Master Tutorial - 2

7] Consider a PCM sys with a signal bandwidth of 4KHz. The sampling rate is 10KHz, and each sample is represented using 16 bits. Calculate the following : Bit rate, Nyquist freq, Sig to quantization noise ratio

→ BW = 4 KHz

f_s = 10KHz N = 16

(i) Bit rate $= f_s \times N$

$= 10KHz \times 16$

$= 160$ Kbps

(ii) Nyquist ~~rate~~ freq $= 2 f_m$

$= 8$ KHz

(iii) $(SNR)_{PCM} = 1.8 + 6N$

$= 1.8 + 6(16)$

$= 97.8$ dB

8] Quantization levels $= 2^N$

$= 1024$

Step size $= \Delta = \frac{10}{1024} = 9.76 \times 10^{-3}$ V

Max error $= \frac{\Delta}{2}$

$= \frac{9.76 \times 10^{-3}}{2}$

9] $\Delta = 0.01 \times 2$

$= 0.02$

levels $= \frac{\text{range}}{\Delta} = \frac{4K}{0.02}$

$= 200K$

Levels $= 2^N$

$N = \log_2[\text{level}]$

No. of bits ≈ 18 bits

Bit rate $= 16 \times 18$

$= 288$ kbps

10] Duration of speech sig = 10s

Sampled rate = 8KHz

$(SNR)_o = 40dB$

$1.8 + 6N = 40dB$

$6N = 40 - 1.8$

$N = 6.36$

$N \approx 6$

No. of samples = 8K * 10

= 80K samples

No. of bits in 10s sample = 80K × 6

= 480K bits

= 60 Kbytes

13] ~~2πfm Am <~~

Given : $A_m = 0.5V$ $f_m = 10KHz$

$f_s = 10 \times 2 f_m$

$= 20 f_m$

$f_s = 200KHz$

$2\pi f_m A_m < \Delta f_s$

$\frac{2\pi f_m A_m}{f_s} < \Delta$

$0.157 < \Delta$

14] $A_m = 4V$

(f_s) $SR = 32KHz$

$f_m = 4KHz$

$$2\pi f_m A_m < \Delta f_s$$

$$\frac{2\pi f_m A_m}{f_s} < \Delta$$

$$\frac{2\pi \times 4K \times 4}{32} < \Delta$$

$$3.14 < \Delta$$

15] $f_m = 2KHz$

$f_s = 20,000$

$\Delta = 0.1$

$$2\pi f_m A_m < \Delta f_s$$

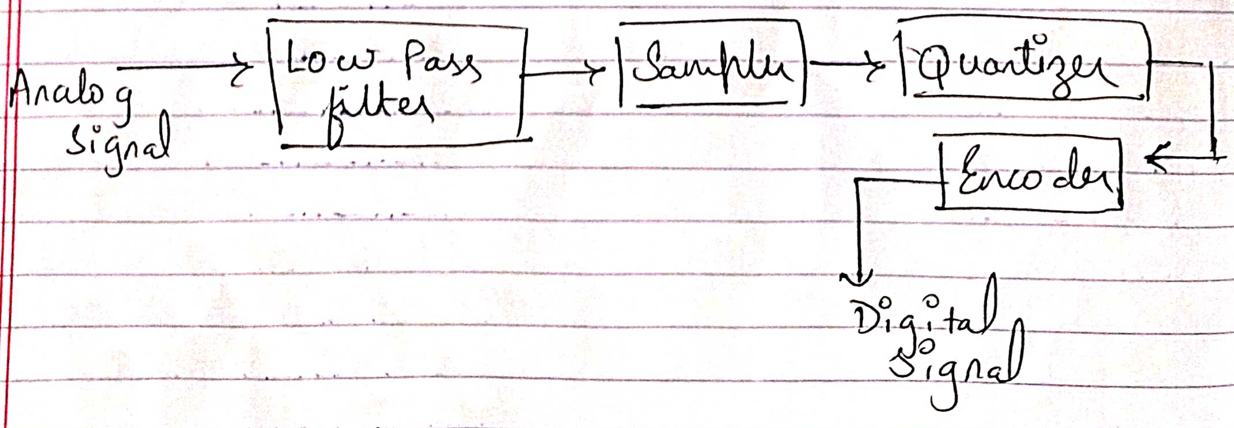
$$A_m < \frac{0.1 \times 20,000}{2\pi \times 2\times10^3}$$

$$A_m < 0.159 V$$

1] The modulation method that converts A to D signal is called Pulse Code Modulation [PCM]

PCM is a discrete time, discrete amplitude waveform coding process by means of which analog signal is directly represented by sequence of coded pulses.

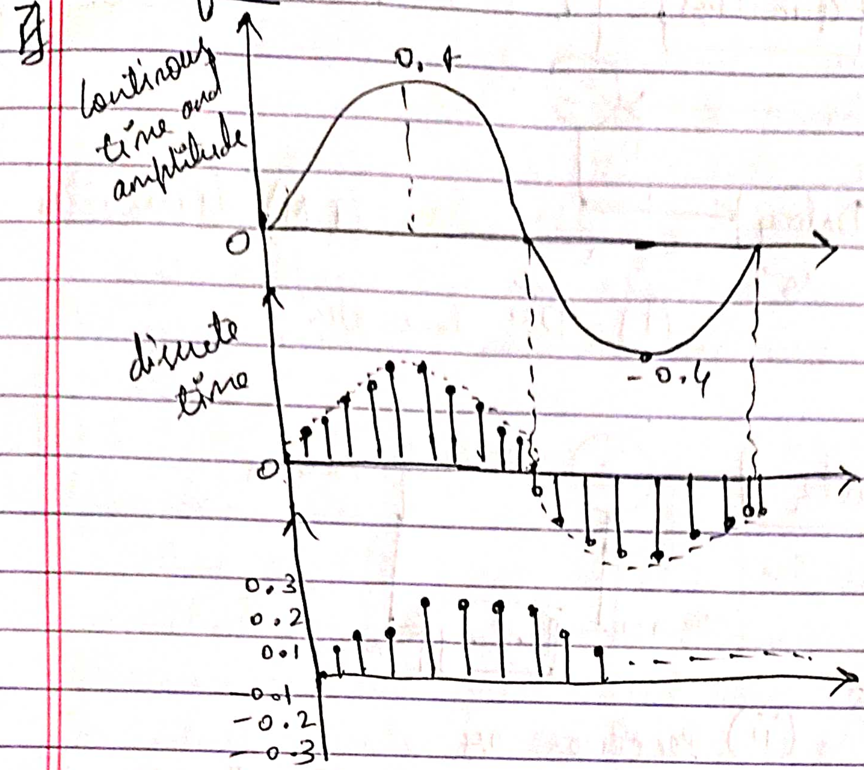
Block diagram



MVP - Q

Waveform

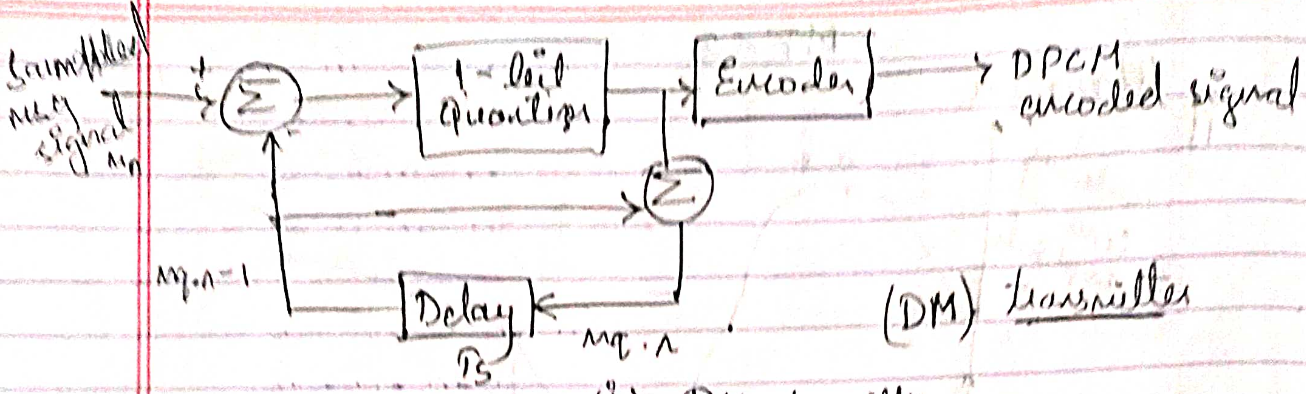
1]



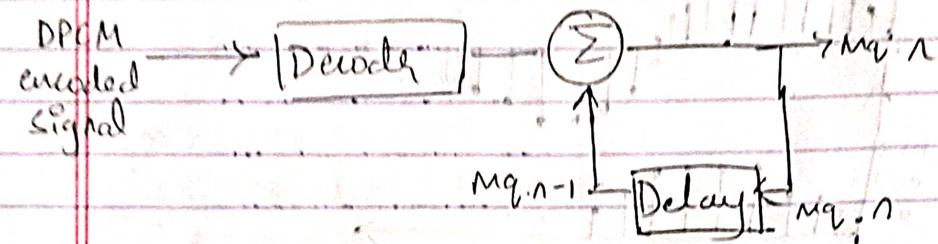
- LPF - Filters out high-frequency components above the Nyquist freq to prevent aliasing during sampling.
- Sampler - Takes discrete samples of the analog sig at a regular rate determined by the clk.
- Quantizer - Approximates each sample value to the nearest quantization level, assigning a binary code.
- Encoder - Converts the quantized values into a digital bitstream.

2] Suitable modulation tech that adjusts its quantization step in response is ~~Adaptive~~ Delta modulation (ADM).
[ADM is a form of differential pulse code modulation where the step ~~of~~ size of the quantizer is adjusted dynamically based on the characteristics of the i/p signal] X

- Delta Modulation: The i/p signal is compared with the previous quantized signal and the difference is encoded and transmitted. The receiver then uses the difference to reconstruct the original signal.

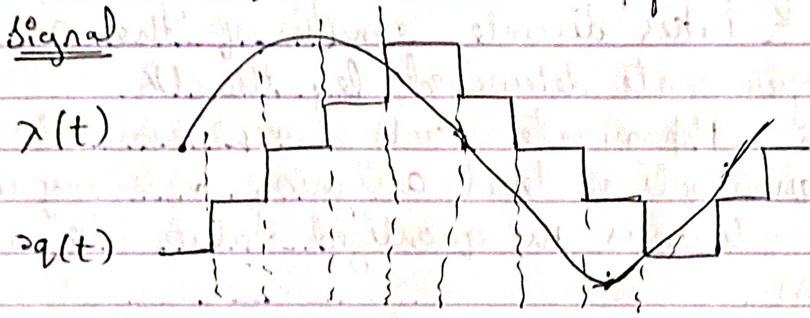


(i) DM transmitter



(ii) Receiver DM

- [Adaptation Mechanism :- The key innovation in ADM is adaptation of the step size. The quantization step size is adjusted dynamically based on the local characteristics of the i/p signal.] ×



Applications

1) wireless communications
2) Speech coding

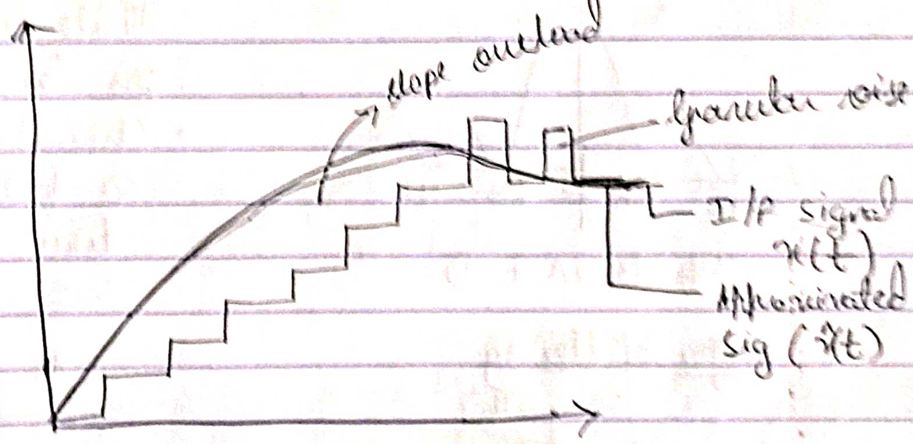
Advantages

1) Improved performance
2) Adaptability.

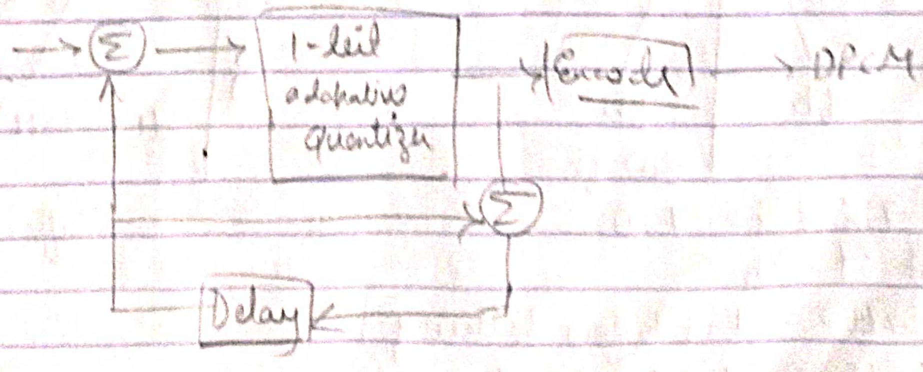
3) The two types of quantisation noise are:

- Slope overload noise - It occurs when the i/p signal exceeds the quantizer's dynamic range. When an i/p signal has a rapid change in voltage, and this change exceeds the amplifier's slew rate, the amplifier cannot accurately reproduce the signal's fast transition.
  This happens due to smaller step size.
  To reduce error, increase the step size or ↑ freq.
  ∴ Slope = $\frac{\delta \text{ height}}{T_s \text{ width}} = \delta f_s$
  (Applications): Audio amplifiers

- Granular noise -
  When an i/p signal $x(t)$ is relatively 'constant' in amplitude and the $\hat{x}(t)$ is bouncing up down.
  It occurs due to finite resolution of quantisation levels.
  To reduce granular noise, ↓ step size.
  But if step size is reduced, slope overload error ↑.
  ∴ In ADM, step size is not variable. If it is made variable then both errors can be controlled.



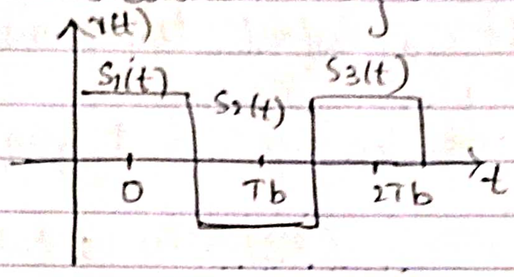
ADM method

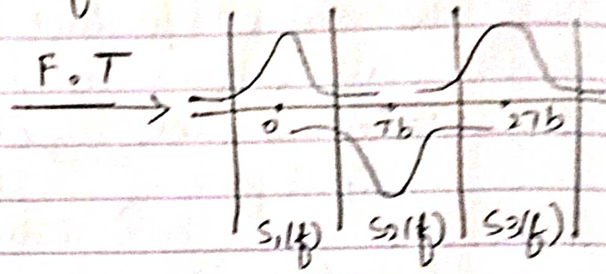


16] Intersymbol Interference (ISI)

→ The spreading of the pulse beyond its alloted time interval T_b causes it to interfere with neighbouring pulses is also known as ISI.

Let us consider a sig which is finite in T.D





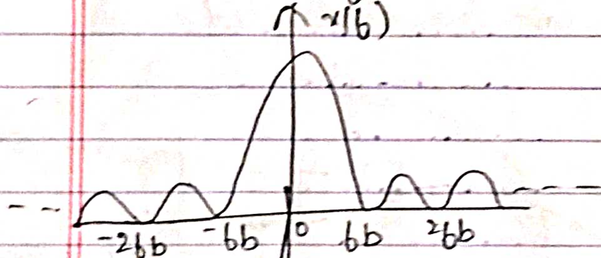
Finite in Time Domain.

Infinite in F.D

At	$S_1(t)$	$S_2(t)$	$S_3(t)$
$t=0$	✓	×	×
$t=T_b$	×	✓	×
$t=2T_b$	×	×	✓

⇒ ∴ No overlap of signal in T.D, hence no ISI

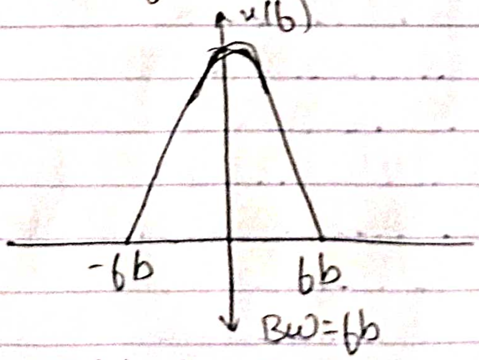
For the signal which is infinite in F.D, practically we need infinite BW. To get that we apply filtering.



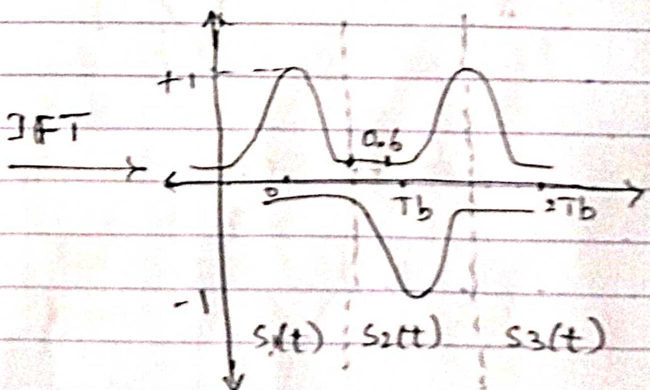
Infinite in F.D

- Finite time pulses are used for baseband transmission
- Channel BW is always finite, so filtering is req before trans.

<u>After filtering</u>



Finite in F.D



Infinite in T.D

<u>Bit Error</u>

At $t = T_b$

$S_2(t) = -1$ [Before filtering]

$S_2(t) = -1 + 0.6 + 0.6$

$= 0.2V > 0$

$= 1$ [After filtering]

∴ Due to ISI, $S_2(t) = 1$

Factors

- Channel dispersion - This dispersion causes the signals to overlap in time, leading to ISI
- Bandwidth limitations - This can result in distortion of transmitted signal.
- Multipath propagation - If diff path has different delays, the symbols can interfere with each other.

Impact

- Reduced Detection Accuracy
- Decreased Data Rate
- Bit Error Rate increase.

Examples of ISI

- Echo in Telephone Lines

In traditional telephone lines, echoes can cause ISI. The delayed reflection of the signal can interfere with subsequent symbols.

- Wireless com in Urban Environment.

With the tall buildings and structures, signals may take multiple paths due to reflections off buildings.

Methods to Mitigate ISI

- Equalization

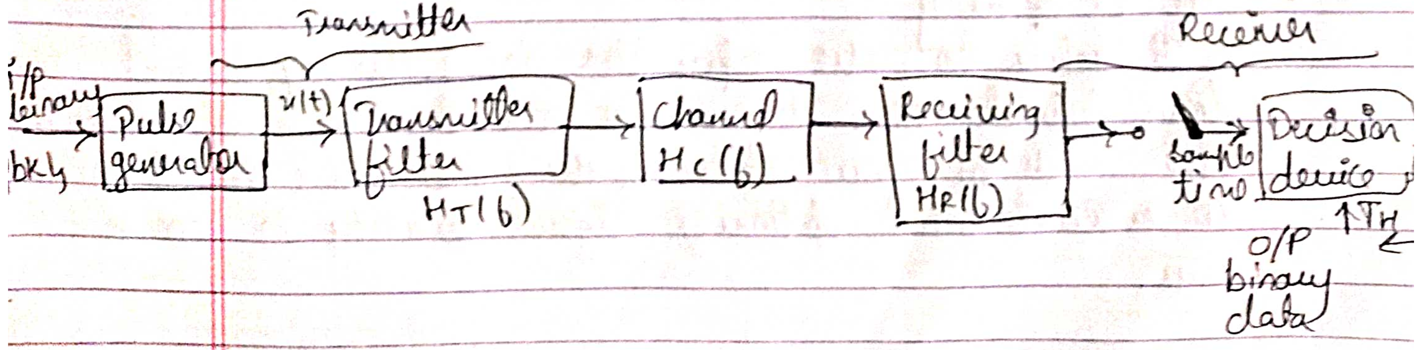
This involves adjusting the amplitude and phase of the received signal to mitigate the effects of channel dispersion.

- Freq-Domain Equalization

This tech involves applying FD processing to compensate for Freq selective nature of channel.

- Guard intervals

This helps in reducing ISI. It allows the signal to settle before the detection of the next symbol.

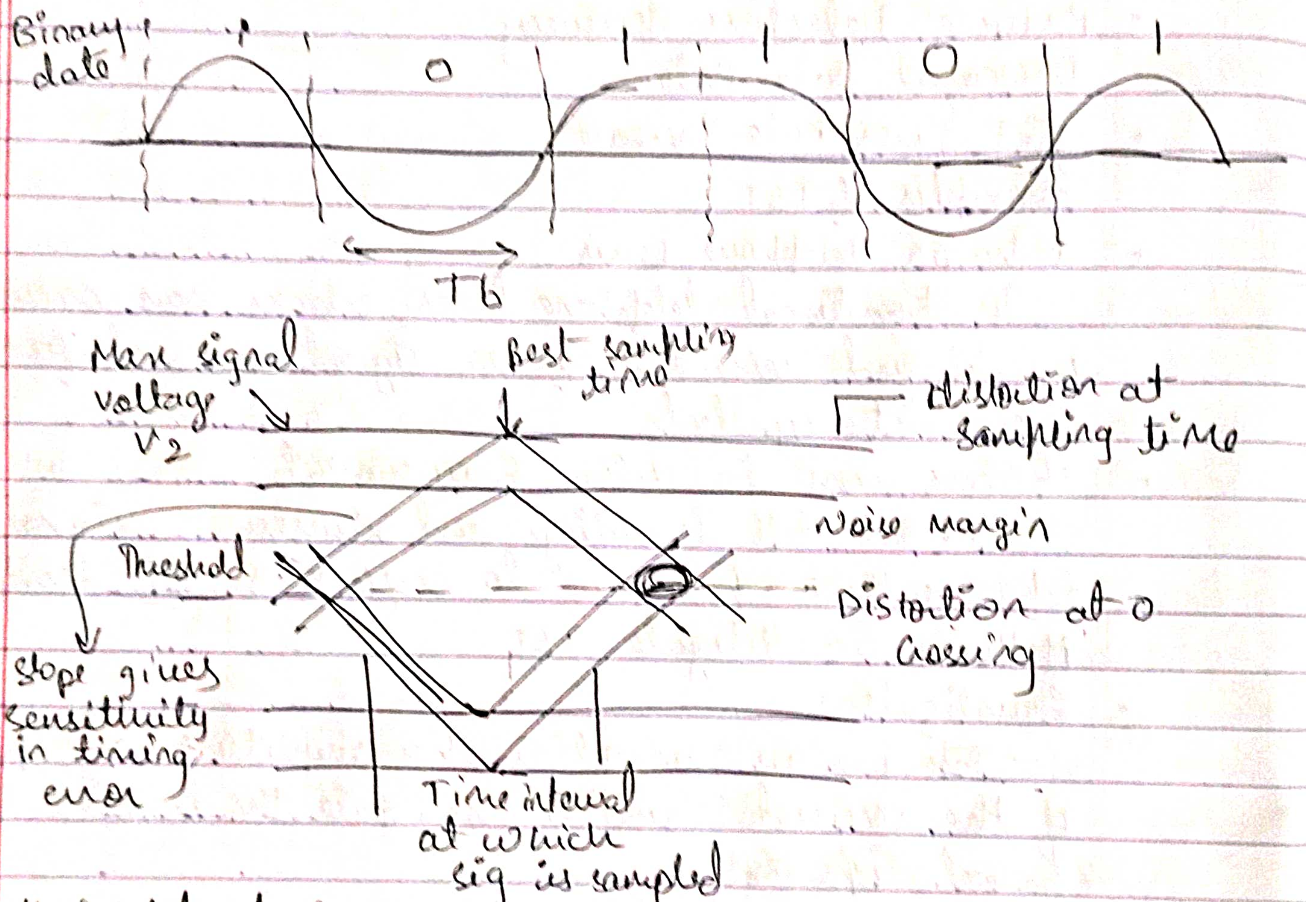


18] Eye pattern

An eye pattern, is a graphical representation of a digital signal, commonly used to assess signal quality and performance.

- The name 'eye pattern' is derived from the shape of the graph, which typically resembles an open eye.

Graph



- Horizontal opening

It refers to the separation or gap b/w the two lobes of the eye pattern. The eye pattern consists of two main lobes, representing the transition from low to high voltage level (0 to 1) and other high to low (1 to 0). The gap b/w these lobes are Horizontal opening.

- Vertical opening

It refers to the separation b/w the upper and lower boundaries. It indicates a better-defined distinction b/w high and low states of the signal. This is done to evaluate and optimize signal quality in comm sys.

- Slope
  It refers to the steepness or gradient of rising and falling edges of the signal waveform. Steeper slopes indicate faster rise and fall times, which are desirable for maintaining signal integrity.
- Distortion.
  It refers to the deviation or alteration in ideal shape of the eye pattern. It can be caused by various factors like dispersion, attenuation, ISI, etc.
- Sampling
  Sampling refers to the process of capturing and measuring signal at specific points in time, typically at the edges of the data eye. By sampling, we can assess factors like noise, rise/fall time.
- Noise
  It refers to the unwanted and random variations in the signal that can affect the clarity and reliability of the eye pattern.

<u>Factors</u>

- Channel characteristics
- Signal integrity
- Jitter
- Noise
- Bandwidth limitations.

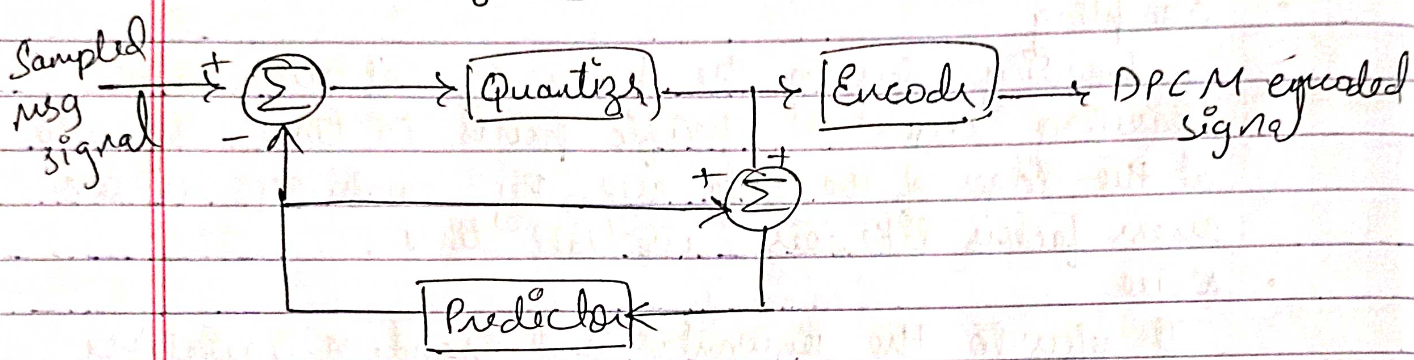
<u>Significance</u>

- Signal integrity assessment
- Fault detection
- BER estimation.
- optimization

4] Identified sys is DPCM

- The modulation techniques works on principle of prediction.
- The value of present sample is predicted from past samples.
- The prediction may not be exact but it is very close to it.

Block diagram



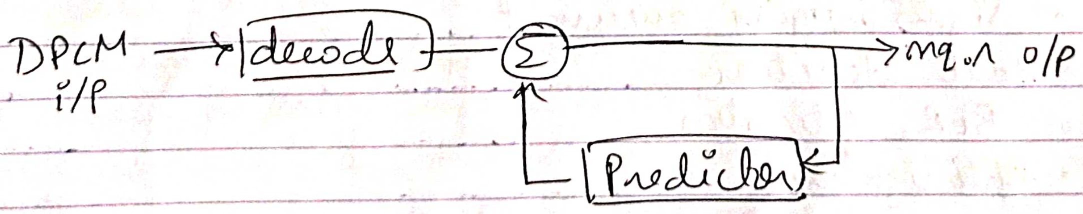
$e_n = m_n - \hat{m}_n$

$e_{q,n} = e_n + q_n$

$m_{q,n} = \hat{m}_n + e_{q,n}$

$= m_n + q_n$

- The decoder ~~seons~~ reconstructs quantized error signal from incoming signal. Then filtered o/p and quantized error are summed to get original sig



$m_{q,n} = q_n + m_n$

$(SNR)_o = \frac{\sigma_m}{\sigma_q^2}$

5) The step-size of quantizer is given by

$$\Delta = \frac{M_{max} - (-M_{max})}{L}$$

$$= \frac{2M_{max}}{L}$$

Let R denote no. of bits per sample.

$$L = 2^R$$

$$\therefore \Delta = \frac{2M_{max}}{2^R}$$

Boundary limit for Q $= -\frac{\Delta}{2} < q < \frac{\Delta}{2}$

$$f_Q(q) = \begin{cases} \frac{1}{\Delta} & -\frac{\Delta}{2} \le q \le \frac{\Delta}{2} \\ 0 & \text{otherwise} \end{cases}$$

$$\sigma_Q^2 = E[Q^2]$$

$$= \int_{-\Delta/2}^{\Delta/2} q^2 f_Q(q)\,dq$$

$$= \frac{1}{\Delta}\int_{-\Delta/2}^{\Delta/2} q^2\,dq$$

$$= \frac{1}{\Delta}\left(\frac{\Delta^3}{24} + \frac{\Delta^3}{24}\right)$$

$$= \frac{2\Delta^{\cancel{3}}}{24} \times \frac{1}{\cancel{\Delta}}$$

$$= \frac{\cancel{2}\Delta^2}{\cancel{24}\,12} = \frac{\Delta^2}{12}$$

$$\sigma_Q^2 = \frac{m^2_{max}}{3 \cdot 2^{2R}}$$

$$\therefore (SNR)_0 = \frac{P}{\sigma_Q^2}$$

$$= \frac{3 \cdot 2^{2R} \cdot P}{M^2_{max}}$$

$$(SNR)_0 = \frac{3P}{m^2_{max}} \cdot 2^{2R}$$

$$\left(P = \frac{Am^2}{2} = \frac{m^2_{max}}{2}\right)$$

$$= \frac{3\cancel{m^2_{max}}/2}{\cancel{m^2_{max}}} \cdot 2^{2R}$$

$$= \frac{3}{2}\, 2^{R}$$

$$(SNR)_0 = \frac{3}{2} \cdot 2^{2R}$$

$$10 \log_{10}(SNR)_o = 10 \log\left(\frac{3}{2}\right) + 10 \log_{10} 2^{2R}$$

$$= 1.8 + 2R \cdot 10 \log_{10} 2$$

$$= 1.8 + 6R \;//$$

6) a) Channel Noise

It any unwanted signal that adds to the transmitted signal during its journey through the comm channel

- Thermal noise
- Interference
- Channel distortion

b) Error probability Analysis

This aims to calculate the prob of incorrect info being received at the destination due to channel noise.

- Modeling channel
- choosing a signal transmission
- Calculating prob of error

Common metrics used in error prob are:

- Bit error rate
- Packet " "
- Symbol " "

Factors affecting:

- Signal to noise Ratio
- Channel characteristics
- Transmission scheme

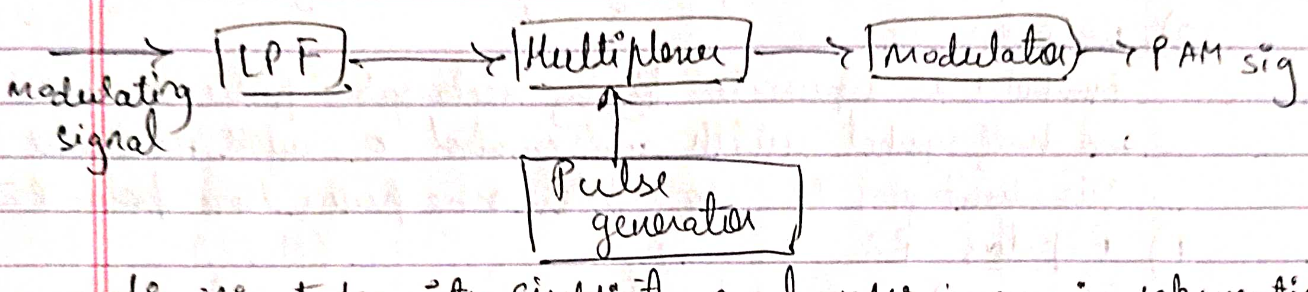
$10 \log_{10}(SN = 10 \log(\frac{3}{2} + ...$

$= 1.8 + 2R \cdot 10 ...$

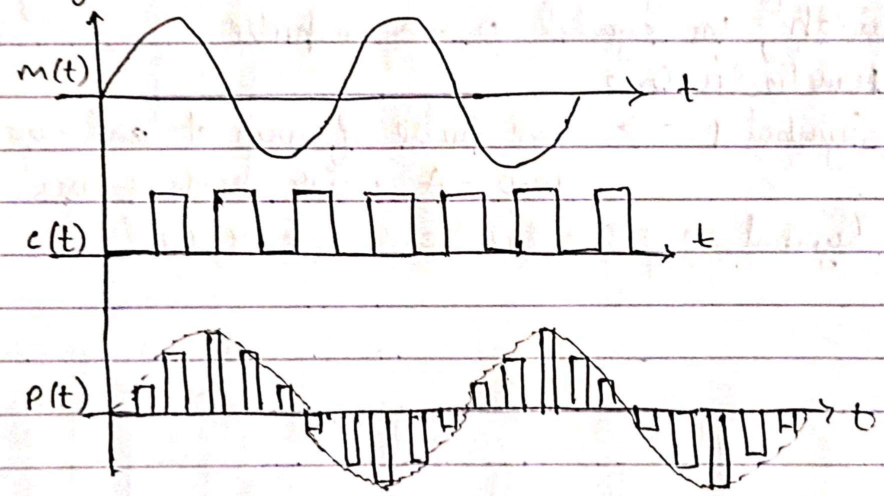
$= 1.8 + 6R$

12) Pulse Amplitude Modulation

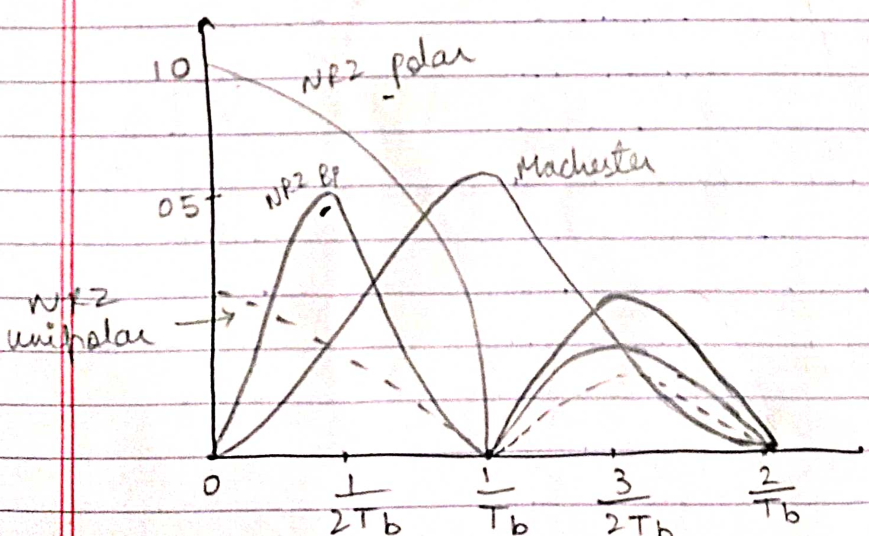
- PAM involves encoding digital data by varying amp of discrete pulse
- PAM is basic form of Pulse modulation. It is a tech in which amp of each pulse is controlled by instantaneous amp of the modulation signal.
- It can be digital as well as analog.



- Significant for its simplicity and efficiency is representing digital symbols.



Factors affects Power spectra of PAM sig



Disadvantage - Waste of power due to transmitted DC level.

1) Unipolar NRZ
Symbol 1 → transmitting pulse of amp A
Symbol 0 → switching off the pulse.
This is also referred as on-off signaling.

2) Polar NRZ
Symbol 1 and 0 are represented by transmitting pulses of amp +A and -A. This is easy to generate.
Disadvantage - Power spec of signal is large near zero freq

3) Unipolar RZ
Symbol 1 is represented by rectangular pulse of amp A and half symbol width and symbol 0 with no pulse.
Disadvantage - It req 3dB more power than polar RZ.

4) Bipolar RZ
It uses 3 amp. +ve and -ve pulses of equal amp. are used for symbol 1, with each pulse having half symbol width, for symbol 0, → no pulse.

5) Manchester Code
Symbol 1 → +ve pulse of amp A and -ve pulse of amp -A with both pulses half symbol wide.
Symbol 0 → Polarities are reversed.

17) Nyquist Criterion

The receiver reconstructs data sequence by extracting and decoding the corresponding sequence of weights from the o/p $y(t)$.

$t = iT_b$

$$p(iT_b - kT_b) = \begin{cases} 1 & \text{for } i = k \\ 0 & \text{" } i \neq k \end{cases} \quad -①$$

$$y(iT_b) = \mu a_i + \sum_{k=-\infty}^{\infty} a_k p(i-k)T_b = \mu a_i$$

FT of impulses:

$$P_\delta(f) = \frac{1}{T_b} \sum_{-\infty}^{\infty} P\left(f - \frac{n}{T_b}\right) \quad -②$$

$$P_\delta(t) = \sum_{-\infty}^{\infty} p(nT_b)\,\delta(t - nT_b)$$

$$P_\delta(f) = F[P_\delta(t)]$$

$$= \int_{-\infty}^{\infty} \sum_{-\infty}^{\infty} p(nT_b)\,\delta(t - nT_b)\,e^{-j2\pi ft}\,dt$$

putting $n = i - k$

$$P_\delta(f) = \int_{-\infty}^{\infty} \sum_{-\infty}^{\infty} p([i-k]T_b)\,\delta(t - [i-k]T_b)\,e^{-j2\pi ft}\,dt$$

Substituting eq ①,

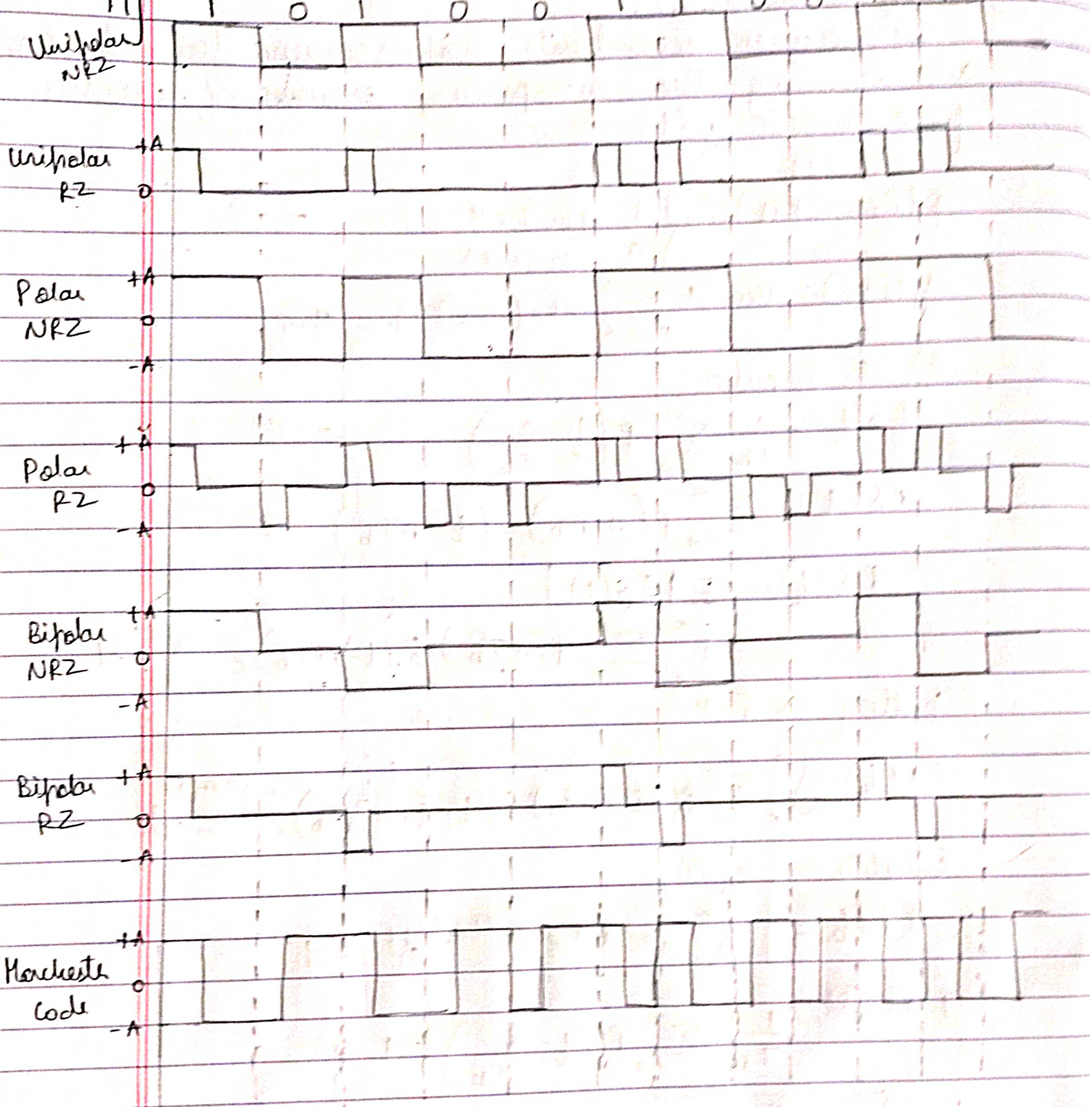
$$\therefore P_\delta(f) = \int_{-\infty}^{\infty} \delta(t)\,e^{-j2\pi ft}\,dt = 1$$

From ②,

$$\frac{1}{T_b} \sum_{-\infty}^{\infty} P\left(f - \frac{n}{T_b}\right) = 1$$

$$\boxed{\sum_{n=-\infty}^{\infty} P\left(f - \frac{n}{T_b}\right) = T_b = \frac{1}{R_b}}$$

11



Symbol 1 -

Symbol 0 -