Q1. What is Equalizer? Explain Adaptive Equalizer?

Ans. Equalizer is a signal processor device used in digital communication system to reduce distortion. It adjusts received signals for channel effect.

Types of Equalizers:

1. Fixed Equalizer.
2. Adaptive Equalizer: it is self-adjusting equalization system. It is commonly used in wireless communication, mobile network, etc.
   - Working:
   i) Uses adaptive algorithm to update its coefficients based on received signals.
   ii) Compares received signals with known reference signal.
   iii) Helps multipath effects and channel distortion.
   - Types of Adaptive Equalizer:
     1. Zero Forcing Equalizer→ Cancels ISI completely.
     2. Minimum Mean Square Error (MMSE)→ balances between ISI reduction & noise suspension.
     3. Decision Feedback Equalizer (DFE)→ uses previously detected symbols to predict & correct errors.
     4. Blind Equalizers→ estimate channels parameters based on statistical properties of received signals.
   - Advantages:
     i. Wireless Communication (Wi-Fi, 5G, 4G).
     ii. High Speed Data Transfer (Optical Fiber).
     iii. Satellite Communication.

Q2. A television signal with bandwidth of 4.2MHz is transmitted using binary PCM. The number of quantization level is 512. Calculate (i)Signalling Rate (ii)Transmission Bandwidth

Given Data:

fm = 4.2MHz = $4.2 * 10^6$; L = 512; n = $log_2 L$

Solution:

1. Calculate no. of bits per sample (n)
   n = $log_2 L$
   n = $log_2(512)$
   n = 9 bits per sample

2. Calculate Nyquist Sampling Rate (fs)

$$f_s = 2f_m$$

f_s = 2*$4.2 * 10^6$

$f_s = 8.4 * 10^6$

$f_s = 8.4MHz$

3. Calculate Signalling rate

   R = n*f_s
   R = 9*$8.4 * 10^6$
   R = 75.6 * 10^6
   R = 75.6 Mbps

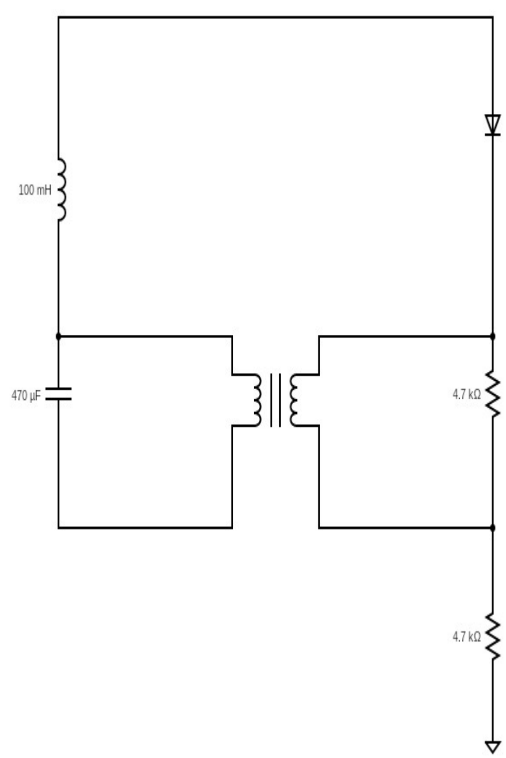
4. Calculate Transmission Bandwidth

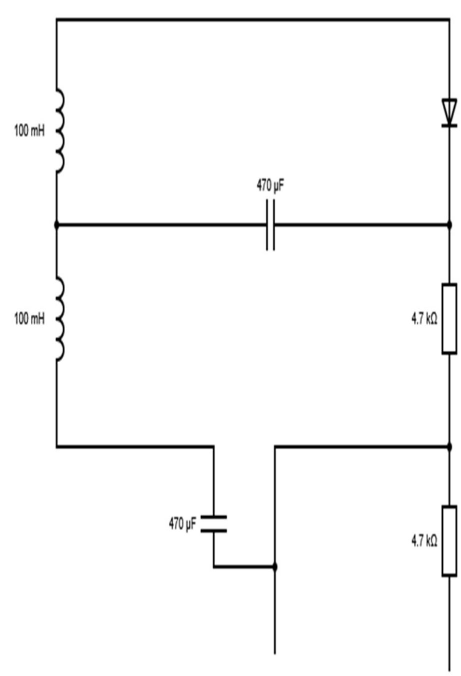
$$B_T = R\ /\ 2$$

   $B_T = 75.6/2$
   $B_T = 37.8MHz$

Q3. Explain why ratio detector is preferred over phase discriminator?

Ans.





➔ Key Reasons:

1. Better AM Noise Rejection:
   - Phase Discriminator is sensitive to both AM & FM.
   - Ratio Detector, however, automatically cancels out AM noise (like static, interference) because of its circuit design.
   - This is especially useful in real-world environment where signals are weak or noisy.
2. No need for limiter stage:
   - Phase Discriminator requires a limiter before it to remove AM noise.
   - Ratio detector doesn't need a limiter, simplifying the design and saving power.
3. Improved Stability:
   - The Ratio Detector is less sensitive to amplitude variations.
   - This leads to more stable and consistent audio output.

➔ Summary Table:

Feature	Phase Discriminator	Ratio Discriminator
AM Noise Rejection	Poor	Good
Needs Limiter?	Yes	No
Circuit Complexity	Simple	Slightly more complex
Stability	Less	More
Common Use	Rare now	Still used in some radios

Q4. Write short note on.

1. Quantization: it is the process of converting continuous range (analog signal) of values into finite range of discrete values (digital signal).
   - In this process, the amplitude of the sampled signal is mapped to the nearest value within a set of fixed levels.
   - This allows digital systems to store, process and transmit analog signals.

2. Quantization Error: it is the difference between the original analog value and its quantized digital value.
    - Occurs due to rounding off during quantization.
    - This error introduces quantization noise.
    - Affects the precision of the reconstructed signals.

Q5. Explain the terms:

1. Unipolar NRZ:
    - Uses only one priority.
    - 0V → Bin. 0
    - 1V → Bin.1
    - Signal not returns to zero between bits.
    - Cons: No sync. For long sequence of 0's.

2. Unipolar RZ:
    - Uses only one polarity.
    - Signal returns to zero after each bit period.
    - Pros: Easier sync. Due to mid-bit transition.
3. Polar NRZ:
    - Uses two polarities.
    - +V → Bin. 1
    - -V → Bin. 0
    - No returns to zero; level is constant throughout the bit.
    - Better than unipolar NRZ in power efficiency.
4. Polar RZ:
    - Signal Level: +V → Bin. 1 and -V → Bin. 0
    - But returns t zero midway through bit period
    - Mid-bit transition helps in synchronization.
    - Cons: Requires more bandwidth.
5. Split-Phase Manchester:
    - A type of Bi-phase Coding.
    - Low = 1 Low-to-High Transition.
    - High = 0 High-to-low Transition.

6. Polar Quaternary:
    - Using four voltage levels to represent 2 bits per symbol.
    - Approx. 0V-3V, -1V

      10+1V, +1V

      11V, +3V

Q6. What is Inter-Symbol Interference (ISI)? Explain Methods to eliminate it?

Ans. Inter-Symbol Interference (ISI) is a type of distortion in digital communication where one transmitted symbol overlaps with another, causing errors in signal detector.

❖ Methods to eliminate ISI:

1. Nyquist Criterion for ISI free transmission:

➔ It states that if channel has bandwidth (B), the maximum data rate without ISI (2B) symbols per second.

➔ Solution: Nyquist pulse shaping to be done.

2. Equalization:

➔ Adaptive equalizers compensate for ISI by adjusting to channel variations.

➔ Types: i) Zero-Forcing Equalizers.

ii) Maximum Mean Square Error (MMSE) Equalizer.

3. Pulse Shaping Technique:

➔ Using raised cosine filters or Gaussian Pulses.

4. Decision Feedback Equalization (DFE):

➔ Uses past detected symbols to predict.

Q7. Draw & explain block diagram of PCM Transmitter.

Ans.

PCM ➔ Pulse Code Modulation is digital modulation technique used for converting analog signal into digital signal for transmission.

→Consists of three main processes: Sampling, quantization and encoding.

Explanation:

1. Low-Pass Filter: removes high frequency components above Nyquist rate

$$f_s \geq 2f_m$$

2. Sampler: samples analog signal at fixed sampling rate (f_s). Uses natural sampling or flat-top sampling.
3. Quantizer: sampled analog values converted in discrete levels. Can be uniform or non-uniform quantization.
4. Encoder: converted into binary code words (8-bit/16-bit). Gray to Binary to minimize errors.
5. Parallel to Serial Converter: Converts parallel binary data → Serial Bitstream.
6. PCM Signal Output: the final digital PCM signal is transmitted over channel.

Q8. Describe Pre-Emphasis and De-Emphasis.

Ans.

1. Pre-Emphasis:
   i. Purpose → boosts high frequency components before transmission.
   ii. Location → applied at the transmitter.
   iii. Effect on Noise → Reduces impact of noise of high frequency.
   iv. Time Constant → 75μsec (USA) and 50μsec (Europe, India)
   v. Frequency Response → high frequency are boosted.
   vi. Whys used? → high frequency more dominant, boosting prevents distortion.
   vii. Circuit used → High pass filter
   viii. Application → FM Transmitter, TV broadcasting and communication system.

2. De-Emphasis:
    i. Purposes → Attenuates high frequency components after reception.
    ii. Location → applied at receiver.
    iii. Effect on noise → original signal restored by reducing amplified high frequency.
    iv. Time Constant → 75μsec (USA) and 50μsec (Europe, India)
    v. Frequency response → high frequency are attenuated.
    vi. Why used? → restores original frequency response & balance audio quality.
    vii. Circuit Used → Low-Pass Filter.
    viii. Application → FM receiver to recover original signal.

Q9. The maximum deviation allowed in FM broadcast system is 75KHz. If modulating signal is single tune sinusoid of 8KHz, determine the bandwidth of FM signal. What will be the bandwidth when modulating signal frequency is doubled.

Ans. Given Data:

Maximum deviation (Δf) = 75KHz

Modulating signal frequency 1 (f_{m_1}) = 8KHz

Modulating signal frequency 2 (f_{m_2}) = $2f_{m_1}$

Formula:

BW = 2(Δf+f_m) ➔ Carson's Rule for FM Bandwidth

Sol.:

1. Calculate bandwidth for $f_{m_1} = 8kHz$
   BW1 = 2(Δf+f_m)
   = 2(75+8) = 2(83)
   BW1 = 166KHz
2. Calculate new modulating frequency f_{m_2}
   fm2 = $2f_{m_1}$
   = 2*8 kHz
   fm2 = 16KHz

3. Calculate Bandwidth for fm2 = 16KHz
   BW2 = 2(Δf+fm2)
   = 2(75+16) = 2(91)
   BW2 = 182KHz

Q10. Find Nyquist rate and Nyquist interval for the following signal:

X(t) = 3cos(200Πt) + 5sin(600Πt) + 10cos(1200Πt)

Ans. Given Data: X(t) = 3cos(200Πt) + 5sin(600Πt) + 10cos(1200Πt)

Formula:

1. Nyquist rate (fs) =2fm
2. Nyquist interval (Ts) = 1/fs
3. Angular Frequency (w) = 2Πf

Sol.:

1. Find frequency of each component.
   a. For 3cos(200Πt), w1 = 200Π, f1 = 200Π/2Π = 100Hz
   b. For 5sin(600Πt), w2 = 600Π, f2 = 600Π/2Π = 300Hz
   c. For 10cos(1200Πt), w3 = 1200Π, f3 = 1200Π/2Π = 600Hz

   fmax = max (100Hz, 300Hz,600Hz) = 600Hz

2. Calculate Nyquist rate (fs = 2fm)
   fs = 2*600Hz
   fs = 1200Hz
3. Nyquist interval:
   Ts = 1/fs = 1/1200 sec = 833.33*10^-6 sec.

Q11. Draw and explain Armstong Method in FM Generation.

Ans. The Armstrong Method is an indirect method used for generating frequency modulated signal (FM).

It employs a phase modulator rather than a direct frequency modulator. This method is widely used as it provides better frequency stability.

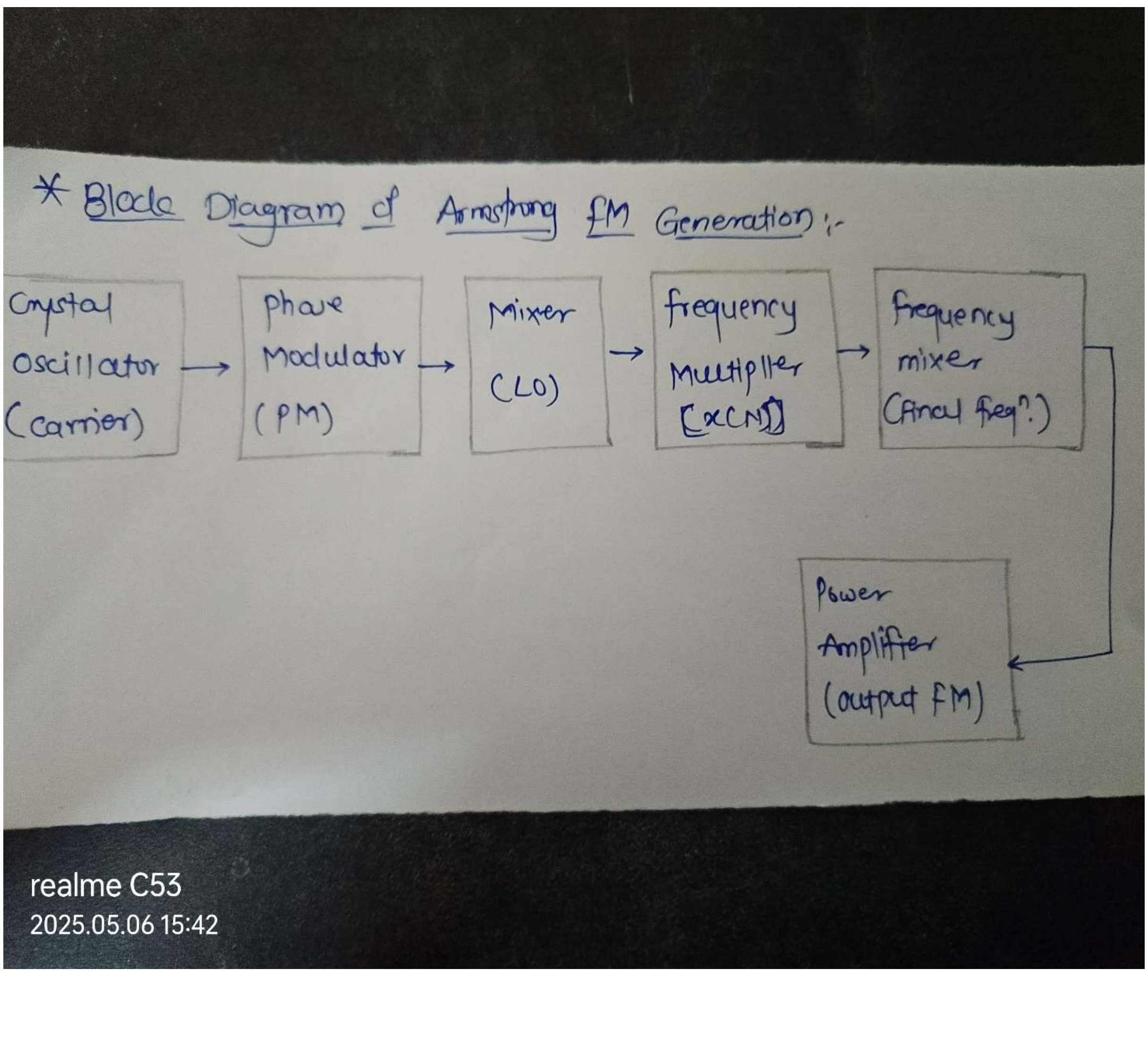
❖ Working Principle of Armstrong FM Generation:

1. Carrier Generation: A crystal oscillator generates a stable carrier signal at high frequency.
2. Phase Modulator (PM): It converts phase modulation (PM) → Frequency modulator (FM).

3. Mixing: The PM signal is mixed with another signal to shift frequency to an intermediate level.
   PM Signal + Another Signal → Shifts Frequency to Specific level.
4. Frequency Multiplication: The signal is passed through frequency multiplier. It increases the frequency deviation to desired level.
5. Frequency Translation: It shifts the signal to final transmission frequency.
6. Power Amplification: The FM signal is amplified using power amplifier before transmission. And also used to avoid signal distortion.

❖ Advantages:
   i. High stable frequency generation.
   ii. Low noise & distortion.
   iii. Better Control.

❖ Disadvantages:
   i. Complex Circuit Design.
   ii. Requires multiple Stages.



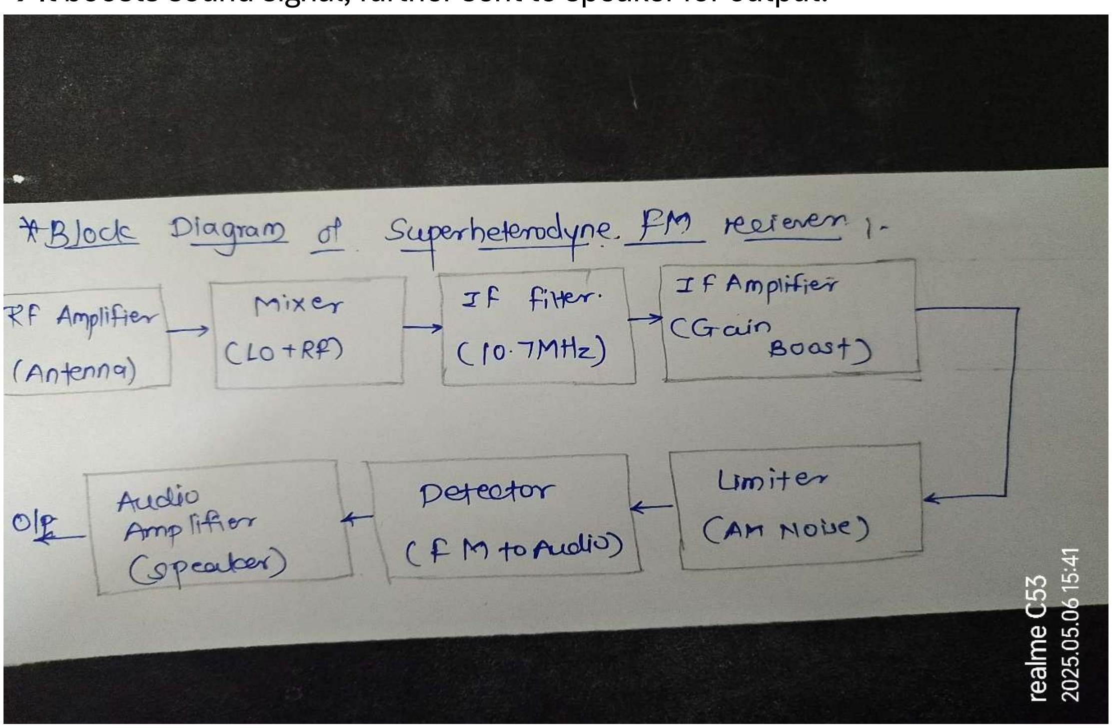
Q12. Draw and explain super-heterodyne FM Receiver.

Ans. The Super-Heterodyne FM receiver is widely used to convert high frequency signals into a lower intermediate frequency (IF) for easy processing.

❖ Working:

1. RF Amplifier:
   →Receives the FM signals.
   →It boosts weak signal while reducing noise.
2. Mixer & Local Oscillator (LO):
   →It mixes the received signal with signal of local oscillator (LO)
   →Produces intermediate frequency (IF) of 10.7 MHz

3. IF Filter & Amplifier:
   →IF filter removes unwanted frequencies, allowing only 10.7MHz signal.
   →IF amplifier boost s the signal strength.
4. Limiter:
   →It removes amplitude variations and ensures only frequency variations remained.
5. FM Detector:
   →converts frequency variations into audio signals.
6. Audio Amplifier:
   →It boosts sound signal, further sent to speaker for output.



❖ Advantages:

→Better Sensitivity.

→Improved selectivity.

→Stable Reception.

❖ Disadvantages:

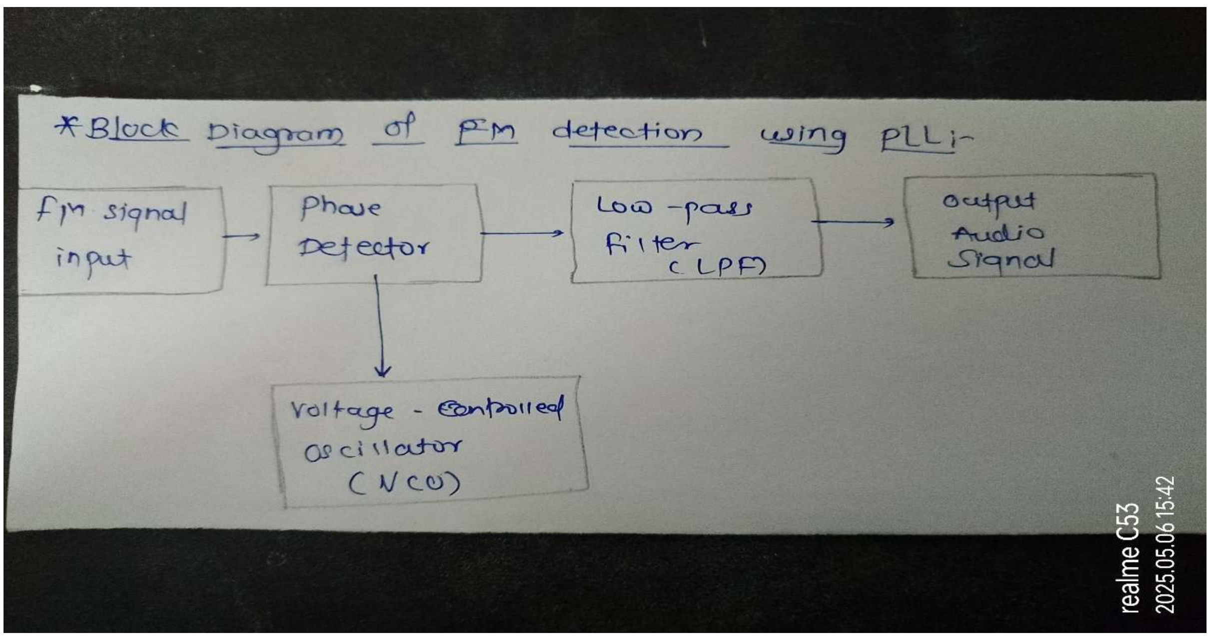
→Complex circuit design compared to simple Fm Detector.

→Requires local oscillator tuning for different frequencies.

Q13. Draw and explain FM detection using PLL.

Ans. A Phase Lock Loop (PLL) is a highly efficient method for FM detection. It converts frequency variations into corresponding amplitude variations, that represents the original message signal.

- Block Diagram:



- Working:

1. FM Input Signal:
   →Received signal is fed into the phase detector of PLL.
2. Phase Detector:
   →compares phase difference between incoming Fm signal and output of voltage-controlled oscillator (VCO).
3. Voltage-Controlled Oscillator (VCO):
   →It generates a frequency that follows the input signal's frequency variations.

4. Low-Pass Filter (LPF):
   →The error voltage is passed through on LPF to remove high-frequency noise.
   →Output is demodulated audio signal, further sent to speaker or audio amplifier.

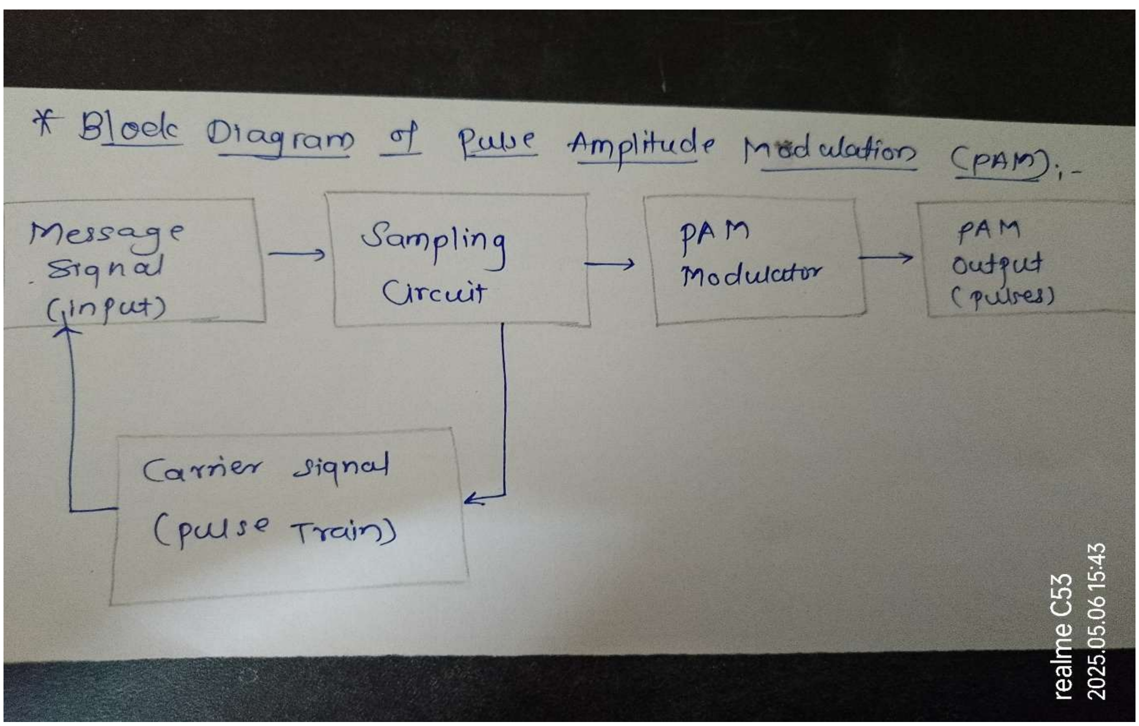
- Advantages:
  →High accuracy and stability.
  →Excellent noise rejection.
  →used in stereo FM receivers and digital communication system.
- Disadvantages:
  →More complex than simple FM.
  →Precise component tunning is required.

Q14. Draw and explain generation of Pulse Amplitude Modulation.

Ans. Pulse Amplitude Modulation is a type of modulation where amplitude varies according to instantaneous values of the modulating signal.

→PAM is commonly used in digital communication system for pulse code modulation (PCM) and Time Division Multiplexing (TDM).

❖ Block Diagram:



❖ Working:

1. Message Signal (input):
   →A continuous time analog signal is taken as modulating signal.
2. Sampling Circuit:
   →A pulse train is used to periodically sample amplitude of message signal.
   →Sampling can be natural sampling or flat-top sampling.
3. PAM Modulator:
   →sampled signal is multiplied with pulse train (carrier signal to generate PAM).
   →Amplitude of each pulse varies according to the original signal.
4. PAM Output:
   →The final output is pulses with varying amplitude.
   →It represents sampled values of original signals.

❖ Advantages:
→Simple generation and demodulation.
→used in digital signal processing.
→efficient for time-distortion multiplexing.

❖ Disadvantages:
→more susceptible to noise.
→Bandwidth requirement is high.
→Signal distortion.

❖ Applications:
→used in ethernet communication.
→intermediate step in PCM.
→Applied in LED dimming control and optical fibre communication.

Q15. Explain Sampling and its types.

Ans. Sampling is the process of converting continuous-time (CT) signal into Discrete-Time (DT) signal.

It is the most important step in pulse code modulation (PCM), Pulse Amplitude Modulation (PAM) and Digital Signal processing (DSP).

There are 3 main types of sampling:

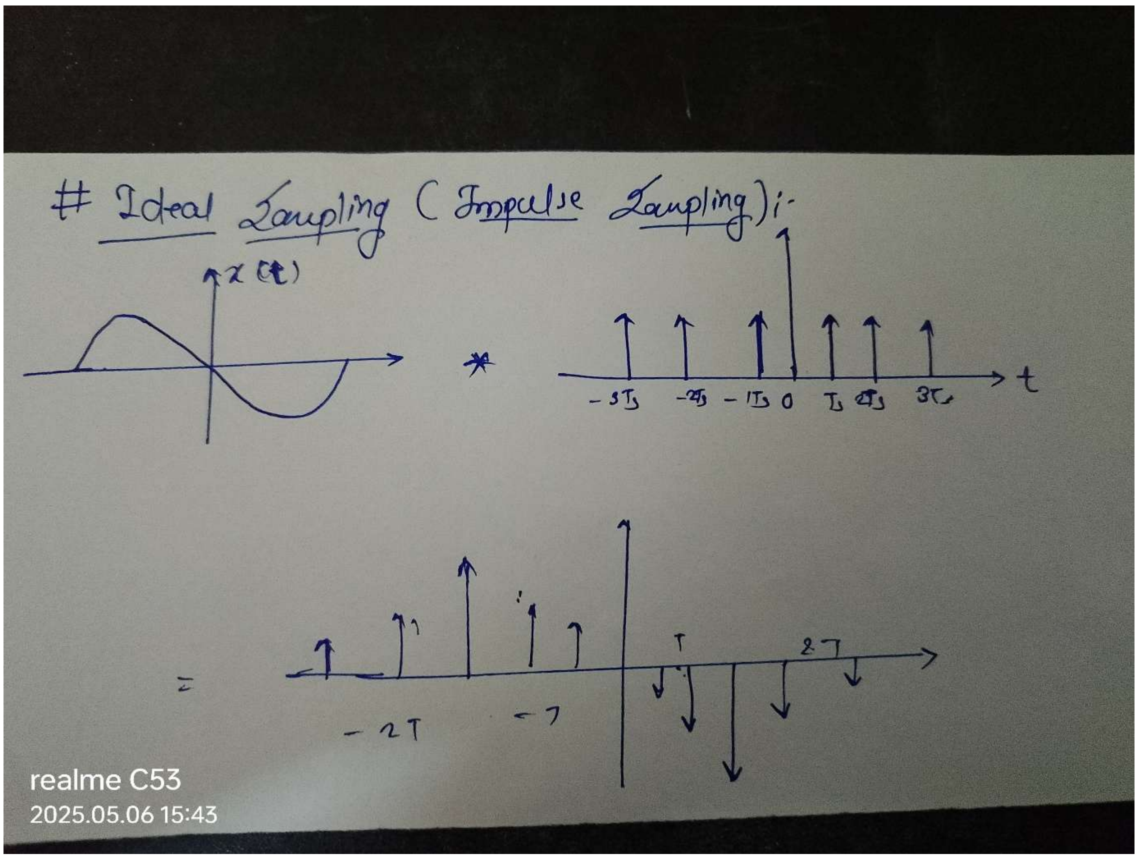
1. Ideal Sampling (Impulse Sampling):

➔ i/p signal + impulse train

➔ It can be performed by multiplying input signal x(t) with impulse train of period 'T'.

➔ Amplitude of impulse changes with amplitude of input signal x(t).

➔ The output of sampling is given by:
y(t) = x(t) * impulse train
y(t) = x(t) * $\sum_{n=-\infty}^{\infty} \quad = \delta(t - nT)$

➔ 

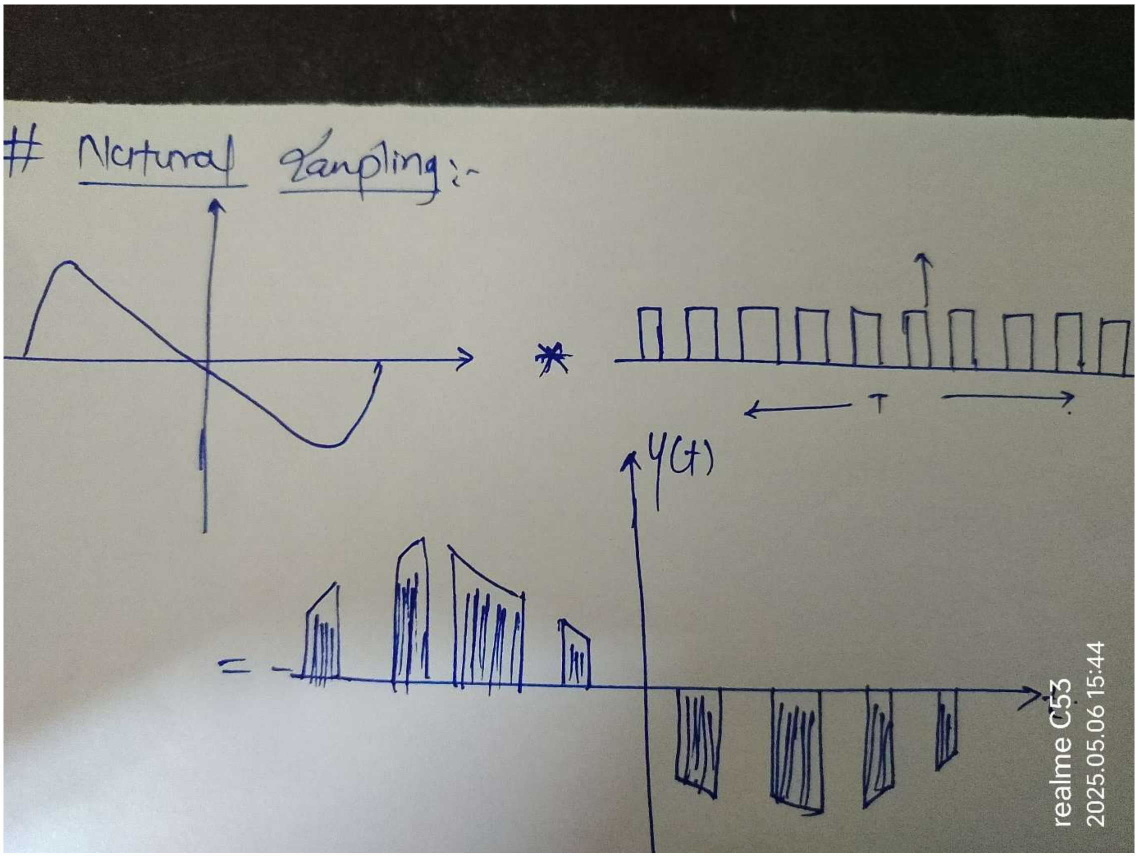
2. Natural Sampling:

➔ It is similar to ideal sampling, except impulse train is replaced by pulse train of period (T).

➔ The output of sampler is:

y(t) = x(t) * p(t)

y(t) = x(t) * $\sum_{n=-\infty}^{\infty} \quad = P(t - nT)$



3. Flat-Top Sampling:

➔ During transmission, noise is introduced at top of transmission pulse which can be easily removed if the pulse is in form of flat-top.

➔ The top of the samples are flat i.e. they have constant amplitude. Hence, it is called as flat-top sampling.

➔ The sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal.

y(t) = p(t) * $y\delta(t)$

➔ To get sampled spectrum, consider Fourier transformation on both sides

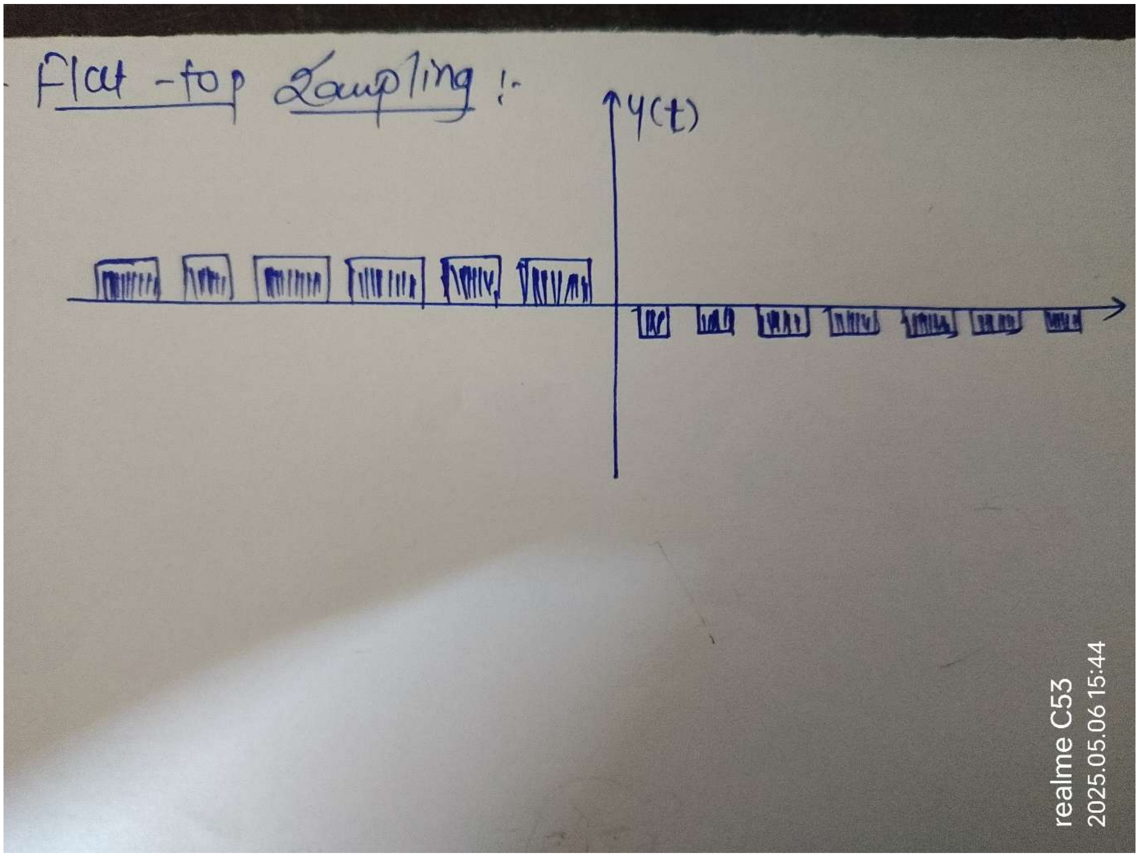
$y[\omega] = FT$ [p(t) * $y\delta(t)$]

➔ By knowledge of convolution property,

$y[\omega] = p(\omega)y\delta(\omega)$

Here,

$p(\omega)$ = TSq$\left(\frac{\omega T}{2}\right)$ = 2sin$\left(\frac{\omega T}{\omega}\right)$



Q16. Compare PAM, PPM and PWM.

Ans.

Aspects	PAM	PWM	PPM
Definition	Amplitude of Pulse varies based on modulating signal	Width of pulse varies based on modulating signal	Position of pulse varies based on modulating signal
Pulse Characteristics	Varies in amplitude, fixed width & position	Varies in width, fixed amplitude & position	Varie sin position, fixed amplitude & width
Complexity	Simple	Moderate	Complex
Bandwidth Requirement	Low	Moderate	High
Noise Immunity	Low	Better than PAM	Best
Power Efficiency	Low	Higher than PAM	Highest
Synchronization	Not needed	May be required	Strict synchronization required
Usage	Analog-to-digital conversion (PCM)	Motor speed control, LED diming, audio amplifier	Secure communication, telemetry & optical communication
Conclusion	PAM is simplest but more prone to noise.	PWM is widely used in power control & motor drivers.	PPM is best for high-noise environments but requires precise synchronization.

Q17. What is aliasing? How to reduce it?

Ans. It is a phenomenon in signal processing where high frequency components of signal get mis-represented as lower frequencies when sampled at an insufficient rate.

It may lead to distortion and incorrect reconstruction of the original signal.

It occurs when sampling frequency (fs) is less than twice the highest frequency (fm) of the signal, violating the Nyquist theorem (fs >= 2fm)

How to reduce it?

1. Increase sampling rate:

➔ Use sampling frequency at least twice the highest frequency of the signal (fs>=2fm)

➔ This ensures proper reconstruction and avoids overlapping of frequency.

2. Use anti-aliasing Filter:

➔ A low-pass filter is applied before sampling to remove high-frequency components.

➔ They ensure that only frequencies under Nyquist level can enter the sampler.

3. Proper Signal Bandlimiting:

➔ Ensures that i/p signal has limited frequency range before sampling.

➔ Remove unwanted high-frequency noise before digitization.

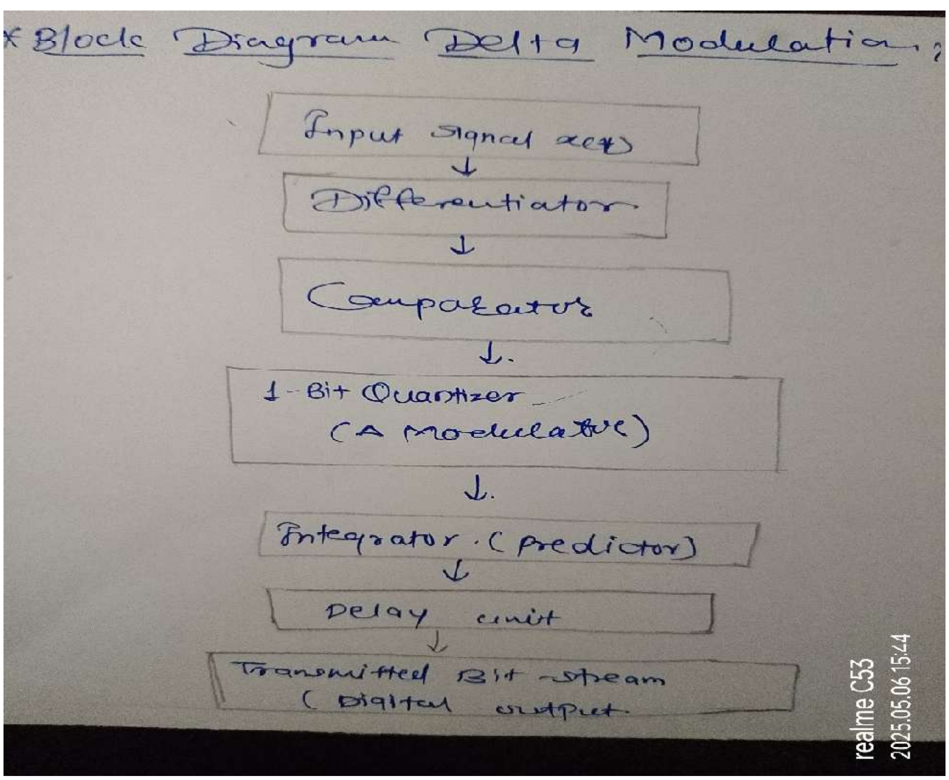
4. Use higher bit resolution:

➔ In digital signal processing, higher bit depth helps in better representation of the signal.

Q18. Draw and explain Delta Modulation.

Ans. It is simplified form of analog-to-digital conversion.

❖ Block Diagram:



❖ Explanation:

1. Input Signal x(t): The analog signal to be converted.
2. Differentiator: