MT-2

7. $BW = 4kHz$
$f_s = 10kHz$
$N = 16$

Q. Consider PCM sys with signal BW of 4kHz. Sampling rate is 10 kHz & each sample is represented using 16 bits. Calculate the following

1. Bit rate (bit/sec) => sampling freq X No of sample

$= f_s \times N$

$= 10k \times 16$

$= 160K$ bps

2. Nyquist frequency $= 2 \times f_m = 2 \times 4kHz$ $(f_m = BW)$

$= 8 kHz$

3. Signal-to-quantization noise ratio

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

$= 1.8 + 6 \times 16$

$= 97.8$ dB.

8. In PCM sys., the i/p signal ranges from $-5V$ to $+5V$. The quantization is performed using a 10-bit ADC. Determine the following:

1. Number of quantization levels

$L = 2^R$

$= 1024$

2. Quantization step size

$\Rightarrow \Delta = \frac{10}{1024} = 9.76 \times 10^{-3}$ v

3. Max quantization error $\Rightarrow \frac{\Delta}{2} = \frac{9.76 \times 10^{-3}}{2} = 4.88 \times 10^{-3}$

9. A PCM sys is used to encode an analog signal with freq range of 0 to 4kHz. If the quantization error should be less than 1% of maximum i/p signal amplitude, calculate the minimum number of bits required for quantization. Additional, determine the bit rate if sampling rate is 16 KHz.

=> $\Delta = e \times 2 = 0.01 \times 2 = 0.02$.

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

$1\% = \frac{\Delta}{2}$ $\therefore \Delta = 0.02$

Step size

L - levels $= \frac{\text{range}}{\Delta} = \frac{4k}{0.02} = 200k$

No. of bits ≈ 18 bits.

↳ No. of bits $= L = 2^N$

Apply log

$\log(L) = N \log 2$

$\log_2(L) = N$

$N = \log_2(200k)$

$N = 17.60$

$N = 18$ bits.

Bit rate → $f_s \times N$

$= 16 \times 18$

$= 288 KHz$

10. A speech signal has total duration of 10s. It is sampled at the rate of 8KHz & then encoded. The signal-to-(quantization) noise ratio is required to be 40dB. Calculate the minimum storage capacity needed to accommodate this digitized speech signal.

→ $f_s = 8 KHz$

$SNR = 40 dB$

$1.8 + 6N = 40 dB$

$6N = 40 - 1.8$

$N = 6.36$

$N \approx 6.$

No. of samples $=$ $f_s \times$ duration

$\cancel{f_s \times} = 8K \times 10$

$= 80K$ samples.

No. of bits in 10s sample $\rightarrow N$

$= 80K \times 6$

$= 480K$ bits

$= 60K$ bytes

$\left[\frac{480}{8} \right.$ in bytes

13. A Delta modulation sys I/P applied 10KHz, 1 Vpp. The signal is sampled ten times more than Nyquist rate. What is min step size required to prevent slope overload?

→

$2\pi f_m A_m < \Delta f_s$

sampling freq → f_s

Step size → Δ

msg freq = f_m

msg Amp = A_m

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

derivative

$\left| \frac{dx(t)}{dt} \right|_{max} < \Delta \times f_s$

$\frac{dx(t)}{dt} = A_m \cos(2\pi f_m t) \cdot 2\pi f_m$

$= 2\pi f_m A_m \cos(2\pi f_m t)$

$\left| \frac{dx(t)}{dt} \right|_{max} = 2\pi f_m A_m$

max of Cos = 1

$A_m = 0.5$ ∵ V_{p-p} Peak to peak = 1 V ÷ 2

$f_m = 10K\,Hz$

$f_s = 10 \times 2 f_m$

$= 200K$

$\Delta = ?$

$2\pi f_m A_m < \Delta f_s$

$\frac{2\pi (10 \times 10^3)(0.5)}{200 \times 10^3} = \Delta$

$\boxed{\Delta = 0.157}$ ⇐ min step size by which slope Overload can be prevented.

14.

$2\pi f_m A_m < \Delta f_s$

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

$\frac{2\pi (4 \times 10^3)(4)}{32 \times 10^3}$

$\Delta = 3.14$

$f_s = 32\,KHz$

$A_m = 4\,V$

$f_m = 4\,KHz$

15. $2\pi f_m A_m < \Delta f_s$

$2\pi \times 2k \times A_m < 0.1 \times 20k$

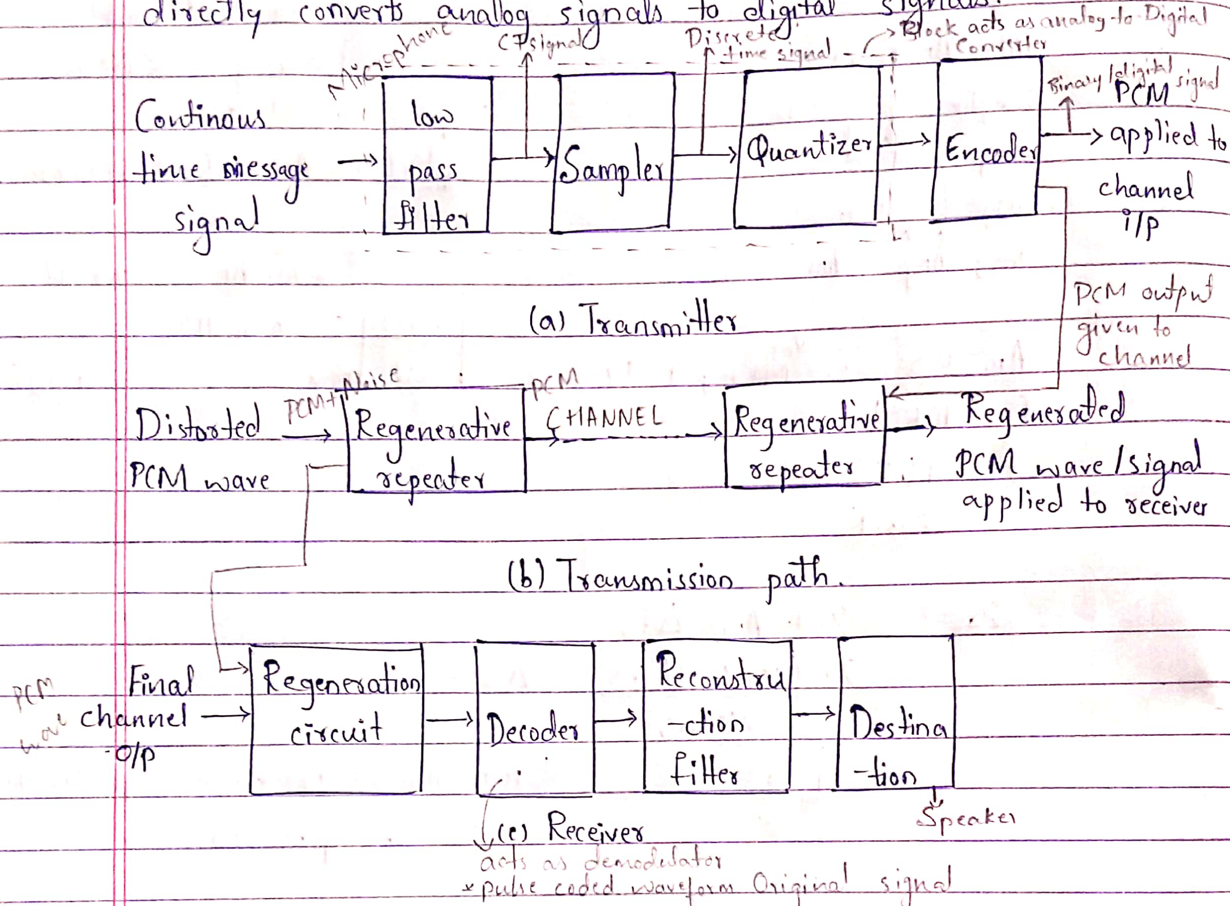
$12.56 \times A_m < 2$

$A_m < 0.159$

$A_m = 0.159\,V.$

1. Identify the modulation method that converts analog signal to digital signal. Explain the same with a neat block dig.

→ Pulse Code Modulation (PCM) is the modulation method that directly converts analog signals to digital signals.



Basic elements of PCM system

a) Transmitter

LPF - eliminates high frequency components which is greater than highest freq of msg signal to avoid aliasing.

Sampler - collects sample data at instant values of msg signal. so as to reconstruct original signal $f_s > 2W$.

$2^n \to$ levels $n \to$ bits.

Quantizer - ~~Reduces~~ Approximates each sample to the nearest allowed value (quantization level), represents it with binary code. Process of approximation or rounding of the signal

Encoder - Converts quantized values into a digital bit (stream suitable for transmission.) & applies line code

b) Transmission Path

Carries the digital message (codes) through a channel (like a wire or radio wave.

c. Receiver:

Decoder : Reads the codes & recreates the rounded step heights.

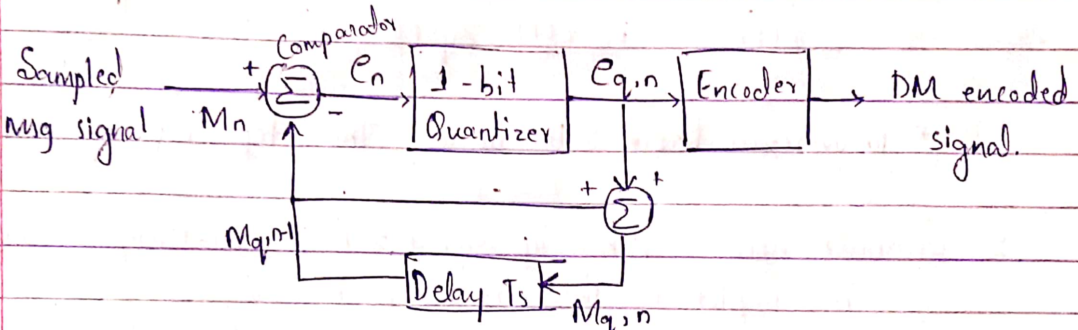
Reconstruction Filter: Connects the steps back into a slope, resembling the original smooth one.

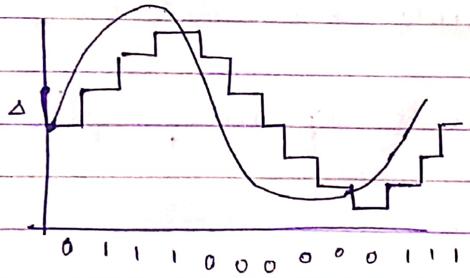
2. Explain a suitable digital modulation technique that adjusts its quantization step in response to changes in signal characteristics effectively encode and transmit information in dynamic signal environments.

→ Delta Modulation is a suitable...

* It adjusts its step size to match signal changes

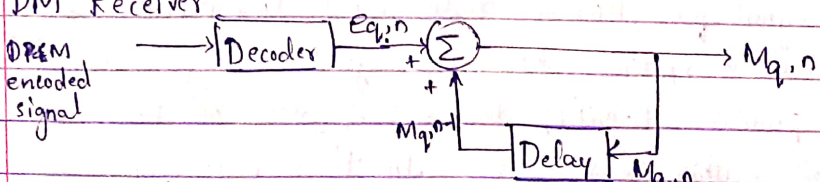
DM Transmitter



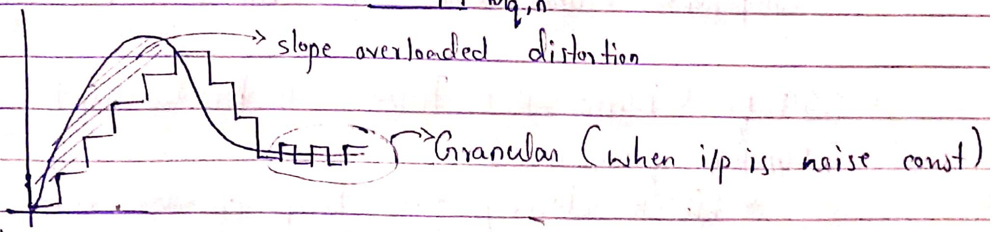


- Look at the current pt on line
- Decides wether to take up or down step based on signal
- & sends 1 → up, 0 → down.

DM Receiver



error

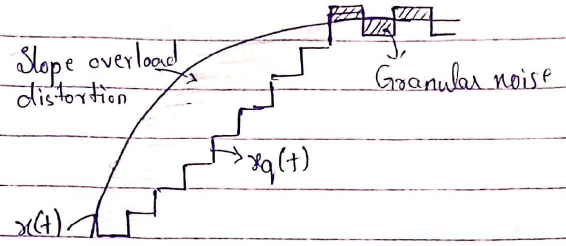


Advantages
- Simple implementation
- Low BW requirements
- Robust to noise

Disadvantages
- Granular error
- Slope overload distortion error

3. List and explain 2 types of quantization noise. Identify the suitable modulation technique to overcome these noise & explain it with appropriate block diagrams.

1. Slope overload Distortion
2. Granular Noise



1. SOD → If the slope of $x(t)$ is much higher than $x_q(t)$ over a long duration then $x_q(t)$ will not be able to follow $x(t)$ ie $x(t) - x_q(t)$

* To overcome these SOD - increase the step size of $x_q(t)$

2. Granular noise → When i/p signal $x(t)$ is relatively constant in amplitude then $x_q(t)$ is bouncing up & down which will cause granular noise distortion.

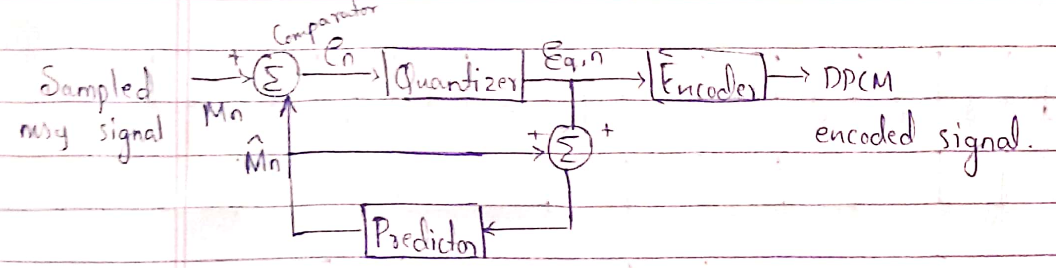
* To overcome this → Decrease step size of $x_q(t)$

4. In CS designed for efficient audio signal transmission & reception, there exists a system whose i/p signal undergoes a predictive encoding process. Identify the sys & describe the various components within this sys with their interconnections.

→ DPCM (Differential Pulse Code Modulation)
(Uniform quantizer.)
* Which cleverly predicts & encodes audio signals.
* Better version of PCM

Application → In speech image & audio compression

a) Transmitter



* Predictor : Takes guess about next sample level/ based on previous ones.
* Quantizer : Measures the difference's strength & rounds it to simple level

Encoder : Assigns a code to rounded difference & sends it, focusing on unexpected changes.

$e_n = M_n - \hat{M}_n$ $\qquad$ $e_n \rightarrow$ sampled msg signal i/p

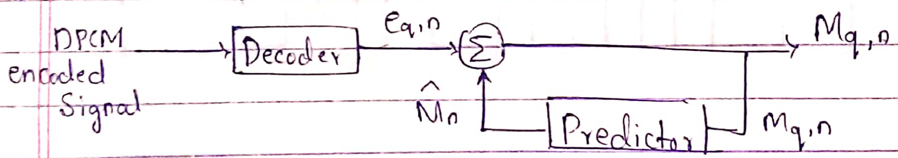
$e_{q,n} = e_n + q_n$ $\qquad$ $\Rightarrow$ Quantized o/p

$M_{q,n} \Rightarrow \hat{M}_n + e_{q,n}$

$= \underbrace{\hat{M}_n + e_n}+q_n$

$M_{q,n} = M_n + q_n$ $\qquad$ [from e_n]

b. Receiver



* Decoder → Reads the code and recreates the level.

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

$= e_{q,n} + m_n - e_n$

$= \cancel{e_n} + q_n + m_n - \cancel{e_n}$

$M_{q,n} = M_n + q_n$

Advantages

1. Bandwidth requirement is less as compared to PCM [less BW]
2. Quantization error can be ↓ ∵ of Predictor.
3. No. of bits used to represent per sample is reduced.

11. 10100110 0110