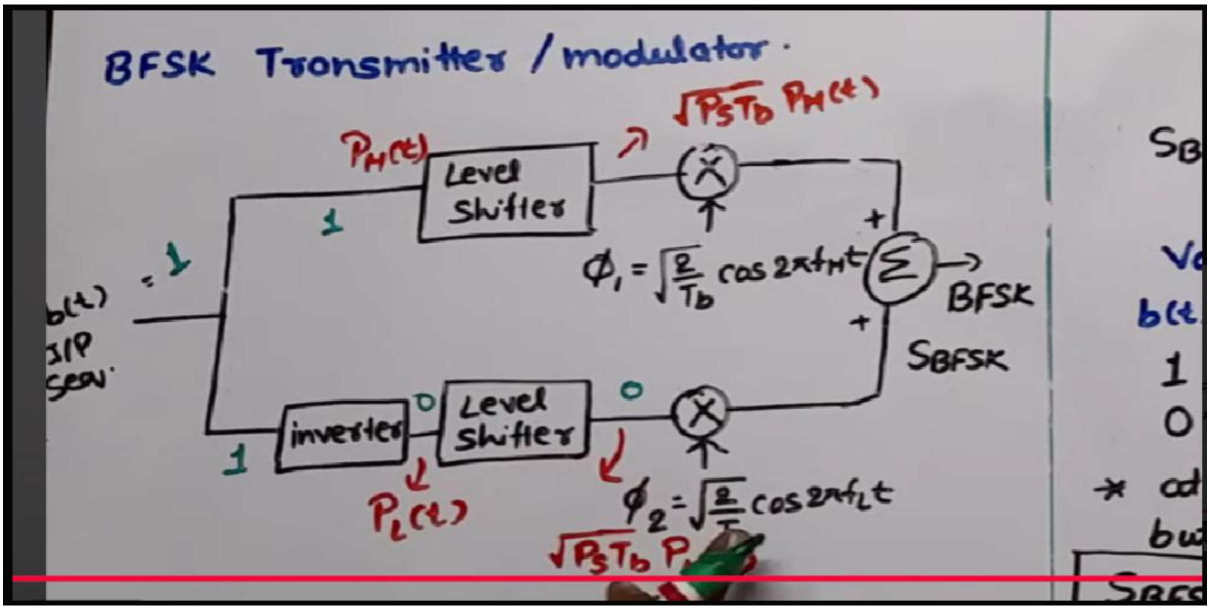
Q1) Draw and explain the transmitter block diagram of BFSK system



🧠 BFSK Transmitter / Modulator Explanation

1️⃣ Input: Binary Data $b(t)$

- The input binary data sequence $b(t)$(1s and 0s) controls which carrier frequency will be transmitted.
- When $b(t) = 1$→ frequency $f_1$is sent.
- When $b(t) = 0$→ frequency $f_2$is sent.

2️⃣ Level Shifter

- The level shifter converts the binary input (0 and 1) into **logic control signals** (e.g., +1 and 0, or 0 V and 5 V).
- It prepares the binary signal for analog multiplication with carrier signals.

3️⃣ Inverter

- The inverter produces the **complement** of $b(t)$.
  - So if $b(t) = 1$, inverter output = 0
  - If $b(t) = 0$, inverter output = 1
- This ensures that **only one path (upper or lower)** is active at a time.

4️⃣ Two Carriers

- The system uses **two orthogonal carriers**:

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_1 t)$$

$$\phi_2(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_2 t)$$

where T_b= bit duration.

- These two carriers correspond to the two different frequencies used for transmitting bits 1 and 0.

5️⃣ Multipliers (Mixers)

- Each carrier is multiplied (modulated) by its respective control signal:
  - Upper path: $P_H(t) \times \phi_1(t)$
  - Lower path: $P_L(t) \times \phi_2(t)$
- Only one multiplier output is nonzero at a time — depending on whether the bit is 1 or 0.

6️⃣ Summing Junction

- The outputs of the two multipliers are added together.
- The resulting signal is the **BFSK waveform**:

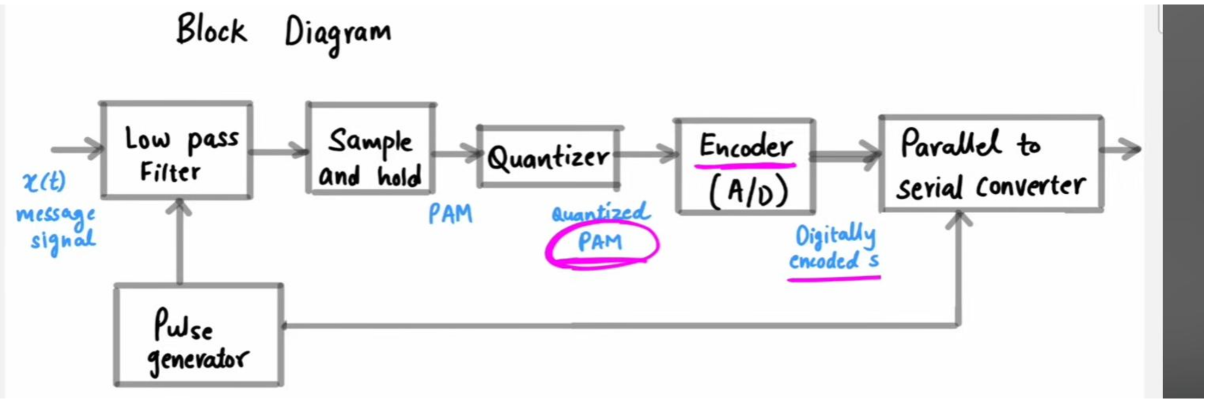
$$s(t) = \begin{cases} \sqrt{2P_b} \cos(2\pi f_1 t), & \text{if } b(t) = 1 \\ \sqrt{2P_b} \cos(2\pi f_2 t), & \text{if } b(t) = 0 \end{cases}$$

- This is the **BFSK modulated output signal**.

7️⃣ Output

- The output labeled **BFSK / SBFK (Signal Band Frequency Shift Keying)** is transmitted.
- The transmitted signal alternates between $f_1$and $f_2$based on the digital input bits.

Q3.1) Draw neat diagram and explain in detail a) PCM transmitter (ii) Delta Modulator



Low Pass Filter (LPF):
Removes high-frequency noise and limits signal bandwidth.

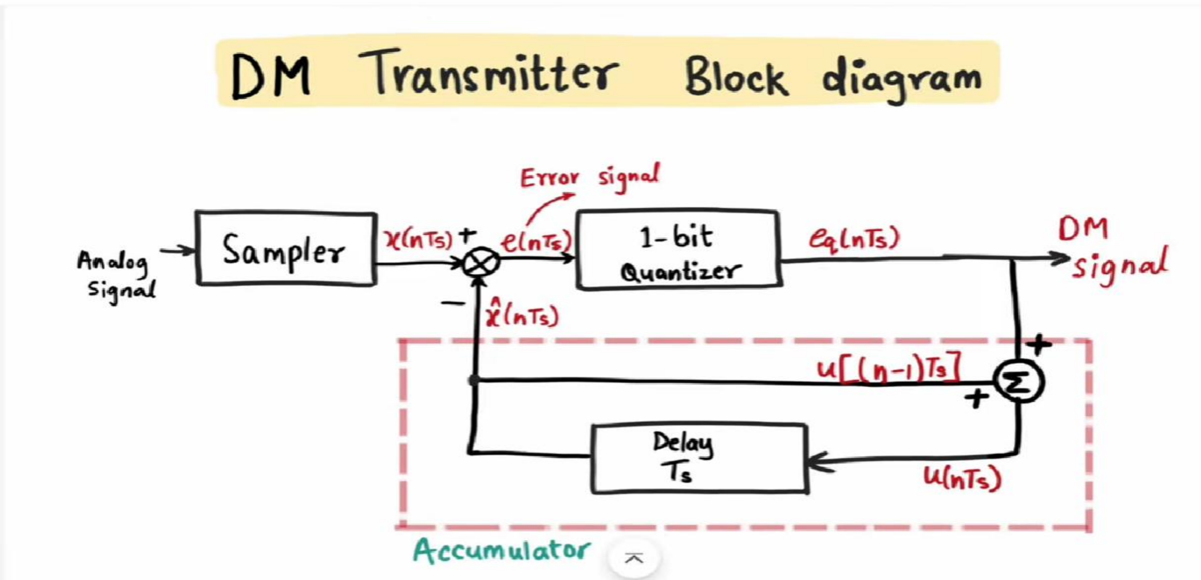
Sample & Hold:
Samples the analog signal at regular intervals and holds each value for processing.
→ Output: PAM signal.

Quantizer:
Converts sampled values into fixed discrete amplitude levels.
→ Output: Quantized PAM.

Encoder (A/D Converter):
Converts each quantized level into a binary code.
→ Output: Digitally encoded samples.

Parallel-to-Serial Converter:
Converts parallel binary data into serial bit stream for transmission.
→ Output: **PCM signal**.

Q3.2)



� **Sampler:**
The analog input signal is sampled at regular intervals to get discrete-time samples.

� **Comparator:**
Each new sample is compared with the previous predicted sample to find the **error signal**, which shows whether the signal has increased or decreased.

� **1-bit Quantizer:**
The error signal is quantized into two levels — **+Δ** for an increase and **–Δ** for a decrease. Thus, only **1 bit** is needed to represent each sample.

� **Accumulator (Predictor):**
Adds the present quantized output to the previous sample to estimate the next signal value. This feedback helps track the signal trend.

� **Output:**
The final output is a **binary sequence** (1s and 0s) representing the slope or direction of the analog signal change.

Q5)

Drawbacks of Delta Modulation

1. **Slope Overload Distortion:**
   Occurs when the input signal changes too rapidly compared to the step size (Δ).
   The modulator cannot follow the steep slope of the signal, causing distortion.
2. **Granular Noise:**
   Happens when the step size (Δ) is too large for slowly varying signals.
   The output signal keeps oscillating around the input, producing a noisy output.
3. **Fixed Step Size Problem:**
   A single fixed step size cannot handle both slow and fast signal variations effectively.
   Small Δ reduces slope overload but increases granular noise, and large Δ does the opposite.
4. **Low Signal-to-Noise Ratio (SNR):**
   Due to quantization and slope errors, DM provides lower SNR compared to PCM.
5. **Limited Accuracy:**
   The system gives only the direction of signal change, not its exact amplitude, leading to less precise reconstruction.

‘

Q6) Comparison between TDM and FDM

Parameter	TDM (Time Division Multiplexing)	FDM (Frequency Division Multiplexing)
Basic Principle	Shares the same frequency channel in **different time slots**.	Shares the same time by using **different frequency bands**.
Type of Signals	Mainly used for **digital signals**.	Mainly used for **analog signals**.
Bandwidth Requirement	Requires **less bandwidth**.	Requires **more bandwidth** as each channel needs a separate band.
Interference	No interference since time slots are separate.	**Crosstalk** may occur due to overlapping frequency bands.
Synchronization	Needs **synchronization** between transmitter and receiver.	**No synchronization** required.
Guard Band	Not required.	**Guard bands** are required between channels.
Example	Used in digital telephony and computer networks.	Used in radio and TV broadcasting.

Q7) For faithfull recovery of communication signal comment on sampling theorem

Sampling Theorem (For Faithful Signal Recovery)

The **Sampling Theorem** states that —

*A continuous-time signal can be completely and faithfully recovered from its samples if it is sampled at a rate at least **twice the maximum frequency** present in the signal.*

Mathematically,

$$f_s \geq 2f_m$$

where:

- f_s= sampling frequency
- f_m= highest frequency component in the signal

Explanation

- If the sampling rate is **less than $2f_m$**, overlapping of signal spectra occurs, known as **aliasing**, which leads to distortion.
- When the signal is sampled at or above **$2f_m$**, all the information of the original analog signal is preserved, allowing **faithful reconstruction** using a low-pass filter.

Conclusion

For **faithful recovery** of a communication signal, it **must be sampled at or above twice its highest frequency component** (i.e., $f_s \geq 2f_m$).

Q8) Explain intersymbol interference and explain the measure to be taken to reduce ISI

Intersymbol Interference (ISI)

Definition:
Intersymbol Interference (ISI) occurs when one symbol in a digital communication system interferes with the next symbol. This happens because the transmitted pulses spread and overlap in time due to **bandwidth limitation** or **channel distortion**.

As a result, the receiver cannot clearly distinguish between consecutive symbols, causing **errors in detection**.

Causes of ISI

1. **Bandwidth limitation** of the channel.
2. **Multipath propagation**, where signals take different paths and arrive at different times.
3. **Dispersion** in optical or wired channels.

Measures to Reduce ISI

1. **Use of Equalizers:**
   - Equalizers are used at the receiver to compensate for channel distortion and reduce ISI.
2. **Pulse Shaping Techniques:**
   - Using raised cosine or Nyquist pulses ensures zero ISI at sampling points.
3. **Increase Channel Bandwidth:**
   - A wider bandwidth allows pulses to pass with less distortion.
4. **Use of Error-Correcting Codes:**
   - Helps to detect and correct errors caused by ISI.

Conclusion

ISI degrades signal quality and increases error rate. It can be minimized by proper **pulse shaping**, **equalization**, and **channel design** to ensure clear and reliable communication.

Q9) Compare BASK, BFSK, BPSK

Parameter	**BASK (Binary Amplitude Shift Keying)**	**BFSK (Binary Frequency Shift Keying)**	**BPSK (Binary Phase Shift Keying)**
i) Bandwidth Requirement	Least bandwidth required	More bandwidth than BASK	Least bandwidth (same as BASK)
ii) Error Probability	High (more errors)	Moderate	Lowest (best performance)
iii) Noise Immunity	Poor (affected by noise easily)	Better than BASK	Best noise immunity
iv) Reception Complexity	Simple receiver	Moderate complexity	Complex receiver and demodulator
v) Bit Rate / Data Rate	Moderate	Moderate	Same as input bit rate

Q10) Duo Binary Encoding

Definition:
Duo-binary encoding is a type of **correlative coding** technique used to reduce bandwidth by introducing **controlled intersymbol interference (ISI)**.

Explanation:
In this method, the present output depends on both the **current bit** and the **previous bit**. It uses three signal levels: **–1, 0, +1**, which helps transmit data efficiently with less bandwidth and without increasing bit error rate.

Example:
If the input bit is 1 and the previous bit was 1, the output may be +1; if previous bit was 0, output may be 0, etc.

Q12) Explain line coding? What are parameters need to be considered for selecting a line code for a specifiic application

Line Coding

Definition:
Line coding is the process of **converting digital data (bits)** into a **digital signal** suitable for

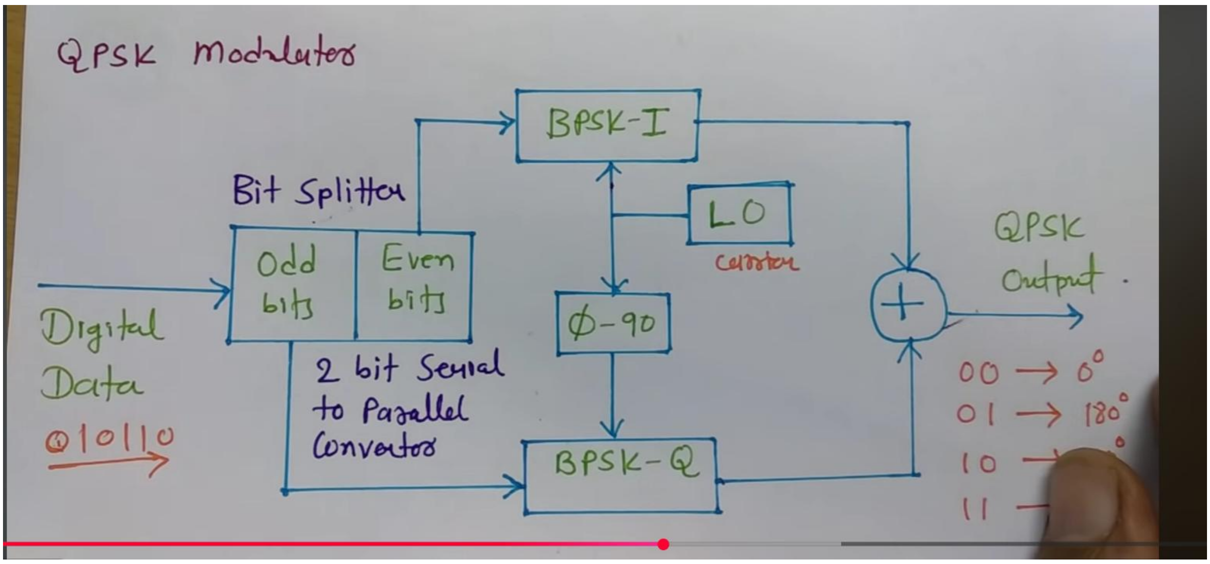
transmission over a communication channel. It defines how 1's and 0's are represented using voltage levels, transitions, or pulse shapes.

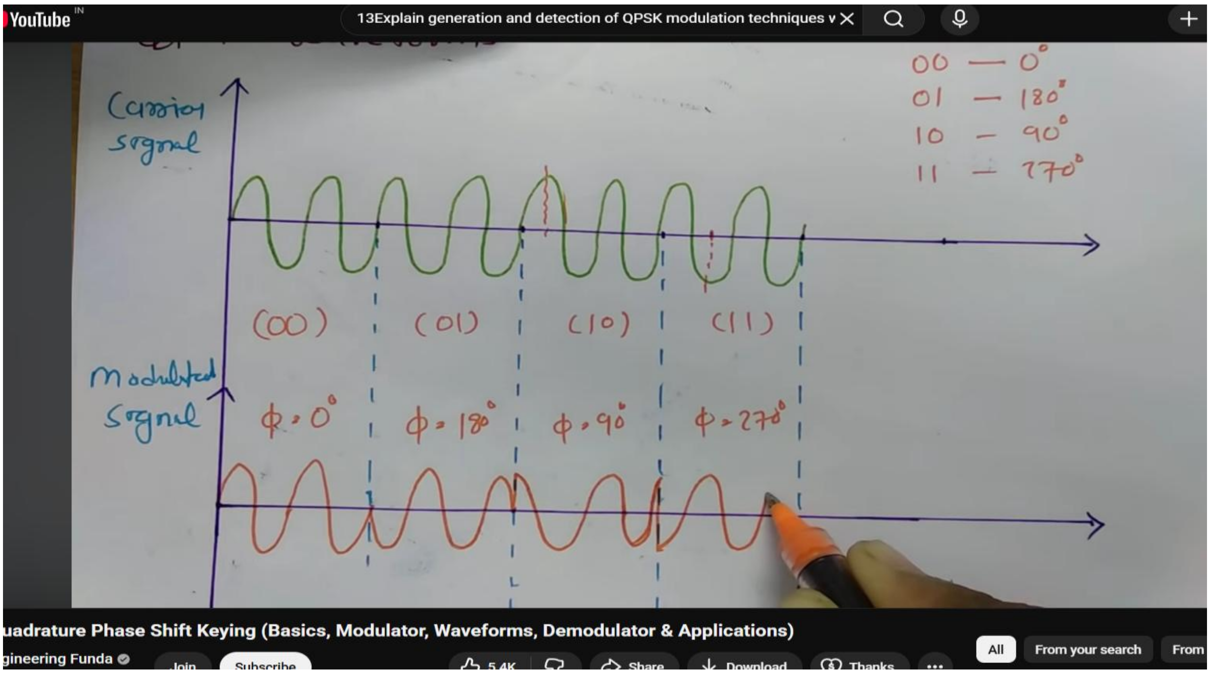
Example:
Common line codes include **NRZ (Non-Return to Zero)**, **RZ (Return to Zero)**, **Manchester**, and **Bipolar** coding.

Parameters to Consider for Selecting a Line Code

1. **Bandwidth Requirement:**
   - The code should occupy minimum bandwidth for efficient transmission.
2. **DC Component:**
   - Codes with less or zero DC component are preferred for AC-coupled systems.
3. **Synchronization:**
   - The line code should provide sufficient transitions to allow easy clock recovery at the receiver.
4. **Error Detection Capability:**
   - Some line codes allow detection of transmission errors (e.g., bipolar violations).
5. **Noise Immunity:**
   - The code should be robust against noise and distortion.
6. **Power Efficiency:**
   - The transmitted signal should use minimum power for reliable detection.

Q13) Explain generation and detection of QPSK modulation techniques with neat diagram and waveform also plot PSD of modulated signal





1. Introduction

Quadrature Phase Shift Keying (QPSK) is a **digital modulation technique** in which **two bits** are transmitted per symbol by changing the **phase** of the carrier signal.
It uses **four distinct phase shifts (0°, 90°, 180°, and 270°)** to represent the four combinations of two binary bits:

Bit Pair	**Phase Shift**
00	0°
01	90°
11	180°
10	270°

2. Generation of QPSK:

Explanation:

- The input binary data stream is divided into **two parallel bit streams** (I and Q).
- Each stream modulates a **carrier signal**:
  - **I-channel** → cos(2πfct)
  - **Q-channel** → sin(2πfct)
- The two signals are added to form the **QPSK signal**:

$$s(t) = \sqrt{\frac{2E_b}{T_b}}[b_I(t)\cos(2\pi f_c t) + b_Q(t)\sin(2\pi f_c t)]$$

- This results in a signal with one of four possible phases.

4. Detection of QPSK:

Explanation:

- The received signal is multiplied with **in-phase** and **quadrature** carriers.
- Low-pass filters extract the baseband signals.
- The I and Q bits are then **combined** to reconstruct the original data.

Q14) Draw and explain the transmitter and receiver block diagram of BPSK system

Binary Phase Shift Keying (BPSK) System

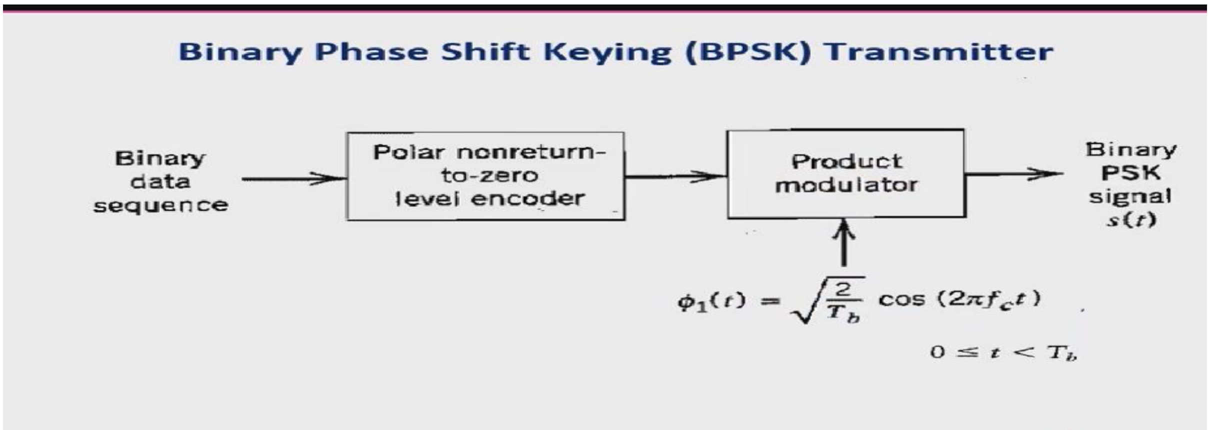
1. BPSK Transmitter (2 Marks)

The **BPSK transmitter** converts binary data (0s and 1s) into phase-shifted carrier signals.

Working:

1. The **binary data sequence** is first passed through a **polar NRZ level encoder**, which converts digital bits into two voltage levels:
   - Logic 1 → +V
   - Logic 0 → –V
2. This encoded signal is then fed to a **product modulator** where it multiplies with a **carrier signal**.
3. As a result, the **phase of the carrier** is shifted by **180°** depending on the input bit:
   - Bit ‘1’ → 0° phase
   - Bit ‘0’ → 180° phase

Hence, two distinct phase states represent binary data.

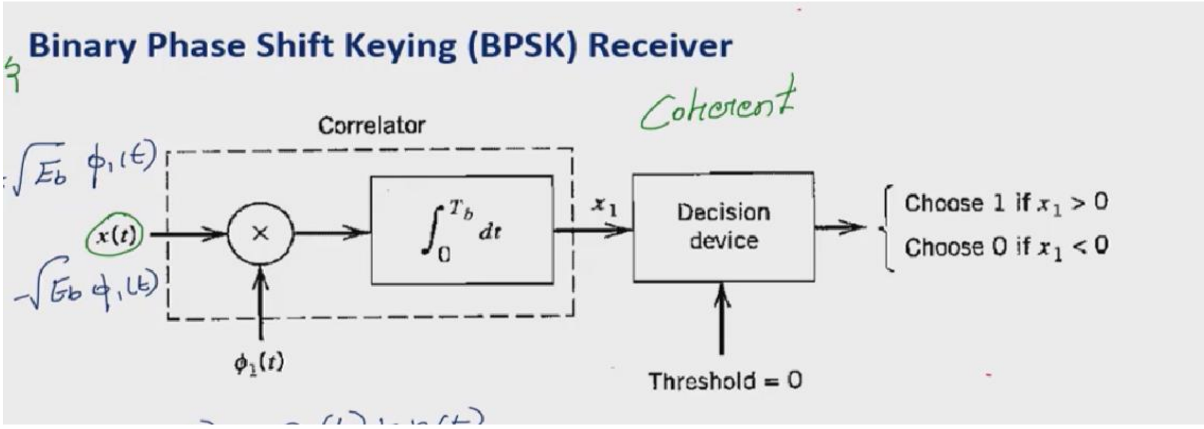


2. BPSK Receiver (2 Marks)

The **BPSK receiver** detects the transmitted signal coherently to recover the original binary data.

Working:

1. The received signal is multiplied with a **locally generated carrier** of the same frequency and phase — this is done in the **correlator**.
2. The correlator output is then **integrated** over one bit period.
3. The resulting value is sent to a **decision device**.
    - If output > 0 → bit is '1'
    - If output < 0 → bit is '0'
4. Thus, the receiver reproduces the original digital data.



Q15)

Pulse Amplitude Modulation (PAM)

Generation:
In PAM, the **amplitude of each pulse** is varied according to the **instantaneous value of the analog input signal**. The signal is passed through a **sampler** controlled by a **pulse generator**, which produces pulses at regular intervals.

Detection:
At the receiver, the PAM signal is passed through a **Low Pass Filter (LPF)** that removes the high-frequency components and **reconstructs the original analog signal**.