0}{4} f_m = \frac{N_0 f_m}{8}$

$(SNR)_0 = \frac{S_0}{N_0} = \frac{C^2 V_c^2 P}{32_4} \times \frac{8}{N_0 f_m} = \frac{C^2 V_c^2 P}{4 N_0 f_m}$ — (3)

$\gamma = \frac{(SNR)_0}{(SNR)_i} = \frac{\frac{C^2 V_c^2 P}{4 N_0 f_m}}{\frac{C^2 V_c^2 P}{4 N_0 f_m}} = 1$

$\boxed{\gamma = 1}$

* figure of merit for AM system using ~~coherent~~ Envelope detection:-




$s(t) = V_c[1 + m x(t)]\cos\omega_c t$

$(SNR)_i = ?$ $\quad V_c\cos\omega_c t = \frac{V_c^2}{2}$

$V_c m x(t)\cos\omega_c t = \frac{V_c^2}{2}\frac{m^2}{V_m^2} P$

$S_i = \frac{V_c^2}{2} + \frac{V_c^2}{2}\frac{m^2}{V_m^2} P$

$S_i = \frac{V_c^2}{2}\left[1 + \left(\frac{m}{V_m}\right)^2 P\right]$

$N_i = 2 \times \frac{N_0}{2} \times 2 f_m$

$N_i = 2 N_0 f_m$

$(SNR)_i = \frac{S_i}{N_i} = \frac{\frac{V_c^2}{2}\left[1 + \left(\frac{m}{V_m}\right)^2 P\right]}{2 N_0 f_m}$ → (1)

$(SNR)_0 = ?$

$f(t) = s(t) + n(t) = V_c[1 + m x(t)]\cos\omega_c t + \underbrace{n_I(t)\cos\omega_c t - n_Q(t)\sin\omega_c t}_{n(t)}$

$f(t) = \cos\omega_c t\,[V_c + V_c m x(t) + n_I(t)] - n_Q(t)\sin\omega_c t$

$\uparrow$ i/p for envelope detector.

o/p $y(t) = \sqrt{(V_c + V_c m x(t) + n_I(t))^2 + n_Q(t)^2}$ $n_Q(t)$ is zero, because envelope detecto allows only inphase components.

$y(t) \cong V_c + V_c m x(t) + n_I(t)$

$y(t) = V_c m x(t) + n_I(t)$ $V_c = 0$

$S_0 = \frac{V_c^2}{2} P \frac{m^2}{V_m^2}$

$N_0 = 2 \times \frac{N_0}{2} \times 2 f_m$

$N_0 = 2 N_0 f_m$

$$\frac{S_0}{N_0} = (SNR)_0 = \frac{\frac{V_c^2}{2} P \left(\frac{m}{V_m}\right)^2}{2 N_0 f_m}$$

$$\gamma = \frac{(SNR)_0}{(SNR)_i} = \frac{\frac{\frac{V_c^2}{2} P \left(\frac{m}{V_m}\right)^2}{2 N_0 f_m}}{\frac{\frac{V_c^2}{2}\left[1 + \left(\frac{m}{V_m}\right)^2 P\right]}{2 N_0 f_m}}$$

$$\boxed{\gamma = \frac{P\left(\frac{m}{V_m}\right)^2}{1 + \left(\frac{m}{V_m}\right)^2 P}}$$

1) Calculate the figure of merit for AM using envelope detector for single tone AM.

A Single tone AM.

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

$x(t) = P$

Avg power of $x(t) = P = \frac{V_m^2}{2}$

$$(SNR)_i = \frac{\frac{V_c^2}{2}\, 1 + \left(\frac{m^2}{V_m^2}\right) \frac{V_m^2}{2}}{2 N_0 f_m} = \frac{\frac{V_c^2}{2} \cdot 1 + \frac{m^2}{2}}{2 N_0 f_m}$$

$$(SNR)_0 = \frac{\frac{V_c^2}{2} \frac{V_m^2}{2} \frac{m^2}{V_m^2}}{2 N_0 f_m} = \frac{\frac{V_c^2}{2} \cdot \frac{m^2}{2}}{2 N_0 f_m}$$

for 100% modulation $m = 1$

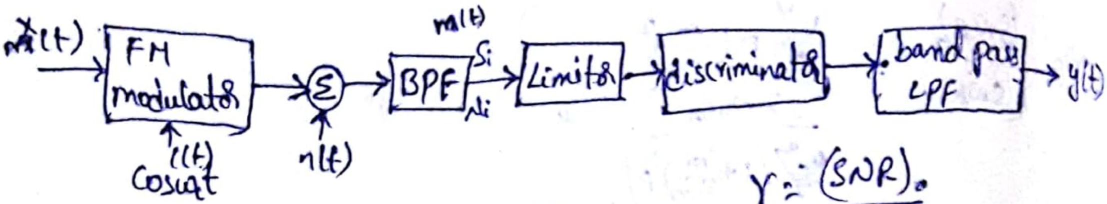
$$\gamma = \frac{(SNR)_0}{(SNR)_i} = \frac{\frac{\frac{V_c^2}{2} \frac{m^2}{2}}{2 N_0 f_m}}{\frac{\frac{V_c^2}{2} \cdot (1 + \frac{m^2}{2})}{2 N_0 f_m}} = \frac{\frac{V_c^2}{2} \cdot \frac{m^2}{2}}{\frac{V_c^2}{2} \cdot 1 + \frac{m^2}{2}}$$

$$\gamma = \frac{\frac{m^2}{2}}{1+\frac{m^2}{2}} \qquad \text{Sub } m=1$$

$$= \frac{1/2}{1+\frac{1}{2}} = \frac{1}{2+1}$$

$$\boxed{\gamma = 1/3}$$

## * Noise in FM Receivers:-




$$\gamma = \frac{(SNR)_o}{(SNR)_i}$$

$$n(t) = n_I(t)\cos\omega_c t - n_Q(t)\sin\omega_c t$$

$$r(t) = \sqrt{n_I^2(t) + n_Q^2(t)}$$

$$\psi(t) = \tan^{-1}\left(\frac{n_Q(t)}{n_I(t)}\right)$$

$$n(t) = r(t)\cos(\omega_c t + \psi(t)) \quad -①$$

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

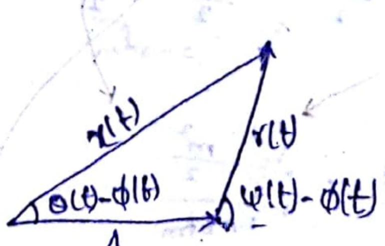
Assume $\phi(t) = 2\pi k_f\int_0^t x(t)dt$

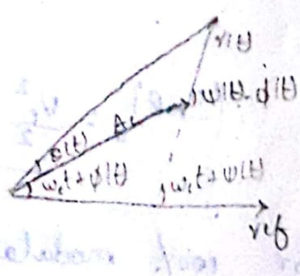
$$s(t) = A_c\cos(\omega_c t + \phi(t)) \quad -②$$

Add ① & ②

$$x(t) = s(t) + n(t)$$

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




$$\theta(t) - \phi(t) = \tan^{-1}\left[\frac{r(t)\sin(\psi(t)-\phi(t))}{A_c + r(t)\cos(\psi(t)-\phi(t))}\right]$$

$$= \frac{r(t)}{A_c}\sin(\psi(t)-\phi(t))$$

independent of x(t)

$$\theta(t) = \frac{r(t)}{A_c}\sin(\psi(t)-\phi(t)) + \phi(t)$$

$$\theta(t) = \frac{r(t)}{A_c}\sin\psi(t) + \phi(t)$$

$$\theta(t) = 2\pi k_f \int_0^t x(t)\,d(t) + \frac{r(t)}{A_c}\sin\psi(t)$$

$$\frac{1}{2\pi}\frac{d\theta(t)}{dt} = \frac{1}{2\pi}\,2\pi k_f\,x(t) + \frac{1}{2\pi A_c}\underbrace{\frac{d}{dt}\,r(t)\sin\psi(t)}_{n_Q(t)}$$

$$\frac{1}{2\pi}\frac{d\theta(t)}{dt} = k_f x(t) + \frac{1}{2\pi A_c}[n_Q(t)]$$

$$v(t) = k_f x(t) + n_d(t)$$

$$S_0 = k_f^2 P \quad -(3)$$

$$\frac{d}{dt}x(t) \xleftrightarrow{FT} j2\pi f\, X(F)$$

$$H(F) = \frac{1}{2\pi A_c}\times j2\pi f = \frac{jf}{A_c}$$

$$|H(f)|^2 = \frac{f^2}{A_c^2}$$

$$P = \int_{-f_m}^{f_m} S_{N_0}(f)\,df$$

$$N_0 = P = \frac{N_0}{\cancel{2}}\frac{1}{A_c^2}\int_{-f_m}^{f_m} f^2\,d(t) = \frac{N_0}{\cancel{2}}\frac{1}{A_c^2}\left[\frac{F^3}{3}\right]_{-f_m}^{f_m} = \frac{N_0}{\cancel{2}}\frac{1}{A_c^2}\frac{2f_m^3}{3} \quad -(4)$$

sub (3) & (4) in $(SNR)_0$

$$(SNR)_0 = \frac{S_0}{N_0} = \frac{3k_f^2 P A_c^2}{2N_0 f_m^3} \quad -(5)$$

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

$$S = \frac{A_c^2}{2} \quad -(6)$$

$$n_i = \frac{N_0}{2}\times 2f_m = N_0 f_m \quad -(7)$$

sub (6), (7) in $(SNR)_i$

$$(SNR)_i = \frac{A_c^2}{2N_0 f_m} \quad -(8)$$

sub (5), (8) in $\gamma$

$$\gamma = \frac{(SNR)_0}{(SNR)_i} = \frac{3k_f^2 P A_c^2 / 2N_0 f_m^3}{\frac{A_c^2}{2N_0 f_m}}$$

$$= \frac{3k_f^2 P A_c^2}{2N_0 f_m^3}\times\frac{2N_0 f_m}{A_c^2}$$

$$\boxed{\gamma = \frac{3k_f^2 P}{f_m^2}}$$

Calculate figure of merit for single tone FM:-

$$A_c\cos\left(\omega_c t + \frac{\Delta f}{f_m}\sin\omega_m t\right)$$

$$\frac{\Delta f}{f_m}\sin\omega_m t = 2\pi k_f\int_0^t x(t)\,dt$$

Diff on b.s

$$\omega_m \frac{\Delta f}{f_m} \cos\omega_m t = 2\pi k_f x(t)$$

$$2\pi f_m \cdot \frac{\Delta f}{f_m} \cos\omega_m t = 2\pi k_f x(t)$$

$$\Delta f \cos\omega_m t = k_f x(t)$$

$$x(t) = \frac{\Delta f}{k_f} \cos\omega_m t$$

Avg power of the modulating signal across $1\Omega$ resistor is

$$P = \left(\frac{\Delta f}{k_f}\right)^2 \frac{1}{2}$$

$$k_f^2 P = \frac{(\Delta f)^2}{2}$$

$$(SNR)_o = \frac{3}{2}\left(\frac{k_f^2 P A_c^2}{N_0 f_m^3}\right) = \frac{3}{2}\left[\frac{\frac{(\Delta f)^2}{2} \cdot A_c^2}{N_0 f_m^3}\right]$$

$$(SNR)_i = \frac{A_c^2}{2N_0 f_m}$$

$$\gamma = \frac{(SNR)_o}{(SNR)_i} = \frac{\frac{3}{2}\left[\frac{(\Delta f)^2}{2} \cdot A_c^2\right] / N_0 f_m^3}{\frac{A_c^2}{2N_0 f_m}} = \frac{3\left[\frac{(\Delta f)^2}{2} \cdot A_c^2\right]}{2N_0 f_m^3} \times \frac{2N_0 f_m}{A_c^2}$$

$$\gamma = \frac{3\frac{(\Delta f)^2}{2}}{\frac{f_m^2}{1}} = \frac{3(\Delta f)^2}{2} \times \frac{1}{f_m^2}$$

$$= \frac{3}{2}\left(\frac{\Delta f}{f_m}\right)^2$$

$$\boxed{\gamma = \frac{3}{2}(m_f)^2}$$

## UNIT-V PULSE MODULATION

**SAMPLING:** The process of converting an analog signal into a discrete signal is known as sampling.

Figure 8.1shows how this conversion can be done. As shown in the figure 8.1a switch position is controlled by the sampling signal. The sampling signal is a periodic train of pulses of unit amplitude and of period $T_s$. The time $T_s$ is known as sampling time and during this time switch is closed so that sampled signal is equal to the input signal. During remaining time switch is open and no input signal appear at the output.

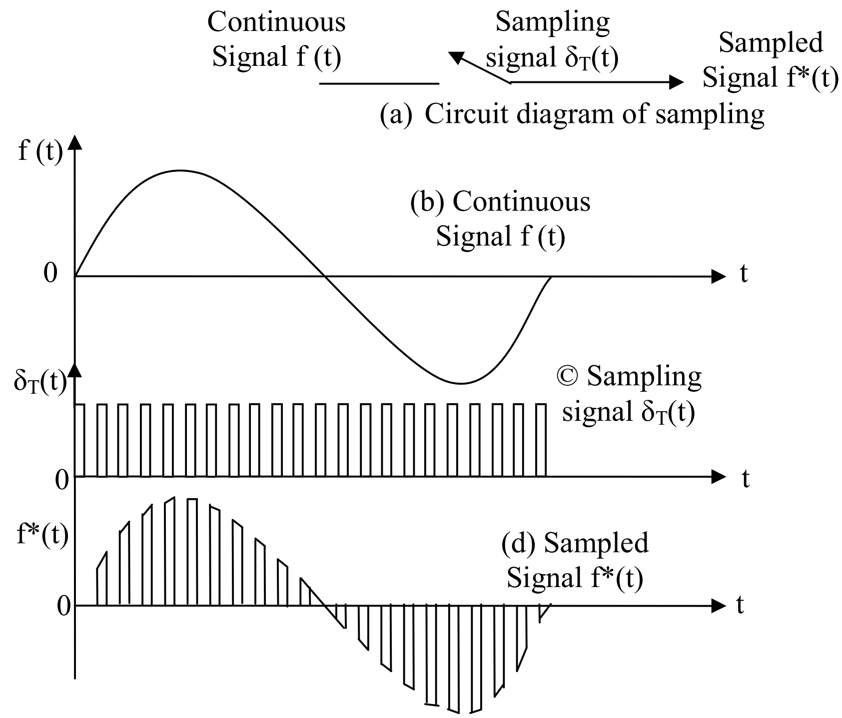
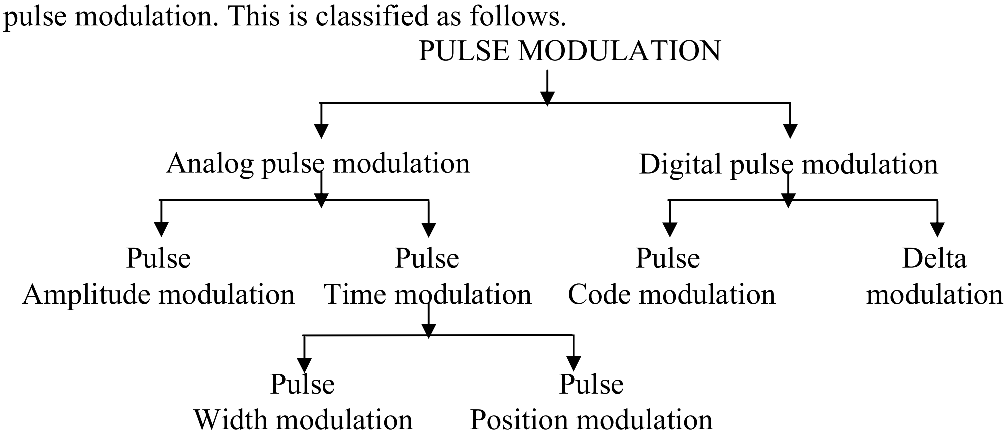



Fig 8.1: Sampling process

**SAMPLING THEOREM:** The sampling theorem states that if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed in the receiver with minimal distortion.

**PULSE MODULATION:** The process of changing any one of the characteristics of train of pulse in according to the amplitude of modulating signal at the time of sampling is called pulse modulation. This is classified as follows.




**GENERATION OF PAM**: The process of changing amplitude of the train of pulse in according to the amplitude of modulating signal at the time of sampling is called pulse amplitude modulation.

The figure 8.2a shows the block schematic of PAM generator. It consists of a low

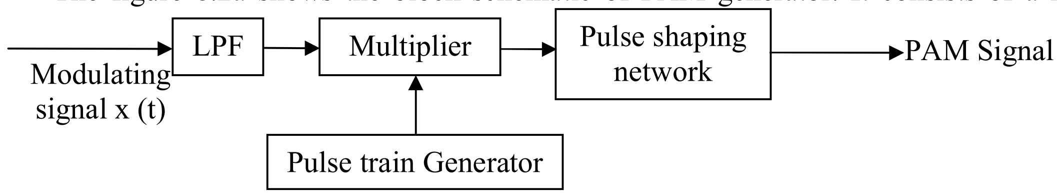



Fig 8.2a: Generation of PAM Signal

Pass filter, a multiplier and a pulse train generator. Initially, the modulating signal x (t) is passed through the low pass filter (LPF). The LPF removes all the frequency components which are higher than frequency $f_m$. This is known as band limiting. The band limiting is necessary to avoid the aliasing effect in the sampling process. The pulse train generator generates a pulse train at a frequency $f_s$, such that $f_s \geq 2f_m$. Thus the Nyquist criterion is satisfied. The pulse sampling network does the shaping work to give flat tops. The figure 8.2b, c, d and e show the waveforms related to the generation of PAM generator.

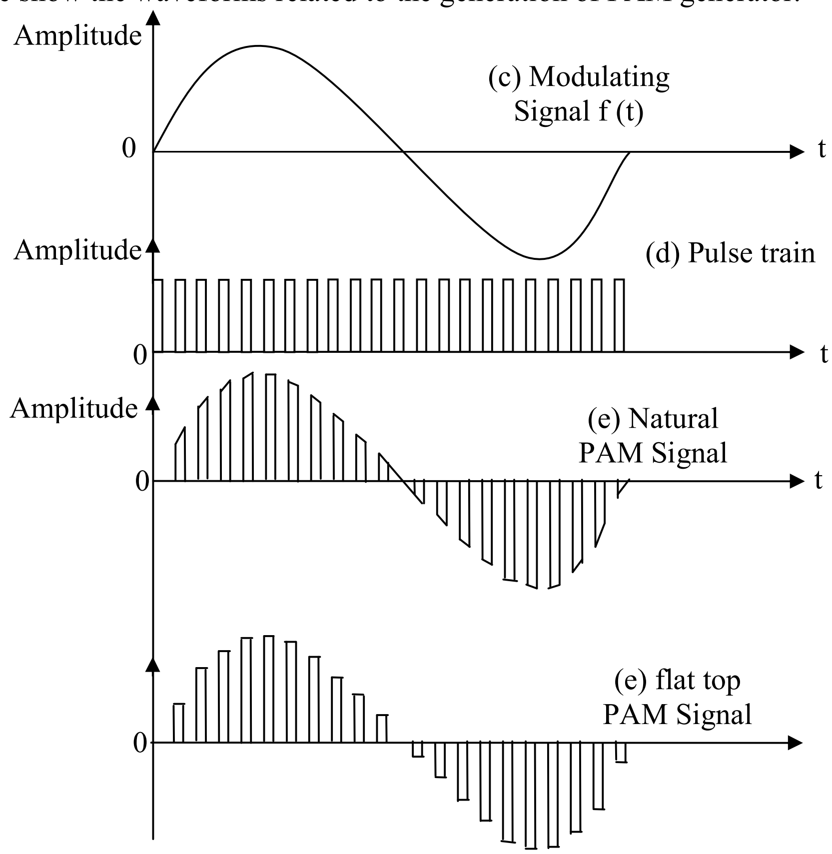



Fig 8.2b: waveforms of PAM

**<u>DETECTION OF PAM</u>**: The original modulating signal can be detected from the natural PAM by passing naturally modulated PAM through a diode detector and a low pass filter. The diode detector detects the envelope of the PAM signal. The low pass filter with cut-off frequency equal to $f_m$ removes high frequency ripple and recovers the original modulating signal. This is illustrated in figure 8.3a.

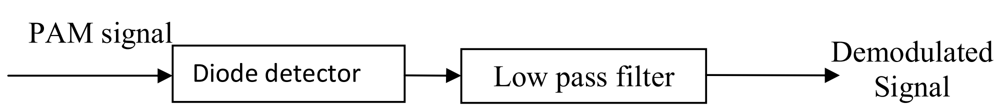



Fig 8.2a: Detection of natural PAM Signal

The demodulated output shown in figure 8.3b is close to the original modulating signal.

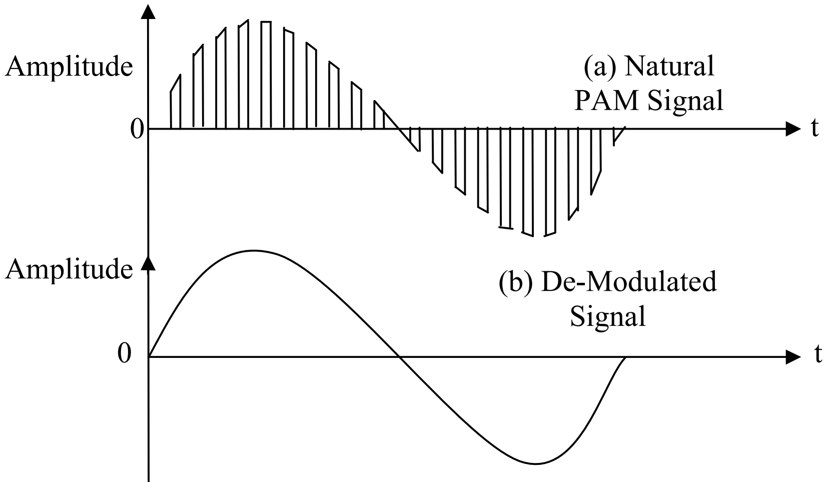



Fig 8.3b: waveforms of PAM detection

In case of flat top PAM to reduce aperture effect, an equalizer is used. As shown in figure 8.3c the receiver consists of low-pass reconstruction filter with cutoff frequency slightly higher than the maximum frequency of message signal. The equalizer compensates the aperture effect. It also compensates the attenuation by a low pas reconstruction filter.

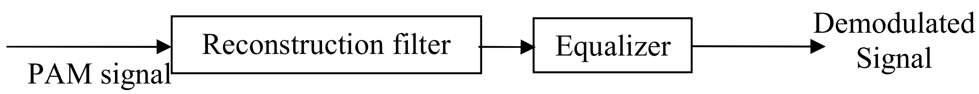



Fig 8.3c: Detection of flat top PAM Signal

**TRANSMISSION BANDWIDTH OF PAM SIGNAL**

The pulse duration $\tau$ is supposed to be very very small compared to time period $T_s$ between the two samples. If the maximum frequency in the signal x (t) is W then by sampling theorem, $f_s$ should be higher than Nyquist rate i.e. $f_s \geq 2W$

$$\frac{1}{fs} \leq \frac{1}{2W}$$

$$T_s \leq \frac{1}{2W} \qquad (\text{since } f_s = \frac{1}{Ts})$$

We know that $\tau \ll T_s$

Therefore $\tau \ll T_s \leq \frac{1}{2W}$ ------ (1)

If ON and OFF time of the pulse is same, then frequency of the PAM pulse becomes,

$f = \frac{1}{\tau+\tau} = \frac{1}{2\tau}$ ----- (2)

Thus figure 8.4 shows that if ON and OFF times of PAM signal are same. Then maximum frequency of PAM signal is given by equation 2 i.e.,

$f_{max} = \frac{1}{2\tau}$ ---- (3)

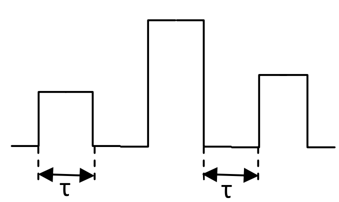



Fig 8.4: maximum frequency of PAM Signal

Therefore bandwidth required for transmission of PAM signal will be equal to maximum frequency $f_{max}$ given by equation (3). This bandwidth gives adequate pulse resolution i.e.,

$$B_T \geq f_{max}$$

Therefore $\qquad B_T \geq \frac{1}{2\tau} \quad$ Since $\tau \ll \frac{1}{2W}$

$$B_T \geq \frac{1}{2\tau} \gg W$$

Transmission bandwidth of PAM signal: $B_T \gg W$

Thus the transmission bandwidth $B_T$ of PAM signal is very very large compared to highest frequency in the signal x(t).

**ADVANTAGES OF PAM:**

1. Generation and detection of PAM is simple

**DISADVATAGES OF PAM:**

1. PAM is less immune to noise.
2. It requires larger transmission power.

**CLASSIFICATION OF PAM BASED ON SIGNAL POLARITY**

The PAM signal can be classified according to signal polarity as Single polarity PAM and Double polarity PAM

The figure shows the single polarity PAM Here, a fixed d.c. level is added to the modulating signal x(t), such that the modulated output i.e. PAM signal is always positive.

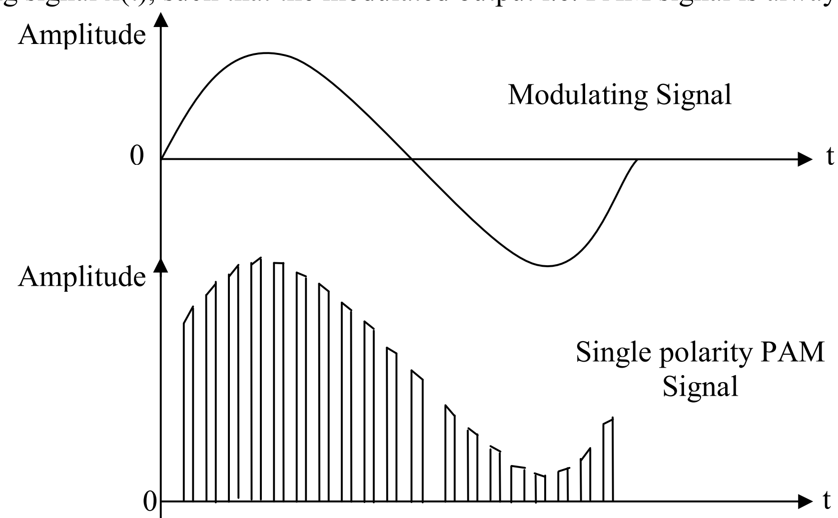



Fig 8.5a: waveforms of single polarity PAM

In double polarity PAM signal, signal has positive as well as negative polarity. It is shown in figure 8.5b.

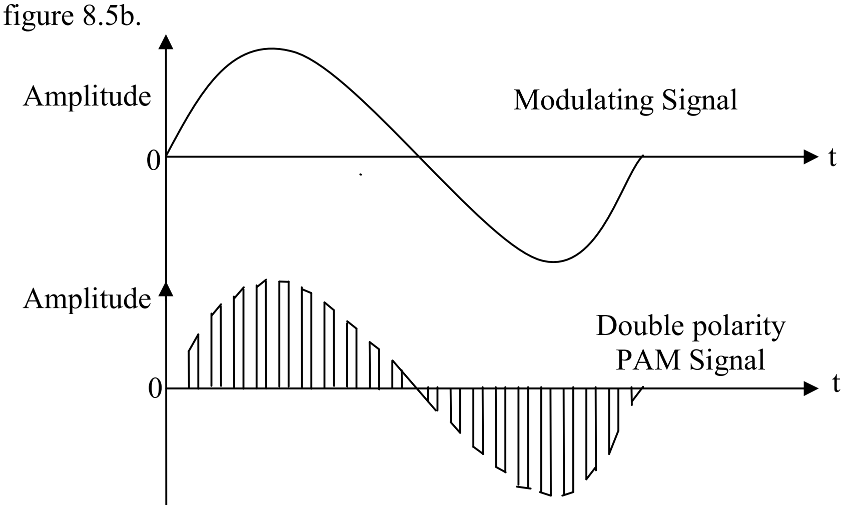



Fig 8.5b: waveforms of double polarity PAM

**GENERATION OF PWM:** The process of changing the width of the train of pulse in according to the amplitude of modulating signal at the time of sampling is called pulse width modulation. Figure 8.6a shows monostable multivibrator circuit to generate pulse width modulated wave.

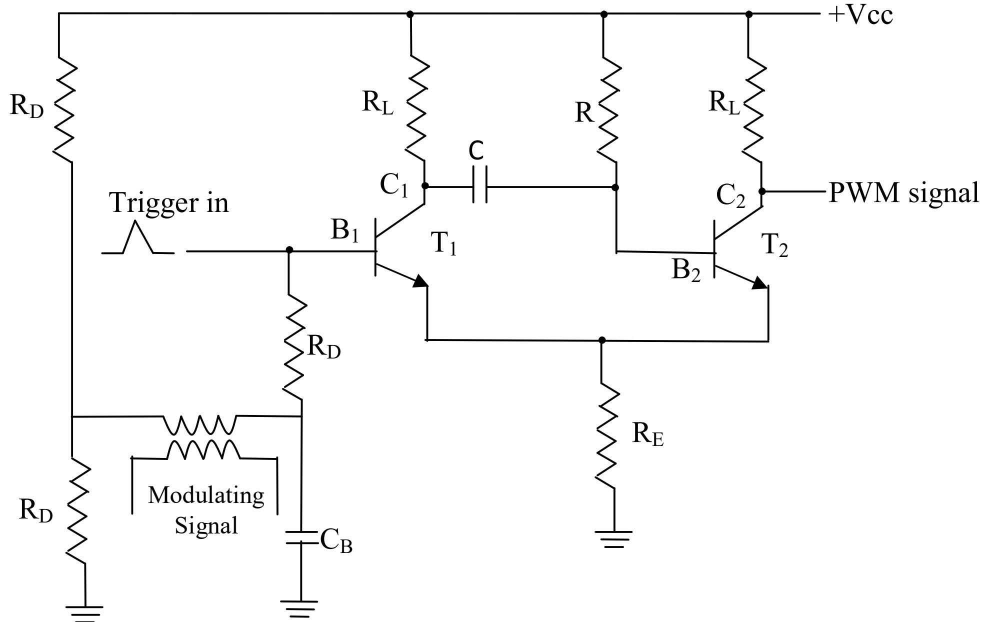



Fig 8.6a: monostable multivibrator circuit to generate pulse width modulated wave.

The stable state for above circuit is achieved when $T_1$ is OFF and $T_2$ is ON. The positive going trigger pulse at $B_1$ switches $T_1$ ON. Because of this, the voltage at $C_1$ falls as $T_1$ now begins to draw the collector current. As a result, voltage at $B_2$ also falls and $T_2$ is switched OFF, C begins to charge up to the collector supply voltage ($V_{CC}$) through resistor R. After a time determined by the supply voltage and the RC time constant of the charging network, the base of the $T_2$ becomes sufficiently positive to switch $T_2$ ON. The transistor $T_1$ is simultaneously switched OFF by regenerative action and stays OFF until the arrival of the next trigger pulse. To make $T_2$ ON, the base of the $T_2$ must be slightly more positive than the voltage across resistor $R_E$. This voltage depends on the emitter current $I_E$ which is controlled by the signal voltage applied at the base of transistor $T_1$. Therefore, the changing voltage necessary to turn OFF transistor $T_2$ is decided by the signal voltage. If signal voltage is maximum, the voltage that capacitor should charge to turn ON $T_2$ is also maximum. Therefore, at maximum signal voltage, capacitor has to charge to maximum voltage requiring maximum time to charge. This gives us maximum pulse width at maximum input signal voltage. At minimum signal voltages, capacitor has to charge for minimum voltage and we get minimum pulse width at the output. With this discussion, we can say that pulse width is controlled by the input signal voltage, and we get pulse width modulated waveform at the output. The waveforms of PWM are shown in figure 8.6b.

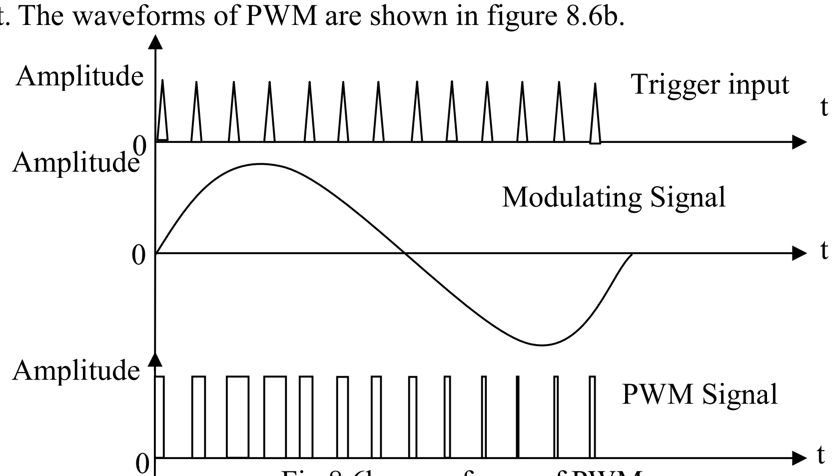



Fig 8.6b: waveforms of PWM

**DEMODULATION OF PWM SIGNAL**:

Figure 8.7a shows the block diagram of PWM detector.

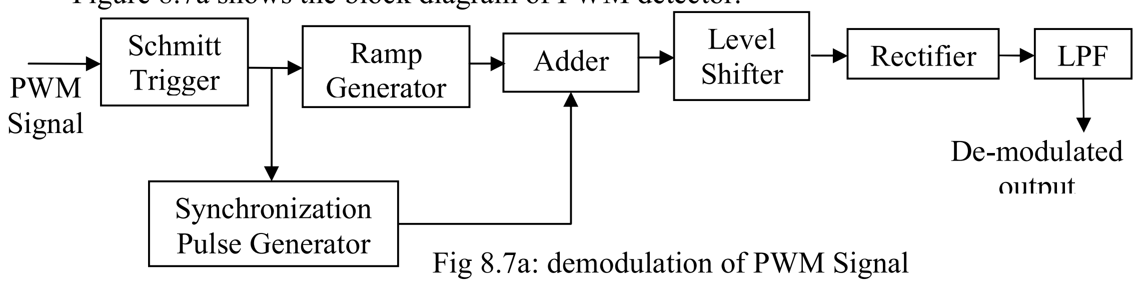



Fig 8.7a: demodulation of PWM Signal

The received PWM signal is applied to the Schmitt trigger circuit. This Schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse detector. The ramp generator produces ramps for the duration of pulses such that heights of ramps are proportional to the widths of PWM pulses. The maximum ramp voltage is retained till the next pulse. The synchronous pulse detector produces reference pulses with constant amplitude and pulse width. These pulses are delayed by specific amount of delay as shown in the figure 8.7b. The delayed reference pulses and the output of ramp generator are added with the help of adder. The output of adder is given to the level shifter. Here, negative offset shifts the waveform as shown in the figure 8.7b. Then the negative part of the waveform is clipped by rectifier. Finally, the output of rectifier is passed through low-pass filter to recover the modulating signal, as shown in the figure 8.7b.

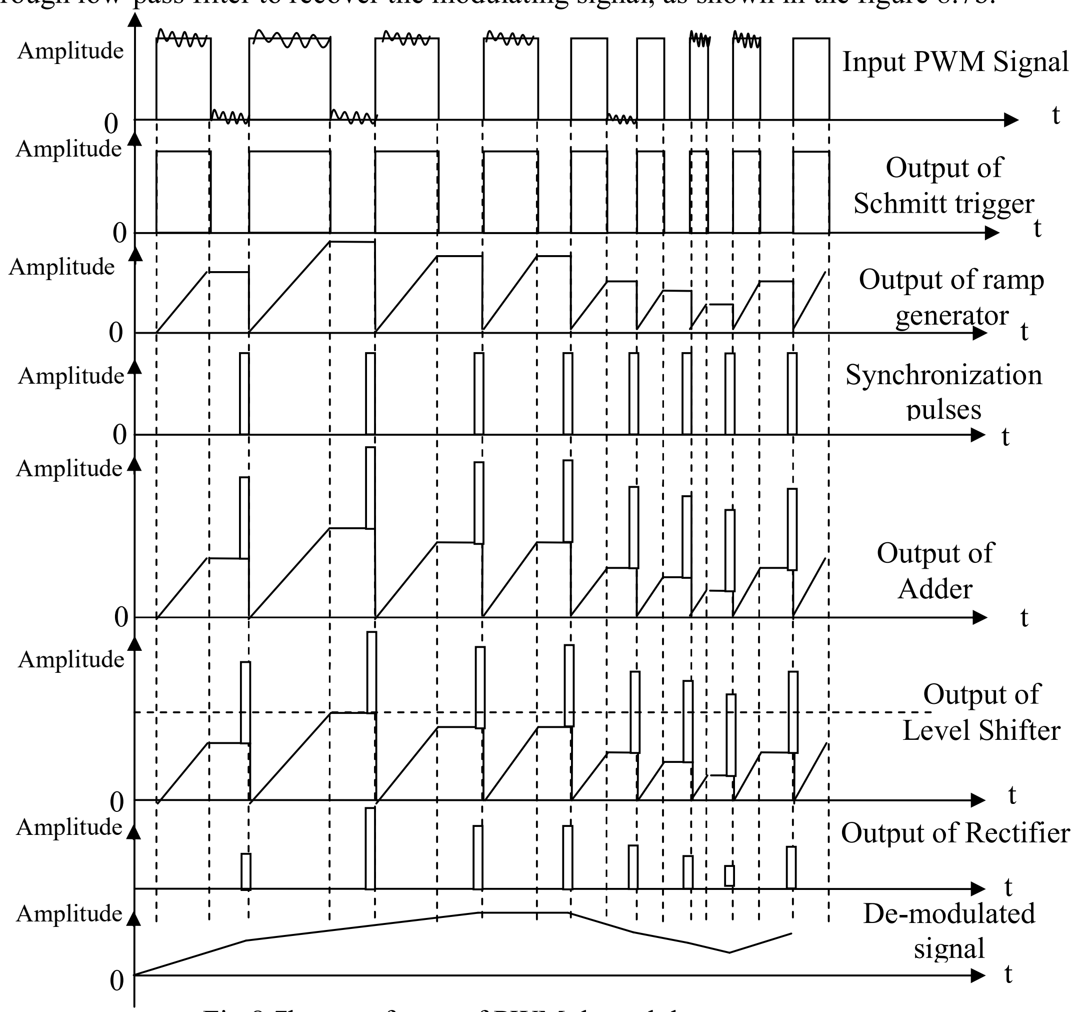



Fig 8.7b: waveforms of PWM demodulator

**ADVANTAGES OF PWM:**

1. noise is less
2. Signal and noise separation is very easy
3. PWM communication does not require synchronization between transmitter and receiver.

**DISADVANTAGES OF PWM:**

1. It requires larger transmission power.
2. Large bandwidth is required for the PWM communication as compared to PAM.

**GENERATION OF PPM:** The process of changing the position of the train of pulse in according to the amplitude of modulating signal at the time of sampling is called pulse position modulation. Figure 8.8a shows the block diagram to generate pulse width modulated wave.

Figure 8.8a shows the PPM generator. It consists of differentiator and a monostable multivibrator. The input to the differentiator is a PWM waveform. The differentiator generates positive and negative spikes corresponding to leading and trailing edges of the PWM waveform. Diode $D_1$ is used to bypass the positive spikes. The negative spikes are used to trigger the monostable multivibrator. The monostable multivibrator then generates the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform, as shown in the figure 8.8b.

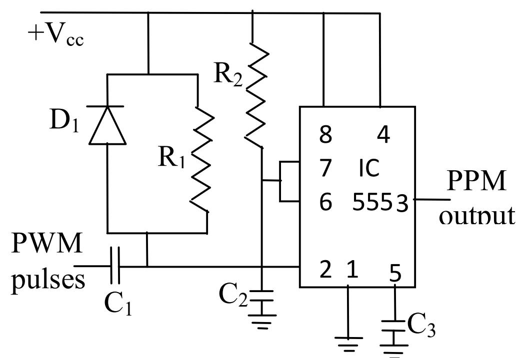



Fig 8.8a: Generation of PPM Signal

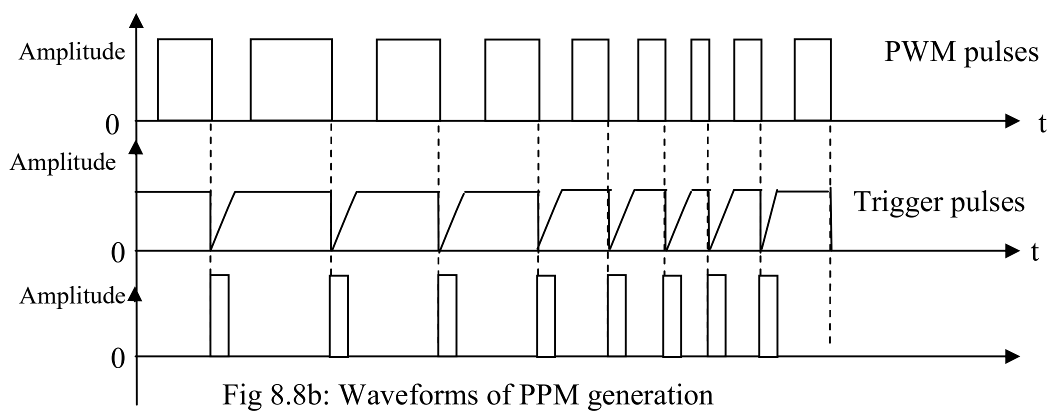



Fig 8.8b: Waveforms of PPM generation

**DEMODULATION OF PPM:**

In pulse position de-modulation, it is required to convert the received pulses that vary in position into pulses that vary in length. The block diagram of PPM demodulator is shown in figure 8.9a.

As shown in figure 8.9a, flip-flop circuit is set or turned 'ON' giving high output when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip-flop circuit is reset or turned 'OFF' giving low output at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip-flop.

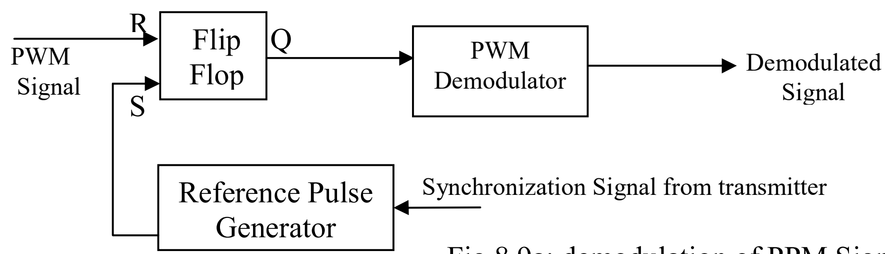



Fig 8.9a: demodulation of PPM Signal

The PWM pulses are then demodulated by PWM demodulator to get original modulating signal. The waveforms of PPM demodulation is shown in figure 8.9b.

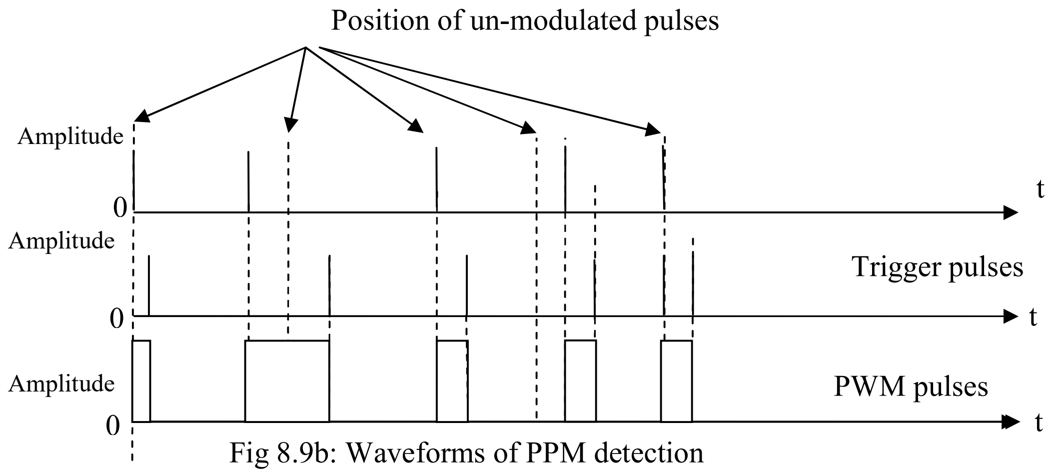



Fig 8.9b: Waveforms of PPM detection

**ADVANTAGES OF PPM**:
1. Noise is less.
2. Signal and noise separation is very easy.
3. Transmission power for each pulse is same.

**DISADVANTAGES OF PPM:**
1. Synchronization between transmitter and receiver is required.
2. Large bandwidth is required as compared to PAM.

**PERFORMANCE COMPARISON OF VARIOUS PULSE ANALOG MODULATION METHODS:**

| S.NO. | PAM | PWM | PPM |
|---|---|---|---|
| 1 | | | |
| 2. | Amplitude of the pulse is proportional to the amplitude of the modulating signal. | Width of the pulse is proportional to amplitude of modulating signal. | The relative position of the pulse is proportional to the amplitude of modulating signal. |
| 3 | Bandwidth of the transmission channel depends on width of the pulse. | Bandwidth of the transmission channel depends on rise time of the pulse. | Bandwidth of the transmission channel depends on rising time of the pulse. |
| 4 | The instantaneous power of the transmitter varies. | The instantaneous power of the transmitter varies | The instantaneous power of the transmitter remains constant. |
| 5 | Noise interference is high. | Noise, interference is minimum | Noise, interference is minimum |
| 6 | Similar to amplitude modulation | similar to frequency modulation | similar to phase modulation. |

## SYNCHRONIZATION IN PULSE MODULATION:

Most pulse modulation systems require synchronization of the receiver to the transmitter. Generally start stop method of synchronization is used. We maintain synchronization on a per frame bases. This method involves transmitting some information in addition to the message bearing pulses, to serve as a time mark with in each frame interval so that certain gates in the receiver structure may be made to open and close at the appropriate instant of time. In some cases the necessary time mark is established by transmitting a distinctive marker per frame, where as in other cases it is established by omitting a pulse in that particular time slot. When markers are used, they must differ from the message bearing pulses in some recognizable fashion.

In PAM system the marker pulse may be identified by making its amplitude more than that of all possible message pulses as shown in figure 8.10a for a PAM system involving three independent message sources. Figure 8.10b shows the sampling times of the message sources and the times of synchronization or marker pulses. Such a marker can be located at the receiver by applying the received pulses to a slicer. With a slicing level that is just in excess of the maximum amplitude of the message pulses so that these pulses produce zero output. An ideal slicer has the property that its output is zero whenever the input exceeds this level as shown in figure 8.11. The pulses observed at the slicer output will thus be due to the markers only.

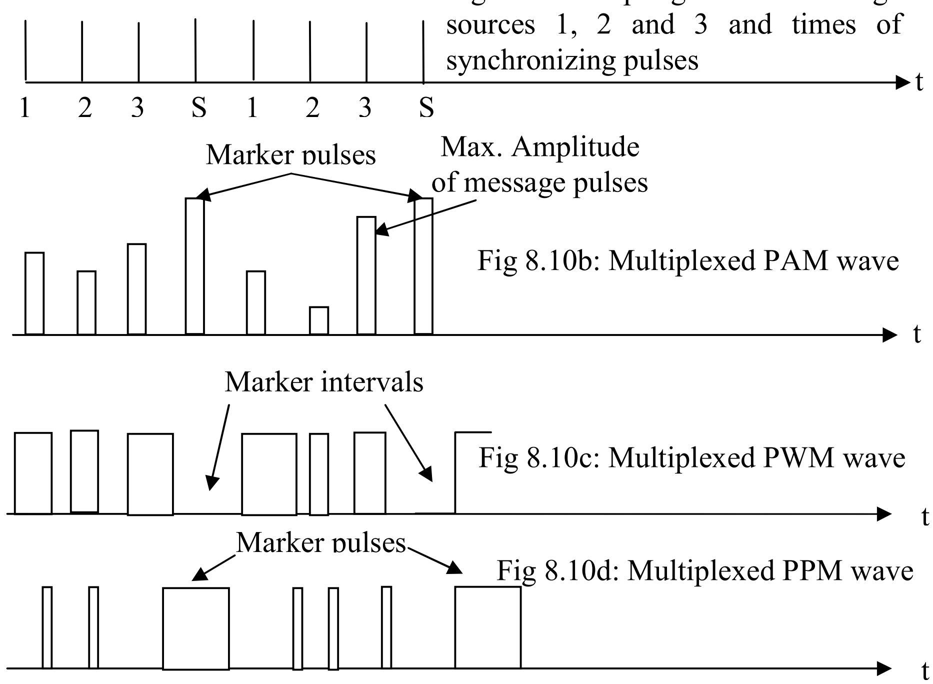



Fig 8.10a: sampling times of message sources 1, 2 and 3 and times of synchronizing pulses

Fig 8.10b: Multiplexed PAM wave

Fig 8.10c: Multiplexed PWM wave

Fig 8.10d: Multiplexed PPM wave

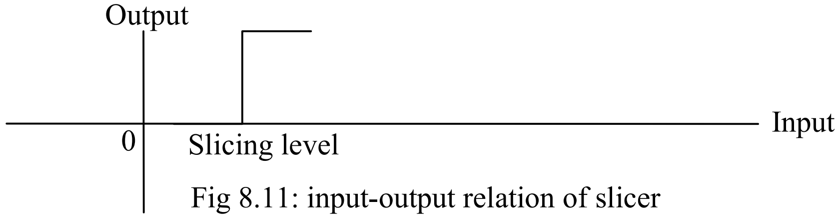



Fig 8.11: input-output relation of slicer

In PWM systems the marker may be identified by omitting a pulse as in figure 8.10c. One method of identifying such a marker in the receiver is to utilize the charging time of a simple resistor capacitor circuit to measure the duration of the intervals between PWM pulses. The time constant of the circuit is chosen so that, during a marker interval, the voltage across the capacitor rises to a value considerable higher than that during the normal charging interval. Thus, by applying the output of the circuit to a slicer with an appropriate slicing level, the presence of a marker is detected.

In a PPM system, the marker pulse may be identified by making its duration several times longer than that of the message pulses, as shown in figure 8.10d. At the receiver, the marker pulses may be separated from the message pulses by using a procedure essentially similar to that described for the PWM system. In this case, the capacitor is charged during the time of occurrence of each pulse, and discharged during the intervening intervals. Accordingly, the voltage across the capacitor reaches its highest value during the presence of a marker pulse and the marker pulses are thereby separated from the message pulses.

## SPECTRA OF PDM AND PPM WAVES:

The spectral analysis of a PDM or PPM wave is complicated. We present here only a qualitative description of the spectra of PDM and PPM waves. Let Ts denote the time separation between the leading edges of duration modulated pulses obtained by natural sampling, with the modulation superimposed on the trailing edges. Then assuming a sinusoidal modulating wave of frequency $f_m$, we find that the spectrum of a naturally sampled PDM wave consists of the following components:

1. A dc component equal to the average value of the pulses.
2. Sinusoidal components of frequencies equal to integer multiples of 1/Ts corresponding to spectral lines at ± n/Ts, where n= 1, 2, 3… These sinusoidal components as well as the dc component are by the un-modulated pulse train which may be regarded as the carrier of the PDM wave.
3. A sinusoidal component to frequency $f_m$ and in phase with the modulating wave, corresponding to spectral lines at $\pm f_m$.
4. Sinusoidal components of frequencies equal to (n/Ts) ± L $f_m$, where n, L=1, 2, 3… corresponding to pairs of side frequencies centered around each spectral line of the un modulated pulse train, except the dc component. These components represent the cross-modulation products between the sinusoidal modulation and sampling frequencies.

The message signal may be recovered by passing the PDM wave through a low pass filter. However, the reconstruction is accomplished with a certain amount of distortion caused by the cross modulation products that fall in the signal band. The frequencies of the important in band distortion components are $(1/T_s)-2f_m$, $(1/T_s)-3f_m$, and so on. To prevent undue distortion of the reconstructed message signal, it is necessary to restrict the maximum excursion of the trailing edge of a duration-modulated pulse. The output of the low pass reconstruction filter contains not only the desired message wave, but also its harmonics. With natural sampling these harmonics are missing. As with natural sampling, the other in band distortion products of the form (1/Ts) - $nf_m$ are present. Thus we may expect a net deterioration of quality in the reconstructed message signal when the sampling is uniform instead of natural.

In the case of a PPM wave, each pulse has a very small duration compared with the sampling interval $T_s$, so that it may be approximated as an impulse. Then it turns out that the spectrum of a PPM wave obtained by natural sampling, and with a sinusoidal modulating wave, is similar in form to that of a PDM wave, except that it contains a component proportional to the derivative of the modulating wave rather than the modulating wave itself. Thus we may demodulate a PPM wave by passing it through a low pass filter and then integrating it to restore the wanted signal component to its original waveform. Greater signal amplitude with less distortion can be obtained at the receiver output.

## COMPARISON OF SAMPLING TECHNIQUES OF PAM

| S.No. | Natural sampling | Flat top sampling |
|---|---|---|
| 1 | It uses chopping principle of sampling | It uses sample and hold circuit Principle |
| 2. | This method is used practically | This method is used practically. |
| 3. | Sampling rate satisfies Nyquist criteria | Sampling rate satisfies Nyquist criteria |
| 4 | Noise interference is minimum | Noise interference is maximum. |

## CROSS TALK:

The interference of the adjacent channels or overlapping of information between adjacent channels is called crosstalk. For faithful communication cross talk must be avoided. In TDM cross talk may occur due to insufficient transmission bandwidth to preserve the shape of the TDM pulses. In FDM the crosstalk may occur when frequency response of filter is not sharp enough. To eliminate or to reduce cross talk a guard band is provided between the adjacent channels.

## TIME DIVISION MULTIPLEXING:

In TDM, each intelligence signal to be transmitted (voice or telemetry data) is sampled sequentially and the resulting pulse code is used to modulate the carrier. The same carrier frequency is used to transmit different pulse sequentially, one after other. Each intelligence, to be transmitted, has been allotted a given time slot. Since only one signal modulates the carrier at any time, no added equipment and no increase in bandwidth is needed when multiplexing. The number of sequential channels that can be handled is limited by the time span required by any one channel pulse and the interval between samples.

Thus, in TDM, each signal occupies the entire bandwidth of the channel. However, each signal is transmitted for only a short period of time, as shown below..

Time Slot

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Signal 1 | Signal 2 | Signal 3 | Signal 4 | Signal 5 | Signal 1 | Signal 2 | Signal 3 |

One Frame

Time

Here five signals are time division multiplexed. Each signal is allowed to use the channel for a fixed interval of time, called time slot. The five signals use the channel sequentially one after other.

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

The concept of TDM is illustrated by the block diagram shown in figure 8.12. Each input message signal is first restricted in bandwidth by a low pass filter to remove the frequencies that are nonessential to an adequate signal representation. The low pass filter outputs are then applied to a commutator. A commutator is a rotating switch which connects the output of each channel modulator to the communication channel input in turn. The commutator is realized with electronic switches since it has to rotate at high speed. The commutator remains at each contact for an interval of time, which is the time slot allotted for each channel. Following the commutation process, the multiplexed signal is applied to a pulse modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the common channel.

At the receiving end of the system, the received signal is applied to a pulse demodulator which performs the reverse operation of the pulse modulator. The narrow samples produced at the pulse demodulator output are distributed to the appropriate low pass reconstruction filters by means of a decommutator, which operates in synchronism with the commutator in the transmitter. This synchronization is essential for a satisfactory operation of the system.

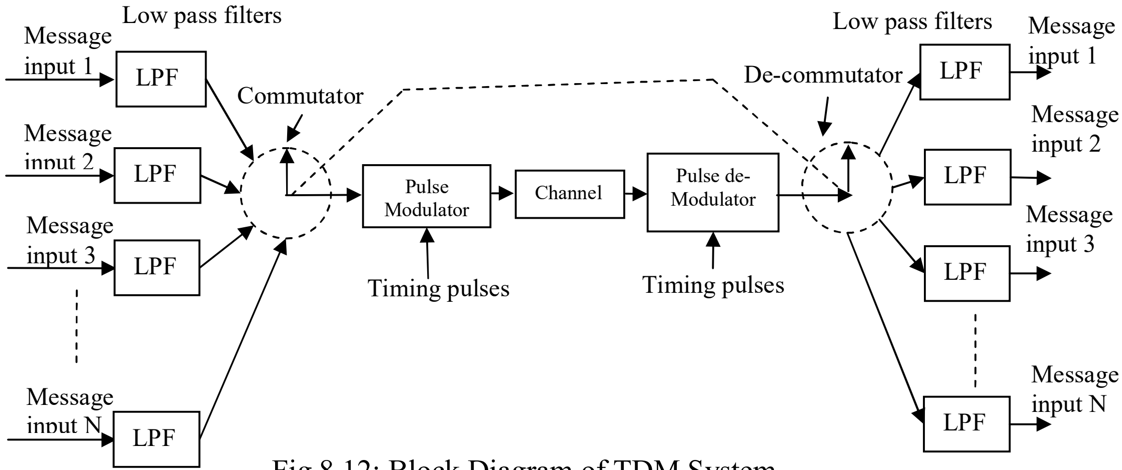



Fig 8.12: Block Diagram of TDM System

## COMPARISON BETWEEN FDM AND TDM:

| S.No. | FDM | TDM |
|---|---|---|
| 1. | Signal separation is in frequency domain | Signal separation is in time domain. |
| 2 | Circuit is more complex. | Circuit is less complex |
| 3 | Cross talk is more | Cross talk is very less or nil. |
| 4 | Performance is medium | Performance is superior. |
| 5 | This is less flexible. | This is more flexible. |
| 6 | This is suitable for analog signals. | This is suitable for digital signals. |
| 7 | This is used in radio telephone, satellite communication. | This is used in digital telephone, satellite communication. |
| 8 | Synchronization between transmitter and receiver is not required | Synchronization between transmitter and receiver is required |
| 9 | It requires modulators, filters and de-modulators | It requires commutator at the transmitting end and de-commutator at the receiving end |

# UNIT – VI  PULSE MODULATION

M. Ragunadh
Assi. proffesor

## Sampling

The process of converting a continuous time signal into discrete-time signal is called sampling.

## Sampling Techniques

There are three types of sampling techniques.

i) Instantaneous (or) ideal sampling

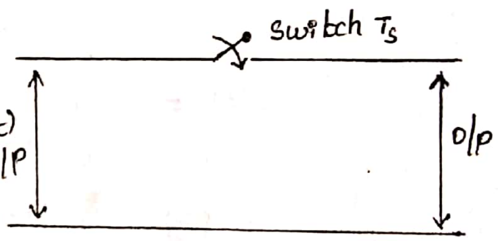
ii) Natural sampling.

iii) Flat top sampling

## Ideal Sampling

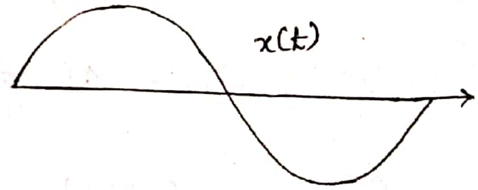
Let $x(t)$ be the msg signal and carrier be an impulse train with time period '$T_s$'. ($T_s$ must satisfy Nyquistrate $T_s \le T/2$)

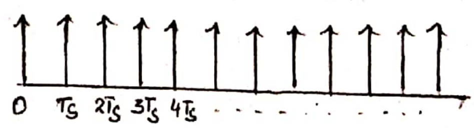
→ To produce ideal sampling we use circuit called switching sampler




→ The switch $T_s$ is closed and opened when approaching zero.

→ Train of impulses are obtained which are having instantaneous value of $x(t)$



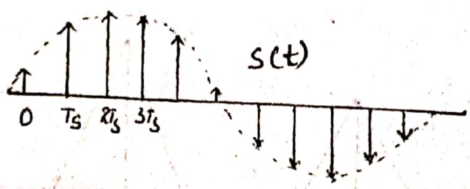




here $c(t) = \delta_{nTs}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$

$\therefore\ s(t) = x(t) \cdot c(t)$

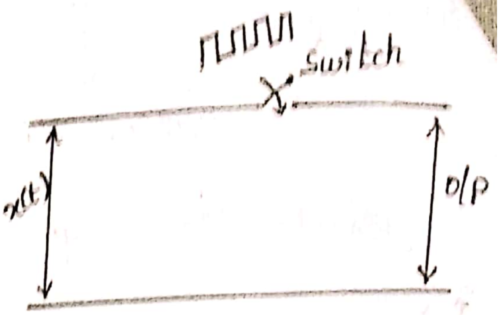
$$= \sum_{n=-\infty}^{\infty} x(t)\, \delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s)$$



# Natural Sampling or Natural Sampled PAM

Let $x(t)$ be message signal and carrier be an rectangular pulse train with time period $T_s$




→ To produce natural sampling we use a circuit called natural sampler

→ The switch is closed for '$\tau$' and opened upto to '$T_s$'. Train of pulses are obtained at output having instantaneous value of $x(t)$.

here $c(t) = \frac{A\tau}{T_s} \sum_{n=-\infty}^{\infty} c_n e^{j2\pi f_s n t}$

where $c_n = \text{sinc}\left(\frac{n\tau}{T_s}\right)$

$\therefore s(t) = x(t) \cdot c(t)$

$= x(t) \cdot \frac{A\tau}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}\left(\frac{n\tau}{T_s}\right) e^{j2\pi n f_s t}$

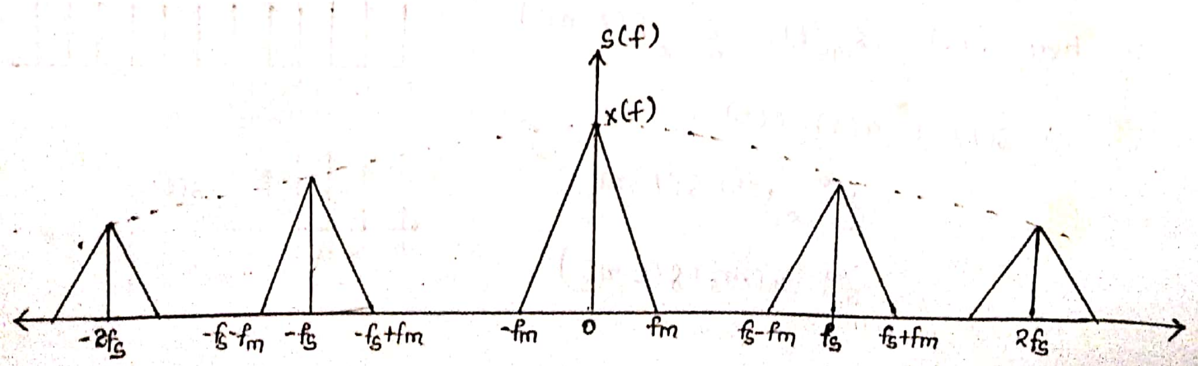
$s(t) = \frac{A\tau}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}\left(\frac{n\tau}{T_s}\right) e^{j2\pi n f_s t} x(t)$

Taking fourier transform on both sides

$S(f) = \frac{A\tau}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}\left(\frac{n\tau}{T_s}\right) F\left[x(t)\, e^{j2\pi n f_s t}\right]$ ($\because$ by frequency shifting property)

$S(f) = \frac{A\tau}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}\left(\frac{n\tau}{T_s}\right) X(f - nf_s)$

Spectrum of $S(f)$

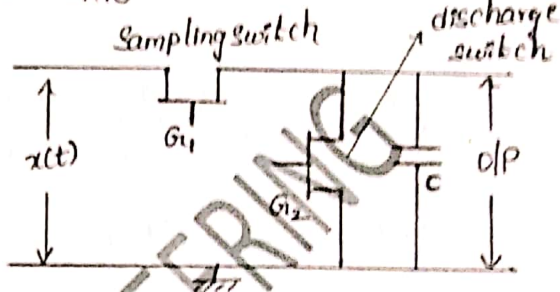


## Flat top Sampling (or) Pulse amplitude modulation (or) Flat top PAM (or) Generation of PAM

②

In flat top PAM, the top of the samples remain constant and equal to instantaneous value of $x(t)$.

Let $x(t)$ be the message signal and $\delta_{nTs}(t)$ be the impulse train.




### Working of Circuit

Sample and hold consists of two gates and a capacitor. The gate $G_1$ is closed for short duration, the capacitor 'c' charges upto peak value of $x(t)$.

Now $G_1$ is opened and capacitor 'c' holds the charge. The discharge switch is closed i.e '$G_2$' due to capacitor discharges to zero volts.

### Analysis

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

$$s(t) = x(t)\,\delta_{nTs}(t)$$

$$s(t) = \sum_{n=-\infty}^{\infty} x(t)\,\delta(t-nT_s)$$

$$s(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(t-nT_s)$$

> Starting edge of the pulse represents instantaneous value of $x(t)$

> To obtain flat-top PAM Convolution of instantaneous sample and a pulse $h(t)$.

$$\therefore\ g(t) = s(t) * h(t)$$

$$\therefore\ g(t) = \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau$$

but $s(\tau) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(\tau - nT_s)$

$$g(t) = \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(\tau-nT_s)\, h(t-\tau)\, d\tau$$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \underbrace{\delta(\tau-nT_s)}_{\delta(t-t_0)}\, \underbrace{h(t-\tau)}_{f(t)}\, d\tau$$

since $\int_{-\infty}^{\infty} f(t)\,\delta(t-t_0) = f(t_0)$

$$\therefore\ g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t-nT_s)$$

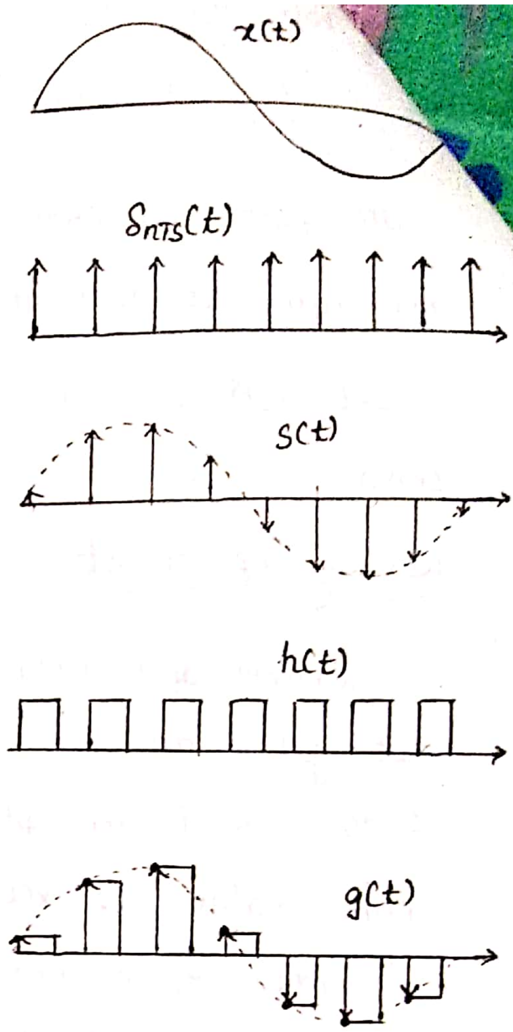
The above equation represent flat top in time domain. Taking fourier transform on both sides

$$g(t) = s(t) * h(t)$$

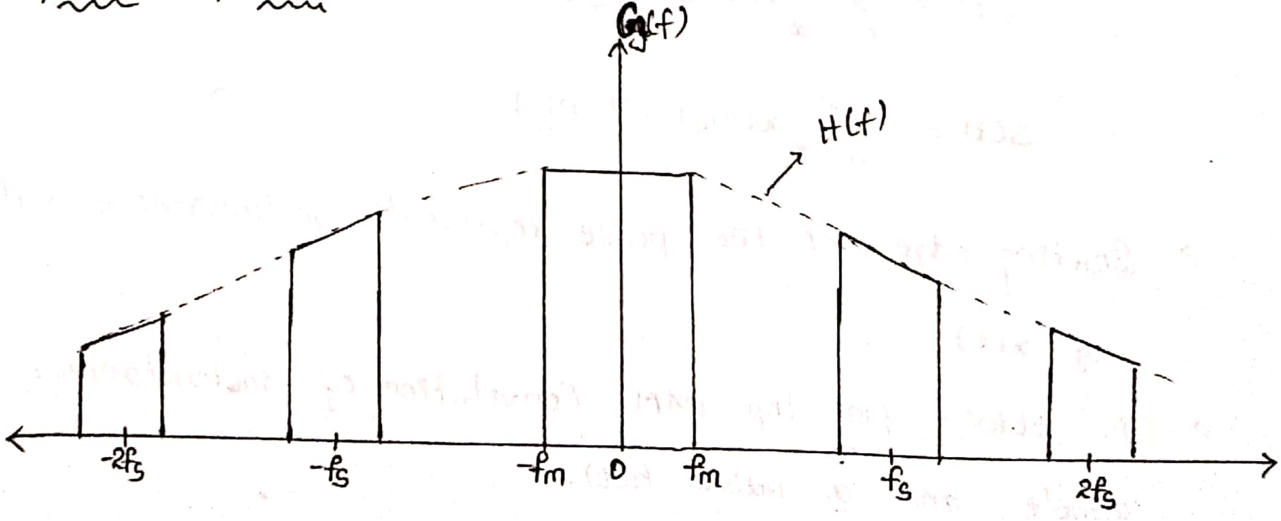
$$G(f) = S(f) \cdot H(f)$$

W.K.T $S(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)$

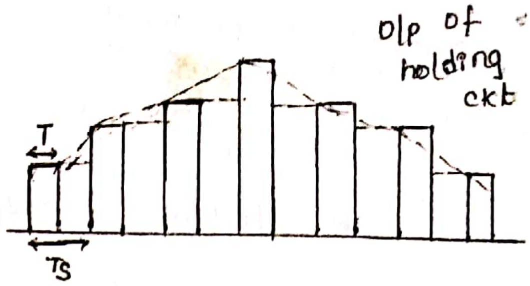
$$\therefore\ G(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s) \cdot H(f)$$



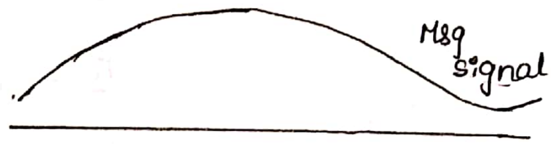

Spectrum of G(f)



Capacitor 'c' is charged to pulse amplitude value and it holds this value between two pulses.




> After holding circuit, output is applied to low pass filter. The output is smoothened in low pass filter.




Drawbacks of PAM

* B.W required for transmission of PAM signal is very large in comparision to maximum frequency present in modulating signal.
* Since PAM pulses varied in accordance with modulating signal. Therefore interference of noise is maximum.
* Amplitude of PAM signal varies, this also varies peak power required by transmitter.

PULSE TIME MODULATION

There are two types of PTM modulation ① PWM ② PPM

① PWM (Pulse width modulation)

It is also known as pulse duration modulation. The width of the modulated pulses varied in accordance with amplitude of modulating signal

## Bandwidth of PAM

$$T << T_S \rightarrow (1)$$

we follow $f_s \geq 2f_m$

$$\frac{1}{T_S} \geq 2f_m$$

$$T_S \leq \frac{1}{2f_m} \rightarrow (2)$$

$$f_{max} = \frac{1}{T+T} = \frac{1}{2T}$$

$$B.W \geq f_{max}$$

$$BW \geq \frac{1}{2T} \rightarrow (3)$$

from (1) and (2) $\quad T << T_S \leq \frac{1}{2f_m}$

$$T << \frac{1}{2f_m}$$

$$\frac{1}{T} >> 2f_m$$

$$\frac{1}{2T} >> f_m \rightarrow (4)$$

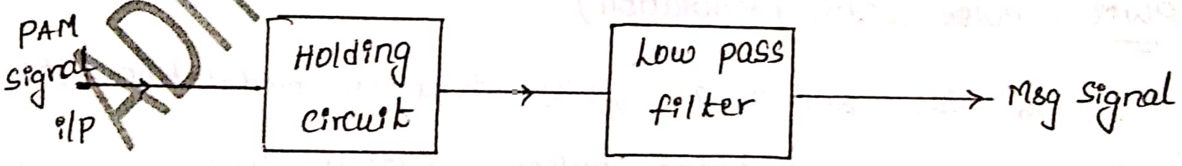
from (3) and (4)

$$BW \geq \frac{1}{2T} >> f_m$$

$$\boxed{BW >> f_m}$$

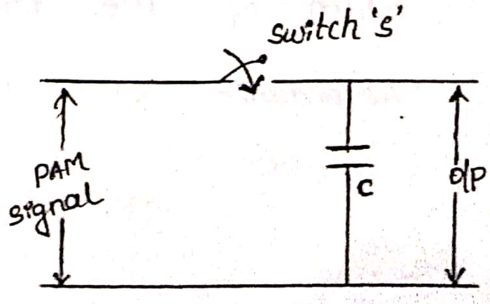


## Demodulation of PAM




### Working

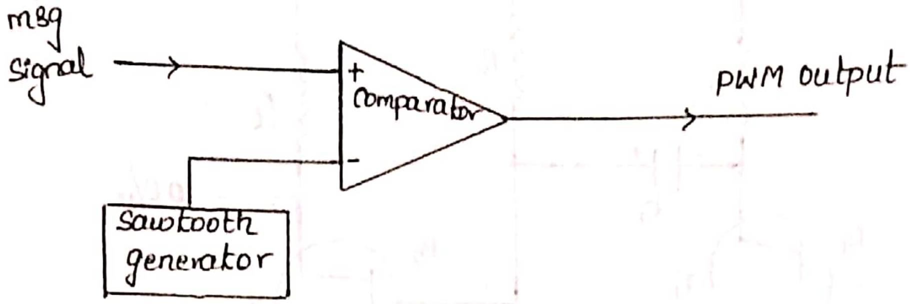
> The switch 's' is closed after the arrival of the pulse and it is opened at the end of the pulse.



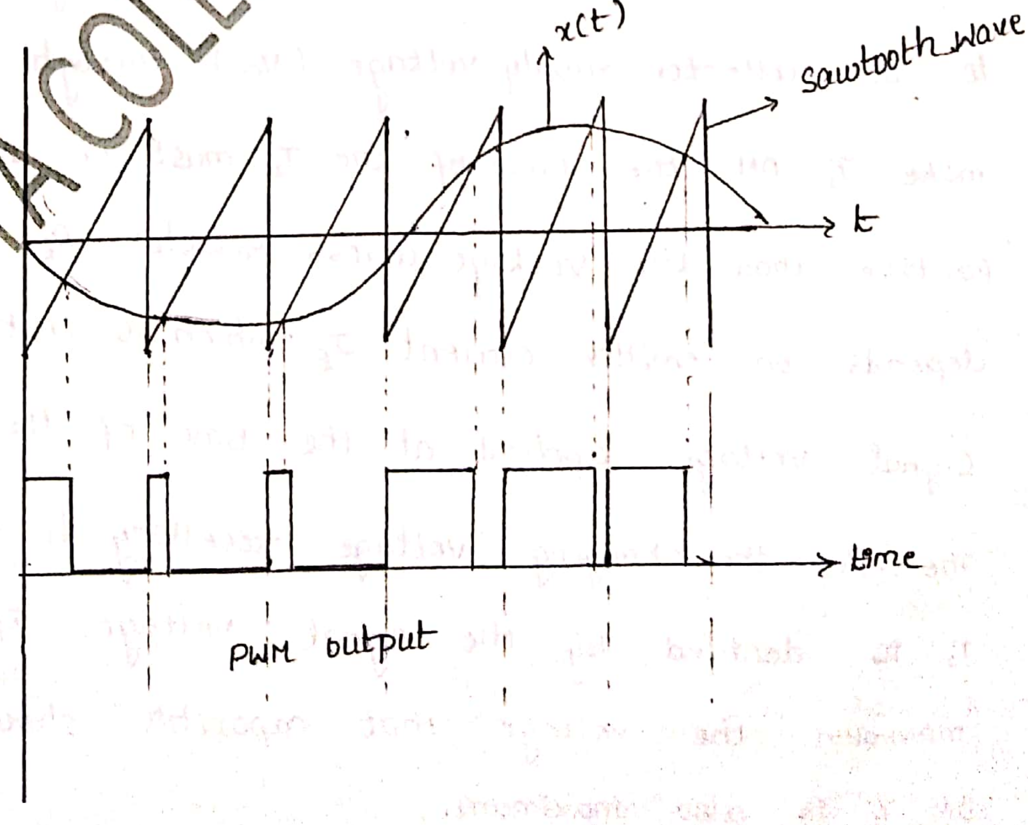

Holding circuit

Generation of PWM





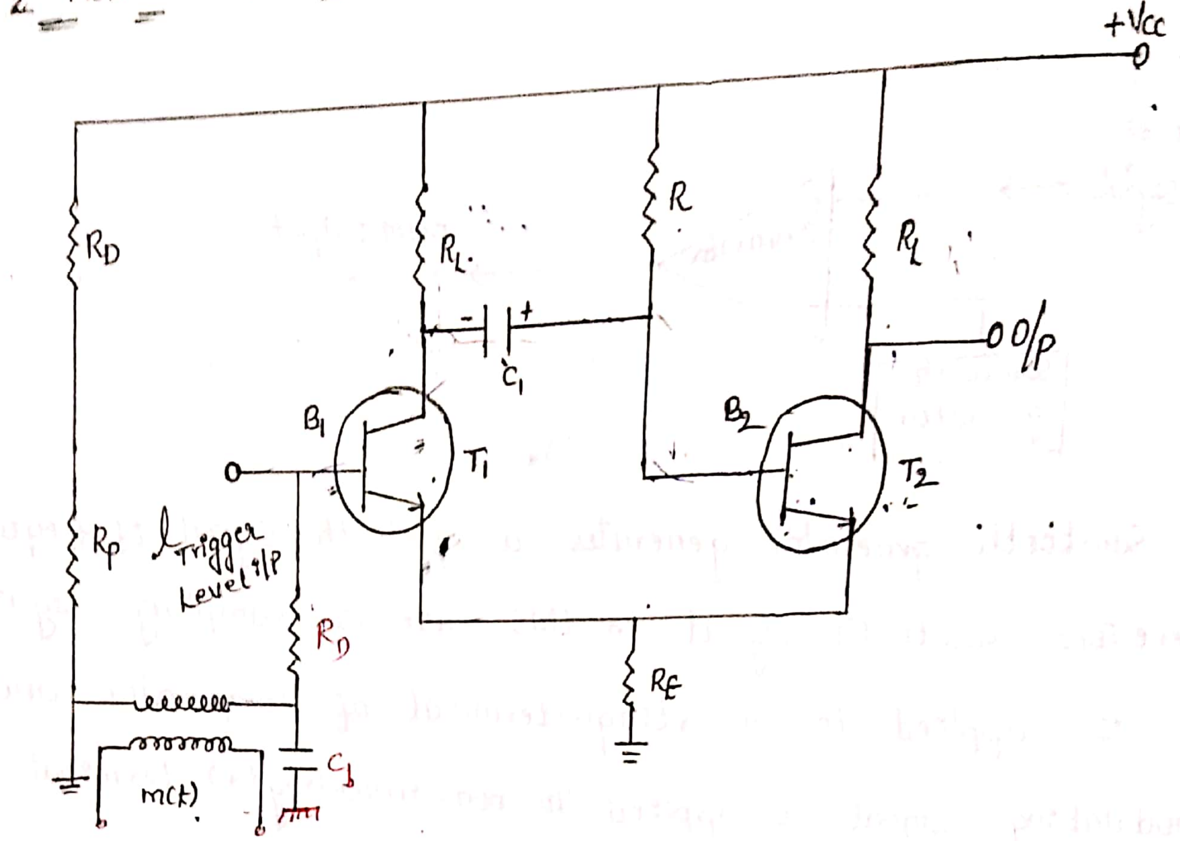

A Sawtooth generator generates a sawtooth signal of frequency fs. Therefore sawtooth signal in this case is sampling signal, it is applied to inverting(-) terminal of comparator and modulating signal is applied to non-inverting(+) terminal of same comparator. Whenever x(t) amplitude is higher than sawtooth signal. The comparator output will be high otherwise output is low, The resultant output waveform is PWM wave form.





## 2nd Method of PWM generation



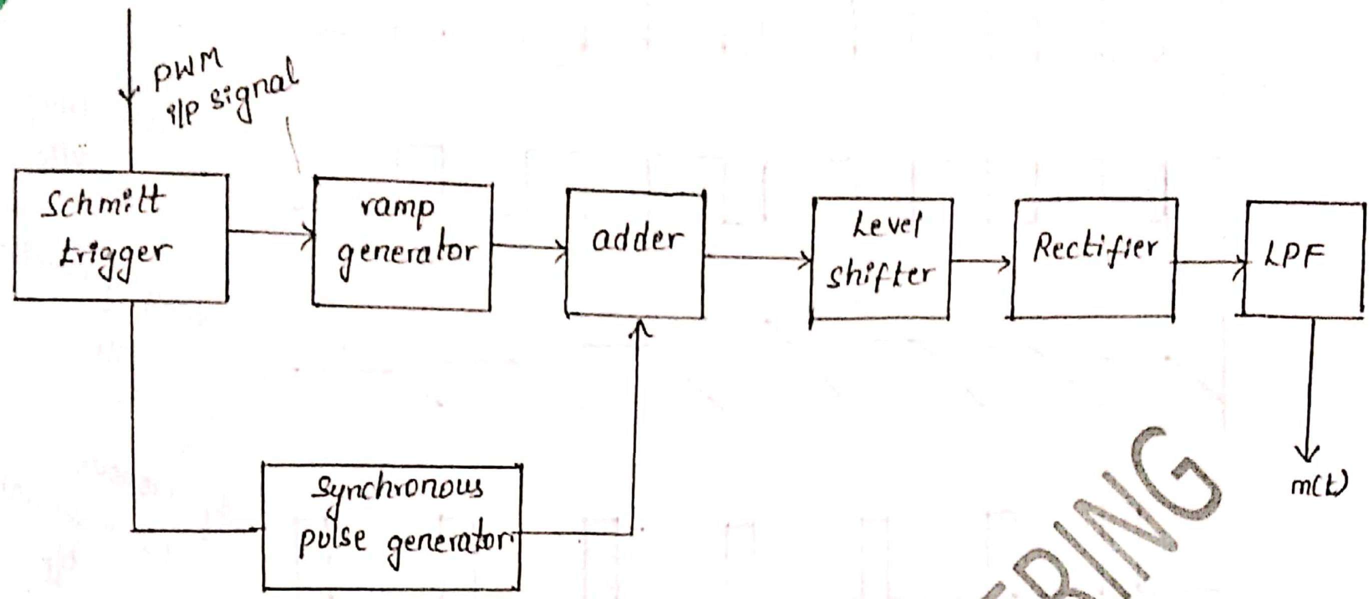

### Operation

The output is obtained when $T_2$ is OFF. Initially $T_1$ is OFF and $T_2$ is ON. The positive going trigger pulse at $B_1$ switches $T_1$ ON. Because of this, the voltage at $C_1$ falls. As a result, voltage at $B_2$ also falls and $T_2$ is switched OFF, C begins to charge up to the collector supply voltage (Vcc) through resistor 'R'. To make $T_2$ ON, the base of the $T_2$ must be slightly more positive than the voltage across resistor $R_E$. This voltage depends on emitter current $I_E$ which is controlled by the signal voltage applied at the base of the transistor $T_1$. Therefore the changing voltage necessary to turn OFF transistor $T_2$ is decided by the signal voltage. If signal voltage is maximum the voltage that capacitor should charge to turn ON $T_2$ is also maximum.

## Demodulation of PWM



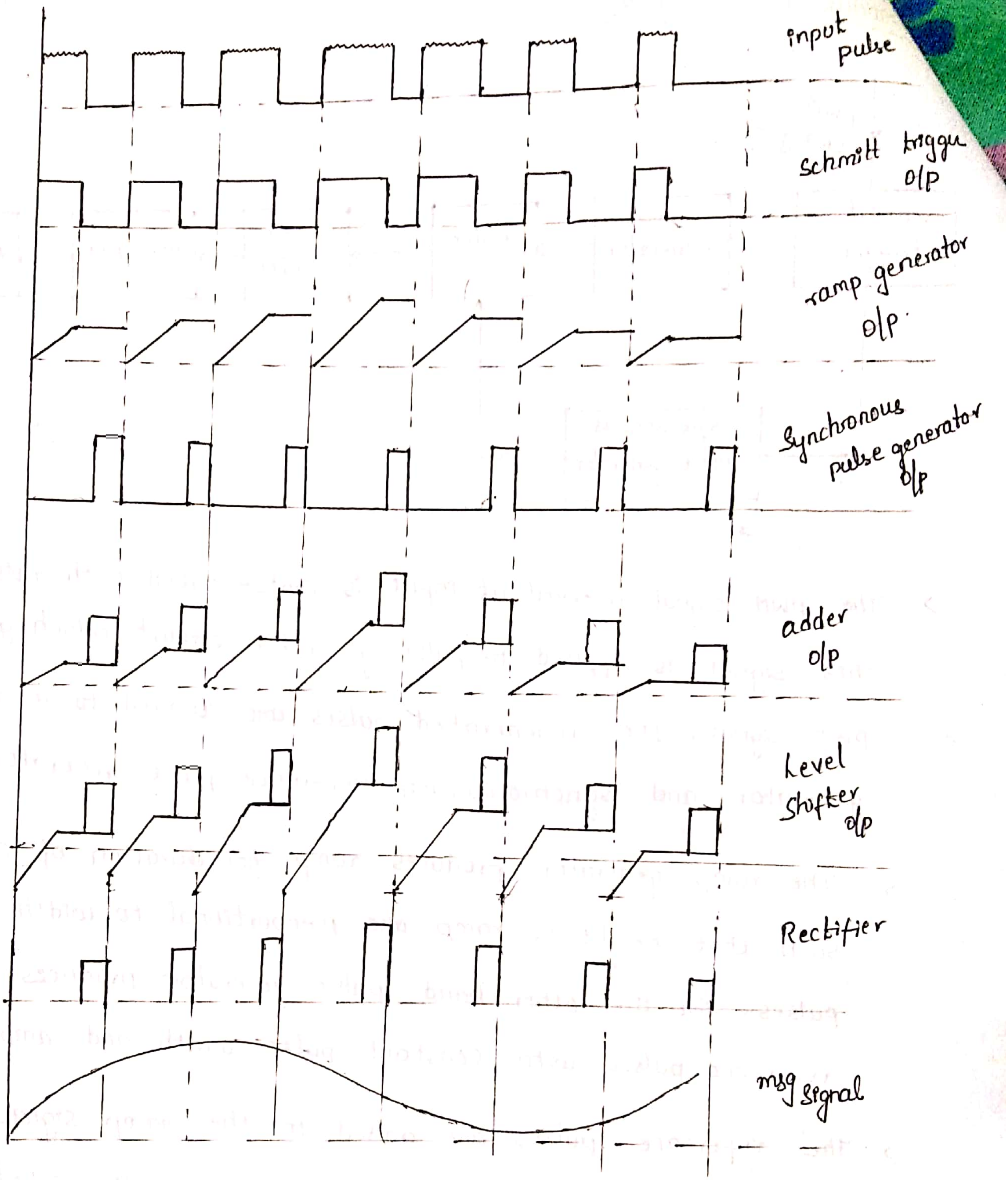



> The PWM signal received at input is contaminated with noise. This signal is applied to pulse generator circuit which generates PWM signal. The regenerated pulses are applied to a ramp generator and synchronization reference pulse generator.

> The ramp generator produces ramp for duration of pulses, such that height of ramp are proportional to width of pulses. On the other hand, pulse generator produces reference pulses with constant pulse width and amplitude.

> The reference pulses are added to the ramp signal. Then output of adder is clipped off in clipper ckt, a Low pass filter is used to recover the modulating signal back from the PAM signal.

> The waveform for the circuits shown below.




Advantages

- Less effect of noise and good noise immunity
- Synchronization between transmitter & receiver is not essential

Disadvantages

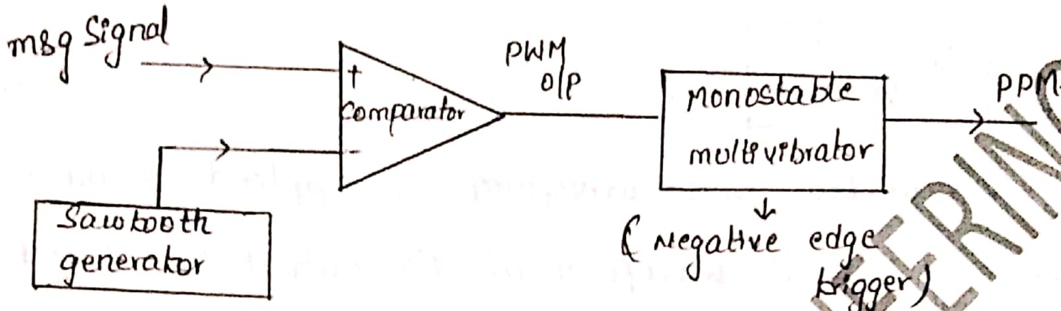
- Due to variable pulse width, pulses have variable power contents, hence transmission must be powerful to handle maximum pulse width.
- Bandwidth required for PWM is large compared to PAM.

## PULSE POSITION MODULATION (PPM)



The position of each pulse is varied in accordance with amplitude of sampled value of message signal is called PPM.

### Generation of PPM



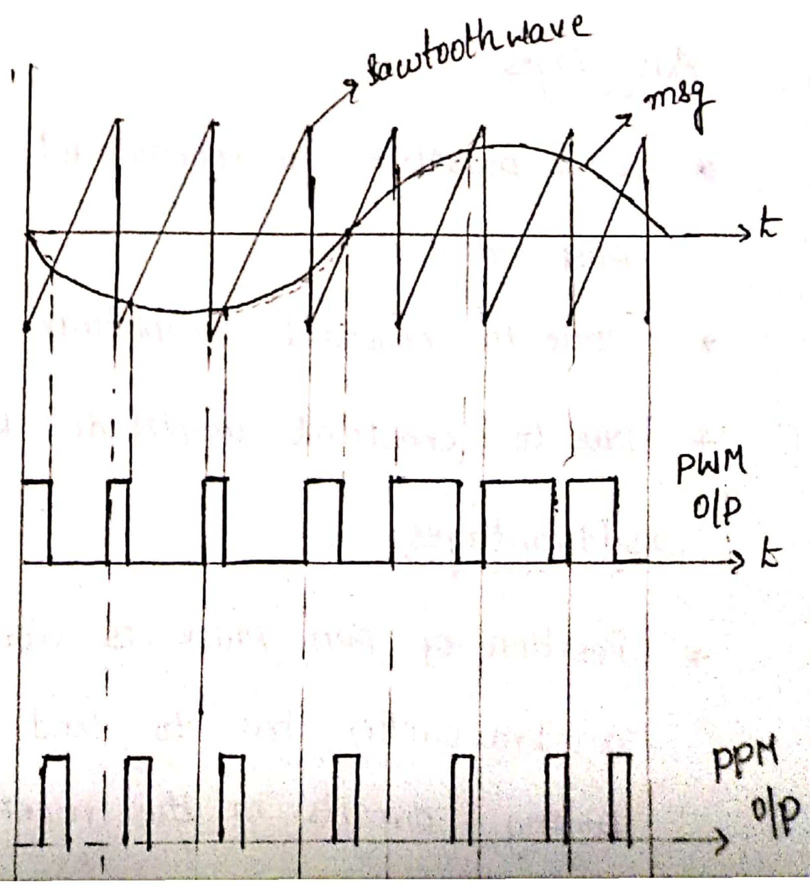

> PPM generator consists of comparator and monostable multivibrator

> The sawtooth generator generates a sawtooth signal and is applied to inverting(-) terminal of comparator and modulating signal to non - inverting (+) terminal.

> The comparator output is PWM output and when it is given to monostable multivibrator with negative edge trigger.

> The pulses are obtained at the negative edge of PWM output.

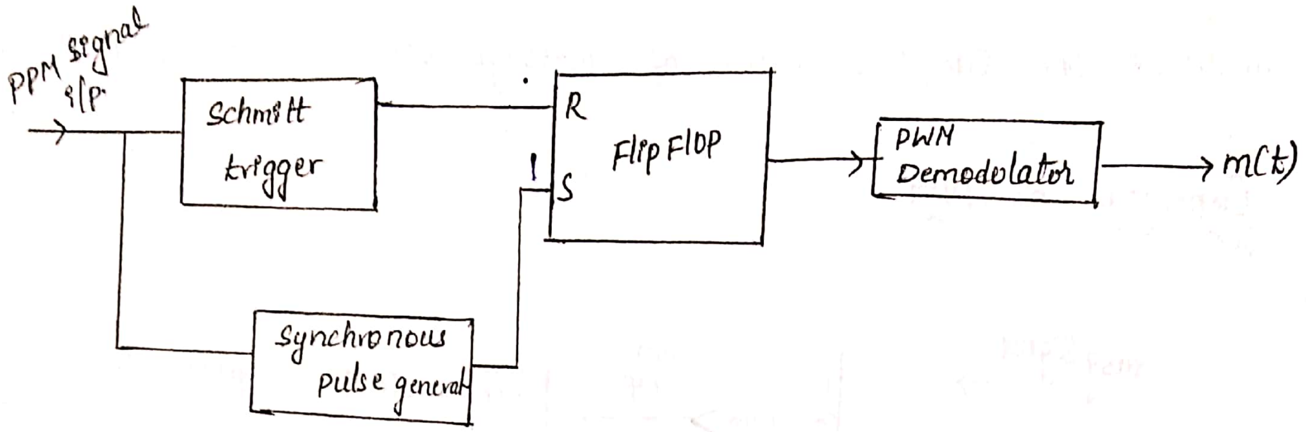
> The waveforms of PPM are shown in here.





## Demodulation of PPM




The noise corrupted PPM waveform is applied to pulse generator it develops a pulsed waveform at its output of fixed duration and applies these pulses to reset pin (R) of flip flop.. A fixed reference pulse is generated from the incoming PPM waveform and thus reference pulse is applied to set pin (S) of flip flop

Due to this Reset & set, we get PWM signal and can be demodulated using PWM demodulator we get original msg signal

### Advantages

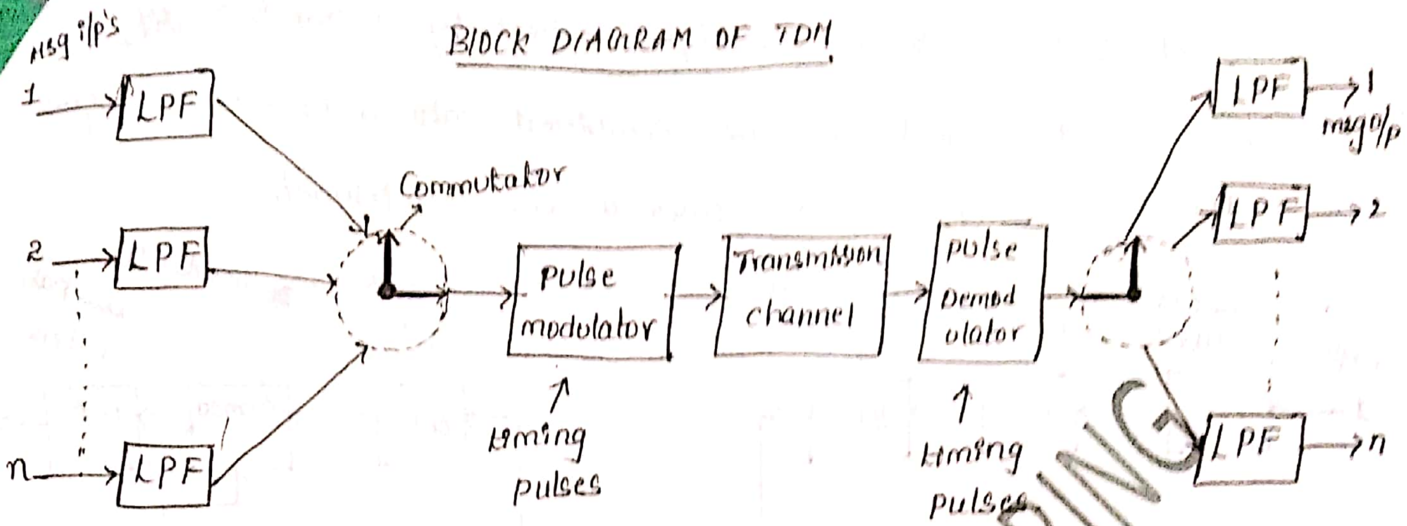
* It is possible to reconstruct PPM signal from noise contaminated PPM signal.
* Due to constant amplitude of PPM pulse, it has good immunity.
* Due to constant amplitude transmitted power always constant.

### Disadvantages

* Position of PPM pulse 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 demodulation not possible.
* Large bandwidth is required to ensure transmission of undistorted pulses.

# ~~ME~~ DIVISION MULTIPLEXING (TDM)

## BLOCK DIAGRAM OF TDM




In this block diagram each input message signal is applied to low pass filter to remove the frequencies that are non-essential to an signal representation.

Low pass filter output is applied to Commutator which is implemented using electronic switching circuitry. The function of Commutator is

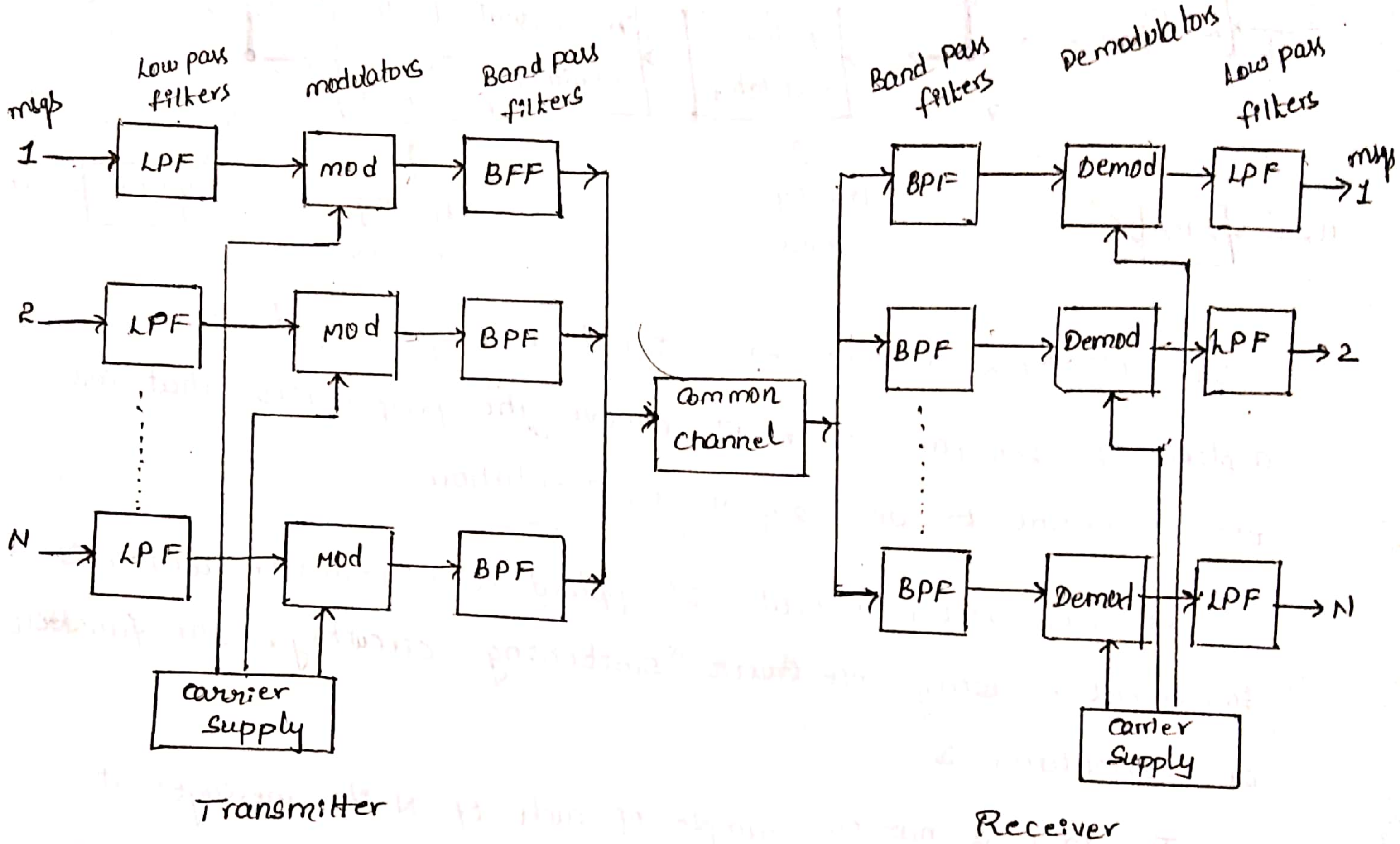
i) To take a narrow sample of each of N i/p messages at a rate $1/T_s$ that is slightly higher than 2W, W-cut off frequency of lowpass filter.

ii) To sequentially interleave these N samples inside a sampling interval $T_s$

The multiplexed signal is applied to pulse modulator, the purpose of which is converting that signal into a form suitable for transmission over the common channel. at receiver pulse demodulator performs reverse operation of pulse modulator. The output at pulse demodulator are distributed to appropriate Low pass filter by means of a decommutator. **TDM** is a standard signal multiplexing formats used in telephony.

# FREQUENCY DIVISION MULTIPLEXING (FDM)

Multiplexing is a technique where by a number of independent signals can be combined into a composite signal suitable for transmission over a common channel.




each input signal is applied to low pass filter to remove high frequency terms, that do not contribute to signal representation. The filtered signals are applied to modulators which shift the frequency range of signals as to occupy mutually exclusive frequency intervals. The necessary carrier signal for modulation is supplied by carrier supply. However most widely used modulation is SSB. The Band pass filters following the modulators are used to restrict the band of each modulated wave to its prescribed range.

The resulting band pass filter outputs are not combined in parallel to form input to common channel. at receiving a bank of B.P. filters with their inputs connected in parallel, is used to separate the message signal on frequency basis.

Finally original message signals are recovered by individual demodulators whose carrier frequency is supplied by carrier supply.

## Comparision of TDM and FDM

| TDM | FDM |
|---|---|
| * It is a technique for transmitting several messages on one channel by dividing time domain slots, one slot for each message. | * In this technique several messages on one channel, message signals are distributed in frequency spectrum such that they do not overlap. |
| * It requires commutator at the transmitting end and a distributor, working in perfect synchronization at receiving end. | * FDM requires modulators filters and demodulators. |
| * Perfect synchronization b/w transmitter and receiver is required. | * Synchronization between transmitter and receiver not required. |
| * Cross talk problem is not severe in TDM. | * FDM suffers from cross talk. |

| | |
|---|---|
| * Preferred for digital signal transmission | * Preferred for analog transmission |
| * It does not require any complex circuitry. | * It requires complex circuitry at transmitter & receiver. |

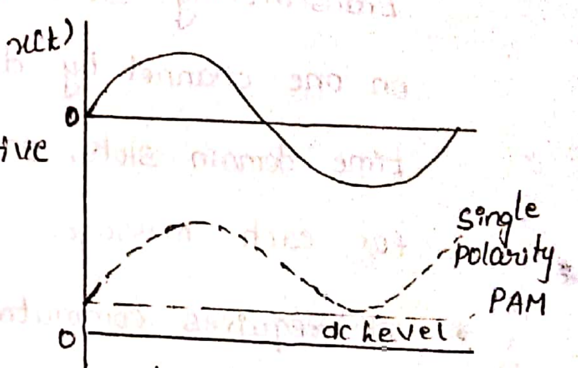
## Single Polarity & Double Polarity PAM

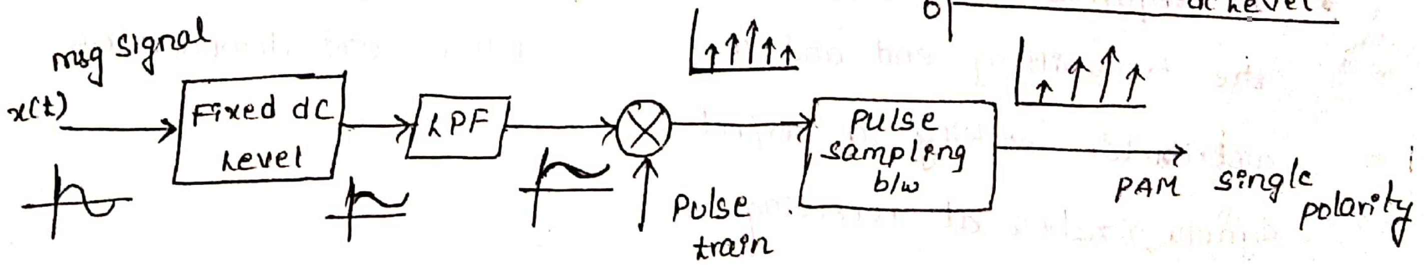
### Single polarity

Single polarity PAM can be generated using a fixed DC level, low pass filter, multiplier, pulse train generator and pulse sampling network.

here PAM signal is always positive thats why called single polarity.

here we are shifting the negative value above '0' dc level.

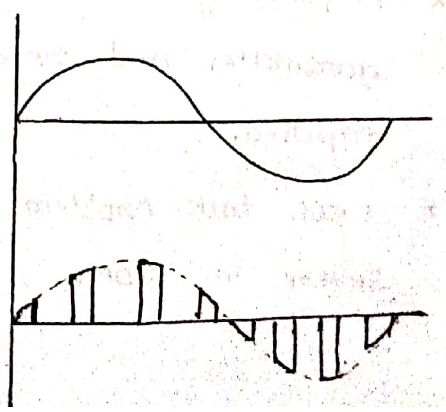



### Double polarity

Double polarity PAM signal has positive as well as negative polarity.

further refer natural PAM



Code No: R1622044 **R16** SET - 1

**II B. Tech II Semester Regular/Supplementary Examinations, April/May - 2019**

**ANALOG COMMUNICATION**

(Electronics & Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answer **ALL** the question in **Part-A**
3. Answer any **FOUR** Questions from **Part-B**

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## PART –A

1. a) Explain the need of modulation in communication system with one example. (3M)
   b) List out few applications of DSB-SC modulation. (2M)
   c) Explain the few Comparisons of FM & AM. (2M)
   d) Define the term fidelity and explain its importance in Receivers. (2M)
   e) Define the term Average noise figure in detail. (2M)
   f) List out different type of Pulse modulation systems in detail. (3M)

## PART -B

2. a) Draw and Describe an expression for AM wave and sketch its frequency spectrum. (7M)
   b) A tone modulated AM-signal with a modulation index of "m" and base band signal Frequency of $\omega_m$ is detected using envelope detector, whose time constant is RC, for Effective demodulation, show that (1/RC) ≥ [m $\omega_m$/(√1-m2 )]. (7M)

3. a) Draw the block diagram and explain generation of DSB-SC signal using balanced modulator. (7M)
   b) Discuss the effect of frequency and phase error in demodulation of DSB-SC wave using synchronous detector. (7M)

4. a) Explain the generation of N.B.F.M using narrow band P.M generator along with necessary equations. (7M)
   b) Draw the circuit diagram of Phase locked loop for detection of FM and explain its operation. (7M)

5. a) Draw the block diagram of Super hetrodyne receiver and Explain the function of each block in detail. (7M)
   b) List out the different Classification of Transmitters and explain any one type in detail. (7M)

6. a) Explain the threshold noise effects in angle modulation system with suitable diagrams? (7M)
   b) Explain the concept about pre-emphasis and de-emphasis along with circuit diagrams. (7M)

7. a) Explain the concept of how a PPM signal can be generated from PWM signal along with circuit diagram. (7M)
   b) Draw the block diagram of Time Division Multiplexing and explain the function of each block in detail. (7M)
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Code No: R1622044 **R16** SET - 2

**II B. Tech II Semester Regular/Supplementary Examinations, April/May - 2019**
**ANALOG COMMUNICATION**
(Electronics &Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answer **ALL** the question in **Part-A**
3. Answer any **FOUR** Questions from **Part-B**

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**PART –A**

| | | | |
|---|---|---|---|
| 1. | a) | Derive $P_t$ in Amplitude modulation. | (2M) |
| | b) | List out few comparisons of SSB and DSB-SC modulation. | (3M) |
| | c) | Define the terms wideband FM and Narrowband FM? | (3M) |
| | d) | Define the terms Image frequency and Image frequency rejection ratio. | (2M) |
| | e) | Explain the different noise sources available in communication receiver. | (2M) |
| | f) | List out few comparisons of PAM and PPM. | (2M) |

**PART -B**

| | | | |
|---|---|---|---|
| 2. | a) | What do you understand of modulation index? What is its significance? | (7M) |
| | b) | Draw the circuit diagram of Switching modulator and explain its operation in detail. | (7M) |
| 3. | a) | Explain the principle of coherent detector of DSB-SC modulated more with a neat block diagram. | (7M) |
| | b) | List out few Applications of different AM Systems in detail. | (7M) |
| 4. | a) | Define angle modulation? Explain different types of angle modulations with mathematical expressions. | (7M) |
| | b) | A Sinusoidal carrier of 10V, 5MHz is frequency modulated by sinusoidal message signal of 50V, 100 kHz and $K_f$=50 kHz/V. Find $\Delta f$, $\beta$, band width and power. | (7M) |
| 5. | a) | Draw the block Schematic for FM broadcast receiver and explain the function of each unit. | (7M) |
| | b) | Draw the block diagram of FM Transmitter and explain the function of each block in detail. | (7M) |
| 6. | a) | Explain the noise performance of DSB-SC receiver and prove its S/N ratio is unity. | (7M) |
| | b) | Discuss the threshold effect for AM with envelope detector along with derivation. | (7M) |
| 7. | a) | Explain the PPM generation from PWM with a neat block diagram and necessary figures. | (7M) |
| | b) | Explain the different methods for generation of PWM along with circuit diagrams. | (7M) |
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Code No: R1622044 **R16** SET - 3

**II B. Tech II Semester Regular/Supplementary Examinations, April/May - 2019**

**ANALOG COMMUNICATION**

(Electronics & Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**

2. Answer **ALL** the question in **Part-A**

3. Answer any **FOUR** Questions from **Part-B**

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## PART –A

1. a) What are the basic constituents of a communication system along with block diagram? (3M)

b) Explain the concept of Vestigial side band modulation along with wave forms. (2M)

c) List out the different Advantages and Applications of FM? (2M)

d) What are the main requirements of AM broadcast transmitters, along with block diagram. (3M)

e) Define the term noise figure and noise equivalent temperature. (2M)

f) Explain the different applications of TDM. (2M)

## PART –B

2. a) With suitable diagram explain the square-law diode modulation method for AM generation? (7M)

b) An amplitude modulated voltage is given by V = 100 (1 + 0.4 cos 100 t + 0.001 cos 3000t) cos $10^6$ t. State all frequency components present in the voltage, and find modulation index for each modulating voltage term. What is the effective modulation index of V? (7M)

3. a) How do you detect SSB waves? Explain with neat diagram. (7M)

b) Find the various frequency components and their amplitudes in the voltage given by v (t) = 100 (1+ 0.6 Cos 5000t- 0.3 Cos 100t) sin $6x10^6$t. Draw the single sided spectrum. Also evaluate the modulated and sideband power. (7M)

4. a) What is angle modulation? Explain frequency deviation, percent modulation, phase deviation and modulation index in detail. (7M)

b) Explain the Armstrong method of FM generation along with circuit diagram. (7M)

5. a) Explain the following terms in detail (7M)

(i) Amplitude limiting (ii) Frequency changing and tracking

b) In a broadcast Super Heterodyne Receiver having no RF amplifier is tuned to 555kHz. The local oscillator frequency is adjusted to 1010kHz and the quality factor is 100. Calculate the intermediate frequency, image frequency and image rejection ratio. (7M)



|''|'||||''|'''||||'|
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6. a) What is noise? Explain the difference between thermal noise and shot noise in detail. (7M)
   b) Explain about noise effect in DSB-SC and obtain necessary expression for figure of merit. (7M)

7. a) Describe the synchronization procedure for PAM, PWM and PPM signals in detail. (7M)
   b) What is FDM? Explain the importance of FDM over TDM along with circuit diagram. (7M)

Code No: R1622044 **R16** SET - 4

**II B. Tech II Semester Regular/Supplementary Examinations, April/May - 2019**
**ANALOG COMMUNICATION**
(Electronics &Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answer **ALL** the question in **Part-A**
3. Answer any **FOUR** Questions from **Part-B**

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## PART –A

| | | | |
|---|---|---|---|
| 1. | a) | List out the few comparisons of TDM and FDM. | (3M) |
| | b) | How can you obtain a DSB-SC signal? What is the Band-width of DSB-SC signal? | (3M) |
| | c) | Define Single tone frequency modulation. | (2M) |
| | d) | Define the terms sensitivity and selectivity. | (2M) |
| | e) | Explain, how noise can be calculated in a communication system. | (2M) |
| | f) | List out the different advantages of FDM over TDM. | (2M) |

## PART -B

| | | | |
|---|---|---|---|
| 2. | a) | Derive an expression for AM wave and sketch its frequency spectrum. | (7M) |
| | b) | Draw the circuit diagram of Envelope detector and explain its operation along with wave forms. | (7M) |
| 3. | a) | List out the methods for generation of SSB-SC signal and explain any one of the method in detail. | (7M) |
| | b) | Discuss the process of generation of VSB waves along with its applications. | (7M) |
| 4. | a) | Explain the detection of FM wave using balanced frequency discrimination along with circuit diagram. | (7M) |
| | b) | Explain about the spectra of NBFM and WBFM along with its applications. | (7M) |
| 5. | a) | Define AGC? Explain the different types of AGC occurred in Receivers. | (7M) |
| | b) | Draw the circuit diagram of Communication Receiver and explain the function of each block in detail. | (7M) |
| 6. | a) | What is FM threshold effect? How threshold reduction is achieved in FM receiver in detail. | (7M) |
| | b) | Write short notes on Noise in AM System and how to avoid it. | (7M) |
| 7. | a) | Define PAM? Explain the generation of PAM along with circuit diagram. Give merits and demerits of PAM. | (7M) |
| | b) | For a PAM transmission of voice signal having maximum frequency equal to fm=8KHz calculate the transmission bandwidth. It is given that the sampling frequency fs=16KHz and the pulse duration τ=0.5 TS | (7M) |
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