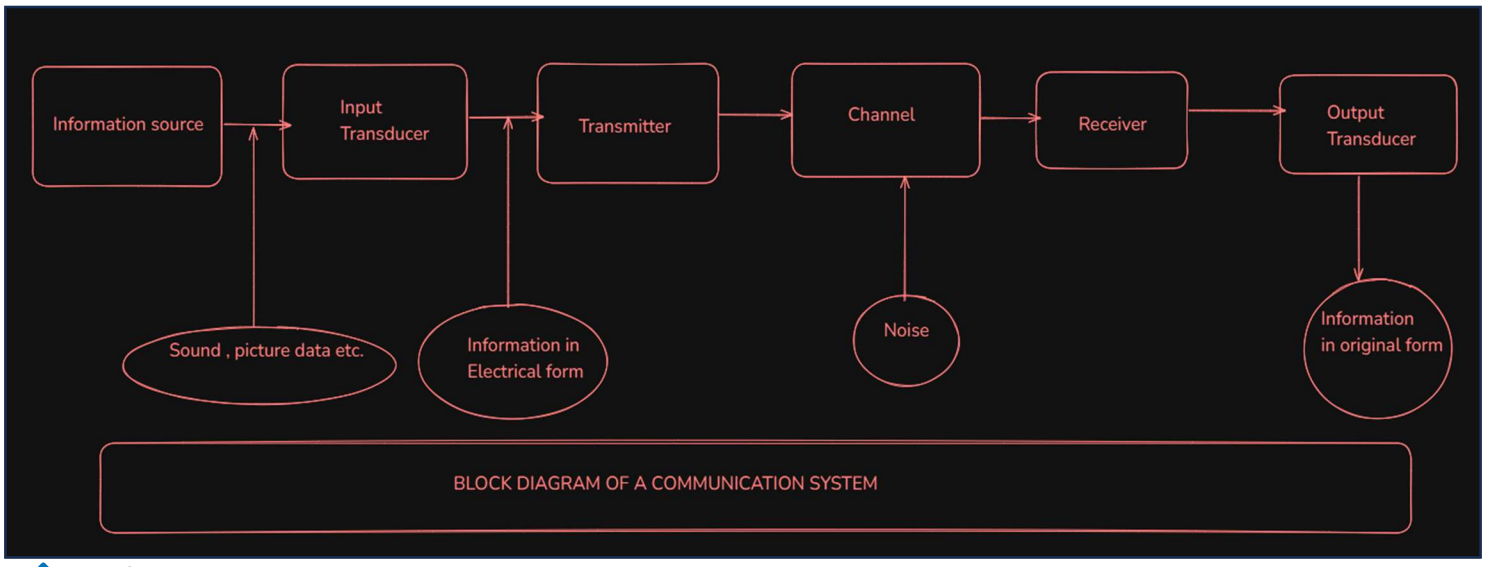
Principle of Communication system

1. **WHAT DO YOU MEAN BY COMMUNCIATION ?**

   ANS: **Communication system** means a setup or process used to **transfer information** from one place (sender) to another (receiver).

2. **DRAW THE BLOCK DIAGRAM OF COMMUNICATION SYSTEM AND BRIEFLY EXPLAIN EACH BLOCK**

   ANS:



1. Information Source

This is where the original message or data comes from.
Example: A person speaking, or a computer sending data.

---

2. Input Transducer

It changes the message into an electrical signal.
Example: A microphone converts voice into an electrical signal.

---

3. Transmitter

It **processes the signal** to make it suitable for sending over long distances. It may **amplify** or **modulate** the signal.
Example: A radio transmitter prepares the signal to be sent via antennas.

---

4. Channel

This is the **medium** through which the signal travels.
Example: Air (in wireless), cables (in wired), or optical fibers.

🔷 **5. Receiver**

It **collects the signal** from the channel and **processes** it to get back the original message. It may **demodulate** or **filter** the signal.

🔷 **6. Destination**

This is the final place where the message is delivered.
Example: A speaker (for sound), a TV (for video), or a computer (for data).

3. WHAT IS BASEBAND TRANSMISSION ?

ANS: **Baseband transmission** is a method of sending the original **low-frequency (unmodulated)** signal **without changing its frequency**.

- It sends the original signal **without modulation**.
- Used for **short-distance** communication.
- Mostly used in **digital communication** like computer networks.

In LAN (Local Area Network), digital signals are sent directly through a cable — this is baseband transmission.

4. WHAT IS MEANT BY MODULATION?

ANS: **Modulation** is the process of **changing some properties of a carrier signal** (like amplitude, frequency, or phase) **according to the message signal**.

In FM radio, your voice is added to a high-frequency signal so it can be transmitted and heard far away.

5. WHAT ARE THE TYPES OF MODULATION?

ANS: Modulation is mainly classified into two types:

🔹 **1. Continuous Wave Modulation (Analog Modulation):**
In this type, a continuous carrier wave is used, and its parameters like amplitude, frequency, or phase are varied according to the message signal.

🔹 **2. Pulse Modulation:**
In this type, the message signal is transmitted in the form of pulses. It is further divided into:

- **a) PAM** – Pulse Amplitude Modulation
- **b) PDM** – Pulse Duration Modulation
- **c) PPM** – Pulse Position Modulation

6. WHAT ARE THE NEEDS OF MODULATION?

ANS:
Modulation is necessary in communication systems for the following reasons:

🔹 **1. Practicality of Antenna Size:**
For transmitting low-frequency signals directly, the required antenna length would be very large. Modulation increases the frequency, so **smaller and practical antennas** can be used.

🔹 **2. To Remove Interference:**

If multiple signals are sent at the same frequency, they interfere. Modulation assigns **different frequency bands** to different signals, **avoiding interference**.

🔹 **3. Reduction of Noise:**

High-frequency signals are **less affected by noise** during transmission. Modulation helps send signals at high frequencies, **improving clarity**.

7. WHAT IS AMPLITUDE MODULATION?

ANS: 🔹 **Definition:**

Amplitude Modulation is a type of modulation in which the **amplitude of the carrier wave** is **varied in proportion** to the **instantaneous value of the message (modulating) signal**, while the **frequency and phase remain constant**.

🔹 **Mathematical Expression:**

Let:

$m(t)$ = message signal

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

Then the **AM signal** is: $s(t)=[A_c+m(t)]\cos(2\pi f_c t)$

Where:

A_c = carrier amplitude

$\mu=A_m/A_c$ = modulation index

- f_c = carrier frequency
- f_m=message frequency

8. DEFINE MODULATION INDEX .

ANS:

✅ **Modulation Index (μ)**

The **modulation index** (denoted as **μ**) in **Amplitude Modulation (AM)** measures the **extent** or **depth** of modulation. It indicates how much the **amplitude of the carrier wave** is varied by the message signal.

🔹 **Definition:**

The modulation index is the ratio of the **amplitude of the message signal** (A_m) to the **amplitude of the carrier signal** (A_c).

$\mu=A_m/A_c$

Where:

- A_m = Amplitude of the message signal
- A_c = Amplitude of the carrier signal

🔹 **Significance of Modulation Index:**

1. **If μ=1**: The carrier is fully modulated (maximum modulation).
2. **If μ>1**: Over-modulation occurs, which leads to **distortion**.
3. **If μ<1**: Under-modulation, meaning the message signal is not fully modulating the carrier.

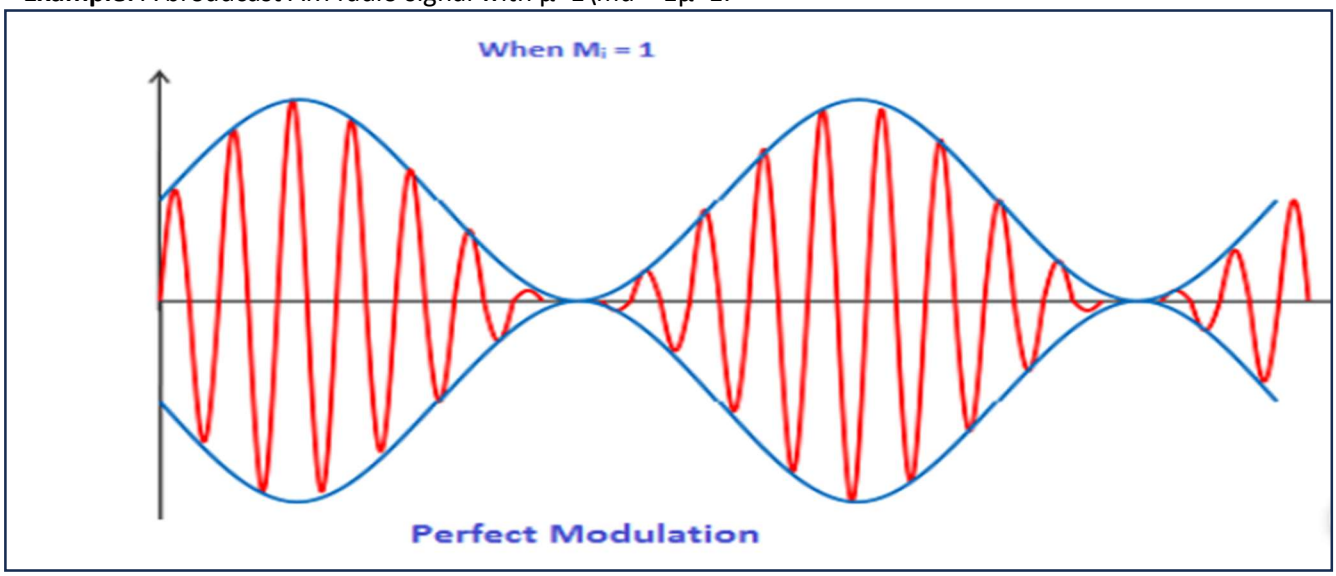
9. WHAT ARE THE TYPES OF AMPLITUDE MODULATED SIGNAL?

ANS: ✅ **Types of Amplitude Modulated (AM) Signals**

Amplitude Modulated signals can be classified into three types based on the **modulation index (μ)**, which is the ratio of the message signal amplitude to the carrier amplitude.

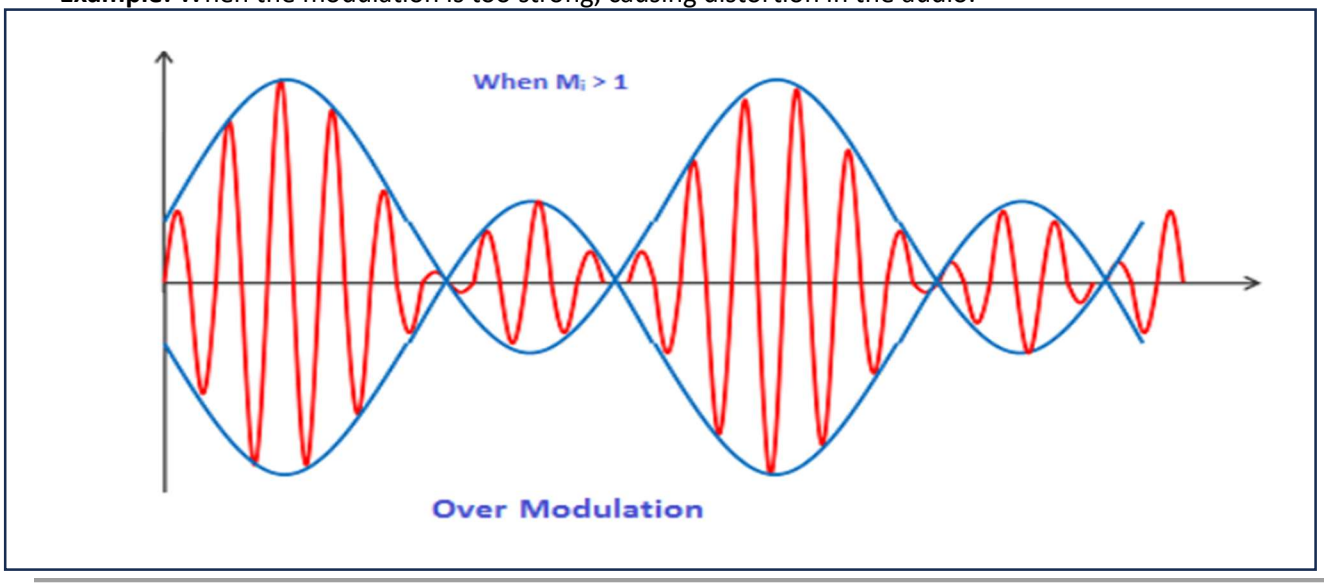
🔹 1. Perfectly Modulated Signal:

- This occurs when the **modulation index** (μ\muμ) is **equal to 1**.
- The **carrier amplitude** is fully modulated by the **message signal**.
- There is no **distortion**, and the AM wave is transmitted with optimal efficiency.
- **Example:** A broadcast AM radio signal with μ=1\mu = 1μ=1.



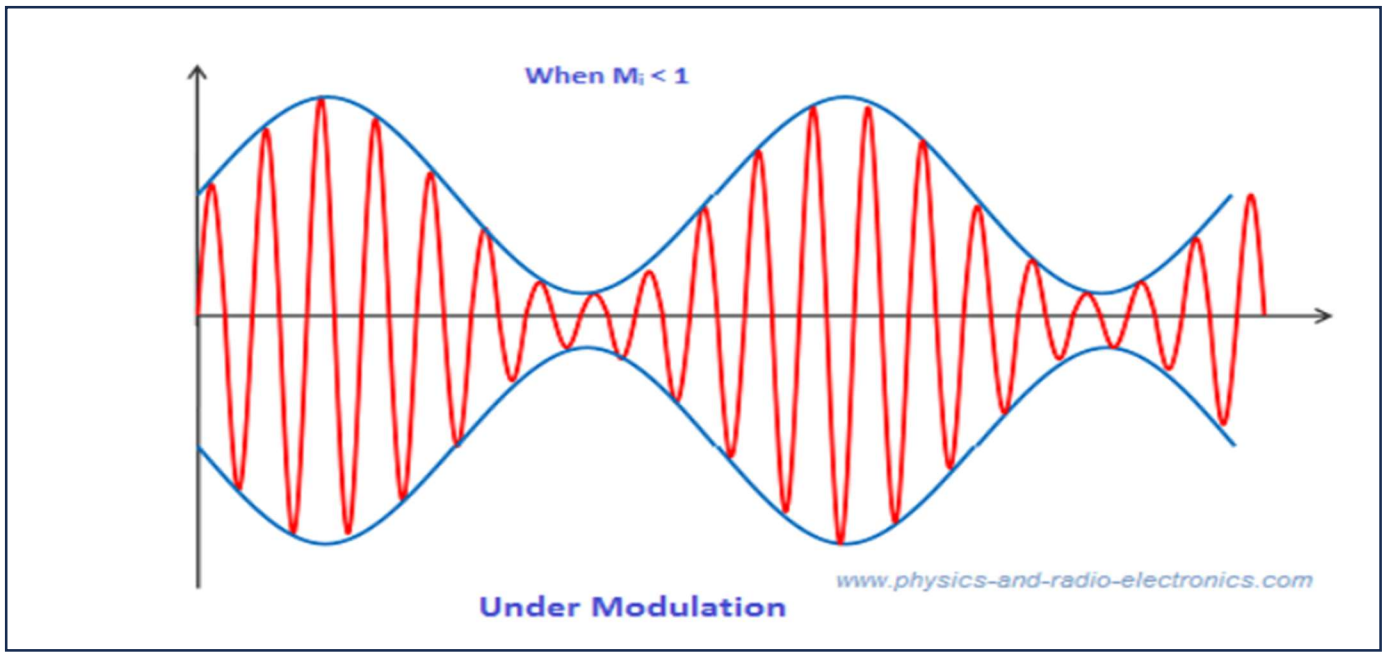
🔹 2. Overmodulated Signal:

- This occurs when the **modulation index** (μ\muμ) is **greater than 1**.
- The **message signal amplitude** is so high that the **carrier wave is over-modulated**.
- **Distortion** occurs because the carrier's amplitude may go negative, causing **sideband overlap** and **signal distortion**.
- **Example:** When the modulation is too strong, causing distortion in the audio.



🔹 3. Under-modulated Signal:

- This occurs when the **modulation index** (μ\muμ) is **less than 1**.
- The **message signal** does not fully modulate the **carrier wave**, so the signal is weak and may not convey the full message clearly.
- **Example:** Weak radio signal or low-frequency sound transmission.



10. DERIVE AN EXPRESSION FOR SINGLETONE AMPLITUDE MODULATED WAVE .

ANS:

Let us consider a single tone modulating signal as :

$x(t) = V_m \cos \omega_m t$ .....(1)

which contains a single frequency ω_m.

Signal x(t) may be a voltage signal or a current signal. Here, we have assumed that x(t) is a voltage signal with maximum amplitude equal to V_m.

Let the carrier signal be :

$c(t) = A \cos \omega_c t$ .....(2)

We know that the general expression for AM signal is :

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

or,

$s(t) = A \cos \omega_c t + x(t) \cos \omega_c t$ .....(3)

Putting the value of x(t), we get

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

or,

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

or,

$s(t) = A \cos \omega_c t \left[1 + \frac{V_m}{A} \cos \omega_m t \right]$ .....(4)

But we know that for AM, the modulation index m_a is given as

$$m_a = \frac{|x(t)|_{max}}{A}$$

where $|x(t)|_{max}$ denotes the maximum amplitude of modulating signal and A is the maximum amplitude of the carrier signal.

In this case, we have

$|x(t)|_{max} = V_m$

Therefore,

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

Putting this value m_a of in equation (4) we get

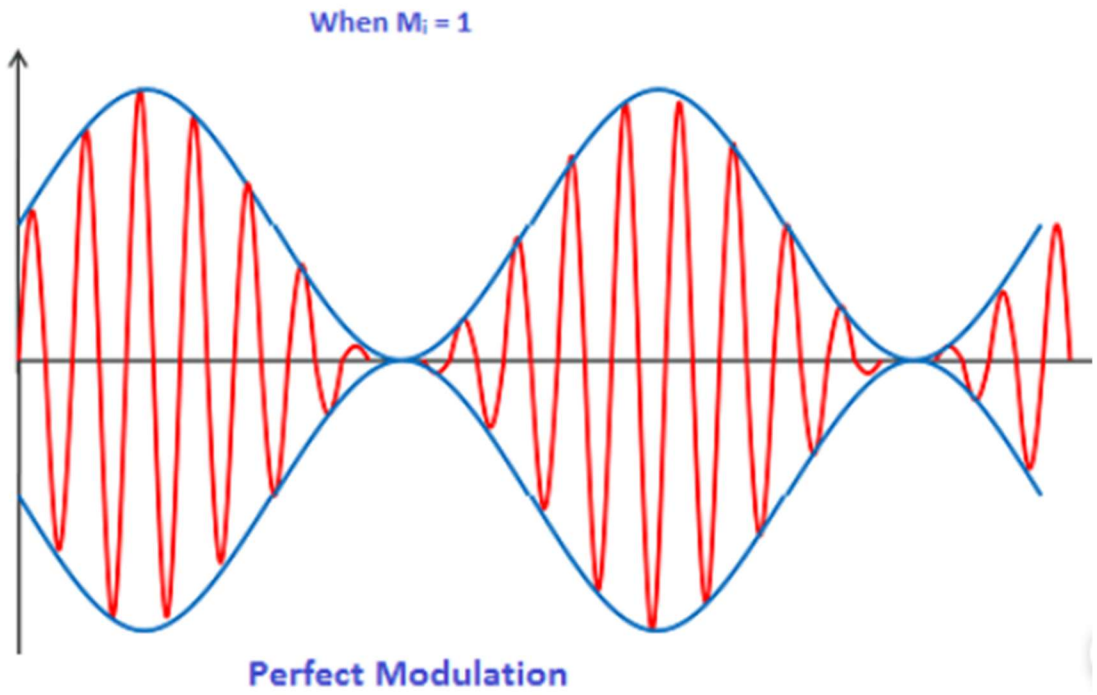
$s(t) = A \cos \omega_c t \left[1 + m_a \cos \omega_m t \right]$ .....(5)

This is the desired expression for single-tone modulated signal.

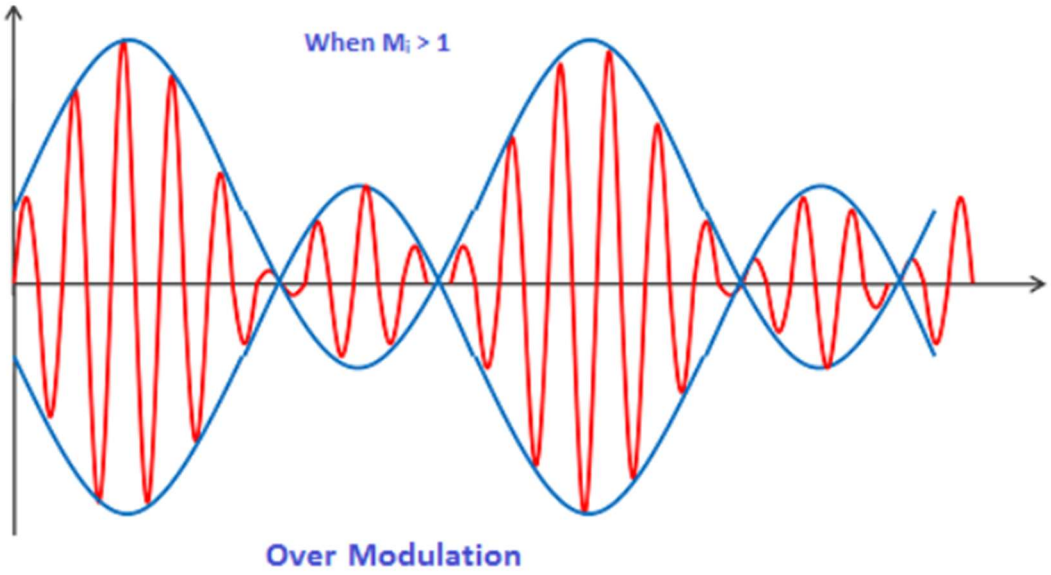
11. **DRAW THE WAVEFORM OF PERFECT-MODULATED , OVERMODULATED & UNDERMODULATED AM WAVE**.

ANS:

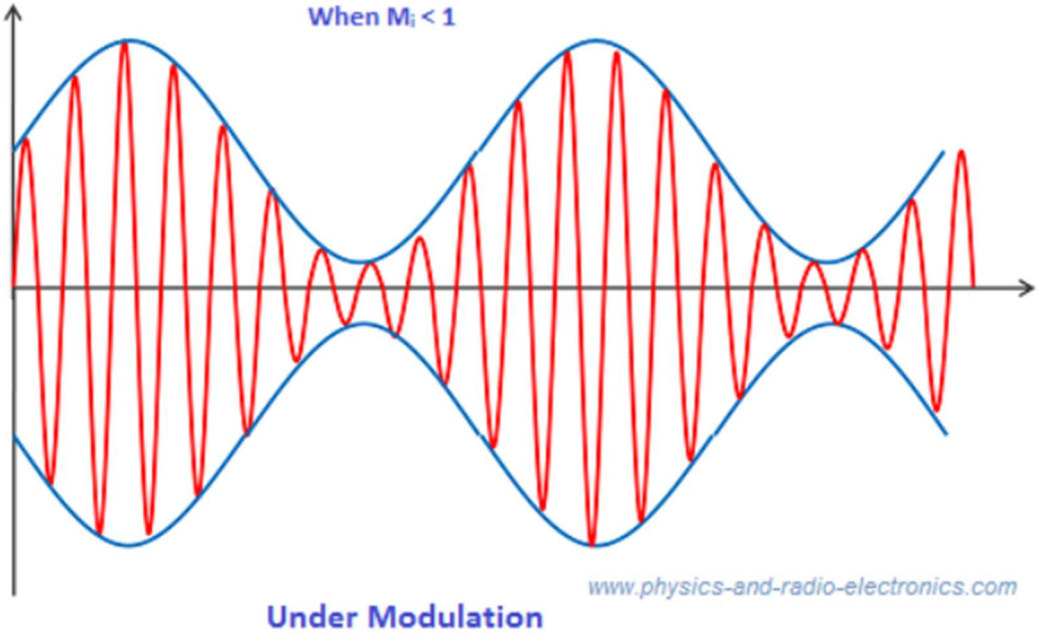
A. PERFECT MODULATED: M=1



B. OVER MODULATED



C. UNDER MODULATED :



12. WHAT ARE THE CODITIONS FOR OVERMODULATTION , UNDERMODULATION AND PERFECT MODULATION:

ANS:

🔹 1. Perfect Modulation

Condition: $\mu = 1$

➡️ The message signal **exactly matches** the carrier amplitude.

➡️ Best quality without distortion.

🔹 2. Overmodulation

Condition: $\mu > 1$

➡️ Message amplitude is **greater than** carrier amplitude.

➡️ Causes **distortion** in the AM wave.

🔹 3. Undermodulation

Condition: $\mu < 1$

➡️ Message amplitude is **less than** carrier amplitude.

➡️ Signal is **transmitted with low strength**.

13. DERIVE THE MATHEMATICAL EXPRESSION FOR MODULATION INDEX :

ANS:

We Know that

E = amplitude of modulated signal

$= A(1 + m_a \cos \omega_m t)$

So, $E_{max} = A(1 + m_a)$ and $E_{min} = A(1 - m_a)$

Now, $\frac{E_{max}}{E_{min}} = \frac{1 + m_a}{1 - m_a}$.

$\Rightarrow E_{max} - m_a E_{max} = E_{min} + m_a E_{min}$.

$\Rightarrow E_{max} - E_{min} = m_a (E_{max} + E_{min})$

$\Rightarrow m_a = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$.

'Therefore' modulation index

$m_a = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$.

14. DERIVE THE EXPRESSION FOR POWER CONTENT OF AM WAVE

ANS:

⇒ Power content in AM wave :-

We Know that the general expression of AM wave is given as

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

The total power P of the AM wave is the sum of the carrier power and sideband power P_s.

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

Now the carrier power P_c is equal to mean-square (ms) value of the carrier term $A\cos\omega_c t$ i.e.

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P_c = mean square value of $A\cos\omega_c t$

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

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

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

$$= \frac{A^2}{4\pi}\int_0^{2\pi}(1+\cos 2\omega_c t)\, dt$$

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

$$= \frac{A^2}{4\pi}\times 2\pi$$

$$= \frac{A^2}{2}$$

Now, the sideband power P_s is equal to the mean square value of sideband term $x(t)\cos\omega_c t$ i.e.

P_s = mean square value of $x(t)\cos\omega_c t$

$$= \overline{(x(t)\cos\omega_c t)^2}$$

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

$$= \frac{1}{2\pi}\int_0^{2\pi} \frac{1}{2}[2\cos^2\omega_c t]\, x^2(t)\, dt$$

$$= \frac{1}{2\pi}\int_0^{2\pi} \frac{1}{2}[1+\cos 2\omega_c t]\, x^2(t)\, dt$$

$$= \frac{1}{2\pi}\int_0^{2\pi} \frac{1}{2} x^2(t)\, dt + \frac{1}{2\pi}\int_0^{2\pi} \frac{1}{2}\cos 2\omega_c t\, x^2(t)\, dt$$

In AM generation, a Bandpass filter or a tuned circuit tuned to carrier frequency ω_c is used to filter out the second integral term.

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'Therefore'

$$P_S = \frac{1}{2\pi}\int_0^{2\pi}\left[\frac{1}{2}x^2(t)\right]dt$$

$$= \frac{1}{2\pi}\int_0^{2\pi}\frac{1}{2}V_m^2\cos^2\omega_m t\, dt$$

$$= \frac{V_m^2}{2\pi}\times\frac{1}{4}\int_0^{2\pi} 2\cos^2\omega_m t\, dt$$

$$= \frac{V_m^2}{2\pi\times 4}\int_0^{2\pi}(1+\cos 2\omega_m t)\, dt$$

$$= \frac{V_m^2}{2\pi\times 4}\times 2\pi$$

$$= \frac{V_m^2}{4}.$$

So, total modulated power $P_t = \frac{A^2}{2} + \frac{V_m^2}{4}$.

$$P_t = \frac{A^2}{2}\left[1+\frac{2}{A^2}\times\frac{V_m^2}{4}\right]$$

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

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

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

15. FIND EXPRESSION FOR TRANSMISSION EFFICIENCY OF AM SIGNAL

ANS:

Transmission Efficiency of Single-tone Amplitude Modulated Signal

We know that the total modulated power of an AM signal is expressed as

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

Out of this total power P_t, the useful message or baseband power is the power carried by the side bands i.e., P_s. The large carrier power P_c is a waste from the transmission point of view because it does not carry any information or message. This large carrier power P_c is transmitted along with the sideband power only for the convenient and cheap detection. Hence P_s is the only useful message power present in the AM wave.

In AM wave, the amount of useful message power P_s may be expressed by a term known as transmission efficiency η.

Hence transmission efficiency of AM wave may be defined as the percentage of total power contributed by the sideband. Mathematically

Transmission efficiency

$$\eta = \frac{P_s}{P_t} \times 100$$

$$\text{or, } \eta = \frac{V_m^2/4}{\frac{A_c^2}{2} + \frac{V_m^2}{4}} \times 100 = \frac{V_m^2}{2A_c^2 + V_m^2} = \frac{m_a^2}{2 + m_a^2} \times 100$$

The maximum transmission efficiency of the AM is only 33.33%. This implies that only 1/3rd power of the total power is carried by sideband and 2/3 is wasted

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16. **ELIMINATED**
17. **DESCRIBE THE GENERATION OF AM SIGNAL USING SQAURE LAW DIODE MODULATION TECHNIQUE:**
    ANS:
18. **HOW IS AM SIGNAL DEMODULATED USING SQ. LAW DETECTOR:**

ANS:

(a) Square-Law Detector :-

(a) Definition: The square law detector circuit is used for detecting modulated signal of small magnitude (i.e. below 1 volt) so that operating region may be restricted to the non-linear portion of the v-i characteristics of the device.

(b) Circuit Diagram:-
Figure below shows the circuit of a square-law detector.

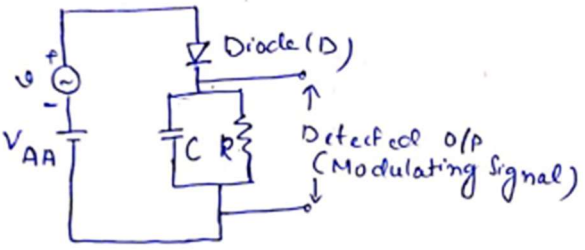


Fig:- Basic circuit of square law diode detector

In the circuit dc supply voltage V_{AA} is used to get fixed operating point in the non-linear portion of v-i characteristic of diode.

The distorted o/p diode current is expressed by the non-linear v-i relationship as

$$i = av + bv^2 \rightarrow (1)$$

Here v is the input modulated voltage.

We know that AM wave is expressed as

$$v = A(1 + m_a \cos \omega_m t) \cos \omega_c t. \rightarrow (2)$$

Substituting the value of v in equation (1) we get-

$$i = a[A(1 + m_a \cos\omega_m t)\cos\omega_c t + b[A(1 + m_a \cos\omega_m t)\cos\omega_c t]^2 \quad \rightarrow (2)$$

Now, if above expression is expanded, we may observe the presence of terms of frequency like $2\omega_c$, $2(\omega_c \pm \omega_m)$, ω_m and $2\omega_m$ besides the input frequency terms.

Hence, this diode current i containing all these frequency terms is passed through a low-pass filter which allows to pass the frequencies below or upto modulating frequency ω_m and rejects the other higher frequency components. Therefore, the modulating or baseband signal with frequency ω_m is recovered from the input modulated signal.

19. HOW IS DSB-SC GENERATED USING A BALANCED MODULATOR

⇨ Double Sideband Suppressed Carrier (DSB-SC) System:-

In AM signal, the carrier contains no information. Any transmitted information lies solely in the sideband. For that reason the carrier may be suppressed and not transmitted. The modulated signal which contains no carrier but two sideband is called double-sideband suppressed carrier (DSB-SC) modulation.

Generation of DSB-SC signal:-

The expression for DSB-SC signal is given by

$$s(t) = x(t) \cos \omega_c t$$

∴ where $x(t)$ = Baseband Signal

$\cos \omega_c t$ = Carrier Signal.

From this expression, we see that a DSB-SC signal is basically the product of the modulating signal and carrier signal. Unfortunately, a single electronic device can not generate a DSB-SC signal. The circuit to achieve the generation of a DSB-SC signal is called a product modulator namely balanced modulator and the ring modulator.

Circuit Diagram:- Fig below shows a balanced modulator circuit using two diodes. A modulating signal $x(t)$ is applied to the two diodes through a centre-tapped transformer with the carrier signal $\cos \omega_c t$.

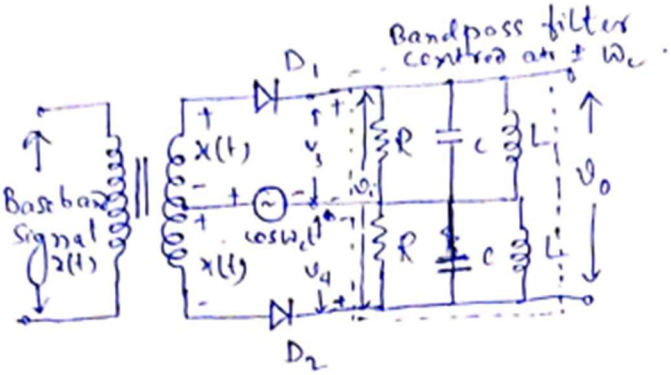


Fig:- Balanced Modulator using Diode.

Mathematical Expression:-

A non-linear V-i relationship is given by

$$i = av + bv^2$$

Here, we have neglected the ~~linear~~ higher power terms.

In the above expression, v is the input voltage applied across a non-linear device and i is the current O/P from the non-linear device.

From the above fig we write the two input voltages v_1 and v_2 across the two diodes as

$$v_1 = \cos\omega_c t + x_0(t)$$
$$v_2 = \cos\omega_c t - x_0(t)$$

So, expression of i_1, Substituting the value v_1 we get.

$$i_1 = a[\cos\omega_c t + x_0(t)] + b[\cos\omega_c t + x_0(t)]^2$$
$$= a\cos\omega_c t + a x_0(t) + b\cos^2\omega_c t + b x_0^2(t) + 2b x(t)\cos\omega_c t \rightarrow (1)$$

Similarly,

$$i_2 = a\cos\omega_c t - a x(t) + b\cos^2\omega_c t + b x^2(t) - 2b x(t)\cos\omega_c t \rightarrow (2)$$

Due to currents i_1 and i_2, the net voltage v_i at the input to bandpass filter is expressed as

$$v_i = v_3 - v_4$$

But from fig. $v_3 = i_1 R$

$$v_4 = i_2 R$$

Therefore $v_i = i_1 R - i_2 R$

$$= R(i_1 - i_2)$$

$$= R[2aX(t) + 4bX(t)\cos\omega_c t]$$

This voltage v_i is the input to the bandpass filter centered around $\pm\omega_c$.

A bandpass filter is that type of filter which allows to pass a band of frequencies. Here since the bandpass filter is centred around $\pm\omega_c$, it will pass a narrow band of frequencies centred at $\pm\omega_c$ with small bandwidth of $2\omega_m$ to pass the sideband.

Therefore, the o/p of BPF centred around $\pm\omega_c$ is given by

$$v_o = 4bRX(t)\cos\omega_c t = KX(t)\cos\omega_c t$$

which is the expression for DSB-SC signal.

20. HOW IS DSB-SC DEMODULATED:

ANS:

Demodulation of DSB-SC signals:-

The DSB-SC signal may be demodulated by Synchronous detection method.

Block Diagram:- Fig below shows the block diagram of synchronous detection method.

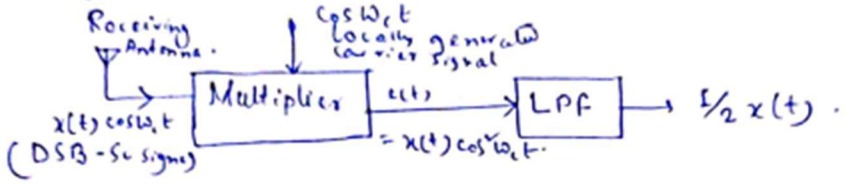


Fig:- Synchronous Detection Method

Working Principle:-

In synchronous detection method, the received modulated or DSB-SC signal is first multiplied by a locally generated carrier signal $\cos \omega_c t$ and then passed through a low-pass filter (LPF). At the o/p of a low-pass filter (LPF), the original modulating signal is recovered.

Mathematically,

$$e(t) = \underbrace{x(t)\cos\omega_c t}_{\text{DSB-SC signal}} \cdot \underbrace{\cos\omega_c t}_{\text{locally generated carrier signal}}.$$

or, $e(t) = x(t)\cos^2\omega_c t = \frac{1}{2}x(t)\left[2\cos^2\omega_c t\right]$.

$$= \frac{1}{2}x(t)\left[1 + \cos 2\omega_c t\right]$$

$$= \frac{1}{2}x(t) + \frac{1}{2}x(t)\cos 2\omega_c t$$

Now, it may be observed that when multiplied signal $e(t)$ is passed through a low-pass filter then the term $\frac{1}{2}x(t)\cos 2\omega_c t$, centred at $\pm 2\omega_c$ is suppressed by LPF and thus at the o/p of LPF, the original modulating signal $\frac{1}{2}x(t)$ is obtained.

Frequency Spectrum:-

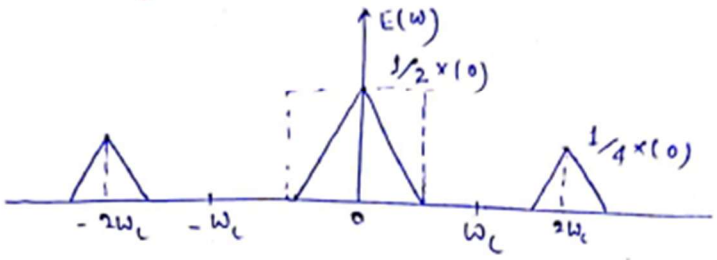
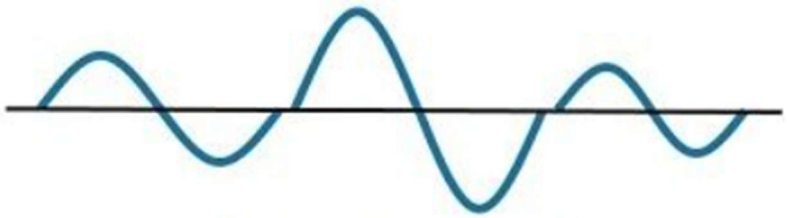


Fig:- Frequency Spectrum of $x(t)\cos\omega_c t$

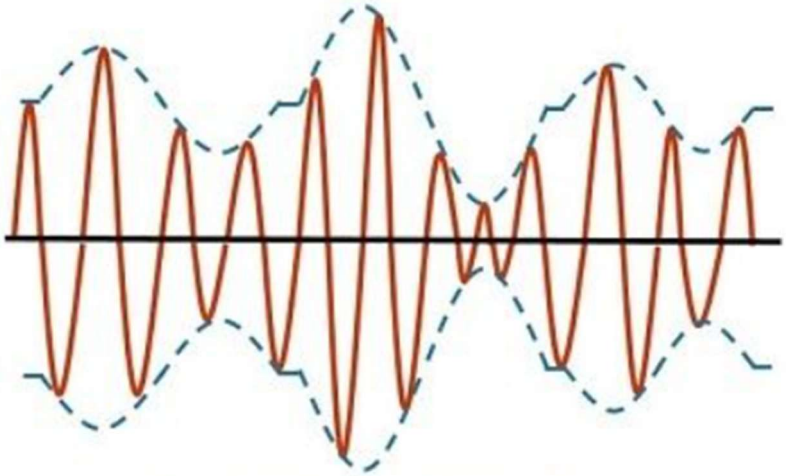
21. ELIMINATED

22. DRAW THE WAVEFORM OF DSB-SC SIGNAL:

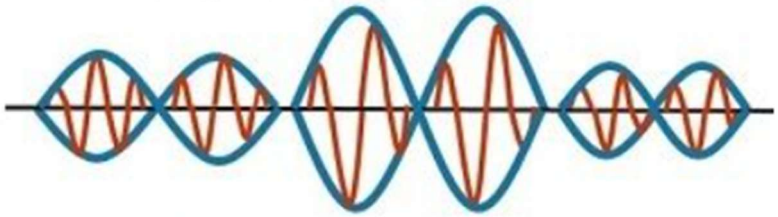
ANS:



Modulating signal



Amplitude modulated wave



Supressed carrier wave

23. **Eliminated**
24. **CALCULATE THE POWER PERCENTAGE POWER SAVING FOR DSB-SC SIGNAL FOR THE PERCENTAGE MODULATION OF A) 100% B) 50%**

    ANS:

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

Solution :-

The total power in AM wave

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

At 100% depth of modulation $m = 1$

$$P_t = 1.5\, P_c.$$

$$\therefore \%\ \text{power saving} = \frac{P_c}{1.5\, P_c} = 66.66\%.$$

At 50% depth of modulation $m = 0.5$

$$P_t = 1.125\, P_c.$$

$$\therefore \%\ \text{power saving} = \frac{P_c}{1.125\, P_c} = 88.88\%.$$

25. WHAT IS THE TRANSMISSION BANDWIDTH OF DSB-SC SIGNAL

ANS:

Transmission Bandwidth

The transmission bandwidth of DSB-FC with with a single-tone modulation is $2f_m$ where f_m is the frequency of modulating signal. Thus the transmission bandwidth of DSB-SC is same as that of standard AM wave.

26. **ELIMINATED**
27. **“**
28. **“**
29. **“**
30. **“**
31. **DEFINE ANGLE MODULATION :**

ANS:

Angle Modulation is a type of analog modulation in which the *angle* (i.e., phase or frequency) of a carrier wave is varied in accordance with the instantaneous amplitude of the message signal, while the *amplitude of the carrier remains constant*.

There are two main types of angle modulation:

1. **Frequency Modulation (FM)** – The frequency of the carrier wave is varied according to the message signal.
2. **Phase Modulation (PM)** – The phase of the carrier wave is varied according to the message signal.

Formula (General Angle Modulated Signal):

$s(t)=A\cos[\omega ct+\phi(t)]$

Where:

- Ac is the carrier amplitude (constant),
- ωc is the carrier angular frequency,

ϕ(t) is the angle (phase) variation caused by the message signal.

Key Features:

- Better noise immunity compared to amplitude modulation.
- Used in FM radio, television audio transmission, and two-way radio.

32. **WHAT ARE THE TYPES OF ANGLE MODULATION:**

ANS:

There are mainly two types:

1. Frequency Modulation (FM):

In FM, the **frequency** of the carrier signal is varied in proportion to the instantaneous amplitude of the message signal.

- **Modulated signal:**

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t)\,dt\right)$$

- f_c: Carrier frequency
- k_f: Frequency sensitivity
- $m(t)$: Message signal

✅ **Applications:** FM radio, audio broadcasting, TV sound transmission, 2-way radio.

2. Phase Modulation (PM):

In PM, the **phase** of the carrier signal is varied in proportion to the instantaneous amplitude of the message signal.

- **Modulated signal:**

$$s(t) = A_c \cos\left(2\pi f_c t + k_p m(t)\right)$$

- k_p: Phase sensitivity

✅ **Applications:** Used in digital communication schemes like QPSK, some radar systems.

33. WHAT IS FREQUENCY DEVIATION

ANS:

📌 **Definition:**

Frequency Deviation (Δf) is the **maximum difference** between the **instantaneous frequency** of the modulated signal and the **carrier frequency**.

🔸 Formula:

$$\Delta f = k_f \cdot |m(t)|_{max}$$

Where:

- Δf = Frequency deviation (in Hz)
- k_f = Frequency sensitivity (Hz per volt)
- $|m(t)|_{max}$ = Maximum amplitude of the message signal

🔸 Example:

If a 1 V message signal causes a ±75 kHz shift in carrier frequency, then:

$$\Delta f = 75\,\text{kHz}$$

🔸 Importance:

- It determines the **bandwidth** of the FM signal (via **Carson's Rule**):

$$\text{Bandwidth} \approx 2(\Delta f + f_m)$$

where f_m is the maximum frequency of the message signal.

34. WHAT DO YOU MEAN BY CARRIER SWING:

ANS:

📌 **Definition:**

> **Carrier Swing** is **twice the frequency deviation** — it represents the full range of frequency change from the lowest to the highest frequency the carrier reaches during modulation.

🔸 **Formula:**

$$\text{Carrier Swing} = 2 \times \Delta f$$

Where:

- Δf = Frequency deviation (maximum shift from center frequency)

🔸 **Example:**

If the frequency deviation is ±75 kHz, then:

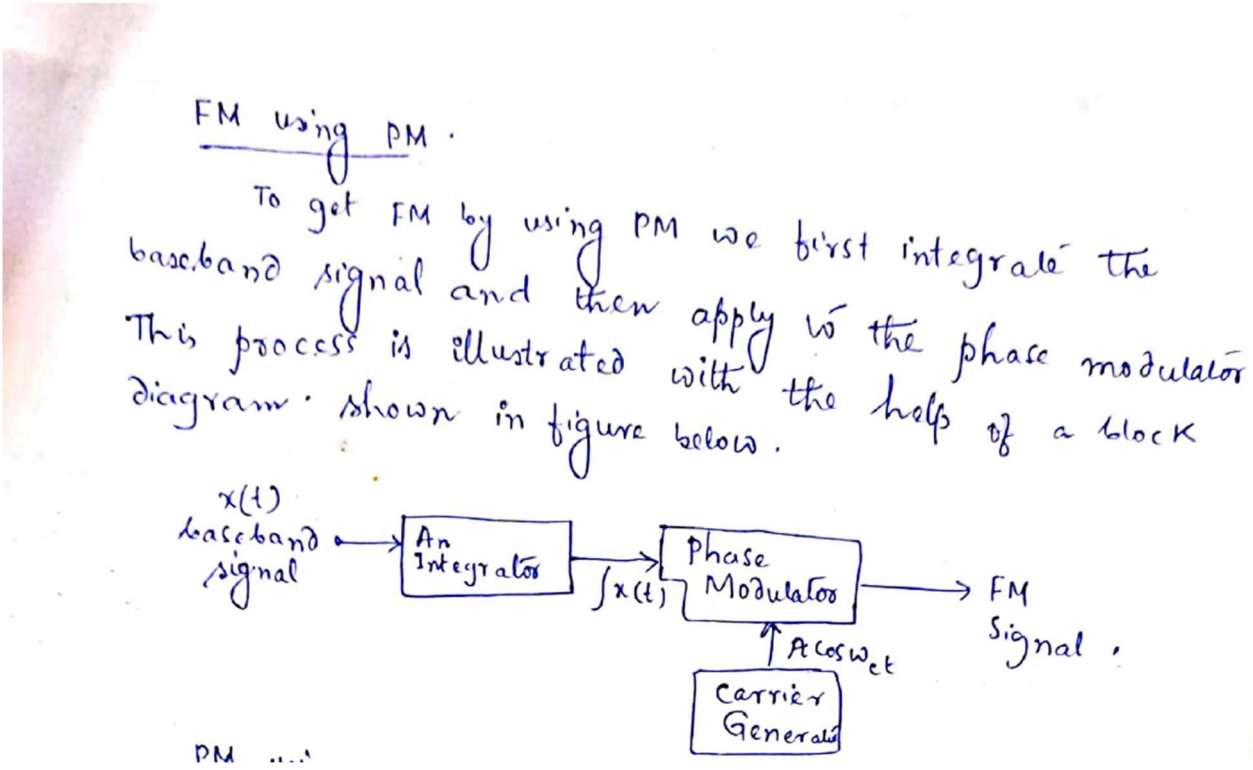
$$\text{Carrier Swing} = 2 \times 75\,\text{kHz} = 150\,\text{kHz}$$

🔸 **Significance:**

- It tells you the **total spread** of the FM signal in the frequency domain due to modulation.
- Important for **spectrum allocation** and avoiding interference.

35. HOW CAN YOU GET A FREQUENCY MODULATED SIGNAL USING A PHASE MODULATOR :

ANS:

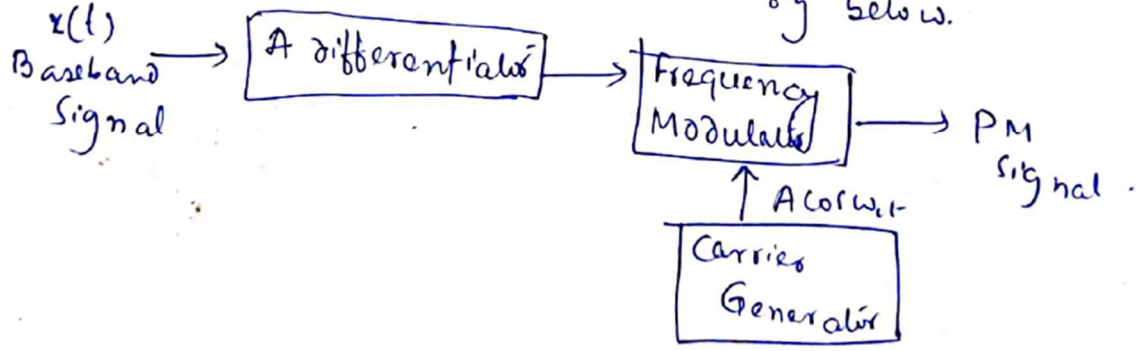


36. HOW CAN WE GET A PHASE MOUDLATION USING A FREQUENCY MODULATOR:

ANS:

PM using FM:-

Similarly, PM wave may be generated by using frequency modulator by first differentiating modulating signal $x(t)$ and then applying to the frequency modulator. This process is shown in fig below.



36. DONE ^

37. WHAT ARE THE TYPES OF FREQUENCY MODULATED SIGNAL:

ANS:

✅ Types of Frequency Modulated (FM) Signals:

There are mainly **two types** of FM signals based on the modulation index (ratio of frequency deviation to modulating frequency):

---

1. Narrowband FM (NBFM)

- **Definition:** When the **modulation index** $\beta < 1$.
- **Frequency deviation** is small.
- **Bandwidth** is limited (approximately $2f_m$, where f_m is max modulating frequency).
- **Used in:** Two-way radio (e.g., police radios, walkie-talkies), communication systems where bandwidth is limited.

✔ Simpler to implement, but lower noise immunity.

---

2. Wideband FM (WBFM)

- **Definition:** When the **modulation index** $\beta > 1$.
- **Frequency deviation** is large.
- **Bandwidth** is much higher (approximately $2(\Delta f + f_m)$ per Carson's Rule).
- **Used in:** Commercial FM radio broadcasting (e.g., 88–108 MHz band), TV audio.

✔ Better sound quality and noise resistance, but re[illegible]es more bandwidth.

38. DERIVE AN EXPRESSION FOR NARROWBAND TRANSMISSION SIGNAL

ANS:

Narrowband FM

We know that the general expression for FM wave in the phasor form is given by

$$c_{FM} = A e^{j(\omega_c t + K_f \cdot \int y(t)]}$$

However, for a narrowband FM, we have $K_f \cdot y(t) << 1$ for all the values of t.

Therefore

$$c_{FM} = A e^{j\omega_c t} \cdot e^{j \cdot K_f \cdot y(t)}$$

$$= A \cdot e^{j\omega_c t} \cdot e^{jK_f \cdot y(t)}$$

$$= A[\cos \omega_c t + j \sin \omega_c t][\cos(K_f \cdot y(t)) + j \sin(K_f \cdot y(t)]$$

So, for narrowband FM the above expression can be written as

$$c_{FM} = A[\cos \omega_c t + j \sin \omega_c t][1 + j K_f \cdot y(t)]$$

$$= A \cos \omega_c t + jA \cos \omega_c t \cdot K_f \cdot y(t) + j \cdot A \sin \omega_c t + j^2 \cdot A K_f \cdot y(t) \sin \omega_c t$$

$$= [A \cos \omega_c t - A K_f y(t) \sin \omega_c t] + j[A \sin \omega_c t + A \cos \omega_c t \cdot K_f \cdot y(t)]$$

39. DISCUSS THE GENERATION OF NARROBAND FM SIGNAL

ANS:

Generation of the narrowband FM

We Know that the expressions for narrowband FM and narrowband PM are given as by

$$S(t)_{FM} = A\cos\omega_c t - AK_f y(t)\sin\omega_c t \rightarrow (1)$$

and $$S(t)_{PM} = A\cos\omega_c t - AK_p x(t)\sin\omega_c t \rightarrow (2)$$

The above two equations provide an idea to generate narrowband FM and narrowband PM respectively. This means that the sideband terms are generated by a balanced modulator as in a DSB-SC System and then the carrier term is added to the sideband terms. Based upon the above equations we can show the method of generation of narrowband FM and narrowband PM. It may be observed that the block diagram shown in Fig satisfy the corresponding equations for narrowband FM and narrowband PM.

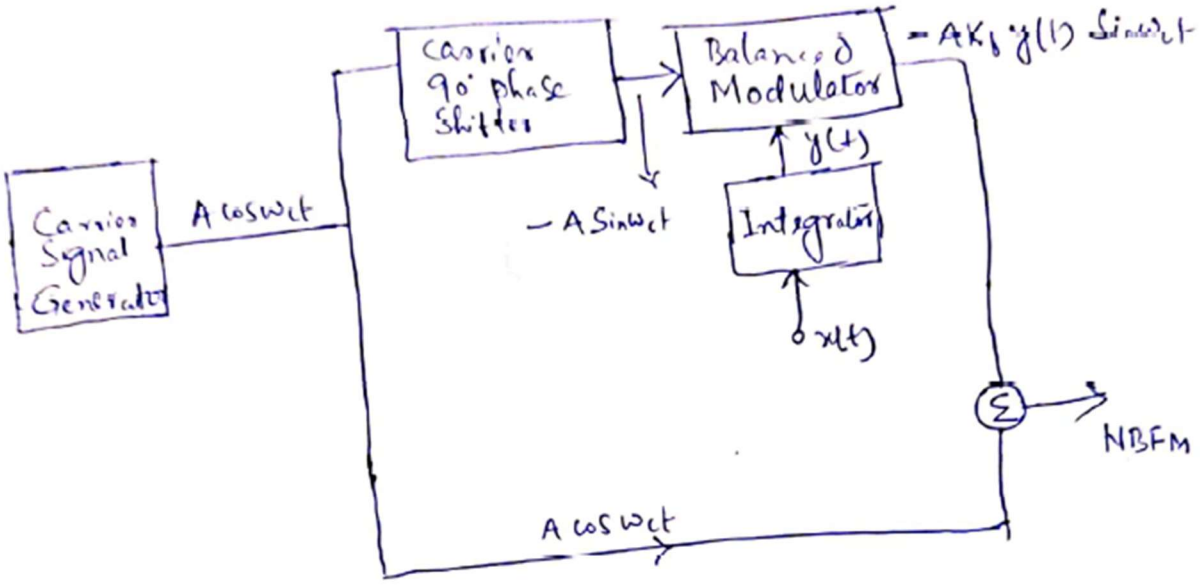


Fig: Narrowband FM Generator

40. STATE CARSON'S RULE

ANS:

Carson's rule or Practical bandwidth

It provides a thumb formula to calculate the BW of single-tone WBFM. According to this rule FM BW is given as twice the sum of the freq deviation and highest modulating freq. However it must be remembered that the rule is just an approximation.

Mathematically

$$BW = 2(\Delta\omega + \omega_m)$$

$$= 2\omega_m\left(1 + \frac{\Delta\omega}{\omega_m}\right)$$

$$= 2\omega_m(1 + m_f)$$

Regarding the carson's rule, we can consider two special cases

(i) $BW = 2(1 + m_f)\omega_m$.

If $\Delta\omega << \omega_m$ i.e., $m_f << 1$ as is the case for NBFM then

$$BW = 2\omega_m.$$

(ii) If $\Delta\omega >> \omega_m$ i.e., $m_f >> 1$ as is the case for WBFM, then

$$BW = 2(m_f)\omega_m.$$

$$= 2\Delta\omega$$

41. EXPLAIN THE FM SIGNAL GENERATION USING VARACTOR DIODE METHOD:

Varactor Diode method for FM generation

The varactor diode is a semiconductor diode whose junction capacitance changes with dc bias voltage. This varactor diode is connected in shunt with the tuned circuit of the carrier oscillator. This arrangement is shown in Fig. below,

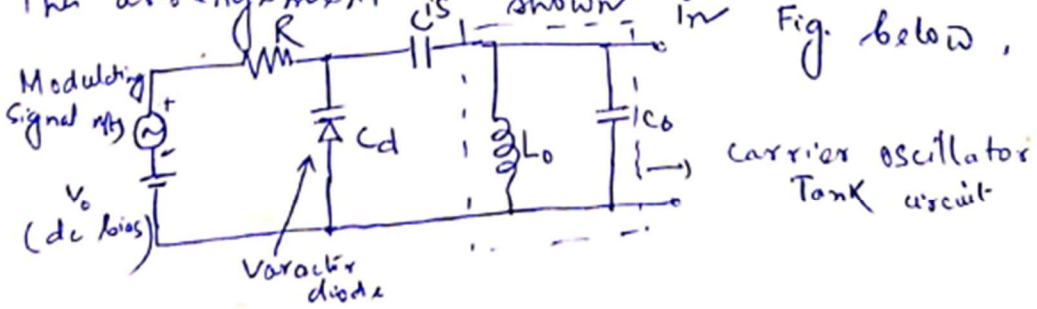


Fig:- Varactor diode method of FM generation

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C is smaller. So voltage from oscillator is across this voltage is small as compared to reverse bias dc.

In varactor diode FM generation arrangement the value of C is made ~~very large~~ much smaller at the operating frequency so that its reactance is very low. As a result, when C is connected in series with the lower capacitance of Varactor diode, the effect is as if diode is connected directly across the tuned circuit. Then the total effective circuit capacitance is the capacitance C_d of the varactor diode in parallel with C_o. This fixes the center carrier frequency

Mathematical Analysis:-

Mathematical Analysis:-

The capacitor C_d of the varactor diode is expressed as

$$C_d = \frac{K}{\sqrt{V_d}} = K V_d^{-1/2}$$

Here V_d is the total instantaneous voltage across the varactor diode and is given by

$$v_D = V_0 + x(t)$$

Also, K is a constant of proportionality.

The oscillator frequency is given as

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

Now, the total capacitance of the oscillator tank circuit will be $C_0 + C_d$ and thus the instantaneous frequency of oscillation ω_i is expressed as

$$\omega_i = \frac{1}{\sqrt{L_0 (C_0 + C_d)}}$$

In the above equation, substituting the value of C_d from we have

$$\omega_i = \frac{1}{\sqrt{L_0 (C_0 + K v_D^{-1/2})}}$$

We conclude that the instantaneous frequency ω_i of FM signal depends upon v_D which in turn depends upon the value of modulating signal $x(t)$. Thus, the instantaneous oscillator frequency ω_i also depends upon the baseband or modulating signal and hence frequency modulated wave is generated.

42. (B) EXPLAIN AMSTRONG METHOD OF FM GENERATION:

ANS:

☞ Indirect Method (Armstrong Method) of FM Generation:-

In the direct method of generation of FM, LC oscillators are to be used. The crystal oscillators can not be used. The LC oscillators are not stable enough for the communication or broadcast purpose. Thus the direct methods can not be used for the broadcast applications. FM can however be indirectly generated using phase modulation where a crystal oscillator is used for stability. This is known as Armstrong method of FM generation. The block diagram of Armstrong method of FM generation is shown in Fig. below.

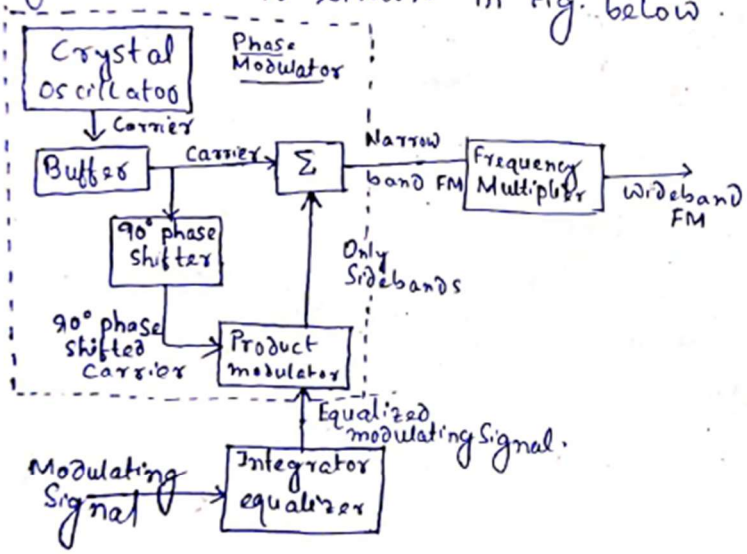


Fig:- Indirect method (Armstrong method) of FM Generation

The operation of the Armstrong method can be summarized in the form of following points.

(i) The crystall oscillator generates the carrier of low frequency typically at 1 MHz. This is applied to the combining network and a 90° phase shifter.

(ii) The modulating signal is passed through an integrater to boost the low modulating frequencies. The modulating signal is then applied to a product modulator.

(iii) The product modulator produces two sidebands such that their resultant is 90° phase shifted with respect to unmodulated carrier.

(iv) The unmodulated carrier and 90° shifted sidebands are added in the combining network.

(v) At the O/P of combining network we get FM wave. This FM wave has a low carrier frequency f_c and low value of modulation index m_f.

(vi) The carrier frequency and the modulation index are then raised by passing the FM wave through the frequency multiplier. Finally we get wideband FM wave.

43. HOW IS A FM SIGNAL DEMODULATED USING BANALNCED SLOPE DETECTOR:

ANS:

Balanced Slope Detector (Balanced Frequency Detector):-

1. Circuit Diagram:-

Figure below shows the circuit diagram of the balanced slope detector. Balanced slope detector consists of two slope detector circuits. The input transformer has a center tap secondary. Hence, the input voltages to the slope detectors are 180° outphase. There are three tuned circuits. Out of them the primary is tuned to IF i.e f_c. The upper tuned circuit of the secondary (T_1) is tuned above f_c by Δf i.e its resonant frequency is $(f_c + \Delta f)$. The lower tuned circuit of the secondary is tuned below f_c by Δf i.e at $(f_c - \Delta f)$.

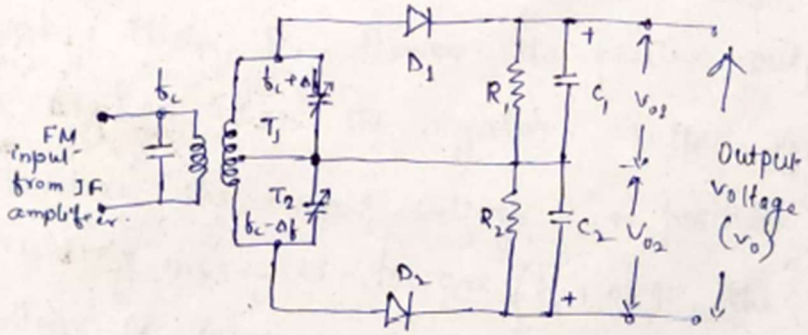


Fig:- Balanced slope detector

$R_1 C_1$ and $R_2 C_2$ are the filters used to bypass the RF ripple. V_{o1} and V_{o2} are the output voltages of the two slope detectors. The final O/P voltage V_o is obtained by taking the substraction of the individual O/P voltages V_{o1} and V_{o2} i.e

$$V_o = V_{o1} - V_{o2}$$

2. Working Operation of the circuit:-

We can understand the circuit operation by dividing the input frequency into three ranges as follows

(i) $f_{in} = f_c$: when the input frequency is instantaneously equal to f_c the induced voltage in the T_1 winding of secondary is exactly equal to that induced in the winding of T_2. Thus input voltages to both diodes D_1 and D_2 will be same. 'Therefore' their dc o/p voltages V_{O1} and V_{O2} will be identical but they have opposite polarities Hence the net o/p voltage $V_o = 0$.

(ii) $f_c < f_{in} < (f_c + \Delta f)$:- In this range of input frequency, the induced voltage in the winding T_1 is higher than that induced in T_2. 'Therefore' input to D_1 is higher than D_2. Hence, the positive output V_{O1} of D_1 is higher than the negative output V_{O2} of D_2. 'Therefore' the output voltage V_o is positive. As the input frequency increases towars $(f_c + \Delta f)$, the ~~out~~ positive o/p voltage increases as shown in Fig. below.

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

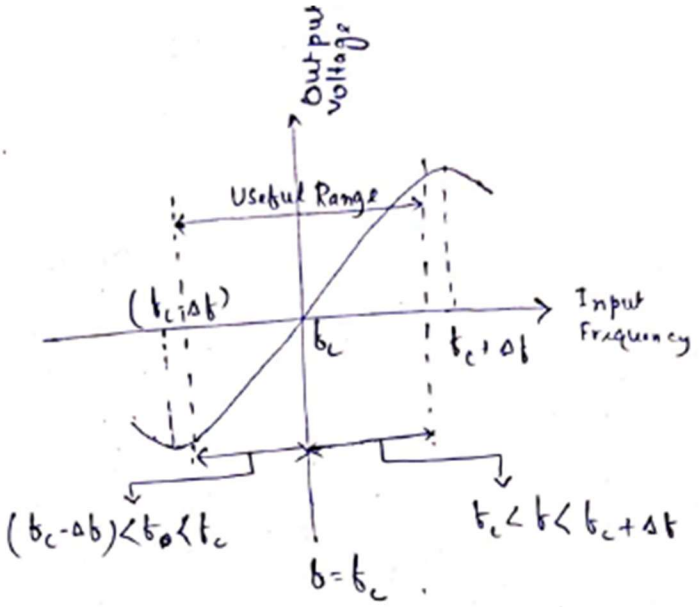


Fig:- Characteristics of the balanced Sloped detector

(iii) $f_c - \Delta f < f_{in} < f_c$:- In this range input to D_1 is less than input to D_2. Hence V_{01} is less than V_{02}. So, O/P voltage V_0 is negative.

44. ELIMINATED

45. STATE SAMPLING THEOREM:

ANS:

Statement: A continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal. i. e.

f_s≥2f_m.

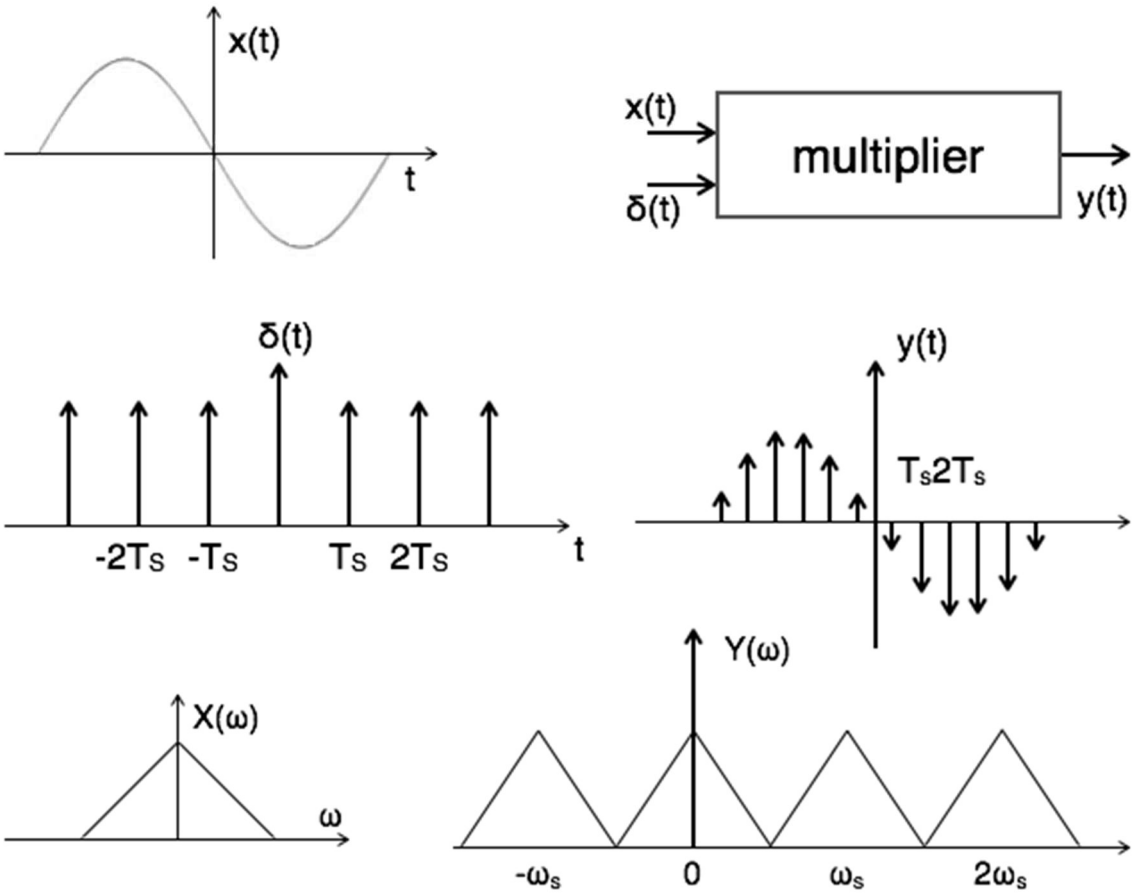
46. PROVE THAT FOR A SUCCESSFUL REPRESENTATION AND RECOVERY OF A CONTINIUOS TIME SIGNAL FROM ITS SAMPLE f_S>=2f_m ;(PRROF OF SAMPLING TH)

ANS:

Proof:

Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to f_m Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period T_s. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) in the following diagrams:



Here, you can observe that the sampled signal takes the period of impulse. The process of sampling can be explained by the following mathematical expression:

Sampled signaly(t)=x(t).δ(t).....(1)Sampled signaly(t)=x(t).δ(t).....(1)

The trigonometric Fourier series representation of δδ(t) is given by

δ(t)=a0+Σ∞n=1(ancosnωst+bnsinnωst).....(2)δ(t)=a0+Σn=1∞(ancosnωst+bnsinnωst).....(2)

Where a0=1TsʃT2−T2δ(t)dt=1Tsδ(0)=1Tsa0=1Tsʃ−T2T2δ(t)dt=1Tsδ(0)=1Ts

an=2TsʃT2−T2δ(t)cosnωsdt=2T2δ(0)cosnωs0=2Tan=2Tsʃ−T2T2δ(t)cosnωsdt=2T2δ(0)cosnωs0=2T

bn=2TsʃT2−T2δ(t)sinnωstdt=2Tsδ(0)sinnωs0=0bn=2Tsʃ−T2T2δ(t)sinnωstdt=2Tsδ(0)sinnωs0=0

Substitute above values in equation 2.

∴δ(t)=1Ts+Σ∞n=1(2Tscosnωst+0)∴δ(t)=1Ts+Σn=1∞(2Tscosnωst+0)

Substitute δ(t) in equation 1.

→y(t)=x(t).δ(t)→y(t)=x(t).δ(t)

=x(t)[1Ts+Σ∞n=1(2Tscosnωst)]=x(t)[1Ts+Σn=1∞(2Tscosnωst)]

=1Ts[x(t)+2Σ∞n=1(cosnωst)x(t)]=1Ts[x(t)+2Σn=1∞(cosnωst)x(t)]

y(t)=1Ts[x(t)+2cosωst.x(t)+2cos2ωst.x(t)+2cos3ωst.x(t).....]y(t)=1Ts[x(t)+2cosωst.x(t)+2cos2ωst.x(t)+2cos3ωst.x(t).....]

Take Fourier transform on both sides.

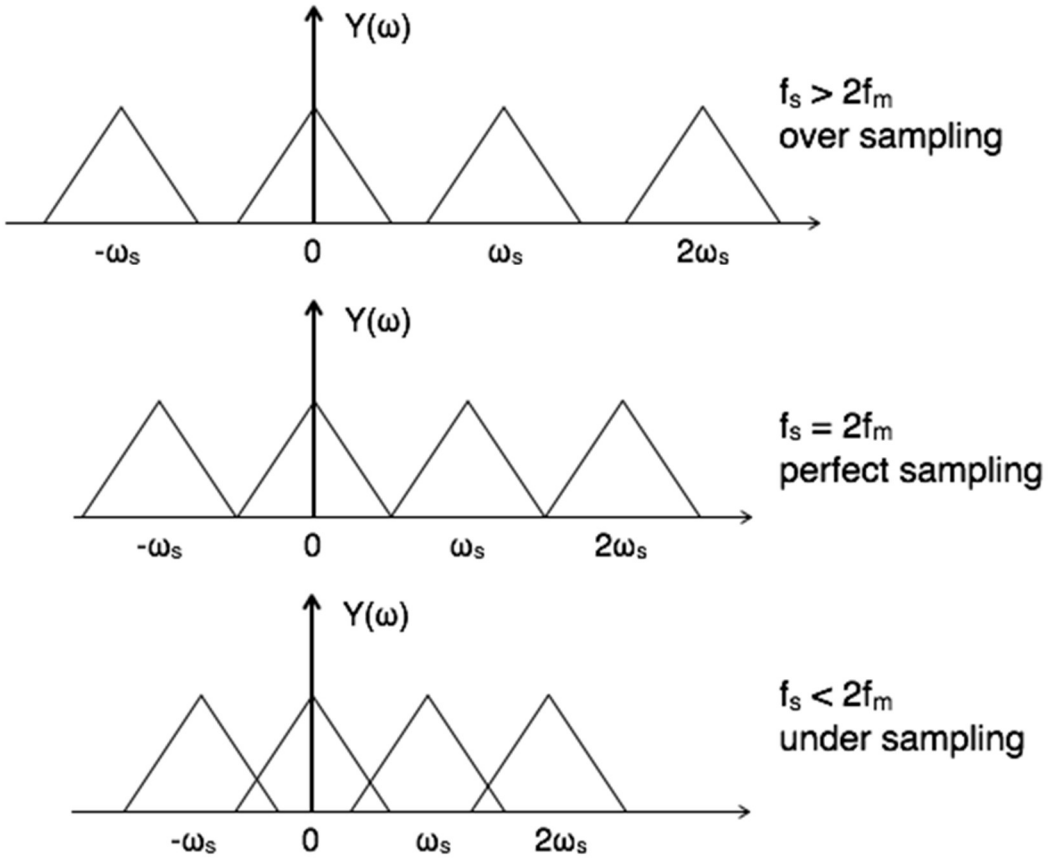
$Y(\omega)=1Ts[X(\omega)+X(\omega-\omega s)+X(\omega+\omega s)+X(\omega-2\omega s)+X(\omega+2\omega s)+...]Y(\omega)=1Ts[X(\omega)+X(\omega-\omega s)+X(\omega+\omega s)+X(\omega-2\omega s)+X(\omega+2\omega s)+...]$

$\therefore Y(\omega)=1Ts\Sigma\infty n=-\infty X(\omega-n\omega s)$

where $n=0,\pm1,\pm2,...\therefore Y(\omega)=1Ts\Sigma n=-\infty\infty X(\omega-n\omega s)$where $n=0,\pm1,\pm2,...$

To reconstruct x(t), you must recover input signal spectrum X(ω) from sampled signal spectrum Y(ω), which is possible when there is no overlapping between the cycles of Y(ω).

Possibility of sampled frequency spectrum with different conditions is given by the following diagrams:



47. WHAT IS NYQUIST RATE AND NYQUIST INTERVAL:

ANS:

Nyquist Rate

It is the minimum sampling rate at which signal can be converted into samples and can be recovered back without distortion.

Nyquist rate $f_N = 2f_m$ hz

Definition:

$$\text{Nyquist Interval} = \frac{1}{2f_{max}}$$

where f_{max} is the highest frequency present in the signal.

Explanation:

- According to the Sampling Theorem, the **minimum sampling rate** must be $f_s \geq 2f_{max}$.
- So, the **maximum sampling interval** (the time between two consecutive samples) is the reciprocal of that rate:

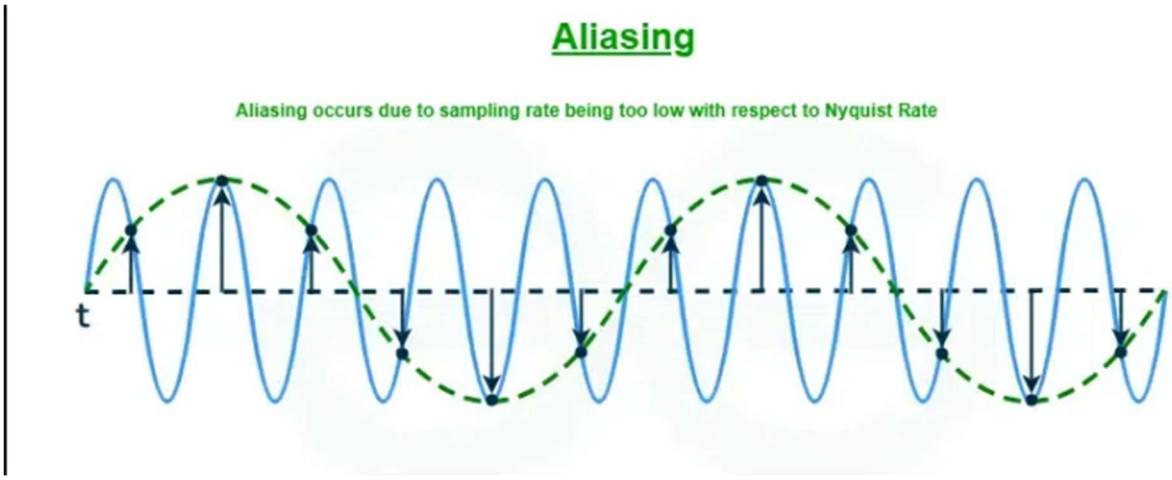
$$T_s = \frac{1}{f_s} \leq \frac{1}{2f_{max}} = \text{Nyquist interval}$$

48. **ELIMINATED**
49. **“**
50. **WHAT IS ALISING AND HOW IS IT REMOVED:**

ANS:

The aliasing effect, also known as aliasing distortion or simply aliasing, is a phenomenon that occurs in signal processing, particularly in digital signal processing (DSP), when a continuous signal is sampled at a frequency that is too low to accurately represent the original signal.

In digital communication, aliasing occurs due to a measurement error in the signal because of an incorrect sampling rate, if the sampling rate is too low aliasing may occur. In simple terms, the Nyquist rate, also known as the Nyquist sampling theorem, is the minimum sampling rate required to accurately sample and reconstruct a signal without distortion. It is defined as twice the maximum frequency present in the signal. Sampling at a rate below the Nyquist rate can lead to aliasing



Prevention of Aliasing

Aliasing should be prevented to stop distortion in any signal Although aliasing can be prevented just by correctly sampling the signal, there are mainly two methods to prevent aliasing:

Method 1: Using Anti-aliasing Filters

Anti-aliasing filters: These are special filters which can prevent or block any frequencies higher than a specific limit. It simply filters out the unwanted frequencies in any signal.

Method 2: Using Oversampling

Oversampling: Oversampling is a technique which is used to measure the signal at a much higher rate than actually needed, which pushes the Nyquist frequency higher. These higher frequencies are then removed without affecting the desired signal. It is a popular method of reducing Aliasing in digital audio and video.

51. **ELIMINATED**

52. “
53. “

54. **EXPLAIN THE PROCESS OF CONVERTING ANALOG SIGNAL INTO DIGITAL SIGNAL USING PCM SYSTEM:**

ANS:

Digital Signal: A digital signal is a signal that represents data as a sequence of discrete values; at any given time it can only take on one of a finite number of values.

Analog Signal: An analog signal is any continuous signal for which the time varying feature of the signal is a representation of some other time-varying quantity i.e., analogous to another time varying signal.

Techniques of Analog-to-Digital Conversion

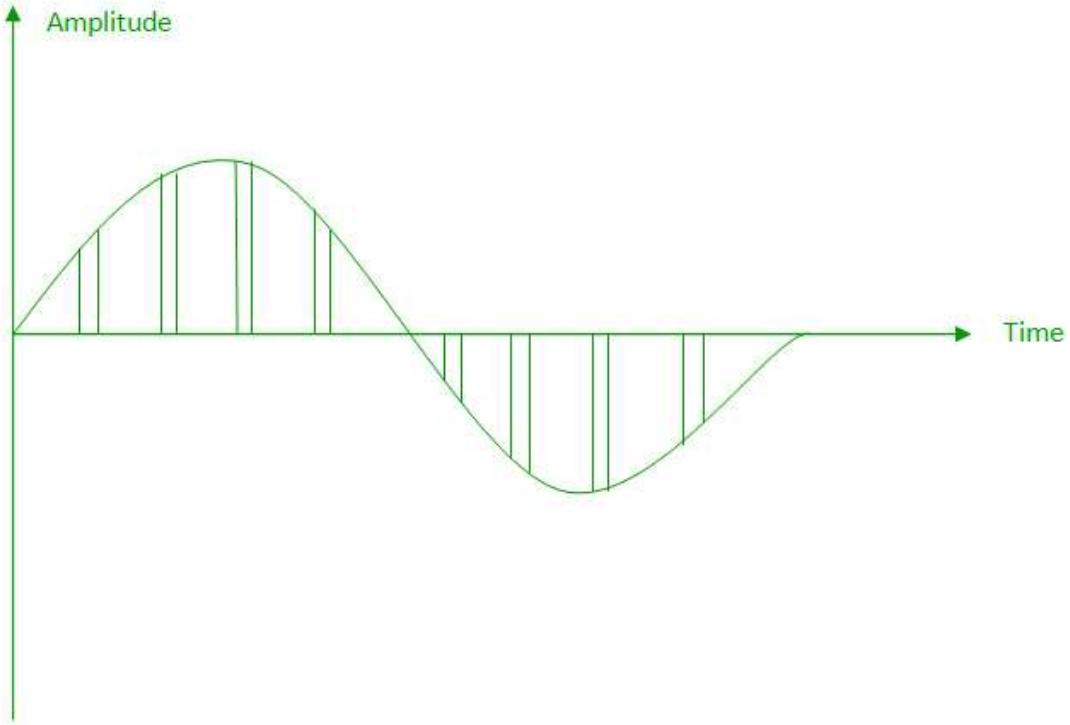
PULSE CODE MODULATION

The most common technique to change an analog signal to digital data is called pulse code modulation (PCM). A PCM encoder has the following three processes:

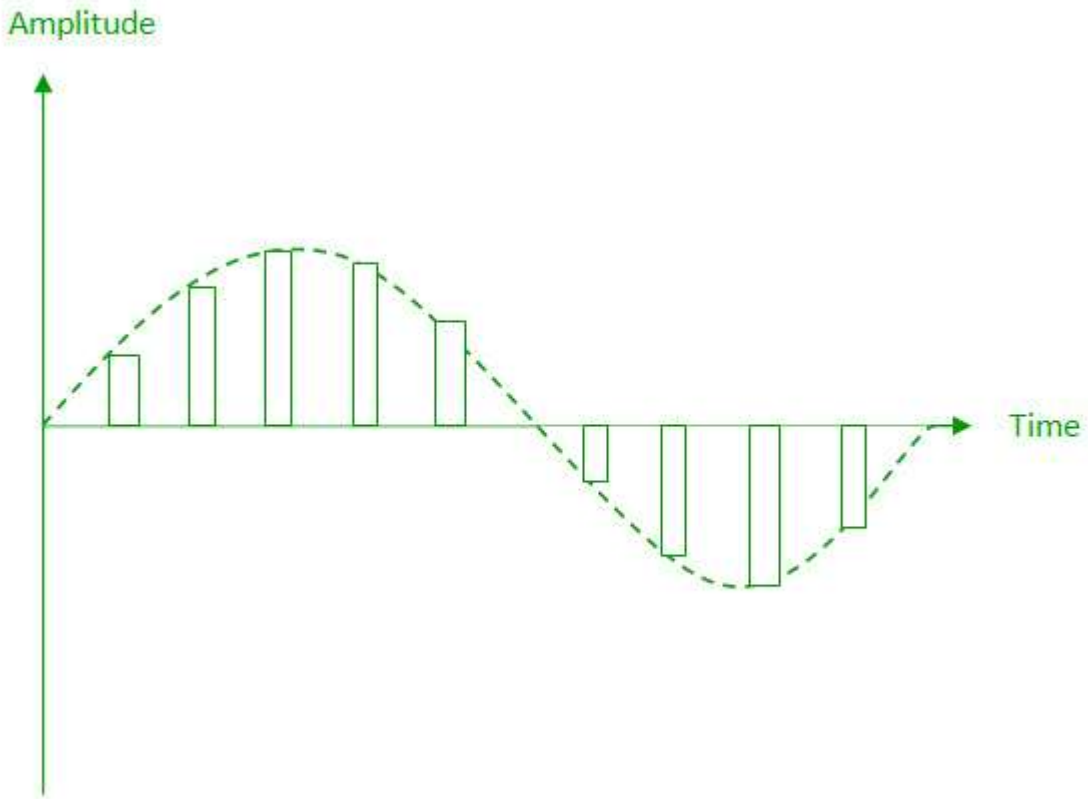
1. Sampling
2. Quantization
3. Encoding

Low pass filter : The low pass filter eliminates the high frequency components present in the input analog signal to ensure that the input signal to sampler is free from the unwanted frequency components. This is done to avoid aliasing of the message signal.

1. **Sampling –** The first step in PCM is sampling. Sampling is a process of measuring the amplitude of a continuous-time signal at discrete instants, converting the continuous signal into a discrete signal. There are three sampling methods: **(i) Ideal Sampling:** In ideal Sampling also known as Instantaneous sampling pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented. **(ii) Natural Sampling:** Natural Sampling is a practical method of sampling in which pulse have finite width equal to T.The result is a sequence of samples that retain the shape of the analog signal.



(iii) Flat top sampling: In comparison to natural sampling flat top sampling can be easily obtained. In this sampling technique, the top of the samples remains constant by using a circuit. This is the most common sampling method used.



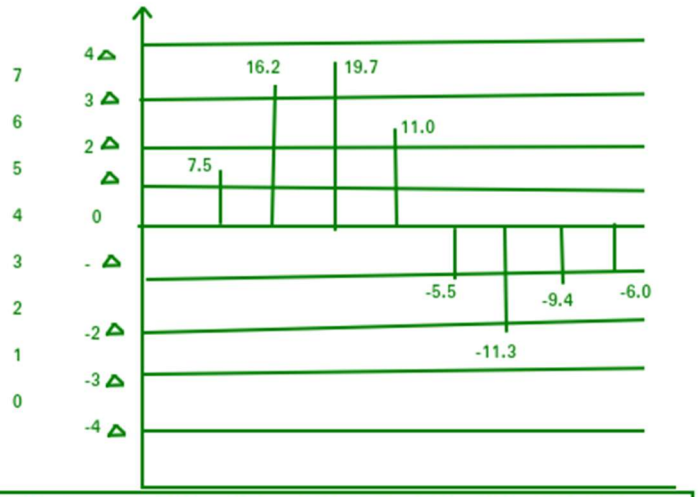
Nyquist Theorem: One important consideration is the sampling rate or frequency. According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal. It is also known as the minimum sampling rate and given by: Fs =2*fh

2. **Quantization –** The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal. The set of amplitudes can be infinite with non-integral values between two limits.

   The following are the steps in Quantization:

   1. We assume that the signal has amplitudes between Vmax and Vmin

   2. We divide it into L zones each of height d where, d= (Vmax- Vmin)/ L



Normalised PAM value :	1.50	3.24	3.94	2.20	-1.10	2.26	-1.88	-1.20
Quantized Value :	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized Error :	0	0.26	-0.44	0.30	-0.40	-0.24	0.38	-0.30
Quantization Code :	5	7	7	6	2	1	2	2
Encoded Words :	101	111	111	110	010	001	010	010

   3. The value at the top of each sample in the graph shows the actual amplitude.
   4. The normalized pulse amplitude modulation(PAM) value is calculated using the formula amplitude/d.
   5. After this we calculate the quantized value which the process selects from the middle of each zone.
   6. The Quantized error is given by the difference between quantized value and normalised PAM value.
   7. The Quantization code for each sample based on quantization levels at the left of the graph.
   8.

3. **Encoding –** The digitization of the analog signal is done by the encoder. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an n bit code. Encoding also minimizes the bandwidth used. Note that the number of bits for each sample is determined from the number of quantization levels. If the number of quantization levels is L, the number of bits is n bit = log 2 L.

55. ELIMINATED

56. WHAT IS QUANTIZATION:

ANS:

After sampling a continuous signal (which gives you a series of amplitudes at discrete time intervals), **quantization** converts each amplitude into one of a limited number of levels. This makes the signal suitable for digital storage or transmission.

57. WHAT ARE THE TYPES OF QUANTIZATION PROCESS:

ANS:

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

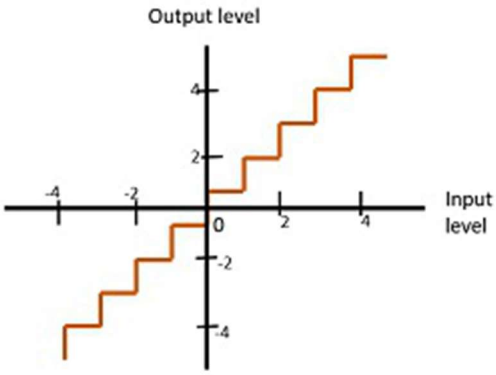


Fig 1 : Mid-Rise type Uniform Quantization

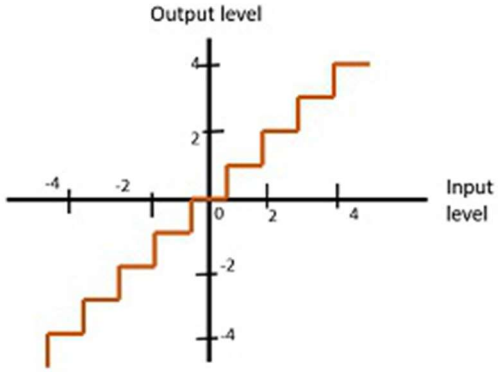


Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

58. EXPLAIN MID-TREAD TYPE QUANTIZER AND MIS-RISE TYPE QUANTIZER:

ANS:

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

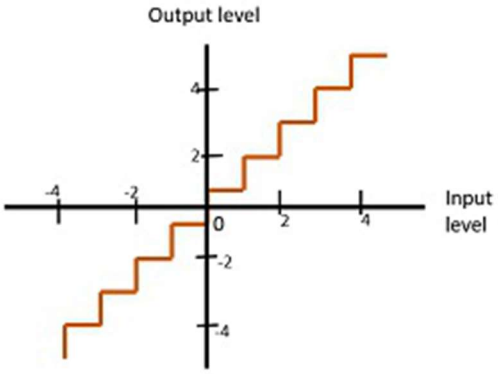


Fig 1 : Mid-Rise type Uniform Quantization

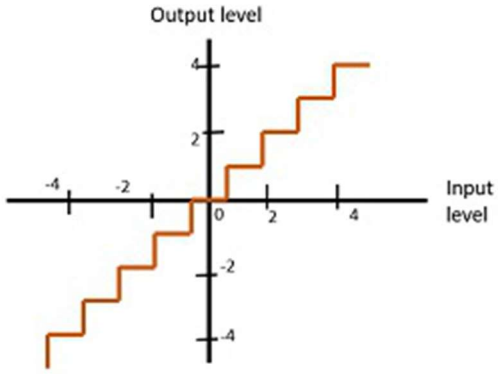


Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

59. ELIMINATED

60. DERIVE AN EXPRESSION FOR TRANSMISSION BANDWIDTH IN PCM SYSTEM:

ANS:

⇨ Transmission Bandwidth in a PCM System:-

In this section, we shall evaluate the transmission bandwidth for PCM system. Let us assume that the quantizer use 'v' number of binary digits to represent each level. Then, the number of levels that may be represented by 'v' digits will be

$$q = 2^v$$

Here 'q' represents total number of digital levels of a q-Level quantizer. For example if 'v' ~~represents~~ = 4 bits, the total number of levels will be

$$q = 2^4 = 16 \text{ levels.}$$

Each sample is converted to 'v' binary bits, i.e number of bits per sample = v.

We know that, number of samples per second = f_s

Therefore, no of bits per second is expressed as

(Number of bits / second) = (Number of bits per samples) x (Number of samples per second)

= v bits per sample x f_s samples per se...

As a matter of fact, the number of bits per second is known as signalling rate of PCM and is denoted by 'r' i.e.

Signalling rate in PCM $r = vf_s$

where $f_s \geq 2f_m$.

Also, since bandwidth needed for PCM transmissi is given by half of the signalling rate.

Therefore transmission bandwidth in PCM

$$BW \geq \frac{1}{2} r$$

But $r = vf_s$

Therefore, $BW \geq \frac{1}{2} vf_s$

Again since $f_s \geq 2f_m$.

Hence, $BW \geq vf_m$

This is required expression for bandwidth of a PCM system.

61. ELIMINATED

62. "

63. WHAT IS COMPANDING:

ANS:

Companding is a technique of achieving non-uniform quantization. It is a word formed by the combination of words **comp**ression and exp**anding**. Companding is done in order to improve SNR of weak signals.

We know if the characteristics of the quantizer is non-linear then it causes the step size to be variable despite being constant then it is known as non-uniform quantization.

64. DRAW AND EXPLAIN COMPRESSOR CHARACTERRISTICS:

ANS:

Compressor characteristics :- Fig. 11 shows the compressor characteristics. As shown in Fig, the compressor provides a higher gain to weak signals and smaller gain to the strong input signals. Thus weak signals are artificially boosted to improve the signal to quantization or noise ratio. It may noted that this compressor characteristics has been shown only for the positive input signal but we can draw it even for the negative input signal using the same principle. Infact compressor is included at the PCM transmitter.

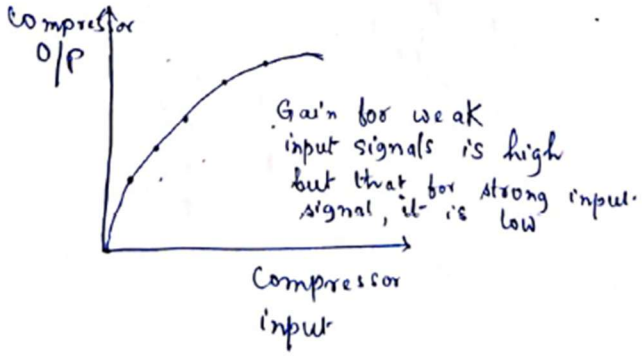


Fig. 11 :- Compressor characteristics

65. EXPLAIN u-LAW COMPNADING AND A-LAW COMPANDING:

ANS:

There are two types of Companding techniques.

A-law Companding Technique

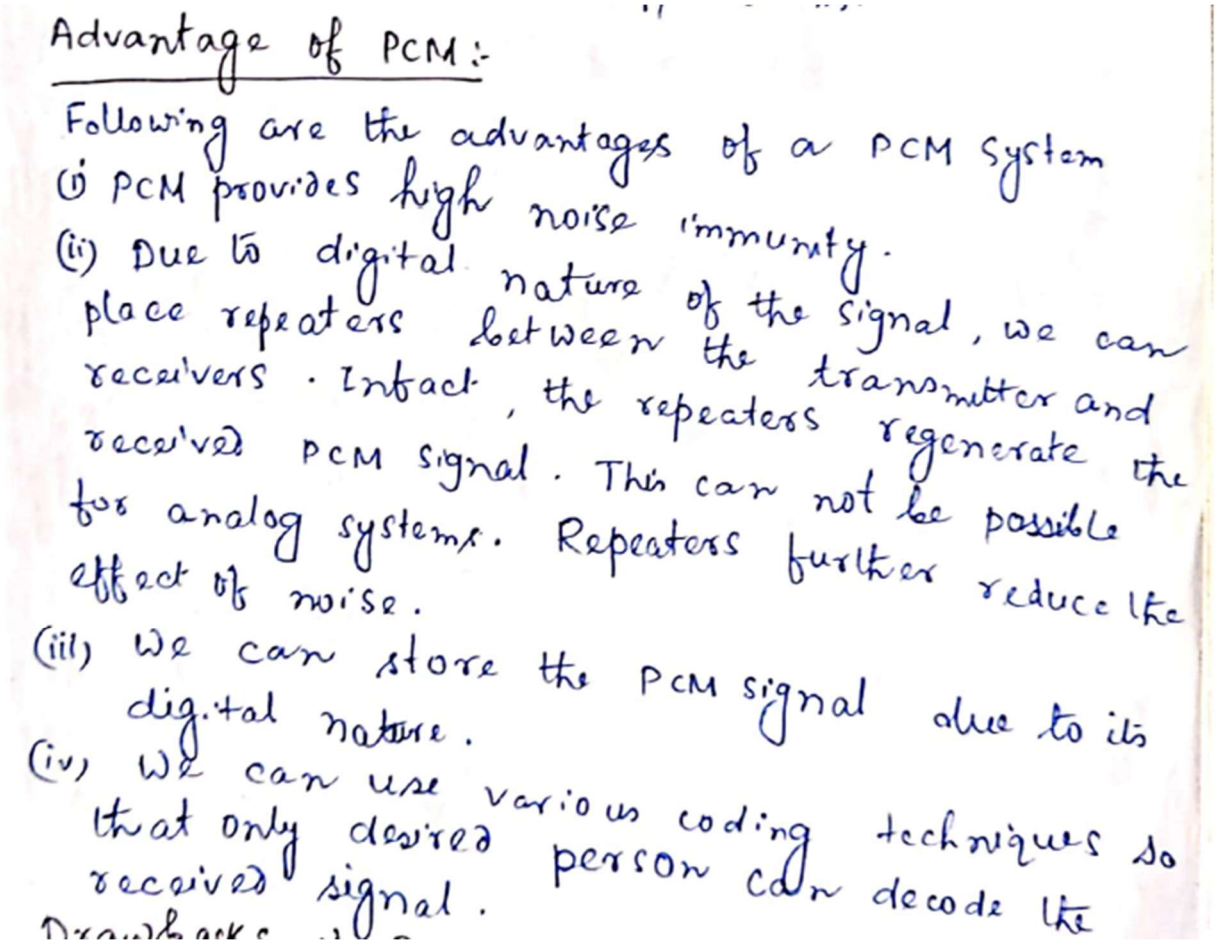
- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and there is no compression.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.
- A-law is used in many parts of the world.

μ-law Companding Technique

- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and there is no compression.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.
- μ-law is used in North America and Japan.

66. What are the advantages of pcm-system:

Ans:



Advantage of PCM :-

Following are the advantages of a PCM system

(i) PCM provides high noise immunity.

(ii) Due to digital nature of the signal, we can place repeaters between the transmitter and receivers. Infact, the repeaters regenerate the received PCM signal. This can not be possible for analog systems. Repeaters further reduce the effect of noise.

(iii) We can store the PCM signal due to its digital nature.

(iv) We can use various coding techniques so that only desired person can decode the received signal.

Drawbacks

67. EXPLAIN DELTA MODULATION IN DETAIL WITH SUITABLE BLOCK DIAGRAM:

ANS:

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of smaller value **Δ**, such a modulation is termed as **delta modulation**.

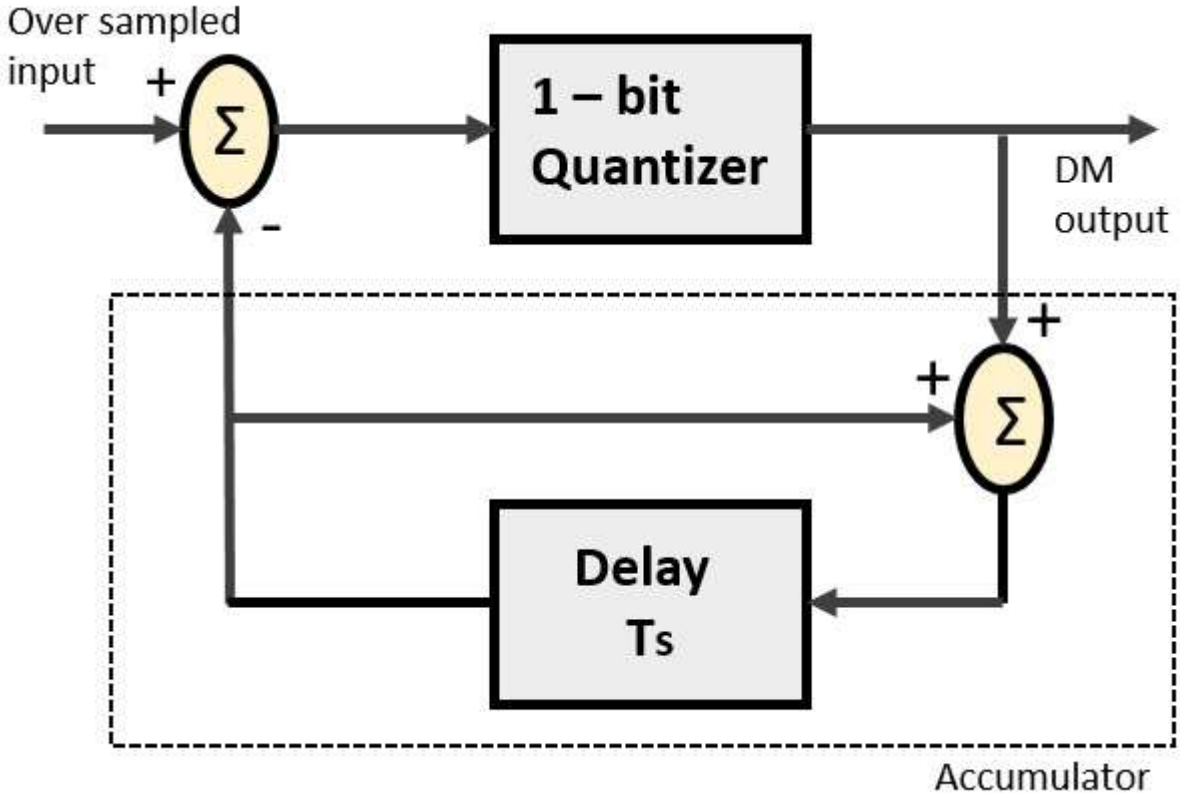
Features of Delta Modulation

- An over-sampled input is taken to make full use of a signal correlation.
- The quantization design is simple.
- The input sequence is much higher than Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., **Δ** (delta).
- The bit rate can be decided by the user.
- It requires simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

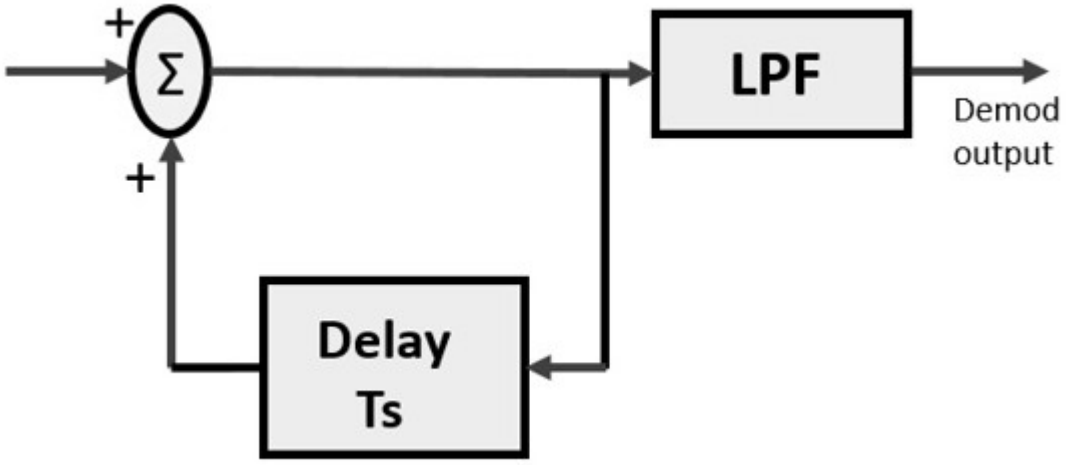
The **Delta Modulator** comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.



A stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.
Following is the block diagram for delta demodulator.



Low pass filter is used for many reasons, but the prominent one is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

68. WHAT ARE SLOPE OVERLOAD DISTORTION & GRANULAR NOISE IN DELTA MODULATION AND HOW ARAE THEY REMOVED:

ANS:

(i) Slope Overload distortion:

This distortion arises because of large dynamic range of the input signal. As can be observed from fig 19, the rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it; the step size Δ becomes too small for staircase signal $u(t)$ to follow the step segment of $x(t)$. Hence there is a large error between the staircase approximated signal and original input signal $x(t)$. This error or noise is known as slope overload distortion. To reduce the error, the step size must be increased when the slope of the signal $x(t)$ is high. Since the step size of delta modulator remains fixed, its max or minimum slope occur along straight lines. Therefore, this modulator is known as Linear Delta Modulator (LDM).

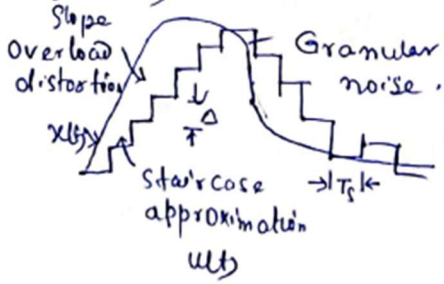


Fig. 19: Quantization errors in delta Modulator

(ii) Granular or Idle noise:-

Granular or idle noise occurs when the step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal the staircase signal is changed by large amount (Δ) because of large step size. Fig. 19 shows that when the input signal is almost flat the staircase signal ult keeps on oscillating by $\pm\Delta$ around the signal. This error between the input and approximated signal is called granular noise. The solution of this problem is to make step size small.

Therefore a large step size is required to accomodate wide dynamic range of the input signal (to reduce slope overload distortion) and small steps are required to reduce granular noise.

Bit Rate (i.e., Signaling Rate) of Delta Modulation:-

Delta modulation bit rate (r) = Number of bits transmitted / Sec

$= f_s \times 1$

$= f_s$

Therefore the delta modulation bit rate is $\frac{1}{v}$ times the bits rate of a PCM system, where v is the number of bits transmitted PCM codeword. Hence we can say that the channel bandwidth for the Delta modulation systems is reduced to a great extent as compared to that for the PCM system.

69. QUESTION AND ANS BOTH ARE BELOW:

Q. Given a sine wave of frequency f_m and amplitude A_m applied to a delta modulator having a step size Δ. Show that the slope overload distortion will occur if

$$A_m > \frac{\Delta}{2\pi f_m T_s}$$

here T_s is the sampling period.

Ans: Let us consider that the sine wave is represented as,

$$x(t) = A_m \sin(2\pi f_m t)$$

It may be noted that the slope of $x(t)$ will be maximum when derivative of $x(t)$ with respect to t will be maximum. The maximum slope of delta modulator may be given as

$$\text{Maximum slope} = \frac{\text{step size}}{\text{Sampling Period}} = \frac{\Delta}{T_s}$$

We know that, slope overload distortion will take place if slope of sine wave is greater than slope of delta modulator; ie,

$$\max\left|\frac{d}{dt}x(t)\right| > \frac{\Delta}{T_s}$$

$$\Rightarrow \max\left|\frac{d}{dt}A_m \sin(2\pi f_m t)\right| > \frac{\Delta}{T_s}$$

$$\Rightarrow \max\left|A_m 2\pi f_m \cos(2\pi f_m t)\right| > \frac{\Delta}{T_s}$$

$$\Rightarrow A_m \cdot 2\pi f_m > \frac{\Delta}{T_s}$$

$$\Rightarrow A_m > \frac{\Delta}{2\pi f_m T_s}$$

70. ELIMINATED

71. WHAT ARE THE DESIRABLE PROPERTY OF LINE CODE:

ANS:

The digital data can be transmitted by various transmission or line codes such as on-off, polar, bipolar and so on. This is called **line-coding**. *Each type of line-code has its advantages and disadvantages.

Thus, among other desirable properties, a line code must have the following properties:

1. Transmission bandwidth

For a line-code, the transmission bandwidth must be as small as possible.

2. Power efficiency

For a given bandwidth and a specified detection error probability, the transmitted power for a line code should be as small as possible.

3. Error detection and correction capability

It must be possible to detect and preferably correct detection errors. For example, in a bipolar case, a signal error will cause bipolar violation and thus can easily be detected.

4. Favourable power spectral density

It is desirable to have zero power spectral density (PSD) at $w = 0$ (*i.e.*, dc) since ac coupling and transformers are used at the repeaters. Significant power in low-frequency components causes dc wander in the pulse stream when ac coupling is used. The a.c. coupling is required since the dc paths provided by the cable pairs between the repeater sites are used to transmit the power required to operate the repeaters.

5. Adequate timing content

It must be possible to extract timing or clock information from the signal.

6. Transparency

It must be possible to transmit a digital signal correctly regardless the pattern of 1's and 0's.

72. THE BINARY DATA 101011001 IS TRANSMITTED OVER A BASEBAND CHANNEL , DRAW THE WAVEFORM FOR THE TRANSMITTED DATA USING FOLLOWING FORMAT : A) POLAR RZ B) SPLIT PHASE MANCHESTAR CODE C) AMS D) POLAR QUTARNARY NRZ SIGNALING ----------ANS WILL BE PROVIDED LATER
73. A DIGITAL DATA 10101101 IS TRANSMITTED USING DIGITAL MODULATION TECHNIQUES , DRAW THE WAVEFORM OF ASK , PSK , AND FSK SIGNAL CORRESPONDING TO THE GIVEN DIGITAL DATA.
ANS: WILL BE PROVIDED