AALIM MUHAMMED SALEGH COLLEGE OF ENGINEERING

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

EC 8491COMMUNICATION THEORY
(FOR IV-SEM-ECE STUDENTS)

PREPARED BY

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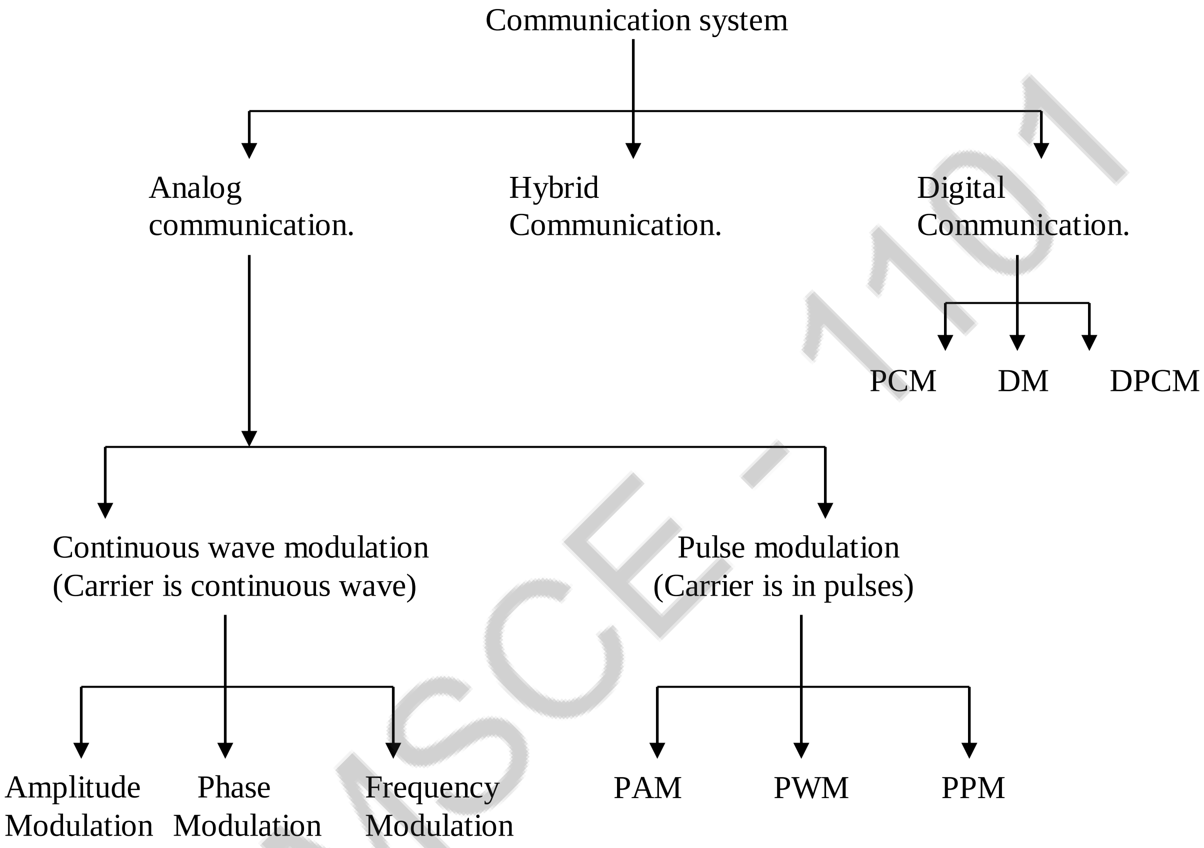
UNIT-I

2-MARK QUESTION AND ANSWERS:

1. What is communication?

- Communication is the process of conveying or transferring messages from one Point to another.

2. What are the types of Communication system?



3. Define modulation?

- Modulation is a process by which some characteristics of high frequency carrier signal is varied in accordance with the instantaneous value of another signal called modulating signal.

4.Determine the Hilbert transform of cosωt. (Nov/Dec-2017)

Hilbert transform shifts the phase of positive frequency components by -90° and that of negative frequency components by +90°. Thus, the Hilbert transform of cosωt is sinωt.

$$H[\cos\omega t] = \sin\omega t.$$

5. Define depth of modulation(or)modulation index.

It is defined as the ratio between message amplitude to that of carrier amplitude.

$m_a = Vm/Vc$

6. What are the degrees of modulation?

Under modulation. m<1
Critical modulation m=1
Over modulation m>1

7. How will you determine the Fourier transform for periodic signal? [Nov-03]

$$\mathscr{F}\{g(t)\} = G(f) = \int_{-\infty}^{\infty} g(t) e^{-2\pi i f t} dt$$

8. If a 10KW amplitude modulated transmitter is modulated Sinusoidally by 50%, what is the total RF power delivered? [Nov-05]

$m_a = 50/100 = 0.5$; $Pc = 10$ kw
$Pt = Pc(1 + m^2/2)$
$= 11.25$kw

9.What is VSB? Where it is used? (Nov/Dec-2017) (APRIL/MAY-2019)

Vestigial Sideband Transmission (VSB)

Vestigial Sideband (VSB) is a type of Amplitude Modulation (AM) technique (sometimes called VSB-AM) that encodes data by varying the amplitude of a single carrier frequency. Portion of one of the redundant sidebands are removed and vestige (portion) of the other sideband is transmitted to form a Vestigial Sideband signal.

Uses of VSB:

- VSB modulation has become standard for the transmission of Television signals.

VSB is used for TV picture broadcasting

10. Define Amplitude modulation?

- Amplitude modulation is the process by which amplitude of the carrier signal is varied in accordance with the instantaneous value of the modulating signal but frequency and phase of carrier wave is remains constant.

11. Define Frequency modulation?

- Frequency modulation is the process by which frequency of the carrier signal is varied in accordance with the instantaneous value of the modulating signal.

12. Define phase modulation?

- Phase modulation is the process by which Phase angle of the carrier signal is varied in accordance with the instantaneous value of the modulating signal.

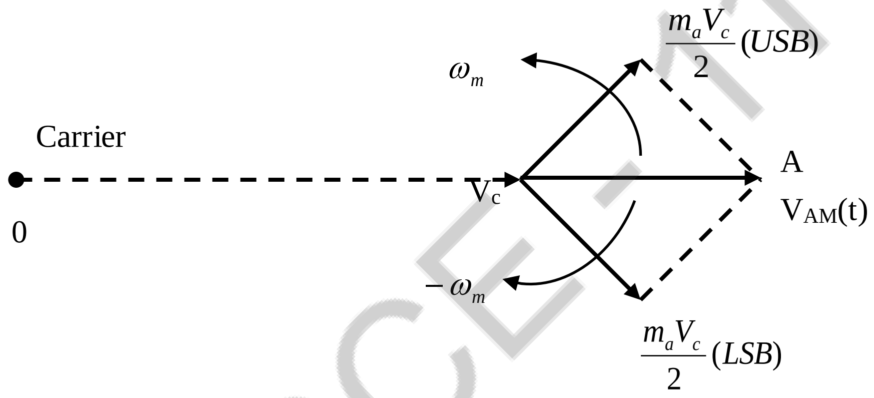
13. As related to AM what is over modulation, under modulation and 100% modulation?

(OR)

When does a carrier is said to be over, under modulated in Amplitude modulation?

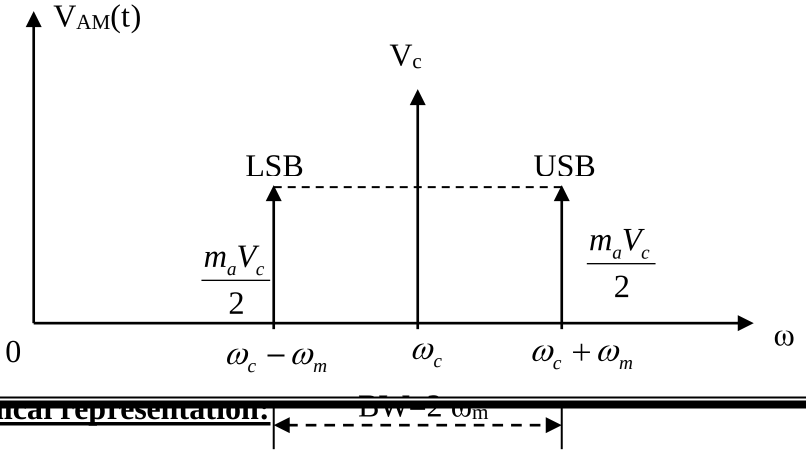
- In the case of Under modulation, modulation index $m_a<1$(i.e.)$V_m<V_c$.Here the envelope of Amplitude modulated signal does not reach the Zero amplitude axis. Hence the Message signal is fully preserved in the envelope of the AM Wave.
- In the case of Over modulation, modulation index $m_a>1$(i.e.)$V_m>V_c$. Here the envelope of Amplitude modulated signal crosses the zero axis.
- In the case critical modulation modulation index $m_a=1$(i.e.)$V_m=V_c$.Here the envelope of the modulated signal just reaches the zero amplitude axis. The message signal remains preserved.

14. Draw the phasor diagram of AM-SC signal.

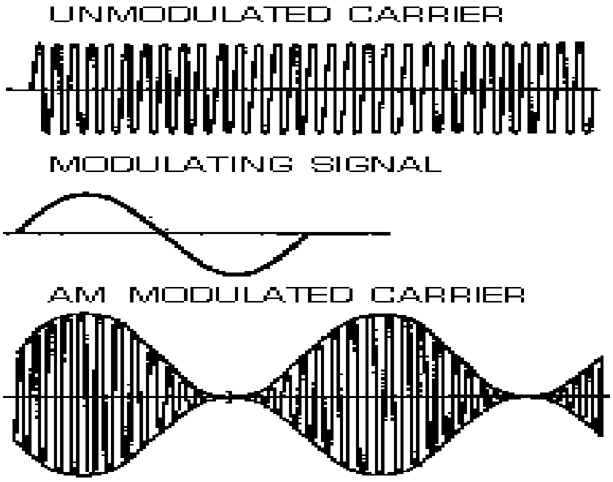


15. Draw the Graphical and frequency spectrum and phasor Representation of AM with carrier.

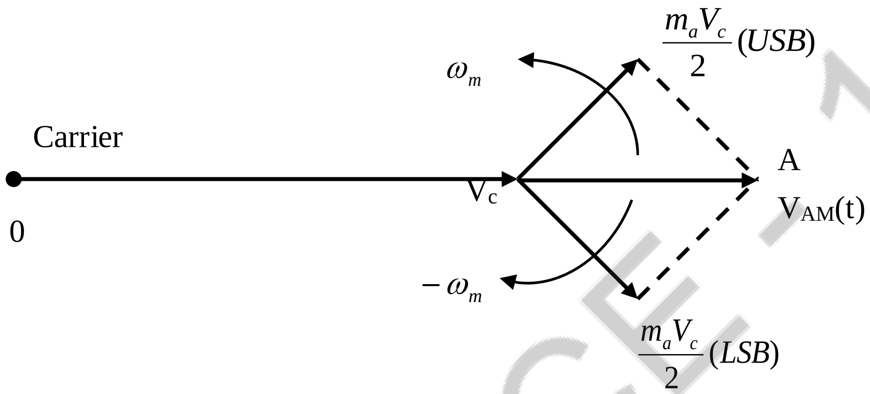
Frequency spectrum of AM with carrier:



Graphical representation:



Phasor representation of AM with carrier:



16.Define carrier swing. (APR/MAY– 2017)

Carrier Swing (CS) is defined as the total variation in frequency from lowest to highest point. Carrier Swing = 2* frequency deviation of the FM signal = 2* $\Delta\omega$

17. What are the advantages of VSB-AM?

- It has bandwidth greater than SSB but less than DSB system.
- Power transmission greater than DSB but less than SSB system.
- No low frequency component lost. Hence it avoids phase distortion.

18. How will you generate DSB-SC-AM?
(OR)
Give the two methods of generating DSB-SC-AM.

There are two ways of generating DSBSC-AM such as
1. balanced modulator 2.ring modulators

19. Compare linear and non-linear modulators.

Sl.no	Linear modulators	Non-Linear modulators
1	Heavy filtering is not required.	Heavy filtering is required

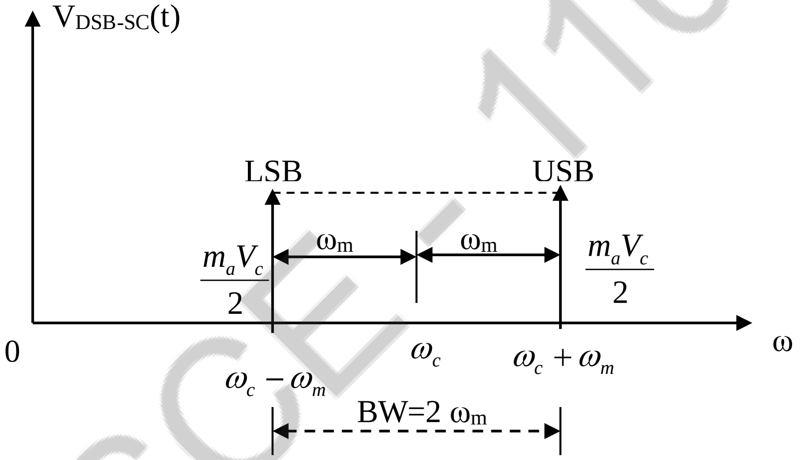
2	These modulators are used in high level modulation.	These modulators are used in low level Modulation.
3	The carrier voltage is very much greater than modulating signal voltage.	The modulating signal voltage is very much greater than the carrier signal voltage.

20. What are advantages of ring modulator?

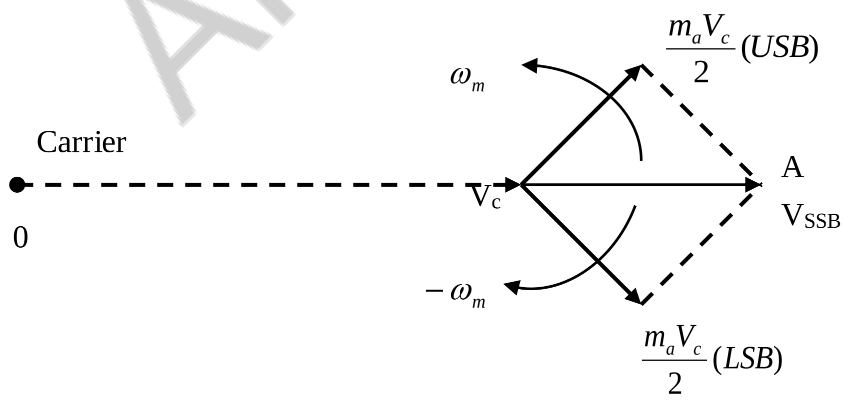
1. Its output is stable.
2. It requires no external power source to activate the diodes.
3. Virtually no maintenance.
4. Long life.

21. Draw the frequency spectrum and phasor Representation of DSB-SC-AM?

Frequency spectrum:



Phasor representation



22. What are the advantages of DSB-SC and SSB-SC.

DSB-SC:

- Suppression of carrier results in economy of power.
- It is commonly used in carrier current telephony system, in which one sideband is filtered out to reduce the width of the channel required for transmission.
- It offers secrecy.

- It increases the efficiency because the carrier is suppressed.

SSB-SC

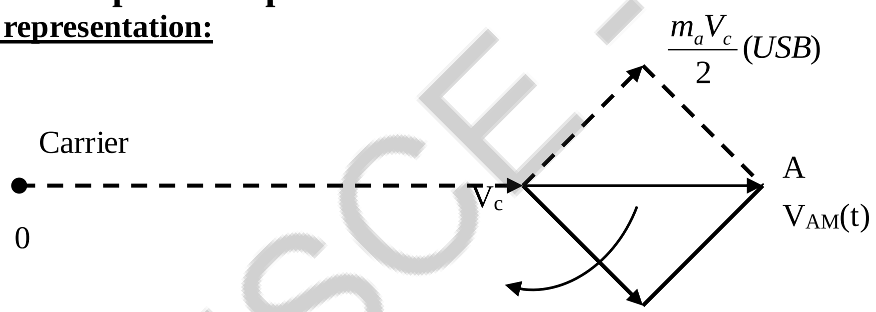
- Bandwidth of SSB is half that of DSB-SC AM. Thus twice the number of channels can be accommodated at a given frequency spectrum.
- No carrier is transmitted, hence possibility of interference with other channels are avoided.
- It eliminates the possibility of fading. Fading occurs due to multipath propagation of electro-magnetic waves.

23. Give the methods of generating SSB-SC-AM. And mention Some applications of SSB-SC

- The two methods of generating the SSB-SC waves are
- Frequency discrimination or Filter method.
- Phase discrimination method.
- **Applications:**
  - ✓ Police Wireless communication.
  - ✓ SSB telegraph system
  - ✓ Point to point radio telephone communication
  - ✓ VHF and UHF communication systems.

24. Draw the phasor representation of SSB-SC-AM?

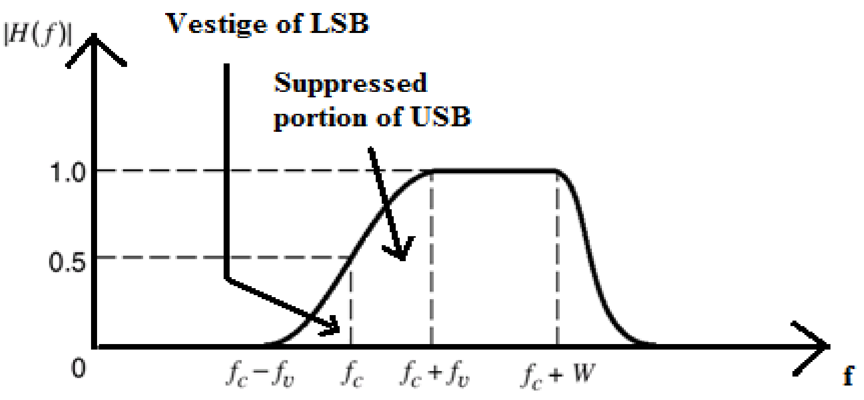
Phasor representation:



25. Draw the frequency spectrum of VSB. Where it is used?

(MAY/JUNE – 2015)

Frequency spectrum of VSB:



26. What is single tone and multi tone modulation?

- If modulation is performed for a message signal with more than one frequency component t modulation is called multi tone modulation.
- If modulation is performed for a message signal with one frequency component then the modulation is called single tone modulation.

27.What are the advantages of converting low frequency signal into high frequency signal? (NOV/DEC – 2016, MAY/JUNE - 2013) (or) why we need modulation?

a) Reduction in the height of antenna

b) Avoids mixing of signals

c) Increases the range of communication

d) Multiplexing is possible

e) Improves quality of reception

To reduce noise and interference

28. Compare AM with DSB-SC and SSB-SC.(APRIL/MAY 2018)

AM signal	DSB-SC	SSB-SC
Bandwidth=$2f_m$	Bandwidth=$2f_m$	Bandwidth=f_m
Contains USB, LSB, carrier	Contains USB,LSB	Contains LSB or USB
More power is required for transmission	Power required is less than that of AM.	Power required is less than AM &DSB-SC

29. What are the types of AM detectors?

1. Nonlinear detectors
2. Linear detectors

30. What are the types of linear detectors?

1. Synchronous or coherent detector.
2 .Envelope or non coherent detector.

31. A transmitter supplies 8 Kw to the antenna when modulated.

Determine the total power radiated when modulated to 30%.

m_a=30/100=0.3;Pc=8 kw

Pt=Pc(1+m^2/2)

=8.36 kw

32. The antenna current of an AM transmitter is 8A when only carrier is sent. It increase 8.93A when the carrier is modulated by a single sine wave. Find the percentage modulatio

Solution:

Given: I_c =8A I_t=8.93A m=0.8

Formula: $I_t=I_c\,(1+m^2/2)^{1/2}$

$8.93=8(1+m^2/2)^{1/2}$

m=0.701

$It=8\,(1+0.8^2/2)^{1/2}$

It=9.1A

33. A 1MHz carrier is amplitude modulated by 400Hz modulating signal to a depth of 50%.The modulated carrier power is 1KW.Calculate the power of the unmodulated signal.

Solution:-

P_c=1KW, m_a=0.5=50%

$$P_t = P_c\left[1+\frac{m_a{}^2}{2}\right] = 1\times10^3\left[1+\frac{(0.5)^2}{2}\right]$$

$$P_t = 1.125KW$$

The increase in power is given by 1.125-1=0.125KW is contained in two side bands.

34. What do you mean by Hibert transform and inverse Hilbert Transform? And write few applications of Hilbert transform?

- It may be observed that the function $x_h(t)$ obtained by providing(-Π/2)phase shift to every frequency component present in x(t),actually represents the Hilbert transform of x(t).This means that $x_h(t)$ is the Hilbert transform of x(t) defined as

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

Also, the inverse Hilbert transform is defined as

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

- Few applications of Hilbert transform.
- For generation of SSB signals,
- For designing of minimum phase type filters,
- For representation of band pass signals.

35. Define super heterodyne principle. (MAY/JUNE – 2015)

The process of mixing two different frequencies to produce a new frequency is called as heterodyning. It is also termed as mixer or converter. It is used to get fixed Intermediate Frequency (IF) in AM detection

- It can be defined as the process of operation of modulated waves to obtain similarly modulated waves of different frequency. This process uses a locally generated carrier wave, which determines the change of frequency.
- And also we can define that a device performs the frequency translation of a modulated signal is known as a frequency mixer. the operation is often called frequency mixing, frequency conversion, or heterodyning.

PART-B

1.Obtain the Derivation of Double side Band Suppressed Carrier system

In DSBSC, the total power consists of only LSB and USB power but not the carrier power, since it is suppressed here.

$$P^1_{total} = P_{LSB} + P_{USB}$$

t

$$\therefore P_{total} = \frac{MV_c}{8R} + \frac{MV_c}{8R} = \frac{MV_c}{4R} = \frac{m}{2}\left[\frac{V_c}{2R}\right] = \frac{m}{2}P_c$$

But in AM, $P_{total} = P_c\left[1 + \frac{m^2}{2}\right]$

Now, Power savings $= \dfrac{P_{total} - P^1_{total}}{P_{total}}$

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

$$\therefore \text{Power savings in \%} == \frac{P_c + P_c.\frac{m^2}{2} - P_c.\frac{m^2}{2}}{P_c\left(1 + \frac{m^2}{2}\right)} \times 100 = \frac{P_c}{P_c\left(1 + \frac{m^2}{2}\right)}$$

$$= \frac{1}{1 + \frac{m^2}{2}} \times 100 = \frac{2}{2 + m^2} \times 100$$

When m=1,

Power Savings in % $= \dfrac{2}{2 + 1} \times 100 = \dfrac{2}{3} \times 100$

=66.67%

In DSBSC, 66.67% of power is saved due to suppression of carrier wave.

2.Describe the Generation of AM

Two basic amplitude modulation principles are discussed. They are square law modulation and switching modulator.

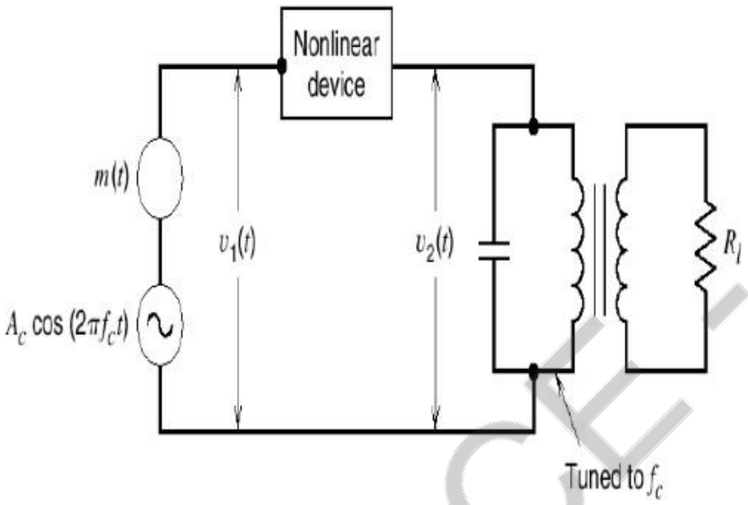
Square Law Modulator

When the output of a device is not directly proportional to input throughout the operation, the device is said to be non-linear. The Input-Output relation of a non-linear device can be expressed as

$$V_0 = a_0 + a_1 V_{in} + a_2 V_{in}^2 + a_3 V_{in}^3 + a_4 V_{in}^4 + \ldots\ldots$$

When the in put is very small, the higher power terms can be neglected. Hence the output is approximately given by $V_0 = a_O + a_1 V_{in} + a_2 V_{in}^2$

When the output is considered up to square of the input, the device is called a square law device and the square law modulator is as shown in the figure



Consider a non-linear device to which a carrier $c(t)=A_c\cos(2\pi f_c t)$ and an information signal m(t) are fed simultaneously as shown in figure 4. The total input to the device at any instant is

$$V_{in} = c(t) + m(t)$$
$$V_{in} = A_c \cos 2\pi f_c t + m(t)$$

As the level of the input is very small, the output can be considered up to square of the input, *i.e.*, $V_0 = a_O + a_1 V_{in} + a_2 V_{in}^2$

$$V_0 = a_0 + a_1[A_c \cos 2\pi f_c t + m(t)] + a_2[A_c \cos 2\pi f_c t + m(t)]^2$$

$$V_0 = a_0 + a_1 A_c \cos 2\pi f_c t + a_1 m(t) + \frac{a_2 A_c^2}{2}(1 + \cos 4\pi f_c t) + a_2[m(t)]^2 + 2a_2 m(t) A_c \cos 2\pi f_c t$$

$$V_0 = a_0 + a_1 A_c \cos 2\pi f_c t + a_1 m(t) + \frac{a_2 A_c^2}{2}\cos 4\pi f_c t + a_2 m^2(t) + 2a_2 m(t) A_c \cos 2\pi f_c t$$

Taking Fourier transform on both sides, we get

$$V_0(f) = (a_0 + \frac{a_2 A_c^2}{2})\delta(f) + \frac{a_1 A_c}{2}\left[\delta(f - f_c) + \delta(f + f_c)\right] + a_1 M(f) +$$

$$\frac{a_2 A_c^2}{4}\left[\delta(f - 2f_c) + \delta(f + 2f_c)\right] + a_2 M(f) + a_2 A_c\left[M(f - f_c) + M(f + f_c)\right]$$

Therefore the square law device output 0 V consists of the dc component at f = 0. The information signal ranging from 0 to W Hz and its second harmonics are signal at f_c and $2f_c$.

$$V_{in} = c(t) + m(t)$$
$$V_{in} = A_c \cos 2\pi f_c t + m(t)$$

As the level of the input is very small, the output can be considered up to square of the input, *i.e.*, $V_0 = a_O + a_1 V_{in} + a_2 V_{in}^2$

$$V_0 = a_0 + a_1[A_c \cos 2\pi f_c t + m(t)] + a_2[A_c \cos 2\pi f_c t + m(t)]^2$$

$$V_0 = a_0 + a_1 A_c \cos 2\pi f_c t + a_1 m(t) + \frac{a_2 A_c^2}{2}(1 + \cos 4\pi f_c t) + a_2[m(t)]^2 + 2a_2 m(t) A_c \cos 2\pi f_c t$$

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$$\frac{a_2 A_c^2}{4}[\delta(f - 2f_c) + \delta(f + 2f_c)] + a_2 M(f) + a_2 A_c[M(f - f_c) + M(f + f_c)]$$

Frequency band centered at f_c with a deviation of $\pm W$, Hz.

The required AM signal with a carrier frequency f_c can be separated using a band pass filter at the out put of the square law device. The filter should have a lower cut-off frequency ranging between 2W and (f_c-W) and upper cut-off frequency between (f_c+W) and 2 f_c

Therefore the filter out put is

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

If $m(t) = A_m \cos 2\pi f_m t$, we get

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

Comparing this with the standard representation of AM signal,

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

Therefore modulation index of the output signal is given by

$$m = 2\frac{a_2}{a_1} A_m$$

The output AM signal is free from distortion and attenuation only when $(f_c - W) > 2W$ or $f_c > 3W$.

Switching Modulator

Consider a semiconductor diode used as an ideal switch to which the carrier signal $c(t) = A_c \cos(2\pi f_c t)$ and information signal $m(t)$ are applied simultaneously as shown figure

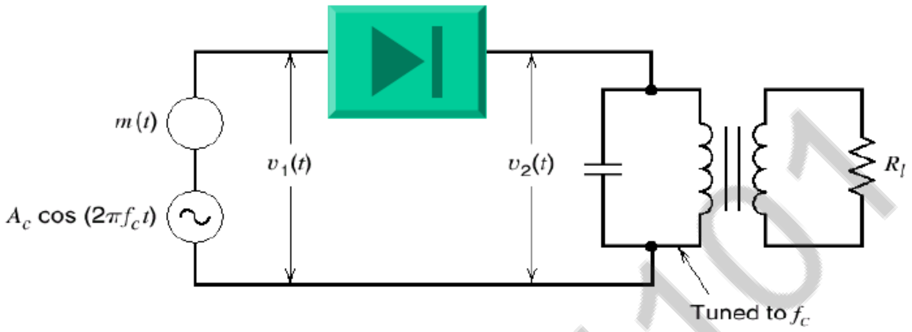


Fig.5. Switching Modulator

The total input for the diode at any instant is given by

$$v_1 = c(t) + m(t)$$
$$v_1 = A_c \cos 2\pi f_c t + m(t)$$

When the peak amplitude of c(t) is maintained more than that of information signal, the operation is assumed to be dependent on only c(t) irrespective of m(t).

When c(t) is positive, v2=v1since the diode is forward biased. Similarly, when c(t) is negative, v2=0 since diode is reverse biased. Based upon above operation, switching response of the diode is periodic rectangular wave with an amplitude unity and is given by

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

$$p(t) = \frac{1}{2} + \underset{n=0,1}{\frac{2}{\pi}}\cos(2\pi f_c t) - \underset{n=-1,+2}{\frac{2}{3\pi}\cos(6\pi f_c t)} + -$$

Therefore the diode response V_o is a product of switching response $p(t)$ and input v_1.

$$v_2 = v_1 * p(t)$$

$$V_2 = \left[A_c \cos 2\pi f_c t + m(t)\right]\left[\frac{1}{2} + \frac{2}{\pi}\cos 2\pi f_c t - \frac{2}{3\pi}\cos 6\pi f_c t + - + -\right]$$

Applying the Fourier Transform, we get

$$\begin{aligned} V_2(f) = {} & \frac{A_c}{4}\left[\delta(f - f_c) + \delta(f + f_c)\right] + \frac{M(f)}{2} + \frac{A_c}{\pi}\delta(f) \\ & + \frac{A_c}{2\pi}\left[\delta(f - 2f_c) + \delta(f + 2f_c)\right] + \frac{1}{\pi}\left[M(f - f_c) + M(f + f_c)\right] \\ & - \frac{A_c}{6\pi}\left[\delta(f - 4f_c) + \delta(f + 4f_c)\right] - \frac{A_c}{3\pi}\left[\delta(f - 2f_c) + \delta(f + 2f_c)\right] \\ & - \frac{1}{3\pi}\left[M(f - 3f_c) + M(f + f_c)\right] \end{aligned}$$

The diode output v_2 consists of

a dc component at $f = 0$.

Information signal ranging from 0 to w Hz and infinite number of frequency bands centered at f, $2f_c$, $3f_c$, $4f_c$, ---------

The required AM signal centred at fc can be separated using band pass filter. The lower cut off-frequency for the band pass filter should be between w and fc-w and the upper cut-off frequency between fc+w and 2fc. The filter output is given by the equation

2.Explain briefly the methods of generation of DSB-SC(APRIL-MAY2019)

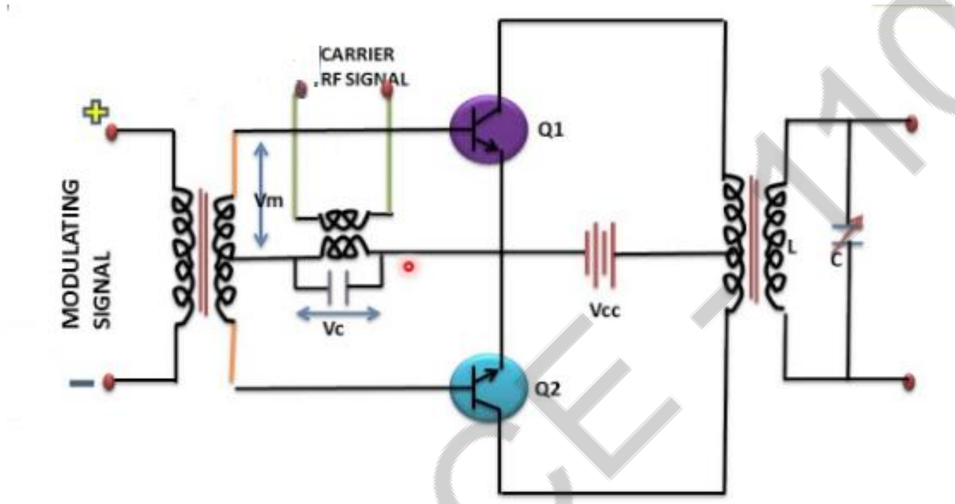
Generation methods for DSBSC Wave

Balanced modulator A balanced modulator consists of two standard amplitude modulators arranged in a balanced configuration so as to suppress the carrier wave as shown in the following block diagram. It is assumed that the AM modulators are identical, except for the sign reversal of the modulating wave applied to the input of one of them.

Here, two non-linear devices are connected in balanced modulator in order to suppress the carrier wave.

Two transistors are indentical and the circuit is symmetrical

This circuit is similar to that of AM except that feeding point of input signals are interchanged.



Analysis

The modulating voltage across the two windings of centre top transformer are equal & opposite in phase i.e. $V_m = -V^1_m$

The input voltage to

Transistor T_1 is $V_{bc} = V_c + V_m = V_c \sin\omega_c t + V_m \sin\omega_m t$

The input voltage to

Transistor T_2 $V^1_{bc} = V_c + V^1_m = V_c \sin\omega_c t - V_m \sin\omega_m t$

Using square law equation, the collector current can be written as

$$i_c = a_1 V_{bc} + a_2 V^2_{bc}$$

$$i^1_c = a_1 V^1_{bc} + a_2 V^{1^2}_{bc}$$

Substitute V_{bc} and V^1_{bc} in above equations,

$$i_c = a_1[V_c \sin\omega_c t + V_m \sin\omega_m t] + a_2[V_c \sin\omega_c t + V_m \sin\omega_m t]^2$$

$$= a_1 V_c \sin\omega_c t + a_1 V_m \sin\omega_m t + a_2 V^2_c \sin\omega^2_c t + a_2 V^2_m \sin\omega^2_m t + 2a_2 V_m V_c \sin\omega_m t \sin\omega_c t$$

$$i^1_c = a_1[V_c \sin\omega_c t - V_m \sin\omega_m t] + a_2[V_c \sin\omega_c t - V_m \sin\omega_m t]$$

$$= a_1 V_c \sin\omega_c t - a_1 V_m \sin\omega_m t + a_2 V^2_c \sin\omega^2_c t + a_2 V_m \sin\omega^2_m t - 2a_2 V_m V_c \sin\omega_m t \sin\omega_c t$$

The output voltage $\vartheta_0(t)$ is given by

$$\vartheta_0(t) = K\left(i_c - i^1_c\right)$$

After substituting i_c and i^1_c in $\vartheta_0(t)$

$$\vartheta_0(t) = k[2a_1V_m \sin\omega_m t + 4a_2V_mV_c \sin\omega_m t \sin\omega_c t]$$
$$= 2ka_1V_m \sin\omega_m t + 4ka_2V_mV_c \sin\omega_m t \sin\omega_c t$$
$$= 2ka_1V_m \sin\omega_m t\left[1 + \frac{2a_2}{a_1}V_c \sin\omega_c t\right]$$
$$\vartheta_0(t) = 2ka_1V_m \sin\omega_m t[1 + m\sin\omega_c t]$$

Where m is modulation index $= \dfrac{2a_2V_c}{a_1}$

This $\vartheta_0(t)$ is the generalized equation for DSBSC using Balanced modulator

3) Explain Ring Modulator or Double balanced modulator

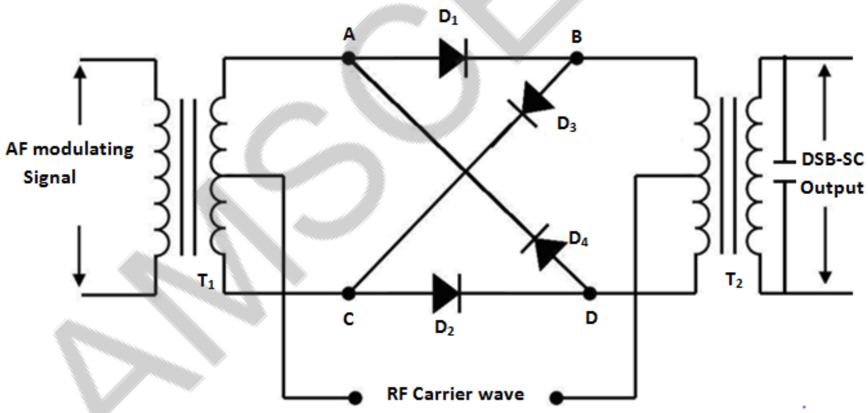
This circuit consists of four diodes and it has a constant forward resistance r_f when switched ON and it has a constant backward resistance r_b when switched OFF.

Diagram

The carrier signal is connected between centre taps of input and output transformers.

During positive half cycle of carrier signal, diodes D_1 & D_2 are switched to their forward resistance and it acts as forward bias. The current divides equally in the upper and lower portions of the primary winding of Tr_2.

The current in the upper part of the winding produces a magnetic field is equal and opposite to the magnetic field produced by the current in the lower half of the secondary. ∴ Magnetic fields cancel each other & no output at the secondary and the carrier is suppressed.



D_1 and D_2 are switched ON to forward resistance during positive half cycle

Diodes D_3 and D_4 are switched to forward resistance during negative half cycle

During negative half cycle of carrier signal, diodes D_3 & D_4 are switched to forward resistance and it acts as forward bias. Then current flows in the secondary winding of Tr_1, and the primary winging of Tr_2.

The equal & opposite magnetic fields cancel each other and there is no output, so the carrier is suppressed.

When both carrier and message signals are present and during positive half cycle of carrier, diodes D_1 and D_2

During negative half cycle of carrier signal diodes D_3 & D_4 will conduct and D_1 and D_2 will not conduct.

Analysis

Message signal $\vartheta_m(t) = V_m \sin\omega_m t$

Carrier signal $\vartheta_c(t) = V_c \sin\omega_c t$

Output signal $\vartheta_{DSBSC}(t) = \vartheta_m(t).\vartheta_c t$

$$= \vartheta_m \sin\omega_m(t).V_c \sin\omega_c t$$

$$\vartheta_{DSBSC}(t) = \frac{V_m V_c}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

This is the generalized equation for DSBSC signal since contains only lower and upper side bands.

4)Describe the Generation methods of SSB Wave

Phase discrimination method for generating SSB wave:

Time domain description of SSB modulation leads to another method of SSB generation using the equations 9 or 10. The block diagram of phase discriminator is as shown in figure 15.

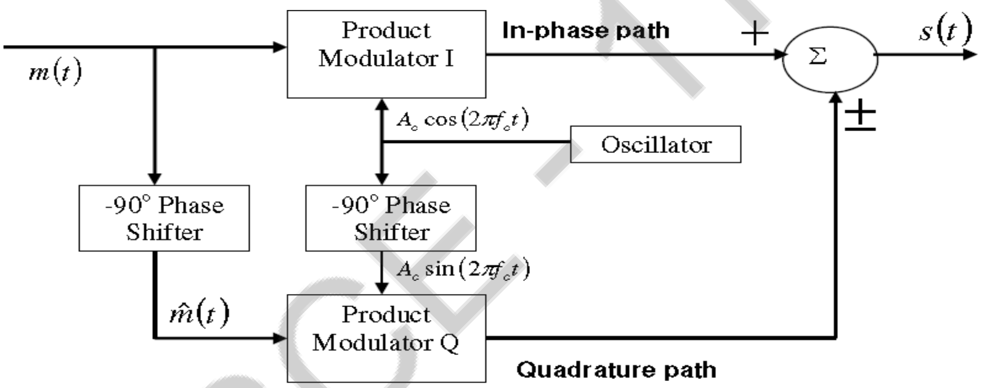


Figure 15 : Block diagram of phase discriminator

The phase discriminator consists of two product modulators I and Q, supplied with carrier waves in-phase quadrature to each other. The incoming base band signal m(t) is applied to product modulator I, producing a DSBSC modulated wave that contains reference phase sidebands symmetrically spaced about carrier frequency fc.

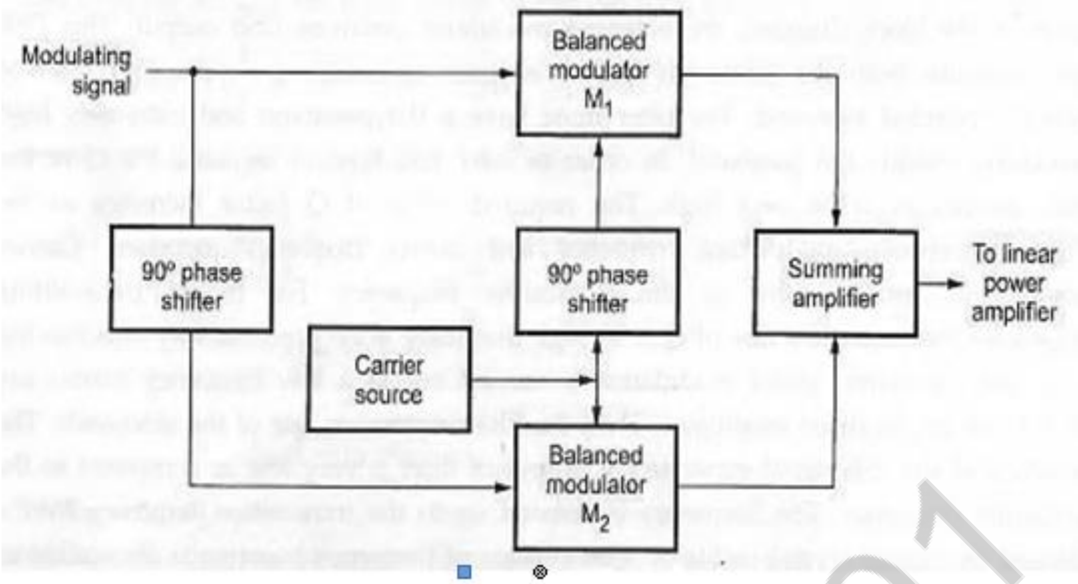
The Hilbert transform mˆ (t) of m (t) is applied to product modulator Q, producing a DSBSC modulated that contains side bands having identical amplitude spectra to those of modulator I, but with phase spectra such that vector addition or subtraction of the two modulator outputs results in cancellation of one set of side bands and reinforcement of the other set.

The use of a plus sign at the summing junction yields an SSB wave with only the lower side band, whereas the use of a minus sign yields an SSB wave with only the upper side band. This modulator circuit is called Hartley modulator.

5)Explain Modified Phase Shift method

Phase Shift Method to Generate SSB

shows the block diagram of phase shift method to generate SSB. The carrier signal is shifted by 90 and applied to the balanced modulator M1. The modulating signal is also directly applied to the balanced modulator M2. The modulating signal is phase shifted by 90 and applied to balanced modulator M2. Both the modulators produce an output consisting of only sidebands. The upper balanced modulator (M1) generates upper sideband and lower sideband, but upper sideband is shifted by +90 whereas lower sideband is shifted by -90.The output of balanced modulators are added by the summing amplifier. Since upper sidebands of both the modulators are phase shifted by $+90^0$, they are in phase and add to produce double amplitude signal. But lower sideband of the balanced modulator s are (+90 , -90) 180 out of phase and hence cancel each other. Thus the output of summing amplifier contains only upper sideband SSB signal. The carrier is already suppressed by balanced modulators.



Message signal $\vartheta_m(t) = V_m \sin \omega_m t$

$\vartheta_o(t) = 2V_o \sin \omega_o t$

$\vartheta_c(t) = 2V_1 \sin \omega_1 t$

o/p of BM 1 $= 2V_o \sin\left(\omega_o t + 90^0\right) V_m \sin \omega_m t$

$= V_m V_o \left[\cos\left((\omega_o t - \omega_m t) + 90^0\right) - \cos\left((\omega_o t + \omega_m t) + 90^0\right)\right]$

o/p of BM 2 $= 2V_o \sin \omega_o t V_m \sin \omega_m t$

$= V_m V_o [\cos(\omega_o t - \omega_m t) - \cos(\omega_o t + \omega_m t)]$

PF in BM 1 & 2 eliminates USB of modulator

o/p pf 2pf$_1$ = $V_m V_o \cos\left((\omega_o t - \omega_m t) + 90^0\right)$

o/p of 2pf$_2$ = $V_m V_o \cos(\omega_o - \omega_m) t$ Assume $V_m = V_o = 1$

o/p of BM 3 $= 2 \sin \omega_c t \cos\left(\omega_o t - \omega_m t + 90^0\right)$

$= \sin\left[(\omega_c + \omega_o - \omega_m) t + 90^0\right] + \sin\left[(\omega_c - \omega_o + \omega_m) t - 90^0\right]$

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

o/p of BM 4 $= 2 \sin\left(\omega_c t + 90^0\right) \cos(\omega_o - \omega_m) t$

$= \sin\left(\omega_c t + 90^0 + \omega_o t - \omega_m t\right) + \sin\left(\omega_c t + 90^0 - \omega_o t + \omega_m t\right)$

$= \sin\left[(\omega_c + \omega_o - \omega_m) t + 90^0\right] + \sin\left[(\omega_c - \omega_o + \omega_m) t + 90^0\right]$

o/p of Summer circuit is

$\vartheta_o(t) = 2 \sin\left(\omega_c + \omega_{o-}\omega_m\right) t + 90^0$

Diagram

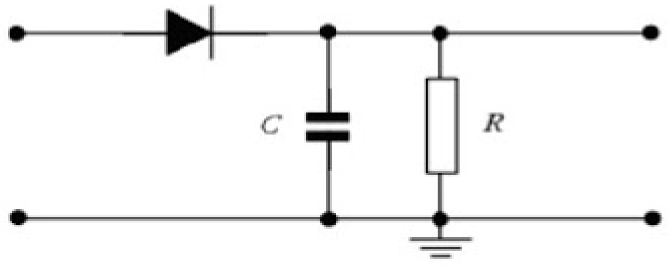
o/p Spectrum for Modified Phase shift method

6)Explain Envelope Detector or Diode Detector

This detector is used to detect message signal from the AM signal

It produces output signal that follows the envelope input AM signal exactly.

Diagram



The general AM wave is applied at the input of Diode.

For every positive half cycle, the diode is forward biased, and it will charge the capacitor to the peak value of input voltage. As soon as, the capacitor charges to the peak value, the diode stops conducting.

Now the capacitor will discharge through R between the positive peaks. The discharging process continues until the next positive half cycle. Then the process repeats.

RC Time Constant

The capacitor charges when Diode is ON and it discharges through R when Diode is OFF.

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

The discharging time constant RC should be long enough capacitor discharges slowly through R.

The spikes are introduced due to charging discharging of capacitor. It can be reduced to a negligible amount by keeping the time constant RC large

If demodulator output contains frequencies that were not present in the message signal during transmission, the demodulator is said to contain Non-linear distortion

There are two types of distortion present in Envelope detector

1) Diagonal peak clipping 2) Negative peak clipping

1) Diagonal Peak Clipping

This distortion occurs when RC time constant of load circuit is too long.

If RC time constant is too long, the discharging curve becomes horizontal. And negative peaks of detected envelope will be partially missing.

The recovered baseband signal i.e. message signal is distorted at negative peaks.

Diagram

The RC time constant should be chosen so that it should compromise between the following two facts.

i) The spikes or fluctuations in envelope detector should be minimum.

ii) Negative peaks of detected envelope should not be missed even partially

2) Negative peak clipping

This occurs due to over modulation effect which is taking place in detector

Diagram

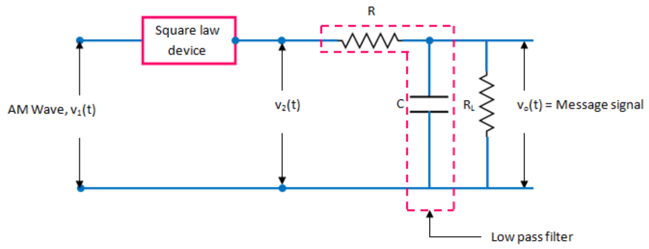
Here, modulation index on the o/p side of detector is higher than that of input side. So that only, over modulation is taking place at output of detector.

The only way to eliminate distortion Is to choose the RC time constant properly.

7) Explain briefly about Square Law Detector

Low level signal can be detected using this method in which the device is operating in non-linear region in order to detect message signal

Diagram



V_d is used to adjust the operating point

The operating is limited to non-linear region due to which the lower half portion of current waveform is compressed. This causes envelope distortion.

Using square law equation,

$i=a_1V+a_2V^2$

$V \rightarrow$ input modulated signal $= V_c\left(1+m\sin\omega_m t\right)\sin\omega_c t$

$$i=a_1\left[V_c\left(1+m\sin\omega_m t\right)\sin\omega_c t\right]+a_2\left[V_c\left(1+m\sin\omega_m t\right)\sin\omega_c t\right]^2$$

After simplifying this equation,

It will have $2\omega_c, 2\left(\omega_c \pm \omega_m\right)$....... besides the input frequency terms.

When is current is passed through LPF, it will pass frequencies upto ω_m and it will suppress the other higher components. $\therefore$ The message signal with frequency ω_m is recovered.

The non-linear characteristics of the diode produces additional frequency components. Frequencies centered about ω_c and $2\,\omega_c$ are easily suppressed using LPF as they are far away from ω_m But $2\,\omega_m$ is very close to ω_m and hence it cannot be totally suppressed by LPF.

$\therefore$ This component $2\,\omega_m$ introduces distortion.

This distortion term $2\,\omega_m$ cannot be completely eliminated and so the square law detector cannot provide a distortionless AM detection.

8)

Write The Comparison Of Amplitude Modulation Systems.(MAY/JUNE 2012)

Description on	AM with carrier	DSB – SC – AM	SSB – SC - AM	VSB - AM
Band width	2fm	2fm	fm	fm <BW<2fm
Power Saving for Sinusoid al	33.33%	66.66%	83.3%	75%

Power Saving for non - Sinusoidal	33.33%	50%	75%	75%
Generation methods	Easier to generate	Not difficult	More difficult to generate	Difficult. But easier to generate than SSB-SC
Detection methods	Simple & Inexpensive	Difficult	More difficult	Difficult

9)Explain Synchronous Detector and obtain the expressions

* This detector is used to detect the message signal from DSBSC and SSB signal.

(i) DSB-SC

First case, we are considering the DSB-SC signal as the incoming signal,

$$\therefore \vartheta_1(t) = V_m V_c \sin\omega_m t \sin\omega_c t \rightarrow \textcircled{1}$$

$$\vartheta_2(t) = V \sin\omega_c t \rightarrow \textcircled{2}$$

$$\text{o/p of mixer} = \vartheta_1(t).\vartheta_2(t) = V_m V_c V \sin^2 \omega_c t \sin\omega_m t$$

$$= V_m V_c V \left[\frac{1-\cos 2\omega_c t}{2}\right] \sin\omega_m t$$

$$= \frac{V_m V_c V}{2} [\sin\omega_m t - \sin\omega_m t \cos 2\omega_c t]$$

Now the LPF passes the first component when attenuating the second component.

$$\therefore \vartheta_o(t) = \frac{V_m V_c V}{2} \sin w_m t$$

Assume the carrier signal $\vartheta_2(t)$ to have a phase difference

$$\textcircled{2} \Rightarrow \vartheta_2(t) = V \sin(\omega_c t + \varphi)$$

$$\vartheta_1(t) = V_m V_c \sin\omega_m t \sin\omega_c t$$

$$\text{o/p of mixer} = \vartheta_1(t)\vartheta_2(t) = V_m V_c V \sin\omega_m t \sin\omega_c t \sin(\omega_c t + \varphi)$$

$$= V_m V_c V \sin\omega_m t \sin\omega_c t \left[\frac{\sin\omega_c t \cos\varphi + \cos\omega_c t \sin\varphi}{\cancel{2}}\right]$$

$$= \frac{V_m V_c V}{\cancel{2}} \sin\omega_m t \sin^2 \omega_c t \cos\varphi + \frac{V_m V_c V}{\cancel{2}} \sin\omega_m t \sin\omega_c t \cos\omega_c t \sin\varphi$$

$$= \frac{V_m V_c V}{\cancel{2}} \sin\omega_m t \left(\frac{1-\cos^2 \omega_c t}{2}\right) \cos\varphi + \frac{V_m V_c V}{\cancel{2}} \sin\omega_m t \sin\omega_c t \cos\omega_c t \sin\varphi$$

$$= \frac{V_m V_c V}{2} \sin\omega_c t \cos\varphi - \frac{V_m V_c V}{2} \sin\omega_m t \cos 2\omega_c t \cos\varphi + V_m V_c V \sin\omega_m t \sin\omega_c t \cos\omega_c t \sin\varphi$$

When it is passed through LPF, it removes all high frequency terms, Then $\vartheta_o(t) = \frac{V_m V_c V}{2} \sin w_m t \cos\varphi$

when $\varphi = 0 \Rightarrow \vartheta_o(t)$ is maximum

$\varphi = 90 \Rightarrow \vartheta_o(t)$ is zero i.e. minimum

ii) SSB-SC

Second case, SSB-SC signal is considered as incoming signal,

$$\therefore \vartheta_1(t) = \frac{V_m V_c}{2} \cos(\omega_c - \omega_m) t$$

$$\vartheta_2(t) = V \sin \omega_c t$$

o/p of mixer

$$= \vartheta_1(t).\vartheta_2(t) = \frac{V_m V_c V}{2} \sin \omega_c t \cos(\omega_c - \omega_m) t \quad \left[\sin A \cos B = \frac{\sin(A - B) + \sin(A + B)}{2}\right]$$

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

Here, the second component is attenuated by the filter

$$\therefore \vartheta_o(t) = \frac{V_m V_c V}{2} \sin w_m t$$

Assume the carrier signal $\vartheta_2(t)$ to have a phase difference,

$$\vartheta_2(t) = V \sin(\omega_c t + \varphi)$$

$$\vartheta_1(t) = \frac{V_m V_c}{2} \cos(\omega_c - \omega_m) t$$

$$\therefore \vartheta_o(t) = \vartheta_1(t).\vartheta_2(t) = \frac{V_m V_c V}{2} \cos(\omega_c - \omega_m) t \sin(\omega_c t + \varphi)$$

$$= \frac{V_m V_c V}{2} [\cos \omega_c t \cos \omega_m t + \sin w_c t \sin w_m t][\sin w_c t \cos\varphi + \cos \omega_c t \sin\varphi]$$

$$= \frac{V_m V_c V}{2} \left[\cos^2 \omega_c t \cos \omega_m t \sin\varphi + \cos \omega_c t \sin \omega_c t \sin \omega_m t \sin\varphi\right]$$

$$= \frac{V_m V_c V}{2} \left[\begin{array}{l} \left(\frac{1 - \cos^2 \omega_c t}{2}\right) \sin \omega_m t \cos\varphi + \cos \omega_c t \sin \omega_c t \cos \omega_m t \cos\varphi + \\ \cos^2 \omega_c t \cos \omega_m t \sin\varphi + \cos \omega_c t \sin \omega_c t \sin \omega_m t \cos\varphi \end{array}\right]$$

$$= \frac{V_m V_c V}{4} \sin \omega_m t \cos\varphi - \frac{V_m V_c V}{4} \cos^2 \omega_c t \sin \omega_m t \cos\varphi + \frac{V_m V_c V}{2} \cos \omega_c t \sin \omega_c t \cos \omega_m t \cos\varphi +$$

$$\frac{V_m V_c V}{2} \cos^2 \omega_c t \cos \omega_m t \sin\varphi + \frac{V_m V_c V}{2} \cos \omega_c t \sin \omega_c t \sin \omega_m t \cos\varphi$$

When it passes to LPF, the o/p will be

$$\vartheta_o(t) = \frac{V_m V_c V}{4} \sin \omega_m t \cos\varphi$$

$$\text{when } \varphi = 0,\ \vartheta_o(t) = \frac{V_m V_c V}{4} \sin \omega_m t$$

10).A transmitter using AM has unmodulated carrier output power of 10KW and can be modulated to a maximum depth of 90% by a sinusoidal modulating voltage without causing overloading. Find the value to which unmodulated carrier power may be increased without resulting in overloading if the maximum permitted modulating index is restricted to 40 %.

Soln:

P_c= 10 KW , m = 0.9

$P_T = P_c\,[\,1+ m^2\,]$

$= 10[1+ (0.9)^2 /2]$

= 14 KW.

This is the maximum power which may be handled by the transmitter without using overload. If the modulation index is changed to 40% then the increased modulated carrier is then given by

$14 = P_c\,[1+ (0.4)^2 /2\,]$

P_c = 12.96 KW.

11).A sinusoidal carrier voltage of frequency 1MHz and amplitude 100 V is modulated by a sinusoidal voltage of frequency 5KHz producing 50% Modulation. Calculate the frequency and amplitude of USB and LSB.

Soln:

Frequency of USB = 1 MHz + 5KHz = 1005KHz.

Frequency of LSB = 1 MHz - 5KHz = 995 KHz.

Amplitude of USB and LSB = $(m_a E_c)/ 2$ = 100 /4 = 25 V.

$$P_{saving} = \frac{P_{DSBFC}-P_{SSB}}{P_{DSBFC}} \times 100$$

$$= \frac{1.125\,P_c - 0.0625\,P_c}{1.125\,P_c}$$

= 94.5 %

UNIT-II

2-MARK QUESTION AND ANSWERS:

1. Define frequency modulation.

Frequency modulation is defined as the process by which the frequency of the carrier wave is varied in accordance with the instantaneous amplitude of the modulating or message signal.

2. Define modulation index of frequency modulation.

It is defined as the ratio of maximum frequency deviation to the modulating frequency.

$m_f = \Delta\,\omega_m = KV_m/\omega_m = \delta$

3.How is narrowband signal distinguished from wideband signal? (Nov/Dec – 2017) (or) Compare narrowband and wideband FM. (MAY/JUNE – 2015, NOV/DEC – 2013, MAY/JUNE – 2011,APRIL/MAY 2019)

S. No.	Narrowband Signal	Wideband Signal
1.	Modulation index is less than 1.	Modulation index is greater than 1.
2.	It contains two sidebands and carrier.	It contains infinite number of sidebands and carrier.
3.	Bandwidth = $2f_m$	Bandwidth = $2(\delta + f_{m\,(max)})$
4.	Maximum Deviation is 5 kHz	Maximum Deviation is 75 kHz
5.	It is used for FM Mobile Communication like Police wireless, ambulance etc.	It is used for Entertainment broadcasting.

4. What do you meant by multitone modulation, Percent modulation?

- Modulation done for the message signal with more than one frequency component is called multitone modulation.
- The term percent modulation as it is used in reference to FM refers to the ratio of actual frequency deviation to the maximum allowable frequency deviation.

Percent modulation $M = \frac{\Delta f_{actual}}{\Delta f_{\max}} \times 100$

5. Define phase modulation.

Phase modulation is defined as the process of changing the phase of the carrier signal in accordance with the instantaneous amplitude of the message signal.

6. What do you mean by angle modulation? And write their types?

- Angle modulation may be defined as the process in which the total phase angle of a carrier wave is varied in accordance with the instantaneous value of the modulating or message signal while keeping the amplitude of the carrier constant.
- Phase modulation (PM) and Frequency modulation (FM) are the types of angle modulation.

7.A frequency modulated signal is given as $s(t) = 20\cos[2\pi f_c t + 4\sin(200\pi t)]$. Determine the required transmission bandwidth. (Nov/Dec – 2017)

Solution:

Given $s(t) = 20\cos[2\pi f_c t + 4\sin(200\pi t)]$

Compare the given FM signal equation with standard

FM signal, $s(t) = E_c\cos[2\pi f_c t + m_f \sin 2\pi f_m t)]$

Here $E_c = 20$ V, $m_f = 4$,

$2\pi f_m t = 200\pi t \Rightarrow f_m$(or) $f_{m(max)} = 100$ Hz

We know that, modulation index of FM

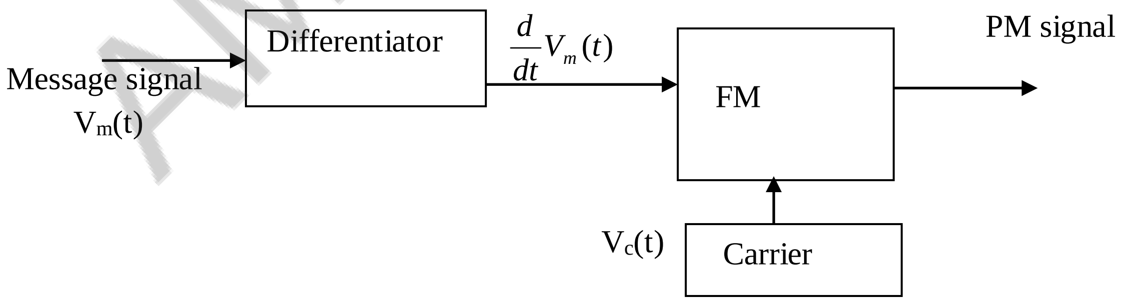
$(m_f) = \delta/f_m$ Hence $\delta = m_f f_m = 4 \times 100$ Hz = 400 Hz.

Bandwidth of FM = $2(\delta + f_{m(max)})$

= 2(400 + 100) = 2(500) = 1000 Hz (or) 1 kHz.

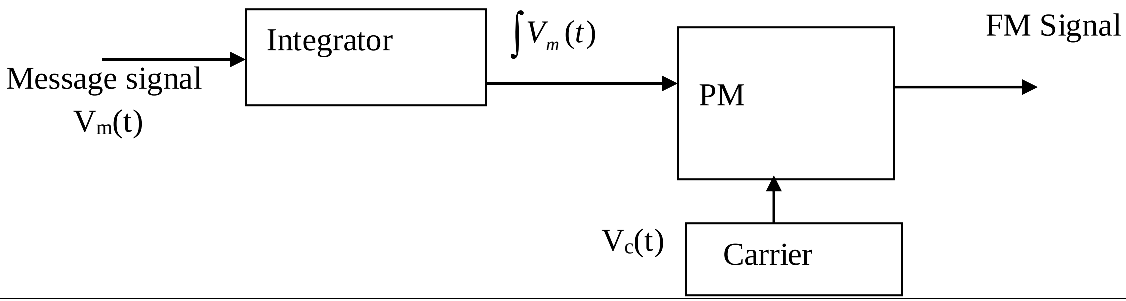
8. How FM wave can be converted to PM wave?

- The PM wave can be obtained from FM by differentiating the modulating signal before applying it to the frequency modulator circuit.



9. How PM wave can be converted to FM wave?

- The FM wave can be obtained from PM by integrating the modulating signal before applying it to the phase modulator circuit.



10..State the Carson's rule. (MAY/JUNE – 2017, MAY/JUNE – 2015, MAY/JUNE – 2013)

FM bandwidth is given as twice the sum of the frequency deviation and the highest modulating frequency.

Carson's rule of FM bandwidth = $2(\delta + f_{m\,(max)})$

Where $f_{m(max)}$ is the maximum modulating signal frequency and δ is maximum frequency deviation.

11. What are the types of Frequency Modulation?

(OR)

What do you mean by Narrow Band and Wide band?

Based on the modulation index FM can be divided into types. They are Narrow band FM and Wide band FM. If the modulation index is greater than one then it is wide band FM and if the modulation index is less than one then it is Narrow band FM

12. Compare FM to PM

* The FM system having greater Modulation index results in larger band width.	PM system generally uses a smaller bandwidth because of smaller modulation index.
* In FM, the modulation index is increased when the modulating frequency is increased & vice versa.	In PM, when the modulating frequency is changed, the modulation index in PM remains constant.

13.Distinguish the features of Amplitude Modulation (AM) and Narrowband Frequency Modulation (NBFM). (MAY/JUNE – 2017)

S. No.	Amplitude Modulation (AM)	Narrowband Frequency Modulation (NBFM)
1.	If the modulation index is less than or equal to 1, then it is called AM	If the modulation index is less than 1, then it is called NBFM.
2.	Most of the power is in carrier hence less efficient.	All the transmitted power is useful.
3.	AM receivers are not immune to noise.	FM receivers are immune to noise.

4.	Adjacent channel interference is present.	Adjacent channel interference is avoided due to guard bands.
5.	It is used in Two-way radios, Very High Frequency aircraft radio, Citizen's Band Radio.	It is used in FM mobile communication like police wireless and ambulances.

14. What is the basic difference between an AM signal and a Narrowband FM signal?

In the case of sinusoidal modulation, the basic difference between an AM signal and a narrowband FM signal is that the algebraic sign of the lower side frequency in the narrow band FM is reversed.

15. Compare Wideband FM and Narrowband FM.[April-04]

Parameter/Characteristics	**Wideband FM**	**Narrowband FM**
Modulation index	Greater than 1	Less than or slightly greater than 1
Maximum Deviation	75KHz	5KHz
Range of Modulating Frequency	30Hz to 15KHz	30 Hz to 3KHz
Maximum Modulation index	5 to 2500	Slightly greater than 1
Bandwidth	Large, about 15 times higher than BW of narrowband FM	Small, approximately same as that of AM
Applications	Entertainment broadcasting	FM Mobile Communication like Police wireless, ambulance etc.

16. What are the advantages of Angle Modulation?

Angle modulation has several inherent advantages over Amplitude modulation.

1. Noise immunity
2. Noise performance and signal-to-noise improvement.
3. Capture effect.
4. Power utilization and efficiency.

17. What is transmission bandwidth of FM?(APRIL-MAY 2019)

- For 'n'side bands the bandwidth of FM wave is given by

$$B.W=2n\omega_m \text{ radians/sec}$$

$$B.W=2nf_m \text{ Hz}$$

18. What are the two methods of producing an FM wave?

Basically there are two methods of producing an FM wave. They are,

i) Direct method

In this method the transmitter originates a wave whose frequency varies as function of the modulating source. It is used for the generation of NBFM

ii) Indirect method

In this method the transmitter originates a wave whose phase is a function of the modulation. Normally it is used for the generation of WBFM where WBFM is generated from NBFM

20.What is the need for pre-emphasis and de-emphasis in FM? (NOV/DEC – 2016, MAY/JUNE – 2016)

Pre-emphasis and de-emphasis circuits are used for the suppression of unwanted noise. The noise has greater effect on higher modulating frequencies than on the lower ones. The effect of noise on the higher frequencies can be reduced by artificially boosting them at transmitter and correspondingly attenuating them at the receiver. This is done with the help of pre-emphasis (boosting at transmitter side) and de-emphasis (attenuating at receiver side) circuits.

.

21. Give the frequency spectrum of Narrow band FM. (NOV/DEC 2018)

22. List the properties of the Bessel function.

The properties of the Bessel function is given by,

i) $J_n(\beta)=(-1)^n J_{-n}(\beta)$ for all n, both positive and negative.

ii) For small values of the modulation index β ,we have

$J_0(\beta)=1$

$J_1(\beta)=\beta/2$

$J_n(\beta)=0, n>2.$

iii) $\sum_{n=-\infty}^{\infty} J^2_n(\beta)=1$

23. What are the types of FM detectors?

Slope detector and phase discriminator.

24. What are the types of phase discriminator?

Foster seely discriminator and ratio detector.

25. What are the disadvantages of balanced slope detector?

1. Amplitude limiting cannot be provided
2. Linearity is not sufficient

3. It is difficult to align because of three different frequency to which Various tuned circuits to be tuned.
4. The tuned circuit is not purely band limited.

26. Define capture effect.

With FM and PM, a phenomenon known as the capture effect allows a receiver to differentiate between two signals received with the same frequency, providing one signal atleast twice as high in amplitude as the other; the receiver will capture the stronger signal and eliminate the weaker signal.

27. What is FM thresholding?

With the use of limiters, FM and PM demodulators can actually reduce the noise level and improve the signal to noise ratio during the demodulation process. This is called FM thresholding.

28. Define Pre-emphasis and de-emphasis.

Noise at the higher-modulating signal frequencies is inherently greater in amplitude than noise at the lower frequencies. i.e the higher - modulating-signal frequencies have a lower signal-to-noise ration than the lower frequencies. To compensate for this, the high- frequency modulating signals are emphasized or boosted in amplitude in the transmitter. To compensate for this boost, the high-frequency signals are attenuated or deemphasized in the receiver after demodulation has been performed.

29. State the Carson's rule.

- Carson's rule provides a thumb formula to calculate the bandwidth of a single tone wide band FM.According to this rule the FM bandwidth is given as twice the sum of the frequency deviation and the highest modulating frequency.however, it must be remembered that this rule is just an approximation.
- Mathematically B.W=$2(\Delta\omega+\omega_m)$

30.A carrier signal is frequency modulated by a sinusoidal signal of 5 V_{pp} and 10 kHz. If the frequency deviation constant is 1 kHz/V, determine the maximum frequency deviation and state whether the scheme is narrowband FM or wideband FM. (MAY/JUNE – 2016)

Solution:

Given $E_m = 5\ V_{pp}$

Where V_{pp} denotes peak to peak amplitude. Then the maximum positive amplitude is given by $E_m = 5/2 = 2.5$ V,

$k_f = 1$ kHz/V = 1000 Hz = 1×10^3 Hz , $f_m = 10$ kHz =

10×10^3 Hz. Maximum frequency deviation, $\delta = E_m \times$

k_f

$= 2.5 \times 1 \times 10^3 = 2500$ Hz.

Since modulation index is less than 1, then this is Narrowband FM.

31. What is the use of crystal controlled oscillator?

The crystal-controlled oscillator always produces a constant carrier frequency there by enhancing frequency stability.

32. In a FM wave the frequency deviation is 25 KHz. What is the Modulation index when the modulating signal frequency is 100Hz &10 KHz?

Soln:

$$\Delta f = 25 \text{ KHz} \quad ; \quad m_f = \frac{\Delta f}{fm} = \frac{\Delta\omega}{\omega_m}$$

when $f_m =$ 100 Hz

$$m_f = \frac{25000}{100} = 250 \text{ radians}$$

when $f_m =$ 10,000 Hz

$$m_f = \frac{25000}{10{,}000}$$

$$m_f = 2.5 \text{radians}$$

33.A carrier is frequency modulated with a sinusoidal signal of 2 KHz resulting in a maximum frequency deviation of 5 KHz. Find (i) Modulation index (ii) Bandwidth of the modulated signal.

Solution:

Given data : Modulating frequency $f_m =$ 2 K Hz
Maximum frequency deviation $= \Delta f =$ 5 KHz

i. Modulation index = $m_f = \frac{\Delta f}{f_m}$

$$= \frac{5\times 10^3}{2\times 10^3} = 2.5$$

ii. Bandwidth of the modulated signal is given by,

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

$$= 2\,(5 \times 10^3 + 2 \times 10^3)$$

= 14 KHz.

34. A 2 KHz audio signal modulates a 50MHz carrier causing a frequency deviation of 2.5KHz.Determine the bandwidth of FM signal.

.

f_m = 2 K Hz

f_c = 50MHz

Δf = 2.5 KHz

Modulation index = m_f $= \frac{\Delta f}{f_m} = \frac{2.5KHz}{2KHz} = 1.25$

B.W=2 f_m=2x2x10^3=4KHz.

35. Determine the bandwidth of a wideband FM; given that carrier Signal of 100MHz frequency modulates a signal of 5 KHz with the 50 KHz as frequency deviation.

B.W for WBFM=2Δf_m

Modulation index = m_f $= \frac{\Delta f}{f_m} = \frac{50KHz}{5KHz} = 10KHz$

Δf_m=50 KHz

f_m = 5K Hz

f_c = 100KHz

B.W=2x50KHz=100KHz

DESCRIPTIVE QUESTIONS:

Unit –II

1)Describe the concept of Angle Modulation (or) Exponential modulation

Angle modulation is the process in which the angle (frequency or phase) of the carrier signal is varied in accordance with the amplitude of message signal.

Angle modulated wave can be mathematically represented as

$$\vartheta_a(t) = V_c \cos(\omega_c t + \theta(t))$$

$$= V_c \cos[\varphi(t)] \rightarrow ①$$

Where $\vartheta_a(t)$ $\rightarrow$angle modulated wave

V_c $\rightarrow$amplitude of carrier signal

$\omega_c t$ $\rightarrow$angular frequency of carrier signal

$\theta(t)$ $\rightarrow$phase of carrier signal

$\varphi(t) = [\omega_c t + \theta(t)]$ $\rightarrow$Angle of carrier signal

Advantages of Angle modulation over Amplitude modulation

*Noise reduction
* Improved system fidelity
* More efficient use of power

Disadvantages of Angle modulation compared to AM
* Requires wide Bandwidth
* Utilizing more complex circle

Applications of Angle modulation
* Television Sound transmission
* Cellular radio
* Two way mobile radio
* Microwave & Satellite communication

2) Explain Frequency modulation (FM) (APRIL-MAY2019)

The frequency of the carrier signal is varied in accordance with the amplitude of message signal.
This is called Frequency modulation.
The positive message signal produces increase in frequency whereas the negative message signal produces decrease in frequency.
In FM, maximum frequency deviation occurs during the maximum positive and negative peaks of message signal.

Let message signal $\vartheta_a(t) = V_m \cos\omega_m t$

Carrier signal $\vartheta_c(t) = V_c \cos\omega_c t$

This carrier signal can also be denoted as

$\vartheta_c(t) = V_c \cos(\omega_c t + \theta)$

$\vartheta_c(t) = V_c \cos \rightarrow ②$

Where, $\vartheta_m \rightarrow$ amplitude of message signal

$\vartheta_c \rightarrow$ amplitude of carrier signal

$\omega_m t \rightarrow$ angular frequency of message signal

$\omega_c t \rightarrow$ angular frequency of carrier signal

$\theta \rightarrow$ phase of carrier signal

$\varphi \rightarrow$ phase angle of carrier signal

$\varphi \Rightarrow \omega_c t + \theta$ from②

Now differentiate this φ w.r.t "t"

$$\frac{d\varphi}{dt} = \omega_c \Rightarrow \varphi = \int \omega_c dt \rightarrow ③$$

In angle modulation, the frequency is varied for FM. So, we have to find instantaneous frequency ω_1. And during FM, the frequency is varied in accordance with instantaneous value of message signal, so

$\omega_1 = \omega_c + K_{FM}\vartheta_m(t)$

$\omega_i = \omega_c + K_{FM} V_m \cos\omega_m t \rightarrow ④$

Where $K_{FM} \rightarrow$ deviation sensitivity for FM

Deviation sensitivity K_{FM} is defined as how much output frequency is varied w.r.t amplitude of message signal $K_{FM} = \frac{\Delta\omega}{V_m}$

Now, similar to equation, the instantaneous phase angle of carrier signal is $\varphi_i = \int \omega_i \, dt \rightarrow ⑤$

$$\varphi_i = \int (\omega_c + K_{FM} V_m \cos \omega_m t) dt$$

$$\varphi_i = \omega_c t + K_{FM} V_m \frac{\sin \omega_m t}{\omega_m}$$

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

where

$\Delta\omega = K_{FM} V_m \rightarrow$ freq. deviation

$\frac{\Delta\omega}{\omega_m} \rightarrow$ modulation index for FM

Mf

$$\varphi_i = \omega_c t + m_f \sin \omega_m t \rightarrow$$

The FM signal can be written as (similar to eqn. 2)

$$\vartheta_{FM}(t) = V_c \cos \varphi_i \rightarrow$$

$$\vartheta_{FM}(t) = V_c \cos[\omega_c t + m_f \sin \omega_m t]$$

w.k.t $\omega_c t = 2\pi f_c t$

$$\therefore \boxed{\vartheta_{FM}(t) = V_c \cos[2\pi f_c t + m_f \sin 2\pi f_m t]}$$

This equation is the general representation of FM signal by considering single frequency component i.e. single tone FM

If modulation is done with more than one frequency component, that is called Multi tone FM. The equation is

3) ExplainPhase modulation (PM)

The phase of the carrier signal is varied in accordance with the amplitude of message signal. This process is called Phase modulation.

(or)

If message signal varies phase of carrier signal, will give rise to phase modulation.

Diagram

In PM, maximum frequency deviation occurs during zero crossing of message signal

Follow the same steps as in FM, and here

$$\vartheta_m(t) = V_m \cos \omega_m t \rightarrow$$

Differentiating this equation w.r.t "t",

$$\frac{d}{dt} \vartheta_m(t) = \frac{d}{dt}[V_m \cos \omega_m t] = -V_m \sin \omega_m t . \omega_m \rightarrow$$

From the diagram, output of frequency modulator is

$$\vartheta_{Pm}(t) = V_c \cos \varphi i \rightarrow ③$$

Where $\varphi i = \int \omega_i dt$, and $\omega_i = \omega_c + K_{PM} \vartheta_m(t)$

④

$\omega_i = \omega_c + K_{PM}[-V_m \sin \omega_m t . \omega_m] \rightarrow ⑤$ where $K_{PM} \rightarrow$ deviation sensitivity of PM.

Deviation sensitivity K_{PM} is defined as how much output phase is varied w.r.t amplitude of message signal $K_{PM} = \frac{\Delta\theta}{V_m}$

Now, sub eqn. ⑤ in ④

$$\varphi i = \int [\omega_c - K_{PM} \omega_m V_m \sin \omega_m t]$$

$$= \omega_c t + K_{PM} \omega_m V_m \frac{\cos \omega_m t}{\omega_m}$$

$$\varphi i = \omega_c t + K_{PM} V_m \cos \omega_m t$$

Eqn. ③ $\Rightarrow \vartheta_{Pm}(t) = V_c \cos[\omega_c t + K_{PM} V_m \cos \omega_m t]$

$$\boxed{\vartheta_{Pm}(t) = V_c \cos[\omega_c t + MP \cos \omega_m t]}$$

where $K_{PM}V_m = m_p \rightarrow$ modulation index in PM.

This is the general representation of PM signal.

4) Explain briefly the types of frequency modulation

Depending on the value of modulation index (m), the frequency modulation is divided into two types

1) Narrow Band FM
2) Wide Band FM

The Bandwidth of FM signal depends on modulation index.

If modulation index (m) is large compared to one radian the bandwidth of FM signal will be very large. This is called Wideband FM.

If modulation index(m) is small compared to one radian the bandwidth of FM signal will be narrow. This is called Narrowband FM.

consider the general FM signal.

$$\vartheta_{FM}(t) = V_c \cos\left[2\pi f_c t + m_f \sin 2\pi f_m t\right]$$

Expanding this equation, $\because \left(\cos(A+B) = \cos A \cos B - \mathrm{Sin}A\mathrm{Sin}B\right)$

$$V_c\left[\cos 2\pi f_c t.\cos\left(m_f \sin 2\pi f_m t\right) - \sin 2\pi f_c t.\sin\left(m_f \sin 2\pi f_m t\right)\right]$$

①

In Narrow Band FM, the modulation index m_f is small compared to one radian,

$$\therefore \ \cos\left(m_f \sin 2\pi f_m t\right) \approx 1$$

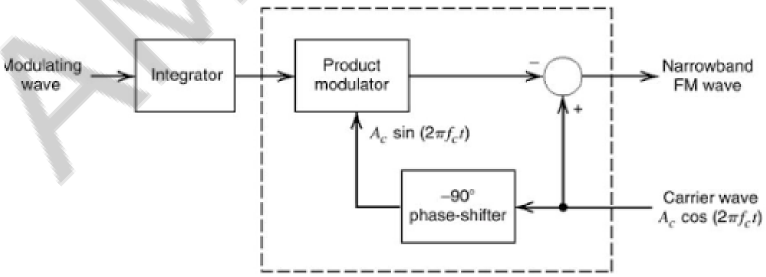
$$\text{and} \sin\left(m_f \sin 2\pi f_m t\right) \approx m_f \sin 2\pi f_m t$$

$$\text{i.e.} \sin(\theta) = \theta$$

$$\therefore \text{eqn ①} \Rightarrow \vartheta_{NBFM}(t) = V_c\left[\cos 2\pi f_c t - \sin 2\pi f_c t.m_f \sin 2\pi f_m t\right]$$

$$\vartheta_{NBFM}(t) = V_c \cos 2\pi f_c t - m_f V_c \sin 2\pi f_c t \ \sin 2\pi f_m t \rightarrow ②$$

The equation defines the approximate form of Narrow Band FM signal. This modulator involves splitting the carrier signal in two paths, one path is direct and another path contains -90^0 phase shifting network and a product modulator.



The carrier signal $V_c \cos 2\pi f_c t$ is applied in two paths, one is shifted i.e. $V_c \sin 2\pi f_c t$ and another one is given to the summer directly. Now, message signal $V_m \cos 2\pi f_m t$ is applied to the integrator and then to the product modulator. This modulator gives the DSBSC modulated signal $m_f V_c \sin 2\pi f_c t \ \sin 2\pi f_m t$ by the combination of message signal and the shifted carrier signal. Now, the difference between this DSBSC signal and the carrier signal gives the Narrow Band FM signal with some distortion.

$$\vartheta_{NBFM}(t) = V_c \cos 2\pi f_c t - m_f V_c \left[\frac{\cos 2\pi(f_c - f_m)t - \cos 2\pi(f_c + f_m)t}{2} \right]$$

$$\vartheta_{NBFM}(t) = V_c \cos 2\pi f_c t - \frac{m_f V_c}{2} \cos 2\pi(f_c - f_m)t - \frac{m_f V_c}{2} \cos 2\pi(f_c + f_m)t$$

Comparing this NBFM equation with AM equation, i.e

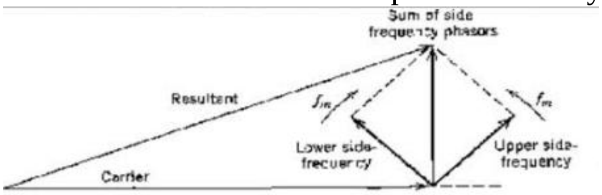
$$\vartheta_{AM}(t) = V_c \cos 2\pi f_c t - \frac{mV_c}{2} \cos 2\pi(f_c - f_m)t - \frac{mV_c}{2} \cos 2\pi(f_c + f_m)t$$

We can know, the difference between the AM signal and a Narrow Band FM signal is that the algebraic of LSB frequency in NBFM is reversed.

Thus NBFM signal requires the same transmission Bandwidth as that of AM signal. i.e. $2f_m$

The NBFM signal can be represented with a phasor diagram, where carrier signal is used as reference.

The resultant of two sideband phasors is always at right angles to carrier phasor.



The effect of this to produce a resultant phasor representing the NBFM signal that is approximately of the same amplitude as the carrier phasor but out of phase w.r.t it.

2) Wideband FM or High index FM

Consider the general FM signal,

$$\vartheta_{FM}(t) = V_c \cos\left[2\pi f_c t + m_f \text{ in } 2\pi f_m t\right] \rightarrow ①$$

This Above equation can be rewritten using exponential form as

$$\vartheta_{WBFM}(t) = V_c.e^{j(2\pi f_c t + m_f \sin 2\pi f_m t)}$$

$$= R_e\left[V_c.e^{j(2\pi f_c t + m_f \sin 2\pi f_m t)}\right]$$

$$= R_e\left[V_c.e^{j2\pi f_c t}.e^{m_f \sin 2\pi f_m t}\right]$$

$$\vartheta_{WBFM}(t) = R_e\left[\vartheta(t).e^{j2\pi f_c t}\right] \rightarrow ②$$

where $\vartheta(t) = V_c.e^{jm_f \sin 2\pi f_m t} \rightarrow ③$

$\vartheta(t)$ is a periodic function of time with fundamental frequency equal to modulation freq f_m

$\therefore$ This eqn. can be expanded by Fourier series.

$$\vartheta(t) = \sum_{n=-\infty}^{+\infty} c_n e^{jn2\pi f_m t} \rightarrow ④$$

The complex fourier coefficient C_n is given by

$$C_n = f_m \int_{-\frac{1}{2}f_m}^{\frac{1}{2}f_m} \vartheta(t).e^{-jn2\pi f_m t}.dt \rightarrow ⑤$$

$$= f_m \int_{-\frac{1}{2}f_m}^{\frac{1}{2}f_m} V_c.e^{-jm_f \sin 2\pi f_m t}.e^{-jn2\pi f_m t}.dt$$

$$C_n = f_m V_c \int_{-\frac{1}{2}f_m}^{\frac{1}{2}f_m} e^{j[m_f \sin 2\pi f_m t - n 2\pi f_m t.]}.dt$$

Now, define a variable x= $2\pi f_m t$

$$\frac{dx}{dt} = 2\pi f_m$$

$$C_n = f_m V_c \int_{-\pi}^{\pi} e^{j[m_f \sin x - nx]}.\frac{dx}{2\pi f_m}$$

$$C_n = \frac{V_c}{2\pi} \int_{-\pi}^{\pi} e^{j[m_f \sin x - nx]}.dx$$

The integral on the RHS of the above eqn. is recognized as nth order Bessel function of the first kind. This eqn. is now expressed as

$$C_n = V_c.J_n(mf) \rightarrow ⑥$$

Where $J_n(mf) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j[m_f \sin x - nx]}.dx$

Now, substitute eqn. ⑥ in ④

$$\vartheta(t) = \sum_{n=-\infty}^{+\infty} V_c.J_n(mf)e^{jn 2\pi f_m t} \rightarrow ⑦$$

To get WBFM signal, sub eqn. ⑦ in ②

$$\vartheta_{WBFM}(t) = R_e\left[\sum_{n=-\infty}^{+\infty} V_c.J_n(m_f).e^{jn2\pi f_m t}.e^{jn2\pi f_c t}\right]$$

$$= V_c \sum_{n=-\infty}^{+\infty} J_n(m_f)\cos\left(2\pi f_c t + n2\pi f_m t\right)$$

$$= V_c \sum_{n=-\infty}^{+\infty} J_n(m_f)\cos 2\pi\left(f_c + nf_m\right) \rightarrow ⑧$$

$$\vartheta_{WBFM}(t) = V_c \sum_{n=-\infty}^{+\infty} J_n(m_f)\cos\left(\omega_c t + n\omega_m t\right)$$

This is the desired from for the fourier series representation of single tone FM signal, for an arbitrary value of m_f. For wideband FM, the modulation index is greater than 1.

The discrete spectrum of $\vartheta_{WBFM}(t)$ is obtained by taking Fourier transform on both sides,

$$\vartheta_{WBFM}(t) = \frac{V_c}{2} \sum_{n=-\infty}^{+\infty} J_n\left(m_f\right)\left[\partial\left(f - f_c - nf_m\right) + \partial\left(f + f_c + nf_m\right)\right]$$

We can plot Bessel function $J_n\left(m_f\right)$ versus modulation index m_f, for various positive integer values of n.

Diagram

Properties of Bessel function

1) $J_n\left(m_f\right) = \left(-1\right)^n J_{-n}(m_f)$, for all n, both positive & negative

2) For small values of modulation index, we have

$$J_o\left(m_f\right) = 1$$

$$J_1\left(m_f\right) = \frac{m_f}{2}$$

$$J_n\left(m_f\right) = 0, n > 2$$

3) $\sum_{n=-\infty}^{+\infty} J^2{}_n\left(m_f\right)=1$

5) What are the methods of generation of FM and Explain.

There are two methods to generate FM signals.

1) Direct method → a) Varactor Diode modulator
→ b) Reactance tube modulator

2) Indirect method → Armstrong method

1) Direct method

In this method, the frequency of carrier signal is varied in accordance with input signal

Varactor Diode modulator

The oscillator whose frequency is controlled by a modulating voltage is called Voltage Controlled Oscillator (vco).

The frequency of vco is varied w.r.t message signal by shunting variable capacitor along with oscillator circuit or tuned circuit. This voltage variable capacitor is known as Varactor.

Variable Capacitor i.e. Varactor is nothing but a varactor diode c_d, which is a specially fabricated PN junction diode. This acts as Variable Capacitance in Reverse Bias condition.

Varactor diode is connected in parallel with resonant circuit of an oscillator through coupling capacitor C_c.

The message signal is fed in series with RPS, then the effective bias to the varactor diode equals the algebraic sum of D.C. Bias voltage V and instantaneous value of message signal.

Finally, the Variable Capacitor C varies with message signal resulting in frequency modulation of the oscillator circuit.

The capacitance of the diode is

$c_d = k(V_D)^{\frac{-1}{2}} \rightarrow ①$

Now, diode voltage V_D is

$V_D = V_o + V_m \sin\omega_m t \rightarrow ③$

The total capacitance of the the oscillator circuit is (c_o+c_d), then instantaneous frequency of oscillation is

$V_D \rightarrow$ total instantaneous voltage across the diode

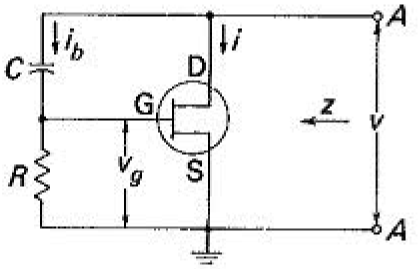
$k \rightarrow$ proportionality constant

$V_o \rightarrow$ polarizing voltage to maintain reverse bias across diode.

$$\omega_i = \frac{1}{\sqrt{L_o\left(c_o + c_d\right)}} \rightarrow$$

$$,\ \omega_i = \frac{1}{\sqrt{L_o\left(c_o + kv_d^{\frac{-1}{2}}\right)}}$$

The instantaneous frequency ω_i is depending on V_D which in turn depends on message signal $V_m(t)$ $\therefore$ It means that oscillator frequency ω_i is $\because$ dependent on message signal $V_m(t)$ & thus frequency modulation is generated.



FET reactance modulator circuit is shown in fig. This behaves as reactance across the terminals A-B.

The terminals A-B of this circuit can be connected across the tuned circuit in order to get FM output.

Here, the varying voltage V across the terminals A-B changes reactance of FET, and this change in reactance may be inductive or capacitive.

The equivalent capacitance value have to be determined.

Gate voltage $V_g = I_b.R$

Base current $I_b = \dfrac{v}{R - jX_c}$

$$\therefore V_g = \frac{V.R}{R - jX_c} \rightarrow$$

The drain current $I_d = g_m x V_g$

$$I_d = \frac{g_m.R.V}{R - jX_c} \rightarrow ②$$

The impedance between the terminals A-B is

$$Z = \frac{V}{\dfrac{g_m R.V}{R - jX_c}} = \frac{R - jX_c}{g_m.R}$$

$$Z = \frac{V}{I_d} \rightarrow ③ \quad Z = \frac{1}{g_m}\left(\frac{R - jX_c}{R}\right) = \frac{1}{g_m}\left(1 - \frac{jX_c}{R}\right) \rightarrow ④$$

$$\rightarrow \text{If } X_c >> R, \text{then}$$

$$Z = \frac{-jX_c}{R} \rightarrow ⑤$$

This impedance is clearly a capacitive reactance which may be rewritten as,

$$Z = \frac{X_c}{g_m.R} = \frac{1}{g_m R.2\pi fc} \qquad X_c = \frac{1}{2\pi fc}$$

$$Z = \frac{1}{2\pi fc_{eq}} \qquad \text{where } c_{eq} = g_m RC$$

This eqn. shows that FET is equivalent to Variable Capacitance C_{eq}.

The equivalent capacitance C_{eq} depends on transconductance g_m, where $g_m = \dfrac{I_d}{V_g}$. Hence C_{eq} can be changed by changing V_g

If condition $X_c >> R$ is not satisfied, then FET will not be purely reactive. It will have resistance part in it.
If $X_c = nR$, where n lies between 5 and 10

$$X_c = \frac{1}{\omega_c} = nR \Rightarrow C = \frac{1}{\omega nR} = \frac{1}{2\pi fnR}$$

Sub. eqn 8 in 7

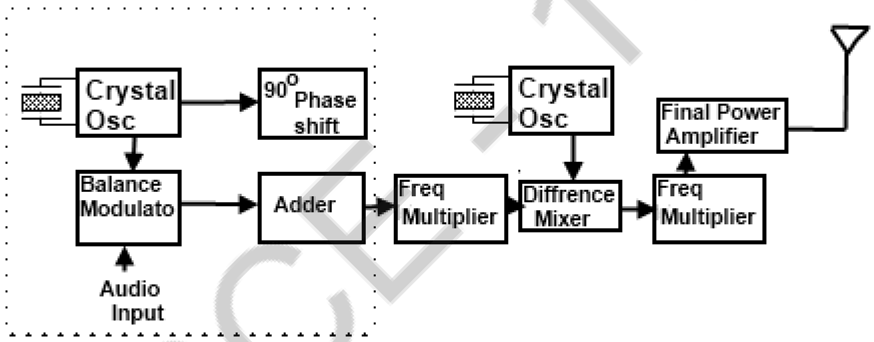
$$C_{eq} = g_m R . \frac{1}{2\pi fnR} = \frac{g_m}{2\pi fn}$$

$$C_{eq} = \frac{g_m}{2\pi fn}$$

2) <u>Indirect method</u>

<u>Armstrong method</u>

Here, the message signal is integrated and then phase modulated with carrier signal so that FM signal is obtained.
The obtained FM signal is NarrowBand FM signal, but frequency multipliers are used to get Wideband FM signal.
The multiplication process is performed in several stages in order to increase the carrier frequency as well as frequency deviation.



ARMSTRONG PHASE MODULATOR

Carrier signal is generated by crystal oscillator.
This carrier signal is applied to 90^0 phase shifter & also to combining Network.
The 90^0 phase shifter produces 90^0 phase shifted carrier. This is applied to the balanced modulator along with the message signal.
The output of Balanced modulator is DSBSC signal, which consists of two sidebands with their resultant in phase with 90^0 shifted carrier.
Now the two sidebands and the carrier signal from crystal oscillator without any phase shift are applied to combining network.
The output of combining network is the resultant of vector addition of carrier and two side bands.
When modulation index is increased, the amplitude of sidebands will also increase, so that amplitude of their resultant increases. This will increase the angle φ made by the resultant with unmodulated carrier.
But the angle φ decreases with reduction in modulation index. Thus the resultant at the output of combining network is phase modulated.
The FM signal is now produced at the output of phase modulator which has low carrier frequency and low modulation index. They can be increased to a certain value with the help of frequency multipliers and mixers.
The power level of the signal is increased by

Derivation

The phase modulated signal is given by

$$\vartheta_{PM}(t) = V_c \cos\left(\omega_c t + m_p \cos\omega_m t\right)$$

The instantaneous angular frequency ω_p is given as

$$\omega_p = \frac{d\theta(t)}{dt}$$ $V_c \rightarrow$ amplitude of carrier signal

Where $\theta(t) = \omega_c t + m_p \sin\omega_m t$ $m_p \rightarrow$ modulation index in PM

$$\frac{d\theta(t)}{dt} \Rightarrow \omega_c + m_p \cos\omega_m t.\omega_m = \omega_p$$

In terms of linear frequency

$$f_p = f_c + m_p f_m \cos\omega_m t$$

The second term in above eqn. represents the frequency shift w.r.t centre frequency

$$f_p = f_c + \Delta f$$

Where $\Delta f = m_p f_m \cos\omega_m t$

Hence, maximum deviation, $\Delta f = m_p f_m$

If message signal frequency f_m remains constant, the frequency deviation will be directly proportional to modulation index m_p.

Thus, as long as message signal frequency does not change, phase modulation produces FM output.

6) What are the methods of demodulation of FM and Explain.

There are two steps involved in demodulation of FM signals.

i) First step converts FM into AM signal by frequency dependent circuits i.e. output voltage depends on input frequency. This is done by frequency discriminators.

ii) In second step, original message signal is recovered from AM signal using Envelope Detector.

FM Demodulation methods

1) Slope Detector
- a) Single tuned or slope Detector
- b) Stagger tuned or Balanced slope Detector

Its operation depends on slope of frequency response characteristics of freq. selective networks.

2) Phase Discriminator
- c) Foster-seeley Discriminator
- b) Ratio Detector

1) <u>Single tuned or slope Detector</u>

Diagram

An FM signal is applied at the input of tuned circuit.

The centre frequency of the FM signal is f_c and frequency deviation is Δf .

Then the resonant frequency of the tuned circuit is adjusted to $f_c + \Delta f$.

Diagram

The amplitude of the output voltage of the tank circuit depends on freq. deviation of input signal.

The circuit is tuned so that its resonant frequency is lower than the carrier freq. f_c

When the signal frequency increases above f_c with modulation, the amplitude of carrier voltage drops. When the signal frequency decreases below "f_c", the carrier voltage rises.

The change of voltage results because of the change in the magnitude of the impedance in the tuned circuit as a function of frequency and results in the effective conversion of frequency modulation into amplitude modulation.

The modulation is recovered from the amplitude modulation by means of normal operation of the circuit i.e. The original message signal is recovered from amplitude modulated signal using Envelope Detector.

2) <u>Balanced slope Detector</u>

This Balanced slope detector consists of two slope detectors.

The transformer is centre tapped transformer. $\therefore$ The input voltages to the two slope detectors are 180^0 out of phase.

The upper tuned circuit of secondary T_1 is tuned above f_c by Δf i.e. $f_c + \Delta f$. Similarly, the lower tuned circuit of secondary is tuned below f_c by Δf i.e. $f_c - \Delta f$.

R_1C_1 and R_2C_2 are the filters used to bypass the RF ripple. V_{01} and V_{02} are the output voltages of two slope detectors. The final output voltage V_0 is obtained by taking subtraction of individual output voltages V_{01} and V_{02} .

<u>Operation of the circuit</u>

i) $f_{in} = f_c$

When input frequency is equal to carrier frequency, the induced voltage in the T_1 winding of secondary is exactly equal to that induced in T_2 winding.

$\therefore$ Input voltage to both the diodes D_1 and D_2 will be same. $\therefore$ Their DC output voltage V_{01} and V_{02} will identical but with opposite polarities. $\therefore$ Net output voltage $V_0 = 0$.

ii) $\underline{f_c < f_{in} < (f_c + \Delta f)}$

Here, the induced voltage in winding T_1 is higher than that in T_2. $\therefore$ Input voltage to D_1 is higher than D_2.

Hence, the positive output V_{01} of D_1 is higher than the negative output V_{02} of D_2.

$\therefore$ Output voltage V_0 is positive.

iii) $\underline{(f_c - \Delta f) < f_{in} < f_c}$

If input frequency is in this range, then induced voltage in winding T_2 is higher than that in T_1. $\therefore$ Input voltage to diode D_2 is higher than that in D_1.

$\therefore$ Output voltage is negative.

If output frequency goes outside the range of $(f_c - \Delta f)$ to $(f_c + \Delta f)$, then output voltage will fall due to reduction in tuned circuit response.

3) <u>Foster-Seeley Discriminator</u>

Diagram

The FM input signal V_1 is applied to the primary winding.

The primary and secondary windings both are tuned to the same centre frequency f_c of the incoming signal.

The phase shift between the primary and secondary voltages of transformer is a function of frequency. The secondary voltage lags primary voltage by 90^0 at centre frequency.

The primary voltage is coupled to centre tap on the secondary through C_p. RFC produces high impedance to frequencies of FM.

The secondary voltage V_2 is equally divided across upper half and lower half of secondary coil.

The voltage across diode D_1 is $V_{D1} = V_1 + \frac{1}{2}V_2$ & the voltage across diode D_2 is

$V_{D2} = V_1 + \frac{1}{2}V_2$

The output voltage $V_0 = V_{D1} - V_{D2}$

<u>Operation of the circuit</u>

i) $\underline{f_{in} = f_c}$

The individual output voltages of two diodes will be equal and opposite. $\therefore$ output $V_o=0$.

ii) $f_{in} > f_c$

V_2 leads V_1 by less than 90^0. $\therefore \frac{1}{2}V_2$ will lead V_1 by less than 90^0 and $-\frac{1}{2}V_2$ will lag behind V_1 by more than 90^0.

The voltage V_{D1} is greater than V_{D2}. Hence output voltage is positive.

iii) $f_{in} < f_c$

V_2 leads V_1 by more than 90^0. Hence $\frac{1}{2}V_2$ will lead V_1 by more than 90^0 and $-\frac{1}{2}V_2$ will lag behind V_1 by less than 90^0.

The voltage V_{D2} is greater than voltage V_{D1}. $\therefore$ output voltage V_o is negative.

Even though the primary & secondary tuned circuit are tuned to the same centre frequency, the voltages applied to the two diodes D_1 and D_2 are not constant.

They vary depending on the frequency of the input signal. This is due to change in phase shift between the primary and secondary windings depending on the input frequency.

4)Ratio Detector

Comparing this circuit with Foster seeley discriminator, we can determine the following changes,

i) The direction of diode D_2 is reversed.
ii) A large value capacitor C_5 is included.
iii) Output is taken somewhere else.

Output voltage due to diode D_1

$$V_o = V_{01} - \frac{VR}{2}$$

$$= V_{01} - \left(\frac{V_{01} + V_{02}}{2}\right) \qquad \because V_R = V_{01} + V_{02}$$

$$V_o = \frac{V_{01} - V_{02}}{2} \rightarrow ①$$

Output voltage due to diode D_2

$$V_o = -V_{02} + \frac{VR}{2}$$

$$= -V_{02} + \left(\frac{V_{01} + V_{02}}{2}\right) \qquad \because V_R = V_{01} + V_{02}$$

$$V_o = \frac{-V_{01} + V_{02}}{2} \rightarrow ②$$

Output voltage of Ratio Detector

Adding equations ① and ②

$$2V_0 = \left(\frac{V_{01} - V_{02}}{2}\right) + \left(\frac{-V_{02} + V_{01}}{2}\right)$$

$$2V_0 = \frac{V_{01}}{2} - \frac{V_{02}}{2} - \frac{V_{02}}{2} + \frac{V_{01}}{2}$$

$$2V_0 = V_{01} - V_{02}$$

$$V_0 = \frac{V_{01} - V_{02}}{2}$$

since $V_{01} = V_{D1}$ and $V_{02} = V_{D2}$,

$$V_0 = \frac{1}{2}\left[V_{D1} - V_{D2}\right]$$

This eqn. shows that the output of ratio detector is half compared to that of Foster seeley discriminator.

$\rightarrow$ When input frequency f_{in} increases above f_c, $V_{D1} > V_{D2}$, $\therefore$ output V_0 is negative.

7)Obtain the Transmission Bandwidth of Angle Modulated Waves.

For a given message signal frequency, FM wave cannot be accommodated in narrower BW since it has infinite number of side bands.

Bandwidth for an angle modulated wave is a function of modulating signal frequency and the modulation index.

Angle modulated wave can be classified on the basis of modulation index (m0.

i) Low index $\rightarrow$ m < 1

ii) Medium index $\rightarrow$ 1 < m < 10

iii) High index $\rightarrow$ m > 10

With low index angle modulation,

BW =$2f_m$

To determine BW for high index modulation, quasi-stationary approach is used.

BW= $2\Delta f$

Actual Bandwidth required to pass all the sidebands is equal to two times the product of highest modulating signal frequency and the number of side bands. Using Bessel function,

BW = 2(n X f_m)

By Carson's rule, BW is equal to twice the sum of peak frequency & highest modulating signal frequency.

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

For low index, $f_m >> \Delta f$

BW=$2f_m$

For high index, $\Delta f >> f_m$

$$BW = 2\Delta f$$

8)Determine the peak frequency deviation Δf and mod. Index (m) for an FM modulator with a deviation sensitivity $K_{FM} = 5KHz / v\ \xi\ a$ mod. Signal

$$\vartheta_m(t) = 2\cos(2\pi 200t)$$

<u>Soln:</u>

Peak Freq. Deviation $\Delta f = K_{FM}V_m = 5k \times 2$

$= 10$ KHz

Mod. Index m=$\frac{\Delta f}{f_m} = \frac{10k}{2000} = 5$

b> Determine the peak phase deviation (m) for a PM modulator with a deviation sensitivity

k=2.5 rad/ϑ ξ a mod. Signal V_m(t)=2cos(2π 2000t)

Peak Phase deviation = Mod. Index
$m=kv_m=2.5x2=5$ rad
m=5 radians

9) If the signal $\vartheta(t)=20\cos\left(6.28\times10^6 t+10\sin 6.28\times10^3 t\right)$ represents a phase modulated signal, determine
a) Carrier freq
b) modulating freq
c) modulation index
d) peak phase deviation
ii) Write the expression for freq. modulated & phase modulated signals if the mod. Freq is doubled?

Soln:

$$\vartheta_p(t)=E_c\cos\left(\omega_c t+m\sin\omega_m t\right)$$

$$V(t)=20\cos\left(6.28\times10^6 t+10\sin 6.28\times10^3 t\right)$$

a) carrier freq $\omega_c=6.28\text{x}10^6$

$$2\pi f_c=6.28\times10^6$$

$$f_c=1\text{MHz}$$

b) Mod. Freq $\omega_m=6.28\text{x}10^3$

$$2\pi f_m=6.28\times10^3$$

$$f_m=1\text{KHz}$$

c) Mod. Index $m_p=\dfrac{kV_m}{\omega_m}=10$

d) Peak phase deviation = mod. index
$\Delta\theta$ = 10 radians

ii) $V_p(t)=E_c\cos\left(\omega_c t+m\sin\omega_m t\right)$

$$\vartheta_f(t)=E_c\cos\left(\omega_c t+\frac{\Delta f}{f_m}\sin\omega_m t\right)$$

10.For an AM modulator with a peak freq. deviation Δf =10KHz, V_c=10V ξ a 500KHz carrier, determine
a) Actual min. BW from Bessel table
b) Approximate min. BW using Carson's rule

Soln:

a) Using Bessel table

$$BW=w(n\times fm)$$

$$m=\frac{\Delta f}{f_m}=\frac{10k}{10k}=1$$

m=1 will yield 3 set of side freq.

$$\therefore n=3$$

BW=2(3x10KHz)
=60 KHz

b) Using Carson's rule

$$BW=2\left(\Delta f+f_m\right)=2\left(10+10=40\text{KHz}\right)$$

11.. Given FM & PM modulator with the following parameters.
FM modulation
deviation sensitivity K_{FM}=1.5KHz/v

carrier frequency f_c=500KHz
modulating signal $\vartheta_m = 2\sin(2\pi 2kt)$

PM modulation

deviation sensitivity K_{PM}=0.75rad/v
carrier frequency f_c=500KHz
modulating signal $\vartheta_m = 2\sin(2\pi 2kt)$

a) Determine the modulation index & sketch the o/p spectrum for both the modulator
b) Change the modulating signal amplitude for both the modulators to $4V_p\xi$ repeat step (a).

Soln:

a) FM modulator

$$m = \frac{K_{FM} V_m}{f_m}$$
$$= \frac{1.5K \times 2}{2K}$$
$$m = 1.5$$

PM modulator

$$m = K_{PM} V_m$$
$$= 0.75 \times 2$$
$$m = 1.5$$

b) FM modulator

$$m = \frac{K_{FM} V_m}{f_m}$$
$$= \frac{1.5K \times 4}{2K}$$
$$m = 3$$

PM modulator

$$m = K_{PM} V_m$$
$$= 0.75 \times 4$$
$$m = 3$$

UNIT-III

RANDOM PROCESS

2 MARK QUESTION AND ANSWERS:

1. Define noise.

Noise is defined as any unwanted form of energy, which tends to interfere with proper reception and reproduction of wanted signal.

2. Give the classification of noise.(NOV/ DEC 2018)

Noise is broadly classified into two types. They are External noise and internal noise. External noise may be defined as that type of noise which is generated external to the communication system. And it can be classified into

1. Atmospheric noise
2. Extraterrestrial noises
3. Man –made noises or industrial noises

Internal noise may be defined as that type of system which is generated internally or within the communication system or receiver. it can be classified into

1. Thermal noise

2. Shot noise
3. Transit time noise
4. Miscellaneous internal noise

3.State Central Limit Theorem. (Nov/Dec – 2017, Nov/Dec – 2016, May/June – 2016)

Central Limit theorem: It provides mathematical justification for using Gaussian process for large no. of individual events

$$V_N = \frac{1}{\sqrt{N}} \sum_{i=1}^{N} Y_i$$

When random variable N approaches infinity, then probability distribution of V approaches normalized Gaussian distribution.

When infinitely large number of identically and independently distributed random variables is added, the resultant is Gaussian distributed.

4. Define Atmospheric noise and industrial noise?

- Atmospheric noise, which is also called static, is produced by lightning discharges in thunderstorms and other natural electrical disturbances which occur in the atmosphere.
- The industrial noise or man-made noise is that type of noise which is produced by such sources as automobiles and aircraft ignition, electrical motors, switch gears and leakage from high voltage transmission lines and several other heavy electrical equipments.

5. Define shot noise?

- Shot noises arises in active devices due to random behaviour of charge carriers. In electron tubes, shot noise is generated due to random emission of electrons from cathodes, whereas in semiconductor devices shot noise is generated due to random diffusion of minority carriers or simply random generation and recombination of electron-hole pairs.

6. Define partition noise?

- Partition noise is generated in a circuit when a current has to divide between two or more paths. This means that partition noise results from the random fluctuations in the division.

7. Define flicker noise (or) Low frequency noise.

- Flicker noise is the one appearing in transistors operating at low audio frequencies. Flicker noise is produced at low frequencies (below few KHz).This noise is also called as flicker noise (1/f noise).

8.What is meant by ergodic process? (Nov/Dec – 2017)

A random process is called ergodic process if time averages are equal to ensemble averages. Thus for ergodic process, $m_X(t) = m_x(T)$ and $R_X(t_1, t_2) = R_x(\tau,T)$

Where $m_X(t)$ is ensemble mean, $m_x(T)$ is time mean, $R_X(t_1, t_2)$ is ensemble autocorrelaion and $R_x(\tau,T)$ is time autocorrelation

9. Define Transit-Time noise or high frequency noise.

- It is generally observed in semiconductor devices, when transit-time of charge carriers crossing a junction is comparable with time period of the signal.
- Some charge carriers diffuse back to the source, this process gives rise to the input admittance and it affects the conductance with increase in frequency. This conductance produces transit time noise.

10. Define Avalanche noise?

- The reverse bias characteristic of a diode shows a region where the reverse current increases rapidly with a slight increase in magnitude of the reverse bias voltage. That is voltage increase current also increases.
- This is because the holes and electrons in the depletion region gain sufficient energy from reverse bias to ionize atoms by collision. This collision provides spikes in current in avalanche region. This noise is called as avalanche noise.

11. Define Thermal noise?

- The thermal noise or white noise or Johnson noise is the random noise which is generated in a resistor or the resistive component of complex impedance due to rapid and random motion of the molecules, atoms and electrons.

12. Write an expression for thermal noise generated in a resistor.

- The expression for maximum noise power output of a resistor may be given as

$$P_n \propto T.B = K.T.B$$

Where K-Boltzmann's constant=1.38×10^{-23}Joule/deg.K

T-absolute temperature.

B-bandwidth of interest in Hz.

13.List the necessary and sufficient conditions for the process to be WSS. (MAY/JUNE – 2017)

A process X(t) is Wide-Sense Stationary (WSS) if the following conditions are satisfied:

- Mean of random process is equal to the expected value of random process.

$m_X(t) = E[X(t)]$ is independent of t.

Autocorrelation $R_X(t_1, t_2)$ depends only on the time difference $\tau = t_1 - t_2$ and not on t_1 and t_2 individually

14.Define WSS.

A random process is said to be Wide Sense Stationary or Weak Sense Stationary (WSS), if it's mean and autocorrelation functions are independent of shift of time origin

15. When a random process is called to be strict sense or strictlystationary? (MAY/JUNE – 2010)(APRIL-MAY2019)

When the statistical properties of a random process do not change with time, then it is called stationary random process.

16. Define non-stationary process.

A process is said to be non-stationary if the statistical properties are function of time.

17. State the properties of autocorrelation function.

a. The autocorrelation function is an even function of τ. (i.e.,) $R_X(\tau) = R_X(-\tau)$

b. The autocorrelation function has its maximum value of $\tau = 0$ (i.e.,) $|R_X(\tau)| \leq R_X(0)$

c. The autocorrelation function shows conjugate symmetry.

d. The autocorrelation function $R_x(\tau)$ and energy spectral density $\psi(f)$ form a Fourier transform pair.

18. State the properties of Gaussian process.

a. If a Gaussian process is applied at the input of stable linear filter, then the output is also Gaussian.

b. Let the Gaussian process be sampled at times $t_1, t_2, \ldots, t_n$. Then the set of random variables is obtained $X(t_1), X(t_2), \ldots, X(t_n)$ due to sampling. This set of random variables is jointly Gaussian for any n.

c. If the Gaussian process is stationary, then Gaussian process is also strictly stationary.

d. If the random variables obtained due to sampling, the Gaussian process are uncorrelated, then they are statistically independent.

19. List out the properties of Power Spectral Density (PSD).

a. Autocorrelation function $R(\tau)$and Power Spectral Density S(f) form a Fourier transform pair. (i.e.,)

b. The area under the Power Spectral Density function S(f) gives average power (P) of the signal. (i.e.,)

20. Define random process.

A Random process X(s,t) is a function that maps each element of a sample space into a time function called sample function. Random process is a collection of time functions.

21. Define transmission of a random process through a LTI filter.

When the stationary random process X(t) with mean m_x and autocorrelation function $R_x(\tau)$ is passed through a Linear Time – Invariant (LTI) filter of impulse response h(t), producing a new random process Y(t) at the filter output.

22.Difference between random variable and random process. (APRIL-MAY2019)

S. No.	Random variable	Random process
1.	A random variable is a mapping from the sample space to the set of real numbers.	A Random process X (s,t) is a function that maps each element of a sample space into a time function called sample function.
2.	It is not the function of time.	It is the function of time.
3.	Random variables are not further classified.	Random processes can be stationary or ergodic.
4.	Only ensemble averages can be calculated.	Ensemble as well as time averages can be calculated.

23.Define jointly Wide Sense Stationary processes.

The two processes X(t) and Y(t) are called jointly Wide Sense Stationary if each of X(t) and Y(t) is Wide Sense Stationary (WSS) and their cross-correlation depends only upon time difference τ. (i.e.,)

$$R_{XY}(t, t+\tau) = E[X(t)\,Y(t+\tau)] = R_{XY}(\tau)$$

Where $R_{XY}(t, t+\tau)$ and $R_{XY}(\tau)$ are crosscorrelation functions.

24.Difference between SSS and WSS processes.

S. No.	Strict Sense Stationary (SSS) process	Wide Sense Stationary (WSS) process
1.	All statistical properties do not change with time.	Mean and autocorrelation do not change with time.
2.	Ideally, this process does not have start and end.	This process does have start and end at finite times.
3.	Such processes are not physically possible.	Such processes are physically possible.
4.	It appears stationary at all the times.	It appears stationary over certain period of time.

25.. Give the representation of narrowband noise in terms of envelope And phase components.

Narrowband noise in terms of envelope and phase components as

$$n(t) = r(t)Cos[2\pi f_c t + \psi(t)]$$

where

$$r(t) = [n_I{}^2(t) + n_Q{}^2(t)]^{\frac{1}{2}}$$

and

$$\psi(t) = \tan^{-1}\left[\frac{n_Q(t)}{n_I(t)}\right]$$

The function r (t) and ψ (t) are called envelope and phase of n (t).

26. What is the figure of merit of DSB-SC system?

- The figure of merit of a DSB-SC system is 2.

27. What is the figure of merit of SSB-SC system?

The figure of merit of an SSB-SC system is 1.

DESCRIPTIVE QUESTIONS:

1. Describe theProperties of Power Spectral density (APRIL-MAY2019)

1) The power spectral density of stationary process for zero frequency value is equal to the total area under the graph of auto correlation function.

Substitute f=0 in eqn. 1

$$S_X(o) = \int_{-\infty}^{\infty} R_X(z)\,dz$$

2) The mean square value of stationary process equals the total area under the graph of power spectral density.
Substitute z=0 in eqn 2

$$R_X(0) = \int_{-\infty}^{\infty} S_X(f)df$$

$$= E\left[X^2(t)\right]$$

$$\text{i.e.} E\left[X^2(t)\right] = \int_{-\infty}^{\infty} S_X(f)df$$

3) Power Spectral Density of stationary process is non-negative.
$S_X(f) \geq 0$ for all f

4) PSD of real valued random process is an even function of frequency $S_X(-f) = S_X(f)$

Cross Spectral Densities

If X(t) and Y(t) are two stationary process, then its cross correlation function are denoted as $R_{XY}(z)$ and $R_{YX}(z)$. Then the Cross Spectral densities are given by

$$S_{XY}(f) = \int_{-\infty}^{\infty} R_{XY}(z)\exp\left(-j2\pi fz\right)dz$$

$$S_{YX}(f) = \int_{-\infty}^{\infty} R_{YX}(z)\exp\left(-j2\pi fz\right)dz$$

2.In rolling six face dice problem, find the probability of occurrence of 4 if it is known that event face has appeared.

Given

Probability of occurence of 4 : P(X)= $\frac{1}{6}$

Probability of occurence of even space : P(Y)= $\frac{3}{6} = \frac{1}{2}$

Solution

Probability of an event X when event Y is known is given by

$$P(X/Y) = \frac{P(X \cap Y)}{P(Y)} \qquad \text{w.k.t } P(X \cap Y) = P(X)$$

$$= \frac{1/6}{1/2} = \frac{1}{6} \times \frac{2}{1}$$

$$P(X/Y) = \frac{1}{3}$$

3.Describe the concept of RANDOM VARIABLE

It is a function which maps every outcome of a sample space to a real number.

Two types

i) Discrete Random Variable

Eg. A coin is tossed 10 times. The random Variable X is no. of tails that are noted. X can take the values from 0,1,2,3……10. $\therefore$ X is a discrete random variable.

ii) Continous Random Variable

Eg. A light bulb is burner until it burns out. The random variable Y is life time of the bulb in hours. Y can take any positive real value, so Y is continuous random variable.

Mean or Expected Value

The mean of a random variable is its average value or central value. It is defined by μ or E(X).

If X is a discrete random variable with possible values $X_1, X_2 \ldots\ldots X_n$ and $P(X_i)$ denotes $P(X=X_i)$, then

$$\mu = E(X) = \sum_{i=1}^{n} X_i P(X = X_i)$$

If X is a continuous random variable with probability density function f(x), then

$$\mu = E(X) = \int_{-\infty}^{+\infty} X f_x(x) dx$$

Properties of Mean

i) E[AX+B]=AE(X)+B

ii)Mean value exists only when

$$|E(X)| < \infty$$

iii) For even $f_X(X)$

$E(X) = 0$

Variance

The variance of a random variable gives an idea of how the values of random variable are spreaded widely. It is denoted by Var(X) or σ^2.

$$\mathrm{Var}(X) = \sigma^2 = E\left[X - E(X)\right]^2$$

$$= E(X^2) - [E(X)]^2$$

Properties of Variance

i) Var[AX] = A^2Var[X]

ii) Var[X] = E[X^2]-M^2_x

<u>Probability Distribution Function</u>

Or

Cumulative Distribution Function

$$F_X(X) = P(X \le x)$$

<u>Properties:</u>

i) $F_X(\infty) = 1$

ii) $F_X(-\infty) = 0$

iii) $0 \le F_x(X) \le 1$

<u>Probability Density function</u>

$$f_x(X) = \frac{d}{dx} F_x(X)$$

<u>Properties</u>

i) $f_x(X) \ge 0$

ii) $\int_{-\infty}^{+\infty} f_x(X)dx = 1$

iii) $F_x(X) = \int_{-\infty}^{+\infty} f_x(\lambda)d\lambda$

iv) $P\{X_1 < X < X_2\} = \int_{X_1}^{X_2} f_x(X)dx$

Two Random Variables

In some situations, there may be two random variables, and that can be explained by Joint distribution function (or) Joint density function.

<u>Joint Distribution function</u>

$$F_{x,y}(X, Y) = P\{X \le x, Y \le y\}$$

<u>Joint Density function</u>

$$f_{xy}(x, y) = \frac{\partial^2 F_{xy}(x, y)}{\partial x \partial y}$$

$$\& F_{xy}(x, y) = \int_{-\infty}^{x}\int_{-\infty}^{y} f_{xy}(x, y)dxdy$$

<u>Joint Movement</u>

$$E\left[XY\right] = \int_{-\infty}^{+\infty}\int_{-\infty}^{+\infty} xy\, f_{xy}(x, y)dx\,dy$$

This is also called Correlation of random variables X & Y

$$R_{XY} = E\left[XY\right]$$

Covariance

$$E\left[\left(X-\mu_x\right)\left(Y-\mu_Y\right)\right]=\int_{-\infty}^{+\infty}\int_{-\infty}^{+\infty}\left(x-\mu_x\right)\left(y-\mu_y\right)f_{xy}(x,y)dx\,dy$$

$$\sigma_{X,Y}:Cov(X,Y)=E\left[\left(x-\mu_x\right)\left(y-\mu_y\right)\right]$$

Correlation Coefficient

$$P_{XY}=\frac{Cov(X,Y)}{\sigma_x\sigma_y}=\frac{\sigma_{xy}}{\sigma_x\sigma_y}$$

Marginal Density function

$$f_X(X)=\frac{dF_X(x)}{dx}$$

$$f_Y(X)=\frac{dF_Y(y)}{dy}$$

Independent Random Variable

I) $F_{XY}(x,y)=F_X(x)F_Y(y)$

ii) $f_{XY}(x,y)=f_X(x)f_y(y)$

iii) $R_{XY}=0$

iv) $R_{XY}=E[X].E[Y]$

v) Random Variables X & Y are uncorrelated then

$$\sigma_{XY}=Cov[X,Y]=0$$

vi) Random Variables X & Y are orthogonal, if

$$R_{XY}=E[X,Y]=0$$

4. Describe the concept of Random process

If the outcome of any event composed of functions of time, it is called Random process.

Auto Correlation function X(t)

This is the expectation of the product of two random variables $X(t_1)$ and $X(t_2)$ obtained by observing the process X(t) at times t_1 and t_2.

$$R_X(t_1,t_2)=E\left[X(t_1)X(t_2)\right]$$

$$=\int_{-\infty}^{+\infty}\int_{-\infty}^{+\infty}X_1X_2\,f_{x(t_1)x(t_2)}(x_1,x_2)dx_1,dx_2$$

Cross Correlation function

The cross correlation function of X(t) and Y(t) are given by

$$R_{XY}(t,u)=E\left[X(t).Y(u)\right]$$

or

$$R_{YX}(t,u)=E\left[Y(t).X(u)\right]$$

Correlation Matrix

$$R(t,u)=\begin{bmatrix}R_X(t,u) & R_{XY}(t,u)\\ R_{YX}(t,u) & R_Y(t,u)\end{bmatrix}$$

Einstein-Wiener-khintchine Relations

Power spectral density : $S_X(f)=\int_{-\infty}^{\infty}R_X(z)\exp\left(-j2\pi fz\right)dz\rightarrow$

Auto correlation function: $R_X(z)=\int_{-\infty}^{\infty}S_X(f)\,\exp\left(-j2\pi fz\right)df\rightarrow$

5..Describe briefly about Gaussian process

Assuming random process X(t) over the interval o to T, and we can weight the random process X(t) with some function K(t). Integrating the product X(t) and K(t) will give new random variable Y, which is

$$Y = \int_{o}^{T} K(t).X(t)\,dt$$

The weighting factor K(t) is selected such that Y should be finite & it should be a Gaussian distributed random variable.

Probability Density function of Y is

$$f_Y(y) = \frac{1}{\sqrt{2\pi}\sigma y}\exp\left[-\frac{(y-\mu_X)^2}{2\sigma^2_y}\right] \qquad \mu y - \text{Mean}$$

$$\sigma^2_y - \text{Variance}$$

Assume random variable is normalized

i.e. $\mu y = 0$ and $\sigma^2_y = 1$

$$\therefore f_Y(y) = \frac{1}{\sqrt{2\pi}}\exp\left(-\frac{y^2}{2}\right)$$

Diagram

Properties

1) If input random process X(t) is applied to a stable linear filter is Gaussian, then the output random process is also Gaussian.

2) X(t) $\rightarrow$ random process. Assume $X(t_1)$, $X(t_2)$….$X(t_n)$ are obtained by observing X(t) at times $t_1, t_2 \ldots t_n$. If random process X(t) is Gaussian, then this set of random variables are jointly Gaussian for any n.

3) If the random variables $X(t_1)$, $X(t_2)$….$X(t_n)$ obtained by sampling a Gaussian process at time $t_1, t_2 \ldots t_n$ are uncorrelated, then these random variables $X(t_1)$, $X(t_2)$….$X(t_n)$ are statistically independent.

6.Consider the random process X(t)=A cos $(2\pi f_c t + \theta)$ where θ is uniformly distributed over the interval $[-\pi, \pi]$ Find power spectral density.

Solution:-

Auto correlation function of the given x(t) is

$$R_X(\tau) = \frac{A^2}{2}\omega_s(2\pi f_c t)$$

$$\text{PSD } S_x(t) = \int_{-\infty}^{\infty} R_x(\tau)\exp(-j2\pi f\tau)\,d\tau$$

$$= \frac{A^2}{2}\int_{-\infty}^{\infty}\cos(2\pi f_c t)e^{-j2\pi f\tau} - d\tau$$

We known that

$$F[\cos 2\pi f_c t] = \frac{1}{2}[\delta(f-f_c)+\delta(f-f_c)]$$

Then

$$\delta_x(f) = \frac{A^2}{2}\frac{1}{2}[\delta(f-f_c)+\delta(f-f_c)]$$

$$\delta_x(t) = \frac{A^2}{4}[\delta(f-f_c)+\delta(f-f_c)]$$

7.. **Consider two random process x(t) and y(t) which have zero mean and individed stationary consider the sum random Z(t) = X(t) + Y(t) .Determine PSD of Z(t)**

Solution

The auto correlation function z(t) is given by

$R_Z(t,u) = E[z(t), z(u)]$

$= E[X(t) + Y(t).X(u) + y(u)]$

$= E[X(t) + X(u) + X(t).Y(u).Y(t).X(u) + Y(t).Y(u)]$

$= E[X(t) + X(u)] + E[X(t).Y(u)] + E[Y(t).X(u)] + [Y(t).Y(u)]$

$= R_X(t,u) + R_{XV}(t,u) + R_{YX}(t,u) + R_Y(t,u)$

Define $\tau = t - u$

$= R_X(t-u) + R_{XV}(t-u) + R_{XY}(t-u) + R_Y(t-u)$

$R_Z(\tau) = R_X(\tau) + R_{XY}(\tau) + R_{YX}(\tau) + R_Y(\tau)$

Taking Fourier transform on the both sides of the above equation (b)

$S_Z(f) = S_x(f) + S_{xy}(f) + S_{yx}(f) + S_t(f)$

Special case:-

When X(t) and Y(t) are uncorrelated the cross densities.

$S_{xy}(f) = S_{yx}(f) = 0$

Then $S_z(f) = S_x(f) + S_y(f)$

Power spectral density of their sum is equal to the sum of their individual power spectral densities

8. The random process X(t) is the input to the input to the filter of impulse response Let V(t) and z(t) denote the random process at the respective filter outputs. Find the cross correlation function of V(t) and z(t)?

The cross correlation function of V(t) and z(t) is

$R_{VZ}(t,u) = E[v(t).z(u)]$

$$= E\left[\int_{-\infty}^{\infty} h_1(\tau_1) X(t-\tau_1)\, d\tau_1 \int_{-\infty}^{\infty} h_2(\tau_2) Y(u-\tau_2)\, d\tau_2\right]$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_1(\tau_1) h_2(\tau_2) E[X(t-\tau_1) Y(u-\tau_2) d\tau_1 d\tau_2]$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_1(\tau_1) h_2(\tau_2) R_{XY}(t-\tau_1, u-\tau_2) d\tau_1 d\tau$$

$$= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_1(\tau_1) h_2(\tau_2) R_{XY}(t-\tau_1-u+\tau_2) d\tau_1 d\tau_2$$

set $\tau = t - u$

$$R_{VZ}(\tau) = \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_1(\tau_1) h_2(\tau_2) R_{XY}(\tau-\tau_1+\tau_2) d\tau_1 d\tau_2$$

Taking Fourier transform of the above equation

(b) we get

$S_{VZ}(f) = H_1(f).H_2(f).S_{xy}(f)$

Where $H_1(f)$ and $H_2(f)$ are frequency response of respective filters.

9.The density function of a continuous random variable is $\text{Find}\, E[X]$

<u>Solution:-</u>

$$E[X] = \int_{-\infty}^{\infty} x\, f_x\,(x)$$

$$= \int_0^4 x\, \frac{x}{8} dx$$

$$= \frac{1}{8}\int x^2\, dx = \frac{1}{8}\left[\frac{x^3}{3}\right]_0^4 = \frac{1}{8\times 3}(4^3)$$

$$E[X] = \frac{8}{3}$$

10.The joint density function of two continuous random variable is f(x,y)= $\begin{cases} cxy & 0 < x < 2\ 0 < y < 3 \\ 0 & \text{otherwise} \end{cases}$ **. Find (a) c; b) p(0<x<1), (1<y<2) (c) p(x<1, y>2)**

<u>Solution:-</u>

(a) $\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} f\,(x, y)\, dx\, dy = 1$

$$\int_0^2\int_1^2 cxy\ dx\, dy = 1 \quad \text{This gives}\, c = \frac{1}{8}$$

$$b)\, P(0 < x < 1,\, 1 < y < 2) = \int_0^1\int_2^1 \frac{xy}{8}\ dx\, dy = \frac{3}{32}$$

$$c)\, P(X < 1, Y > 2) = \int_0^1\int_2^3 \frac{xy}{8}\ dx\, dy$$

$$= \frac{5}{32}$$

11.. A random variable has an exponential PDF given by $f_x\,(x) = a\, e^{-b|x|}$ **where a & b are constant**

Find a) Relationship between a & b

b) Distribution function of x

<u>Solution</u>

(a) $\int_{-\infty}^{\infty} f_x\,(x)\, dx = 1$

∞ $-b|x|$

$$\int_{\infty} ae^{-bx} dx + \int_0 ae^{-bx} dx = a\left[\frac{e}{b}\right]_{-\infty}^{+} a\left[\frac{e}{-b}\right]_0$$

$$= = \frac{a}{b}(1-0) - \frac{a}{b}(0-1) = \frac{a}{b} + \frac{a}{b}$$

$$1 = \frac{2a}{b}$$

b=2a

(b) For x<0

$$F_x(x) = \int_{-\infty}^{\infty} f_x(u)du$$

$$= \int_{-\infty}^{\infty} ae^{bu}du = a\left[\frac{e^{bu}}{b}\right]_{-\infty}^{x} = \frac{a}{b}\left[e^{bx} - 0\right]$$

$$= \frac{a}{b}e^{bx}$$

We know that b=2a

Then $F_x(x) = \frac{a}{2a}e^{bx}$

$$\text{For } x > o = \frac{1}{2}e^{bx}$$

$$F_x(x) = \int_{-\infty}^{x} f_x(u)du = \int_{-\infty}^{x} ae^{-b|u|}du$$

$$= \int_{-\infty}^{0} a\, e^{bu}du + \int_{0}^{x} ae^{-bu}du$$

$$= a\left[\frac{e^{bu}}{b}\right]_{-\infty}^{0} + a\left[\frac{e^{-bu}}{-b}\right]_{0}^{x}$$

$$= \frac{a}{b}(1) - \frac{a}{b}\left(e^{-bx} - 1\right)$$

$$= \frac{a}{2a} - \frac{a}{2a}\left(e^{-bx} - 1\right) \quad b = 2a$$

$$= \frac{1}{2} - \frac{1}{2}e^{-bx} + \frac{1}{2}$$

$$= 1 - \frac{1}{2}e^{-bx}$$

$$\text{Thus } F_x(x) = \begin{cases} \frac{1}{2}e^{bx} & x < 0 \\ 1 - \frac{1}{2}e^{-bx} & x < 0 \end{cases}$$

UNIT-IV

2-MARK QUESTION AND ANSWERS:

1. Compare the noise performance of an AM and FM system?

The figure of merit of AM system is 1/3 when the modulation is 100 percent and that of FM is (3/2) mf^2.The use of FM offers improved noise performance over AM when $(3/2)mf^2 > 1/3$.mf –modulation index in FM.

2. What is threshold effect in AM receivers?

- The loss of the message signal x (t) in an envelope detector due to the presence of the large noise is known as the Threshold effect.
- The threshold effect is also defined as when a noise is large compared to the signal at the input of the envelope detector, the detected output has a message signal completely mingled with noise. it means that if the i/p signal to noise ratio(S_i/N_i) is below a certain level called threshold level.

3. Write the figure of merit (γ) for an AM system?

(OR)

What is the $\dfrac{S_0/N_0}{S_i/Ni}$ **for AM (with envelope detection in small noise Case?**

The figure of merit $\gamma = \dfrac{\overline{x^2(t)}}{A^2 + \overline{x^2(t)}}$

If $x(t) = ACos\omega_m t$

then

$x^2(t) = \dfrac{A^2}{2}$

Therefore

The figure of merit $\gamma = \dfrac{\frac{A^2}{2}}{A^2 + \frac{A^2}{2}} = \dfrac{1}{3}$

4.Define the term noise equivalent temperature. (Nov/Dec – 2017) (APRIL-MAY2019)

The equivalent noise temperature of a system is defined as the temperature at which a noisy resistor has to be maintained such that, by connecting the resistor to the input of a noiseless version of the system; it produces the same available noise power at the system output as that produced by all the sources of noise in the actual system. Noise Equivalent Temperature is denoted by T_e. Thus, $T_e = (F – 1) T$

Where F is the noise factor and T is the temperature in Kelvin

5.List the external sources of noise. (Nov/Dec – 2017)

There are number of external sources of noise. These are grouped into three categories,

- Atmospheric noise i.e., Natural noise
- Extraterrestrial noise i.e., Natural noise
  - ✓ Solar noise
  - ✓ Cosmic noise
- Industrial noise i.e., Manmade noise

6. Define capture effect in FM? (MAY/JUNE – 2016, MAY/JUNE – 2015, MAY/JUNE – 2012)

- When the interference is stronger then it will suppress the desired FM input. When the interference signal and FM input are of equal strength, the receiver fluctuates back and forth between them .This phenomenon is known as the capture effect.
- We may also define as in low noise case; the distortion produced by the noise at the o/p of FM detector is negligible in comparison to the desired modulating signal. And noise almost suppressed by the signal. This phenomenon is called as capture effect.

7. Define FM threshold effect?

- As the input noise power is increased the carrier to noise ratio is decreased the receiver breaks and as the carrier to noise ratio is reduced further crackling sound is heard and the output SNR cannot be predicted by the equation. This phenomenon is known as threshold effect.
- It is also defined as when the SNR becomes even slightly less than unity, an impulse of noise is generated. This noise impulse appears at the output of the detector in the form of click sound. If the SNR ratio is further decreased so that the ratio is moderately less than unity, the impulses are generated rapidly and clicks merge in to spluttering sound. This phenomenon is known as “threshold effect”.

8. How is threshold reduction achieved in FM systems?

- Threshold reduction is achieved in FM system by using an FM demodulator with negative feedback or by using a phase locked loop demodulator. Such devices are referred to as extended-threshold demodulators.

9. What is Pre-emphasis?

- The pre modulation filtering in the transistor, to raise the power spectral density of the base band signal in its upper-frequency range is called pre emphasis (or pre distortion) Pre emphasis is particularly effective in FM systems which are used for Transmission of audio signals.

10. Define de-emphasis.

A de-emphasis in the receiver used to restore relative magnitude of different improvement in AF signal and to suppress noise is called de-emphasis.

11.Comment the role of pre-emphasis and de-emphasis circuit in SNR improvement. (MAY/JUNE – 2017)

Pre-emphasis

It is the process of artificially boosting of high frequency component of message signal in order to improve the overall Signal-to-Noise Ratio (SNR) by minimizing the noise present in the signal. Pre-emphasis is done in the transmitter side before frequency modulation.

De-emphasis

It is the process of decreasing the strength of high frequency component of message signal to get back the original transmitted message signal in order to improve the overall Signal-to-

Noise Ratio (SNR) by minimizing the noise present in the signal. De-emphasis is performed in the receiver side after demodulation.

Thus pre-emphasis and de-emphasis produces a more uniform SNR throughout the message signal frequency spectrum

12. What is Nyquist rate?

The sampling rate of $2f_m$ samples per second for a signal bandwidth of f_m Hertz is called the Nyquist rate. Its reciprocal $1/f_m$ is called the Nyquist interval.

13. Compare AM to FM

* In AM system there are three frequency components and hence the bandwidth is finite.	FM system has infinite number of sidebands in addition to a single carrier. Hence its Bandwidth is infinite.
*The amplitude of modulated wave in AM is dependent of modulation index.	The amplitude of frequency modulated wave in FM is independent of modulation index.
*In AM, most of the transmitted Power is wasted	In FM, noise is very less.

r (t) and $\theta(t)$ are the amplitude and phase of the band pass signal noise. And $\psi(t)$ is the relative phase.

PART B

1)Discuss briefly noise in DSBSC systems using coherent detection

The use of coherent detection requires multiplication of filtered signal x(t) by a locally generated sinusoidal wave $\cos(2\pi f_c t)$ and then low pass filtering the product.

To simplify the analysis, we assume that the amplitude of locally generated sinusoidal wave is unity.

For this demodulation, the local oscillator should be synchronized both in phase and in frequency with the oscillator generating the carrier wave in the transmitter.

We need to calculate Figure of (FOM) Merit in order to know

The modulating signal is given by

$$s(t) = c.A_c \cos(2\pi f_c t) \times m(t) \rightarrow 1$$

Where, $A_c \cos(2\pi f_c t) \rightarrow$ sinusoidal carrier wave

$m(t) \rightarrow$ message signal

C $\rightarrow$ system dependent scaling factor

Assume that m(t) is the sample function of a stationary process of zero mean, whose power spectral density is given by $S_m(f)$ which is limited to a max frequency of W, where $W \rightarrow$ msg. Bandwidth

The average power P of the message signal is the total area under the curve of power spectral density, which is given by

$$P = \int_{-w}^{w} S_m(f)df$$

<u>The average signal power</u>

$$S_c = \left[(s(t))\right]^2 = C^2 A^2{}_c \cos^2(2\pi f_c t) m^2(t)$$

$$= C^2 A^2{}_c m^2(t)\left[\frac{1+\cos 4\pi f_c t}{2}\right] \qquad \cos 4\pi f_c t = 0 \rightarrow \min$$

$$= C^2 A^2{}_c m^2(t)\left[\frac{1}{2}\right] \qquad \underset{\text{Power=}}{\underset{\downarrow}{p}} = \underset{\text{square of msg. s/l}}{\underset{\downarrow}{m^2(t)}}$$

$$= \frac{C^2 A^2{}_c p}{2}$$

<u>The average Noise power</u>

$$N_c = \frac{N_o}{2} \times BW = \frac{N_o}{2} \times 2W$$

$$= WN$$

BW for BDSBSC is 2w

$$\rightarrow (SNR)_{C,DSNBSC} = \frac{\text{Average signal power}}{\text{Average Noise power}} = \frac{S_c}{N_c}$$

$$= \frac{C^2 A^2{}_c P}{sWN_o}$$

<u>To determine $(SNR)_o$</u>

Now filtered noise is Narrow band noise n(t) which is represented as

$$n(t) = n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)$$

$$\therefore \text{From the diagram}, x(t) = s(t) + n(t)$$

$$x(t) = CA_c \cos(2\pi f_c t) m(t) + n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)$$

<u>Output of Product modulator is</u>

$$\vartheta(t) = x(t)\times\cos(2\pi f_c t)$$
$$\vartheta(t) = \left[CA_c\cos(2\pi f_c t)m(t) + n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)\right]\cos(2\pi f_c t)$$
$$= CA_c\cos^2(2\pi f_c t)m(t) + n_I(t)\cos^2(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)\cos(2\pi f_c t)$$
$$= CA_c\left[\frac{1+\cos 4\pi f_c t}{2}\right]m(t) + n_I(t)\left[\frac{1+\cos 4\pi f_c t}{2}\right] - n_Q(t)\left[\frac{\sin 4\pi f_c t}{2}\right]$$
$$= CA_c m(t)\left[\frac{1}{2}\right] + \frac{CA_c m(t)\cos 4\pi f_c t}{2} + \frac{n_I(t)}{2} + \frac{n_I(t)\cos 4\pi f_c t}{2} - \frac{n_Q(t)\sin 4\pi f_c t}{2}$$
$$= \frac{1}{2}CA_c m(t)\frac{1}{2}n_I(t) + \frac{1}{2}\cos 4\pi f_c t\left[CA_c m(t) + n_I(t)\right] - \frac{1}{2}n_Q(t)\sin 4\pi f_c t$$

Output signal y(t) is given by high frequency components using LPF

$$\therefore y(t) = \frac{1}{2}CA_c m(t) + \frac{1}{2}n_I(t)$$

Message signal component at receiver output is $\frac{CA_c m(t)}{2}$

Average signal power $= \frac{C^2 A^2{}_c m^2(t)}{2} = \frac{C^2 A^2{}_c .P}{4} = S_o$

Noise signal component at receiver output is $\frac{n_I(t)}{2}$

Average Noise power $N_o = \frac{n_I(t)}{2}\times BW$ $\qquad$ $BW\,of\,DSBSC = 2W$

$$= \frac{N_o/2}{/}\times BW \qquad n_I(t) = \frac{N_o}{2}$$

$$= \frac{WN_o}{2} \qquad \downarrow\ \text{PSD of Noise}$$

$$(SNR)_{o,DSBSC} = \frac{S_o}{N_o} = \frac{C^2A^2{}_c P/4}{WN_o/2} = \frac{C^2A^2{}_c P}{2WN_o}$$
$$\rightarrow \therefore \text{Figure of Merit } \gamma = \frac{(SNR)_o}{(SNR)_c} = \frac{C^2A^2{}_c P/2WN_o}{C^2A^2{}_c P/2WN_o} = 1$$

Thus FOM for DSBSC system is unity. $\therefore$ SNR at input and output of detector are identical and there is no improvement in SNR.

2)Discuss briefly noise in AM receivers using envelope detection (NOV/DEC2017)

To determine $(SNR)_c = \frac{\text{Avg. signal power}}{\text{Avg. Noise power}} = \frac{S_c}{N_c}$

The modulating signal is given by

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

where $A_c\cos 2\pi f_c t \rightarrow$ carrier signal
$m(t) \rightarrow$ message signal
$m_a \rightarrow$ modulation index of AM signal

Average signal power is given by

$$S_c = A^2{}_c\left[1 + m^2{}_a m^2(t)\right]\cos^2 2\pi f_c t$$
$$= A^2{}_c\left[1 + m^2{}_a m^2(t)\right]\left[\frac{1+\cos 4\pi f_c t}{2}\right] \qquad \cos 4\pi f_c t = 0$$
$$S_c = \frac{A^2{}_c}{2}\left[1 + m^2{}_a P\right] \qquad m^2(t) = P$$

Average Noise power is given by

$$N_c = \frac{N_o}{2}\times BW \qquad BW\,of\,AM = 2W$$
$$= \frac{N_o}{2}\times 2W$$
$$W = WN_o$$

$$\therefore (SNR)_c = \frac{S_c}{N_c} = \frac{\frac{A_c}{2}\left(1+m_a^2P\right)}{WN_o}$$

$$= \frac{A_c^2\left(1+m_a^2P\right)}{2WN_o}$$

<u>To determine $(SNR)_o$</u>

The filtered signal x(t) is given by

X(t) = s(t)+n(t)

$$(t) = A_c\left[1+m_am(t)\right]\cos 2\pi f_ct + \left[n_I(t)\cos 2\pi f_ct - n_Q(t)\sin 2\pi f_ct\right]$$

$$= \left[A_c + A_cm_am(t) + n_I(t)\right]\cos 2\pi f_ct - n_Q(t)\sin 2\pi f_ct$$

The Receiver output y(t)=Envelope of x(t)

$$y(t) = \sqrt{\left[A_c + A_cm_am(t) + n_I(t)\right]^2 + n^2{}_Q(t)}$$

When signal power is large compared to noise power,

$$y(t) = A_c + A_cm_am(t)$$

$A_c \rightarrow$ Carrier amplitude and it is removed blocking capacitor after envelope Detector. Because it is not related with m(t)

$$y(t) = A_cm_am(t)$$

Average signal power at receiver output is given by $S_0 = \frac{A^2{}_cm^2{}_aP}{2}$

Average Noise power is $= \frac{N_o}{2} \times BW$

$$= \frac{N_o}{2} \times 2W$$

$$= N_oW$$

$$(SNR)_0 = \frac{A^2{}_cm^2{}_aP/2}{N_oW} = \frac{A^2{}_cm^2{}_aP}{2N_oW}$$

Figure of Merit FOM= $\frac{(SNR)_o}{(SNR)_c}$

$$FOM = \frac{A^2{}_cm^2{}_aP/2N_oW}{A^2{}_c\left(1+m_a^2P\right)/2N_oW}$$

$$= \frac{m^2{}_aP}{1+m_a^2P}$$

Thus FOM for AM receiver is always less than unity.

3)Obtain the Expressions of noise in FM receiver

Noise $\omega(t)$ is modelled as white Gaussian noise of Zero mean and Power Spectral density $\frac{N_o}{2}$.

In FM, message signal is transmitted by variations of frequency of carrier wave and its amplitude is constant.$\therefore$ Any variations of carrier amplitude at receiver input results from noise or interference.

Amplitude Limiter is used to remove amplitude variations by clipping modulating wave at filter output almost to zero axis.

Discriminator consists of Differentiator and Envelope Detector.

Differentiator produces hybrid modulated wave in which both amplitude and frequency vary in accordance with message signal.

Envelope Detector produces original message signal.

Baseband LPF has Bandwidth that is large to accommodate highest frequency component of message signal. It removes out of band components of noise at discriminator output and keeps effect of output noise at minimum level.

<u>To determine $(SNR)_o$:</u> $(SNR)_0 = \frac{S_o}{N_o}$

The narrow band noise can be represented by

$n(t) = n_I(t)\cos 2\pi f_c t - n_Q(t)\sin 2\pi f_c t \rightarrow (1)$

This eqn. can be written in terms of envelope and phase components as,

$n(t) = r(t)\cos[2\pi f_c t + \varphi(t)] \rightarrow (2)$

Where Noise Envelope of n(t): r(t)= $\sqrt{n_I^2(t) + n_\varphi^2(t)}$

Noise Phase of n(t): φ (t)= $\tan^{-1}\left[\frac{n_\varphi(t)}{n_I(t)}\right]$

FM signal is represented by

$s(t) = A_c\cos[2\pi f_c t + \varphi(t)] \rightarrow (3)$

Where $\varphi(t) = 2\pi k_f \int_0^t m(t)dt \rightarrow (4)$

The Band Pass filter output x(t) is given by

$x(t) = s(t) + n(t)$

Sub. eqn. (3) & (2) $= A_c\cos[2\pi f_c t + \varphi(t)] + r(t)\cos[2\pi f_c t + \varphi(t)]$

The equation x(t) can be represented by means of Phasor diagram. The phase θ of resultant phasor representing x(t) is

$$\theta(t) = \varphi(t) + \tan^{-1}\left[\frac{r(t)\sin(\varphi(t) - \varphi(t))}{A_c + r(t)[\sin\varphi(t) - \varphi(t)]}\right]$$

If r(t) is small to A_c,

$$\theta(t) \approx \varphi(t) + \frac{r(t)\sin(\varphi(t) - \varphi(t))}{A_c}$$

$$\theta(t) = 2\pi k_f \int_0^t m(t)dt + \frac{r(t)}{A_c}\sin[\varphi(t) - \varphi(t)] \rightarrow (5)$$

The Discriminator output $\vartheta(t)$ is equal to derivative of relative phase divided by 2 π

$$\text{i.e. } \vartheta(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt}$$

$$\therefore \vartheta(t) = \frac{1}{\cancel{2\pi}}\frac{d}{dt}\left[\cancel{2\pi} k_f \int_0^t m(t)dt\right] + \frac{1}{2\pi}\frac{d}{dt}\left[\frac{r(t)}{A_c}\sin[\varphi(t) - \varphi(t)]\right]$$

$$\vartheta(t) = K_f.m(t) + n_d(t) \rightarrow (6)$$

$$\text{where } n_d(t) = \frac{1}{2\pi A_c}.\frac{d}{dt}\left[r(t)\sin[\varphi(t) - \varphi(t)]\right]$$

$\therefore \vartheta(t)$ consists of message signal m(t) multiplied by constant factor k_f plus the additive noise component $n_d(t)$.

The discriminator output is given to LPF & its output is given by

Signal power= $K_f.m(t)$ from eqn.(6)

Average signal power $s_o = K_f^2 m^2(t)$

$= K_f^2 p$

<u>Average signal power calculation</u>

PSD of noise = $\frac{N_o f^2}{A_c^2} = SN_o(f)$

$$\text{Average Noise power } = \int_{-w}^{w} S_{N_o}(f)df$$

$$= \int_{-w}^{w} \frac{N_o}{A_c^2} f^2 df = \frac{N_o}{A_c^2}\int_{-w}^{w} f^2.df$$

$$= \frac{2N_o}{A_c^2}\left[\frac{w^3}{3}\right]$$

$$= \frac{2N_o w^3}{3A_c^2}$$

$$\therefore (SNR) = \frac{\text{Average signal power}}{\text{Average Noise Power}}$$

$$\therefore (SNR) = \frac{\text{Average signal power}}{\text{Average Noise Power}}$$

$$= \frac{k^2{}_f p}{2N_o w^3 / 3A^2{}_c}$$

$$(SNR)_o = \frac{3A^2{}_c k^2{}_f p}{2N_o w^3}$$

* <u>To determine $(SNR)_c$</u>

The average signal power is $= \frac{A^2{}_c}{2}$ from eqn. ③

The average Noise power is $= \frac{N_o}{2} \times BW$

$$= \frac{N_o}{2} \times 2W$$

$$(SNR)_c = \frac{\text{Average signal power}}{\text{Average Noise Power}}$$

$$= \frac{A^2{}_c / 2}{N_o W}$$

$$(SNR)_c = \frac{A^2{}_c}{N_o W}$$

<u>Figure of Merit (FOM)</u>

$$FOM = \frac{(SNR)_o}{(SNR)_c}$$

$$= \frac{3A^2{}_c k^2{}_f p / 2N_o W^3}{A^2{}_c / 2N_o W}$$

$$FOM = \frac{3k^2{}_f p}{w^2}$$

4)Explain FM threshold effect

Under low noise, the performance of system is improved but not continued for infinite increase in Bandwidth.

With increase in Bandwidth, input noise power due to circuit components and devices increases.

Consider the message signal as zero and if noise signal is greater than amplitude of carrier signal, the SNR at receiver input is less than unity.

$(SNR)_0$ of FM Receiver is valid only when CNR measured at discriminator input is greater than unity.

When CNR is less than unity, an impulse of noise is generated. This impulse of noise is appeared in the form of click sound at the output of FM discriminator.

<u>Threshold Effect Definition</u>

When CNR is less than unity, the frequency of spike generation is small and each spike produces individual clicking sound in receiver. But when CNR is further decreased, the spikes are generated rapidly and the clickings merge into sputtering sound. This is known as Threshold effect in FM.

<u>ThresholdFM Demodulator</u>

For simplification, the input signal is unmodulator carrier i.e. message signal is zero. The carrier signal is accompanied by white noise.

The white noise is filtered, shaped and converted into narrow band noise by IF filter

The frequency discriminator output is given by

$$x(t) = A_c \cos 2\pi f_c t + n(t)$$
$$= A_c \cos 2\pi f_c t + \left[n_I(t) \cos 2\pi f_c t - n_\varphi(t) \sin 2\pi f_c t \right]$$
$$x(t) = \left[A_c + n_I(t) \right] \cos 2\pi f_c t - n_\varphi(t) \sin 2\pi f_c t$$

Where $n_I(t)$ and $n_\varphi(t)$ are in-phase and quadrature components of narrowband noise

Phasor diagram displays phase relation between various components of x(t)

As amplitude and phase of $n_I(t)$ and $n_\varphi(t)$ changes with time in a random manner, the point P_1 (tip of phasor representing x(t)) wanders around point P_2 (tip of phasor representing carrier signal)

When CNR is large, noise $n_I(t)$ and $n_\varphi(t)$ are smaller than carrier amplitude and so wandering point P_1 spends most of its time near point P_2.

The angle $\theta(t)$ is approximately equal to $\frac{n_\varphi(t)}{A_c}$ which is within multiple of 2π.

<u>Spike generation</u>

When noise phasor r(t) is very large than carrier amplitude, the locus of end point P_1 of the result x(t) moves away from point P_2 and even rotate about origin.

The locus encircles the origin is called as Spike path.

The phase angle $\theta(t)$ changes by multiple of 2π during small time interval. i.e. phase angle $\theta(t)$ increases or decreases by 2π radians.

The discriminator output is given by

Diagram

$$\vartheta(t) = \frac{\theta'(t)}{2\pi}$$

$$\text{where}\ \theta'(t) = \frac{d\theta}{dt}$$

When $\theta'(t)$ changes by 2π, $\frac{d\theta}{dt}$ appears as Sharp spike as impulse with period of 2π.

Area under each spike is 2π and it is obtained by integrating the impulse over the interval t_1 to t_2

$$\text{Area} = \int_{t_1}^{t_2} \frac{d\theta}{dt} dt = \left[\theta\right]_{t_1}^{t_2} = \left[\theta\right]_{-\pi}^{\pi}$$

1) <u>Positive Going Click</u>

This click occurs when noise envelope r(t) and phase $\varphi(t)$ of narrow band noise n(t) satisfy the following conditions

$$r(t) > A_c$$
$$\varphi(t) < \pi \leq \varphi(t) + d\varphi(t)$$
$$\frac{d\varphi(t)}{dt} > 0$$

When x(t) rotates counter clockwise, the phase $\varphi(t)$ changes by 2π and generates positive spikes and if x(t) rotates clockwise, the phase $\varphi(t)$ changes by -2π and generates negative spikes

2) <u>Negative Going Click</u>

This click occurs when noise envelope r(t) and phase $\varphi(t)$ of narrow band noise n(t) satisfy the following conditions.

$r(t) > A_c$

$$\varphi(t) > -\pi > \varphi(t) + d\varphi(t)$$

$$\frac{d\varphi(t)}{dt} < 0$$

5. Explain the types of Noise

Noise is an unwanted electrical signal which affects the message signal at the transmitter and it should be suppressed at the receiver.

Noise Source

Diagram

1) External Noise

Noise created outside the circuit is called External Noise

a) Atmospheric Noise

These noise are caused by lightning, electrical storms and atmospheric distburbances.

This noise is less severe above 30 MHz

b) Man-mad noise

These noise are due to undesired pickups from electrical appliances such as motors, switch gears, automobiles and aircraft ignitions.

Since this noise is under human control, it can be eliminated by removing source of noise.

This noise occurs in the range of 1 MHz to 500 MHz.

2) Internal Noise

These noise are created by active and passive components present inside the circuit.

These are also created due to spontaneous fluctuations present in the physical system.

Thermal motion of free electrons inside the resistor

Random emission of electrons in Vacuum tubes

Random diffusion of electrons and holes in a semi conductor

a) Thermal Noise

This noise is also called electrical noise which arises due to random motion of electrons in a conductor.

Intensity of motion of electrons is proportional to thermal energy supplied to the conductor. That is why this noise is thermal noise.

The net motion of all electrons give rise to an electric current flow through Resistor causing noise.

Thevenin's equivalent circuit of noisy resistor which consists of thermal noise voltage in series with noise resistor.

shows Norton's equivalent circuit of noisy resistor which consists of thermal noise current in parallel with noise conductance.

The noise power $P_n = \frac{V_{rms}{}^2}{R_L + R}$

$$= \frac{\left(V_{TN} / \sqrt{2}\right)^2}{R_L + R}$$

$$= \frac{\left(V_{TN} / \sqrt{2}\right)^2}{2R}$$

$$\Downarrow$$

6.Explain in detail Noise figure in two port network

Noise figure in Two-port Network

$$\text{Noise figure} F = \frac{\text{Total noise power spectral density at output of Two-port Network}}{\text{Total noise power spectral density at output assuming network is noiseless (or) only due to source}} \qquad F = \frac{S_{no}}{S'_{no}} \to ①$$

$S_{no} \to$ Total output noise power is equal to sum of power density due to source alone (S_{no}^{1}) also the power density contributed by the network itself (S_{no}'').

i.e. $S_{no} = S_{no}' + S_{no}''$

i.e. $S_{no} = S_{no}' + S_{no}''$

$$\therefore \text{Eqn.} ① \Rightarrow F = \frac{S_{no}' + S_{no}''}{S_{no}'} = 1 + \frac{S_{no}''}{S_{no}'}$$

Noise figure in terms of gain

Input noise power spectral density $S_{ni} = \frac{KT}{2}$

If two port network is noiseless then

$$S_{no}' = G(\omega) \times S_{ni} = G(\omega) \times \frac{KT}{2}$$

$$① \Rightarrow F = \frac{S_{no}}{S_{no}'} = \frac{S_{no}}{G(\omega)\frac{KT}{2}}$$

Noise Figure in terms of Network Transfer function

$H(\omega) \to$ Network Transfer function

$$Sno' = S_{ni} |H(\omega)|^2$$

$$\therefore ① \Rightarrow F = \frac{S_{no}}{S_{ni} |H(\omega)|^2}$$

Noise Figure in terms of Equivalent input noise temperature

$$F = 1 + \frac{T_e}{T}$$

$T_e \to$ noise temp. of N/w

$T \to$ noise temp. of source

Average noise figure $\left(\bar{F}\right)$

$$\bar{F} = \frac{\int_{w_1}^{w_2} G(\omega) F(\omega) d\omega}{\int_{w_1}^{w_2} G(\omega) d\omega} \qquad W_1 \text{ to } W_2 \to \text{freq.range}$$

Noise figure in terms of SNR

Two port network consists of message signal V_s and noise signal V_n. The output signal is V_o and gain of the network is $G(\omega)$.

The input noise power spectral density is given as

$$S_{ni} = \frac{KT}{2}$$

Let $S_{si} \to$ Power spectral density of input message signal Vs

& $S_{so} \to$ Power spectral density of output

$$\therefore G(\omega) = \frac{S_{so}}{S_{si}} \to ①$$

Noise figure $F = \frac{S_{no}}{S_{no}'} \Rightarrow S_{no} = F.S_{no}'$

$$S_{no} = F.\,G(\omega) S_{ni} \to ②$$

w.k.t

$$S_{no}' = G(\omega) S_{ni}$$

Sub. ① in ②

$$S_{no} = F.\frac{S_{so}}{S_{si}} S_{ni} \Rightarrow F = \frac{S_{no} \times S_{si}}{S_{so} \times S_{ni}}$$

$$F = \frac{S_{si} / S_{ni}}{S_{so} / S_{no}} = \frac{\text{Input SNR}}{\text{Output SNR}} \frac{S_i / N_i}{S_o / N_o} \rightarrow 3$$

$$\text{w.k.t}\, F = 1 + \frac{T_e}{T} \rightarrow 4$$

Equating 3 & 4

$$1 + \frac{T_e}{T} = \frac{S_i / N_i}{S_o / N_o} \rightarrow 5$$

$$S_o = G(\omega) S_i$$

$$5\ \frac{S_i}{N_i} = \left(1 + \frac{T_e}{T}\right) \frac{S_{so}}{S_{no}}$$

$$\frac{S_i}{N_i} = F.\frac{G(\omega) S_i}{N_o} \Rightarrow F = \frac{N_o}{N_i} \times \frac{1}{G(\omega)} \rightarrow 6$$

Output noise power N_o is the sum of output noise power due to input source ($G(\omega) N_i$) and noise power contributed by two port network (N_{tp})

$$\text{i.e.}\, N_o = G(\omega) N_i + N_{tp} \rightarrow 7$$

Sub. eqn. 7 in 6

$$F = \frac{G(\omega) N_i + N_{tp}}{N_i} \times \frac{1}{G(\omega)}$$

$$F = \frac{G(\omega) N_i + N_{tp}}{G(\omega) N_i} = 1 + \frac{N_{tp}}{G(\omega) N_i}$$

$$\frac{N_{tp}}{G(\omega) N_i} = F - 1$$

$$N_{tp} = (F - 1) G(\omega) N_i$$

7.Explain briefly Noise figure in cascaded Amplifier

<u>CASCADED STAGES</u>

Two port network involving more than me stage is called Cascade Amplifier.

Diagram

G_1 and F_1 $\rightarrow$ Gain and Noise figure of first stage.

G_2 and F_2 $\rightarrow$ Gain and Noise figure of second stage.

$N_i \rightarrow$ noise generated by resistor R at input of first stage
Total noise power at final output due to N_i is given by $G_1G_2N_i \rightarrow N_{01}$
First stage introduces its own noise. Due to this noise, the output of first stage is $G_1(F_1\text{-}1)N_i$
The noise is amplified at second stage and appears at output which is given by

$$G_1G_2(F_1\text{-}1)N_i \rightarrow N_{02}$$

Similarly at third stage $N_{03} = G_2(F_2\text{-}1)N_i$

Now total noise power at output is

$N_0 = N_{01} + N_{02} + N_{03}$

$$N_o = G_1G_2N_i + G_1G_2(F_1 - 1)N_i + G_2(F_2 - 1)N_1$$

Dividing by N_i G_1 G_2 in the above eqn.

$$\frac{N_o}{N_i \times G_1G_2} = 1 + F_1 - 1 + \frac{F_2 - 1}{G_1} = F_1 + \frac{F_2 - 1}{G_1}$$

Overall gain of cascaded stage G= G_1G_2

$$\therefore \frac{N_o}{N_i G} = F_1 + \frac{F_2 - 1}{G_1}$$

w.k.t from previous section eqn.(6) $F = \frac{1}{G} \times \frac{N_o}{N_i}$

$$\therefore F = F_1 + \frac{F_2 - 1}{G_1} \rightarrow ①$$

This is the expression of noise figure for two cascaded stages. This can be extended for multistage amplifier.

$$F = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \frac{F_4 - 1}{G_1 G_2 G_3} + \ldots.. \frac{F_n}{G_1 G_2 \ldots.. G_n - 1} \rightarrow ②$$

<u>Equivalent Noise temperature of Cascaded stages</u>

w.k.t $F = 1 + \frac{T_e}{T} \rightarrow ③$

sub. eqn ③ in ①

$$1 + \frac{T_e}{T} = 1 + \frac{T_{e1}}{T} + \frac{1 + \frac{T_{e2} - 1}{T}}{G_1} = 1 + \frac{T_{e1}}{T} + \frac{T_{e2}}{TG_1}$$

$$\frac{T_e}{T} = \frac{T_{e1}}{T} + \frac{T_{e2}}{TG_1}$$

$$T_e = T_{e1} + \frac{T_{e2}}{G_1}$$

This is the expression for equivalent noise temperature for two cascaded stage amplifier. This can be extended for multi stage amplifier.

$$T_e = T_{e1} + \frac{T_{e2}}{G_1} + \frac{T_{e3}}{G_1 G_2} + \ldots.. \frac{T_{en}}{G_1 G_2 \ldots. G_{n-1}}$$

<u>Equivalent noise resistance</u>

$$R_e = R_1 + \frac{R_2}{G^2{}_1} + \frac{R_3}{G^2{}_1 G^2{}_2} + \ldots.. \frac{R_n}{G^2{}_1 G^2{}_2 \ldots.. G^2{}_{n-1}}$$

<u>Noise figure in terms of Noise Resistance</u>

$F = 1 + \frac{R_e}{R_a}$ $R_a \rightarrow$ Resistance of amplifier

Noise figure depends on source impedance ,temperature,DC current and frequency

$\bar{F} = \frac{qI_d R_s}{2KT}$

$qI_d \rightarrow$ PSD of short noise

$R_s \rightarrow$ Source resistance

$2KT \rightarrow$ Output noise voltage

<u>Noise Equivalent Bandwidth (B_N)</u>

Diagram

$\rightarrow B_N = \frac{1}{A}\int_0^\infty |H(\omega)|^2 d\omega$: It is the bandwidth of the ideal band pass system which produces same noise power as Actual system

8.The noise figure of the individual stage of a two stage amplifier 3.03 and 2.54 respectively . The available power gain of the first stage is 62 . Find overall noise figure

<u>Solution:</u>

Given

F_1=3.03

F_2=2.54

G_a=62

For a 2 stage amplifier noise figure given by

$$F_1 = 1 + \frac{T_e}{T} = 1 + \frac{10}{300}$$
$$F_1 = 1.03$$
$$F = F_1 + \frac{F_2 - 1}{G_a} + \frac{F_3 - 1}{}$$
$$= 1.03 + \frac{4-1}{1000} + \frac{16-1}{1000 \times 100}$$
$$F = 1.033$$

UNIT V

PART A

1.State the sampling theorem for band-limited signals of finite energy.

If a finite energy signal g(t) contains no frequency higher than W Hz, it is completely determined by specifying its ordinates at a sequence of points spaced 1/2W seconds apart.

2.What are the advantages of digital transmission?

i. The advantage of digital transmission over analog transmission is noise immunity.

ii. Digital transmission systems are more noise resistant than the analog transmission systems.

iii. Digital systems are better suited to evaluate error performance.

3.What are the disadvantages of digital transmission?

i. The transmission of digitally encoded analog signals requires significantly more bandwidth than simply transmitting the original analog signal.

ii. Analog signal must be converted to digital codes prior to transmission and converted back to analog form at the receiver, thus necessitating additional encoding and decoding circuitry.

4.Define pulse code modulation.

In pulse code modulation, analog signal is sampled and converted to fixed length, serial binary number for transmission. The binary number varies according to the amplitude of the analog signal.

5.What is the purpose of the sample and hold circuit?

The sample and hold circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal.

6.What is the Nyquist sampling rate? *(APRIL-MAY2019)*

Nyquist sampling rate states that, the minimum sampling rate is equal to twice the highest audio input frequency.

7.What is the principle of pulse modulation?

Pulse modulation consists essentially of sampling analog information signal and then converting those discrete pulses and transporting the pulses from a source to a destination over a physical transmission medium.

8.List the four predominant methods of pulse modulation.

i. Pulse width modulation (PWM)
ii. Pulse position modulation (PPM)
iii. Pulse amplitude modulation (PAM)
iv. Pulse duration modulation (PDM)

9.Define quantization.

Quantization is a process of approximation or rounding off. Assigning PCM codes to absolute magnitudes is called quantizing.

10.Define dynamic range.

Dynamic range is the ratio of the largest possible magnitude to the smallest possible magnitude. Mathematically, dynamic range is DR= V_{max} / V_{min}

11.What is nonuniform or nonlinear encoding?

With voice transmission, low-amplitude signals are more likely to occur than large- amplitude signals. Therefore, if more codes are used for lower amplitude, it would increase accuracy and fewer codes are used for higher amplitudes, which would increase quantization error. This type of coding is called nonuniform or nonlinear encoding.

12.What is the advantage and disadvantage of midtread quantization?

Advantage: less idle channel noise
Disadvantage: largest possible magnitude for Q_e

13.What is the necessity of companding?

Companding is the process of compression and then expanding. Higher amplitude signals are compressed prior to transmission and then expanded in the receiver. Companding is themeans of improving dynamic range of communication systems.

14.Define quantization error?

Quantization is the value of which equals the difference between the output and input values of quantizer.

15.What is nyquist rate?

The minimum sampling rate of 2W sample per second for a signal ba

16.What is PAM?

PAM is the pulse amplitude modulation. In pulse amplitude modulation, the amplitude of a carrier consisting of a periodic train of rectangular pulses is varied in proportion to sample values of a message signal.

1.Describe briefly the Elements of Digital Communication Systems:

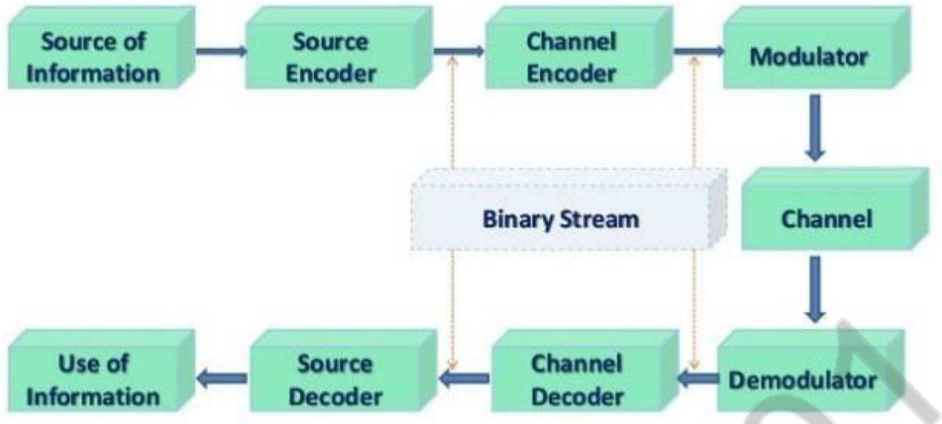


Fig. 1 Elements of Digital Communication Systems

1. **Information Source and Input Transducer:**

   The source of information can be analog or digital, e.g. analog: audio or video signal, digital: like teletype signal. In digital communication the signal produced by this source is converted into digital signal which consists of 1's and 0's. For this we need a source encoder.

2. **Source Encoder:**

   In digital communication we convert the signal from source into digital signal as mentioned above. The point to remember is we should like to use as few binary digits as possible to represent the signal. In such a way this efficient representation of the source output results in little or no redundancy. This sequence of binary digits is called ***information sequence.***

   Source Encoding or Data Compression: the process of efficiently converting the output of whether analog or digital source into a sequence of binary digits is known as source encoding.

3. **Channel Encoder:**

   The information sequence is passed through the channel encoder. The purpose of the channel encoder is to introduce, in controlled manner, some redundancy in the binary information sequence that can be used at the receiver to overcome the effects of noise and interference encountered in the transmission on the signal through the channel.

   For example take k bits of the information sequence and map that k bits to unique n bit sequence called code word. The amount of redundancy introduced is measured by the ratio *n/k* and the reciprocal of this ratio (k/n) is known as *rate of code or code rate.*

4. **Digital Modulator:**

The binary sequence is passed to digital modulator which in turns convert the sequence into electric signals so that we can transmit them on channel (we will see channel later). The digital modulator maps the binary sequences into signal wave forms , for example if we represent 1 by sin x and 0 by cos x then we will transmit sin x for 1 and cos x for 0. (a case similar to BPSK)

5. **Channel:**

The communication channel is the physical medium that is used for transmitting signals from transmitter to receiver. In wireless system, this channel consists of atmosphere , for traditional telephony, this channel is wired , there are optical channels, under water acoustic channels etc.We further discriminate this channels on the basis of their property and characteristics, like AWGN channel etc.

6. **Digital Demodulator:**

The digital demodulator processes the channel corrupted transmitted waveform and reduces the waveform to the sequence of numbers that represents estimates of the transmitted data symbols.

7. **Channel Decoder:**

This sequence of numbers then passed through the channel decoder which attempts to reconstruct the original information sequence from the knowledge of the code used by the channel encoder and the redundancy contained in the received data.

8. **Source Decoder:**

At the end, if an analog signal is desired then source decoder tries to decode the sequence from the knowledge of the encoding algorithm. And which results in the approximate replica of the input at the transmitter end.

9. **Output Transducer:**

Finally we get the desired signal in desired format analog or digital.

2. Explain in detail sampling and types of sampling?

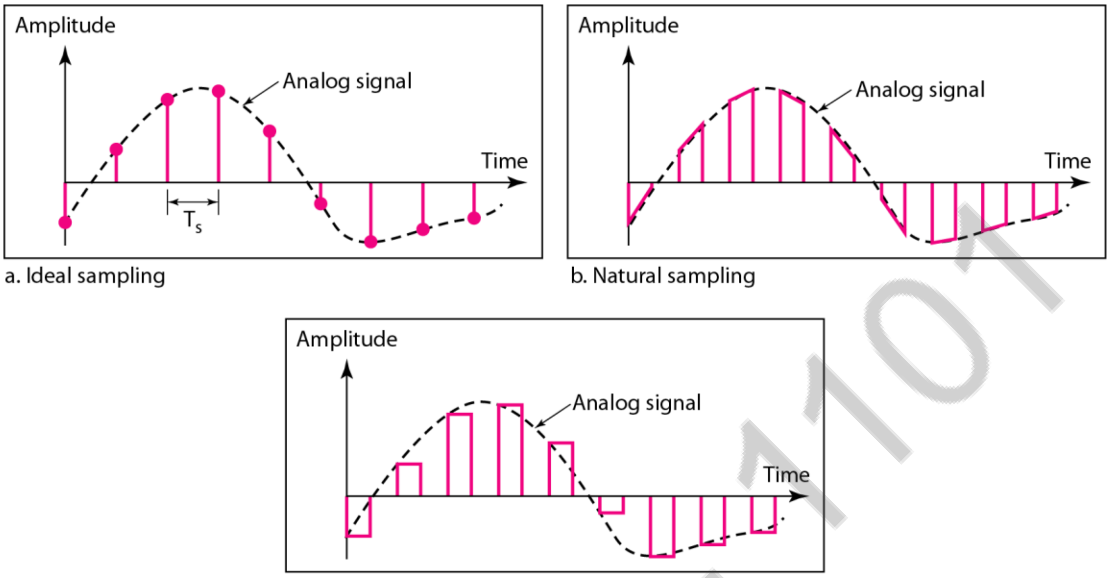
Sampling:

- Process of converting analog signal into discrete signal.
- Sampling is common in all pulse modulation techniques
- The signal is sampled at regular intervals such that each sample is proportional to amplitude of signal at that instant
- Analog signal is sampled every T_s $Secs$, called sampling interval. f_s=1/T_S is called sampling rate or sampling frequency.
- f_s=2f_m is Min. sampling rate called **Nyquist rate.** Sampled spectrum (ω) is repeating periodically without overlapping.
- Original spectrum is centered at ω=0 and having bandwidth of ω_m. Spectrum can be recovered by passing through low pass filter with cut-off ω_m.
- For f_s<2f_m sampled spectrum will overlap and cannot be recovered back.

This is called **aliasing.**

Sampling methods:

- Ideal – An impulse at each sampling instant.
- Natural – A pulse of Short width with varying amplitude.
- Flat Top – Uses sample and hold, like natural but with single amplitude value.



a. Ideal sampling

b. Natural sampling

c. Flat-top sampling

Fig. 4 Types of Sampling

Sampling of band-pass Signals:

- A band-pass signal of bandwidth $2f_m$ can be completely recovered from its samples. Min. sampling rate $=2\times Bandwidth$

$$=2\times 2f_m=4f_m$$

- Range of minimum sampling frequencies is in the range of $2\times BW$ *to* $4\times BW$

Instantaneous Sampling or Impulse Sampling:

- Sampling function is train of spectrum remains constant impulses throughout frequency range. It is not practical.

Natural sampling:

- The spectrum is weighted by a **sinc** function.
- Amplitude of high frequency components reduces.

Flat top sampling:

- Here top of the samples remains constant.
- In the spectrum high frequency components are attenuated due sinc pulse roll off. This is known as **Aperture effect.**
- If pulse width increases aperture effect is more i.e. more attenuation of high frequency components.

3.Explain in detail Pulse Code Modulation (PCM)

PCM system Block Diagram

The band pass filter limits the frequency of the analog input signal to the standard voice-band frequency range of 300 Hz to 3000 Hz. The sample- and- hold circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal.

The analog-to-digital converter (ADC) converts the PAM samples to parallel PCM codes, which are converted to serial binary data in the parallel-to-serial converter and then outputted onto the transmission linear serial digital pulses.

The transmission line repeaters are placed at prescribed distances to regenerate the digital pulses.

In the receiver, the serial-to-parallel converter converts serial pulses received from the transmission line to parallel PCM codes.

The digital-to-analog converter (DAC) converts the parallel PCM codes to multilevel PAM signals.

The hold circuit is basically a low pass filter that converts the PAM signals back to its original analog form.

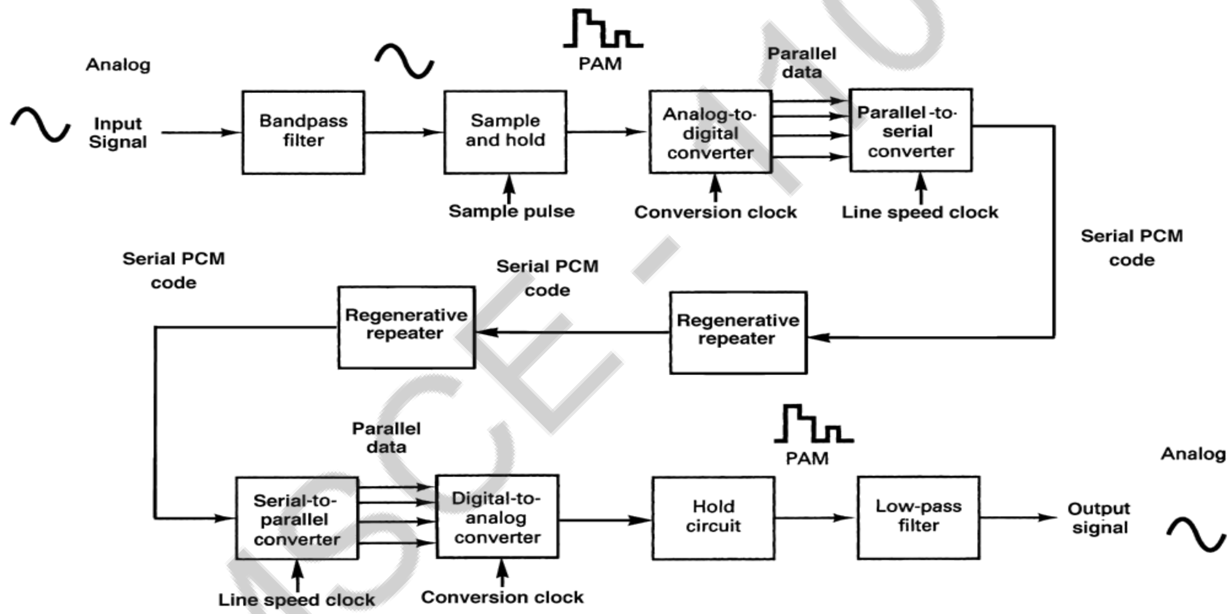


Figure 2.14 (a) Quantization Levels

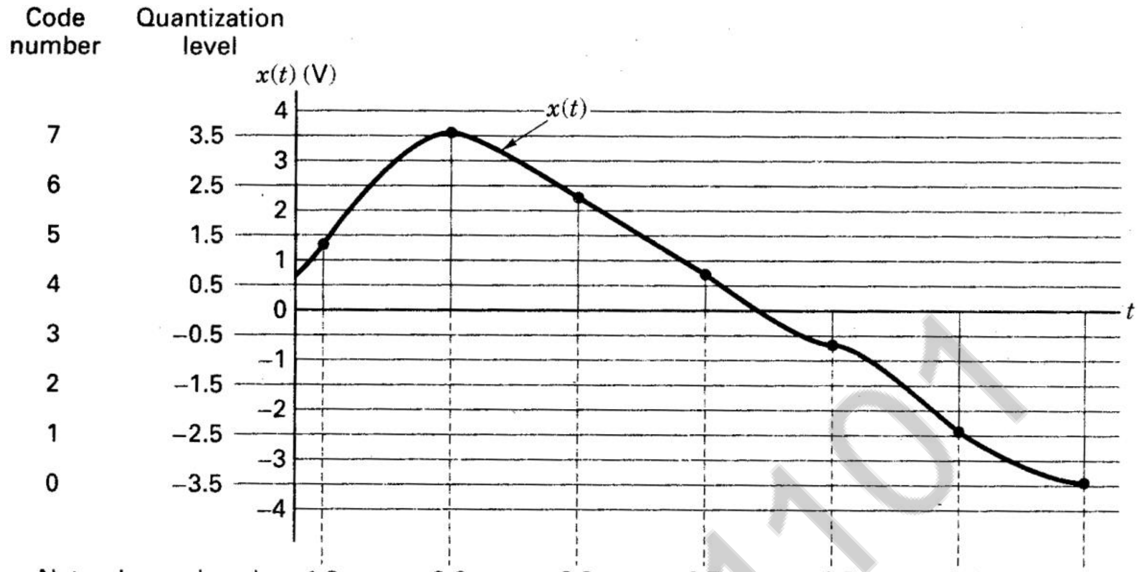
PCM Sampling:

The function of a sampling circuit in a PCM transmitter is to periodically sample the continually changing analog input voltage and convert those samples to a series of constant- amplitude pulses that can more easily be converted to binary PCM code.

A sample-and-hold circuit is a nonlinear device (mixer) with two inputs: the sampling pulse and the analog input signal.

For the ADC to accurately convert a voltage to a binary code, the voltage must be relatively constant so that the ADC can complete the conversion before the voltage level changes. If not, the ADC would be continually attempting to follow the changes and may never stabilize on any PCM code.

Figure shows an analog signal x(t) limited in its excursions to the range -4 to +4V. The stepsize between quantization levels has been set at 1V. Thus, eight quantization levels are employed. These are located at -3.5,-2.5,……+3.5V. Assign the code number 0 to the level at -3.5V, code number 1 to the level at -2.5V, and so on, until the level at 3.5V, which is assigned the code number 7.



Natural sample value	1.3	3.6	2.3	0.7	−0.7	−2.4	−3.4
Quantized sample value	1.5	3.5	2.5	0.5	−0.5	−2.5	−3.5
Code number	5	7	6	4	3	1	0
PCM sequence	101	111	110	100	011	001	000

Each code number has its representation in binary arithmetic, ranging from 000 for code number 0 to 111 for code number 7. The quantile intervals between the levels should be equal. The ordinate in Figure is labeled with quantization levels and their code numbers. Each sample of the analog signal is assigned to the quantization level closest to the value of the sample. There are four representations of x(t) as follows: the natural sample values, the quantized sample values, the code numbers, and the PCM sequence.

Here, each sample is assigned to one of eight levels or a three-bit PCM sequence. Increasing the number of levels will reduce the quantization noise. If we double the number of levels to 16, each analog sample will be represented as a four- bit PCM sequence. But when there are more bits per sample, the data rate is increased, and the cost is a greater transmission bandwidth. Thus, we can obtain better fidelity at the cost of more transmission bandwidth. each sample voltage is rounded off (quantized) to the closest available level and then converted to its corresponding PCM code.

The rounded off error is called the called the *quantization error (Qe).*

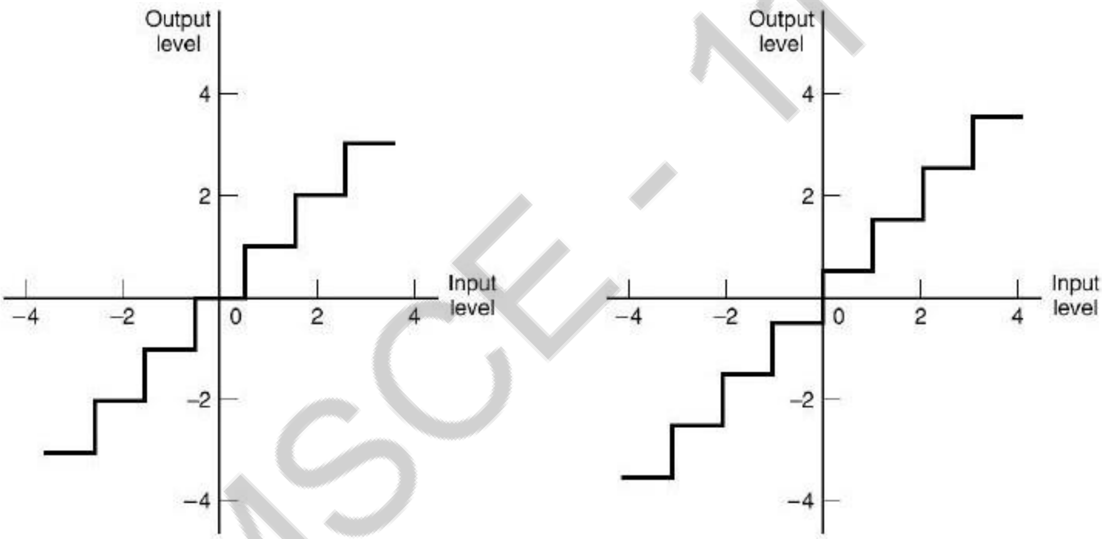
To determine the PCM code for a particular sample voltage, simply divide the voltage by the resolution, convert the quotient to an n-bit binary code, and then add the sign bit.

4.Describe briefly Uniform and non-uniform quantization

In pulse code modulation both the parameters time and amplitude are expressed in discrete form. The sampling process converts the continuous time values of the analog signal into discrete time values. The quantization process converts the continuous amplitude values into a finite (discrete) set of allowable values. This process is called “discretization” in time and amplitude. Here, we shall study about the quantization process. Basically, quantization process may be classified as follows:

Uniform quantization

When the quantization levels are uniformly distributed over the full amplitude range of the input signal, the quantizer is called an uniform or linear quantizer. In uniform quantization, the stepsize between quantization levels remains the same throughout the input range. The quantizer characteristic can also be midtread or midrise type, as shown in the Figure



a) Midtread Type **b) Midrise Type**

(a) For the uniform quantizer of midtread type, the origin lies in the middle of a tread of the staircase like graph.

(b) For the Uniform quantizer of midrise type, the origin lies in the middle of a rising part of the staircase like graph.

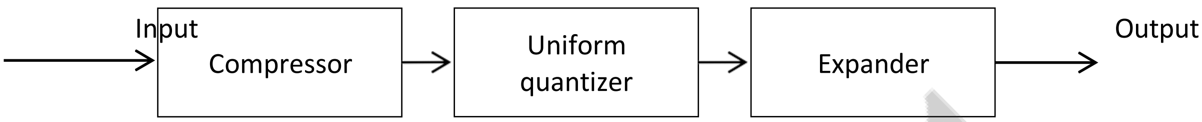
Both the midtread and midrise types of uniform quantizers are symmertric about the origin. Hence they are also called as symmetric quantizer.

Non-uniform quantization

If the quantizer characteristic is nonlinear, then the quantization is known as non-uniform quantization. In non-uniform quantization, the step size is not constant. The step size is variable, depending on the amplitude of input signal.

Companding

The non-uniform quantization is practically achieved through a process called companding. Figure 2.16 shows a companding model. The compressor amplifies weak signals and attenuates strong signals.

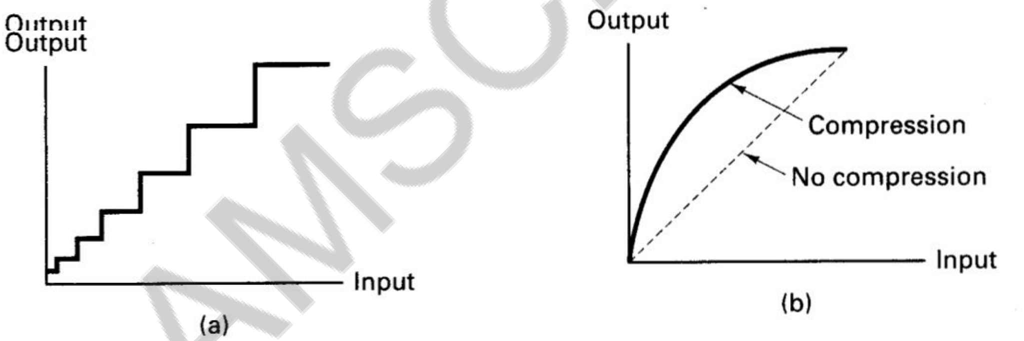


A companding model

This process is called compression. At the receiver, the expander does the opposite function of compression. Thus the expander provides expansion. Therefore, the compression of the signal at the transmitter and the expansion at the receiver is combined to be called as companding.

Companding = Compressing + Expanding

The non-uniform quantizer characteristic is shown in the figure

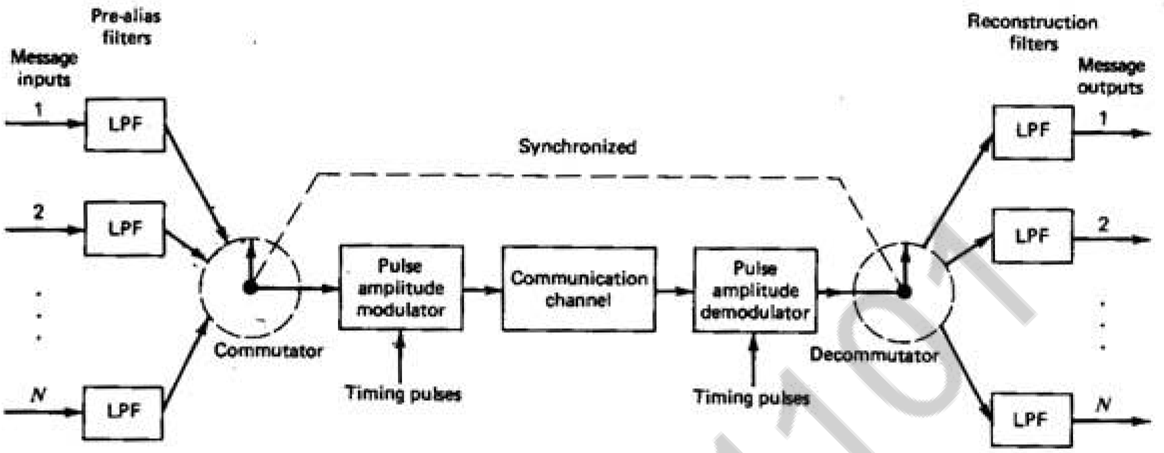


5. Explain in detail about Time Division Multiplexing (TDM)

In TDM, group of signals are sampled sequentially in time at a common sampling rate and then multiplexed for transmission over a common channel. This enables us to combine several digital signals, such as computer outputs, digitized voice signals, digitized facsimile and television signals, into a single data stream with a higher bit rate.

The concept of a PAM / TDM system is shown in the Figure 4.30.

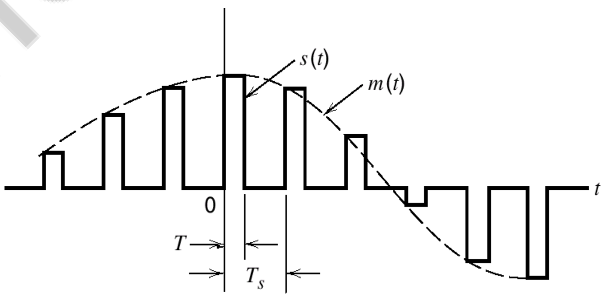
There are N analog message signals in the input. Each message signal is restricted in bandwidth by a low-pass pre-alias filter. The pre-alias filter outputs are applied to a commutator. The commutator is an electronic switching circuitry which takes a narrow sample of each of the N input messages at rate f_s. Such multiplexed samples are then applied to a pulse-amplitude modulator. It transforms the multiplexed signal into a form suitable for transmission over the communication channel.



The received signal is applied to a pulse amplitude demodulator. The short pulses produced at the demodulator output are distributed to the appropriate low- pass reconstruction filters by means of a decommutator. The decommutator operates in synchronism with the commutator. The transmitted message signals are reproduced at the corresponding filter outputs.

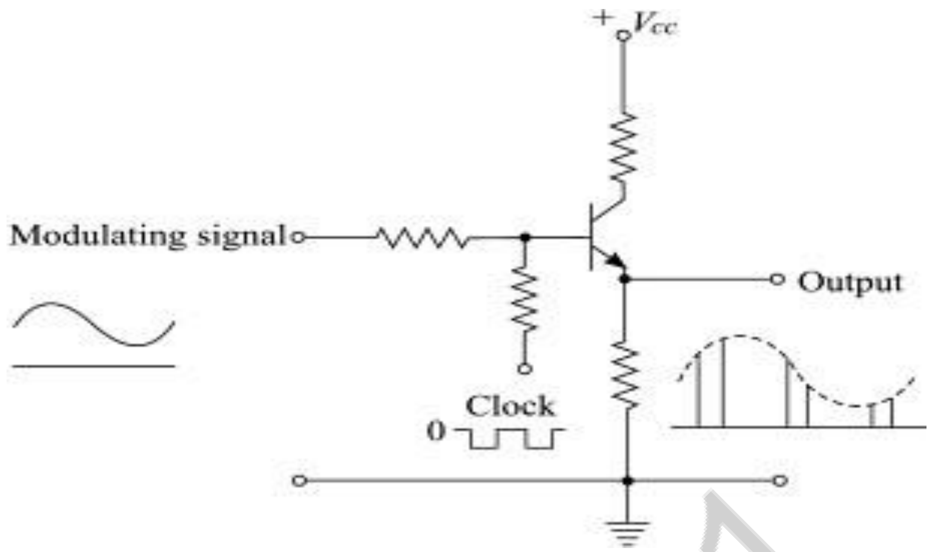
6. Describe brieflyPulse Amplitude Modulation:

In PAM, amplitude of pulses is varied in accordance with instantaneous value of modulating signal.



PAM Generation:

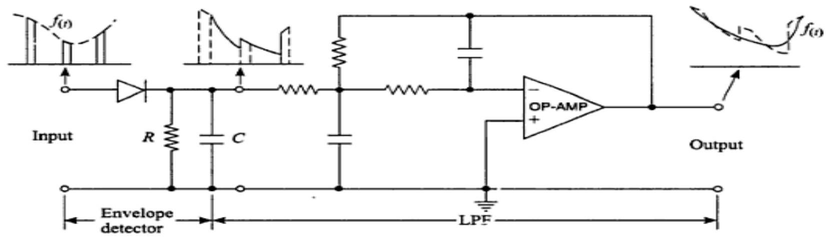
The carrier is in the form of narrow pulses having frequency fc. The uniform sampling takes place in multiplier to generate PAM signal. Samples are placed Ts sec away from each other.



- The circuit is simple emitter follower.
- In the absence of the clock signal, the output follows input.
- The modulating signal is applied as the input signal.
- Another input to the base of the transistor is the clock signal.
- The frequency of the clock signal is made equal to the desired carrier pulse train frequency.
- The amplitude of the clock signal is chosen the high level is at ground level(0v) and low level at some negative voltage sufficient to bring the transistor in cutoff region.
- When clock is high, circuit operates as emitter follower and the output follows in the input modulating signal.
- When clock signal is low, transistor is cutoff and output is zero.
- Thus the output is the desired PAM signal.

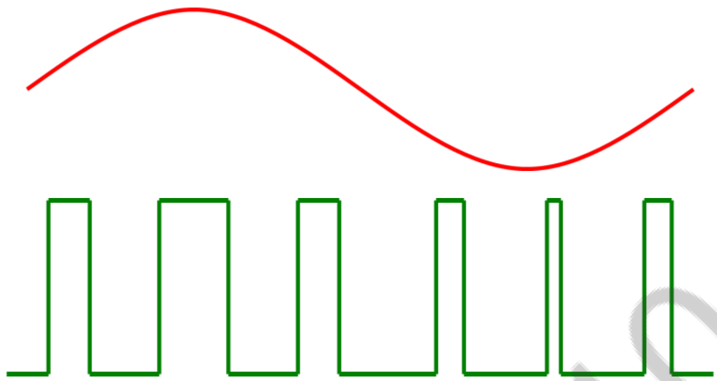
PAM Demodulator:

- The PAM demodulator circuit which is just an envelope detector followed by a second order op-amp low pass filter (to have good filtering characteristics) is as shown below



7.Describe briefly about Pulse Position Modulation and Pulse Width Modulation:

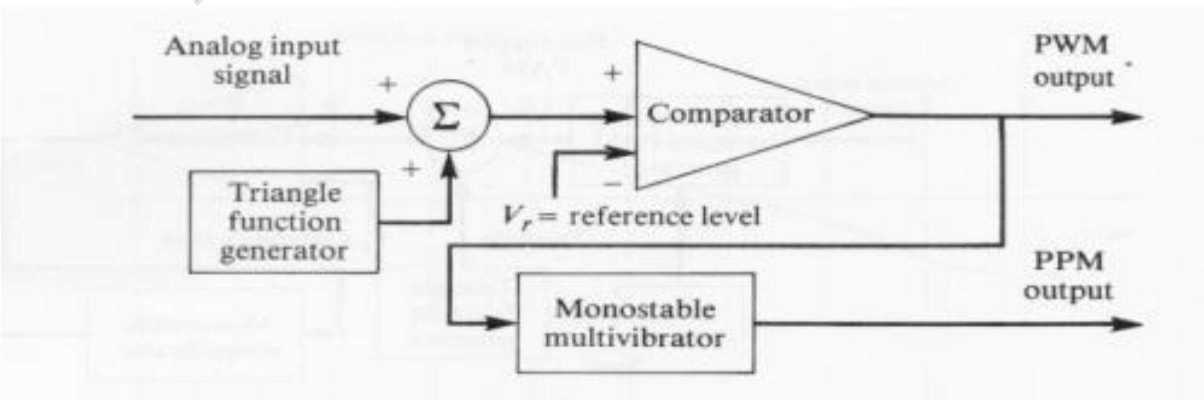
- In this type, the amplitude is maintained constant but the width of each pulse is varied in accordance with instantaneous value of the analog signal.



- In PWM information is contained in width variation. This is similar to FM.
- In pulse width modulation (PWM), the width of each pulse is made directly proportional to the amplitude of the information signal.

- In this type, the sampled waveform has fixed amplitude and width whereas the position of each pulse is varied as per instantaneous value of the analog signal.
- PPM signal is further modification of a PWM signal.

PPM & PWM Modulator:



- The PPM signal can be generated from PWM signal.

- The PWM pulses obtained at the comparator output are applied to a mono stable multi vibrator which is negative edge triggered.
- Hence for each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its RC components.
- Thus as the trailing edges of the PWM signal keeps shifting in proportion with the modulating signal, the PPM pulses also keep shifting.
- Therefore all the PPM pulses have the same amplitude and width. The information is conveyed via changing position of pulses.

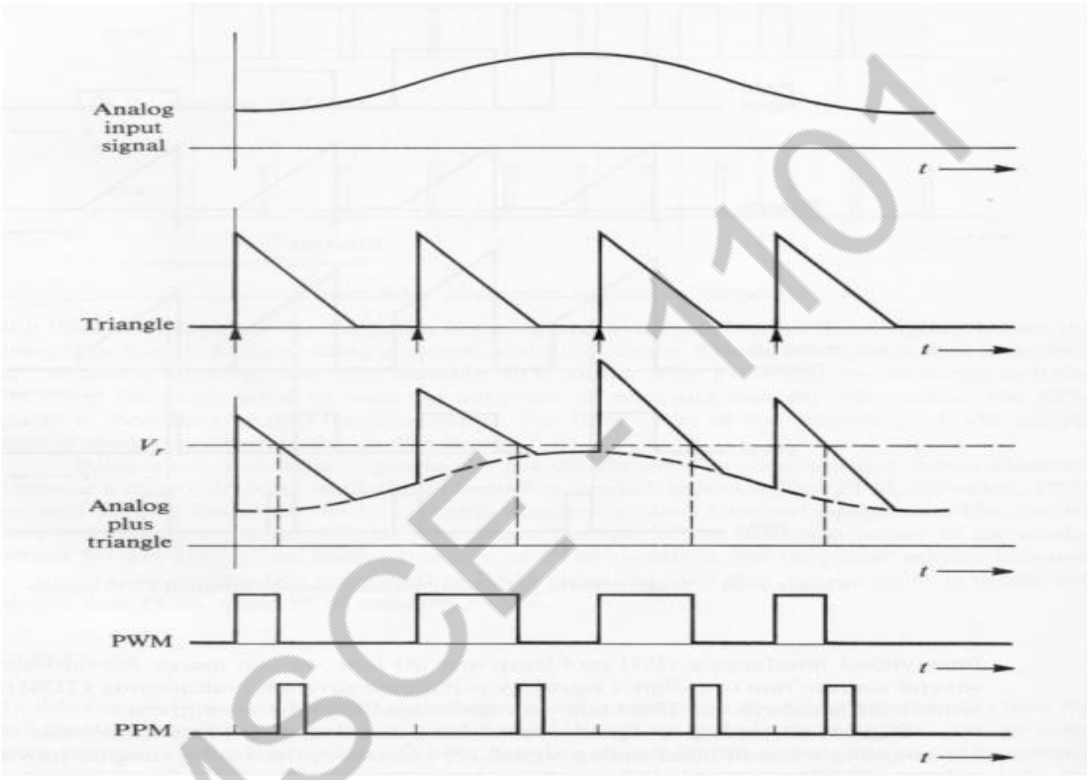
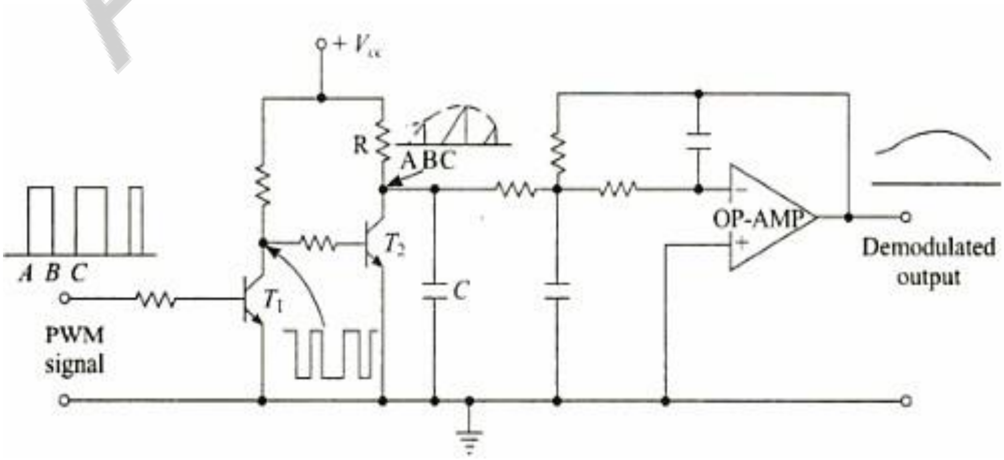


Fig.15. PWM & PPM Modulation waveforms

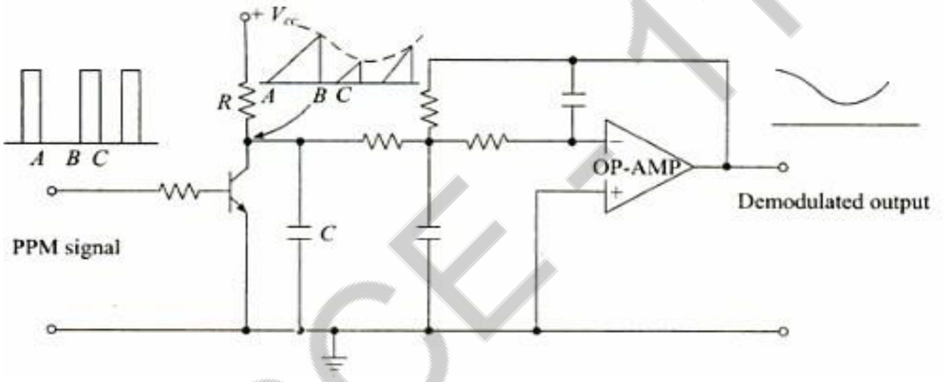
PWM Demodulator:



- Transistor T1 works as an inverter.

- During time interval A-B when the PWM signal is high the input to transistor T2 is low.
- Therefore, during this time interval T2 is cut-off and capacitor C is charged through an R-C combination.
- During time interval B-C when PWM signal is low, the input to transistor T2 is high, and it gets saturated.
- The capacitor C discharges rapidly through T2.The collector voltage of T2 during B- C is low.
- Thus, the waveform at the collector of T2is similar to saw-tooth waveform whose envelope is the modulating signal.
- Passing it through 2nd order op-amp Low Pass Filter, gives demodulated signal.

PPM Demodulator:



- The gaps between the pulses of a PPM signal contain the information regarding the modulating signal.
- During gap A-B between the pulses the transistor is cut-off and the capacitor C gets charged through R-C combination.
- During the pulse duration B-C the capacitor discharges through transistor and the collector voltage becomes low.
- Thus, waveform across collector is saw-tooth waveform whose envelope is the modulating signal.
- Passing it through 2nd order op-amp Low Pass Filter, gives demodulated signal.