27/1/25

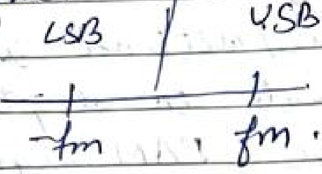
Signal → Analog Data → Digital Processing → reduction of noise
→ storage is more
→ quality of picture is good

Signal → Discrete → Pulse wave.

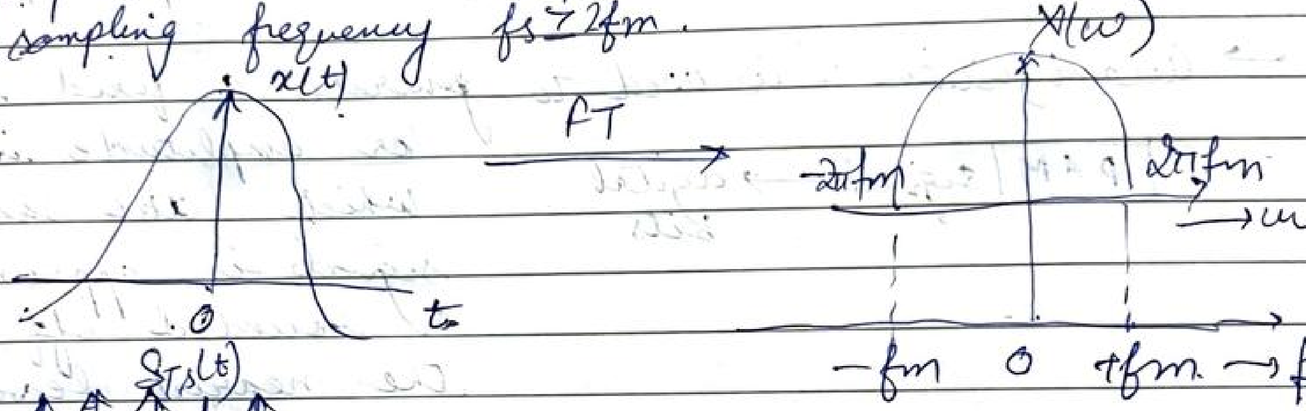
→ To convert continuous time to discrete time by fourier transform
→ Shift from continuous time signal to discrete time signal
• Processing of discrete time signal is more flexible and preferable.
• Conversion of continuous time signal into discrete time signal can be achieved using a fundamental,

mathematical tool extremely important in signal processing called sampling Theorem.

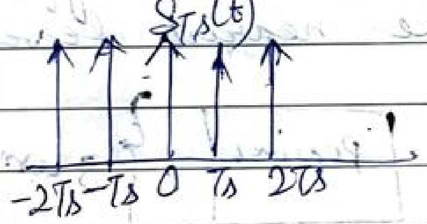
- Bandlimited signal:- There is no frequency component after fm.



- Sampling Theorem:- A BLS of finite energy which has no frequency component higher than fm may be completely represents in its samples and recovered back if the sampling frequency $fs \geq 2fm$.

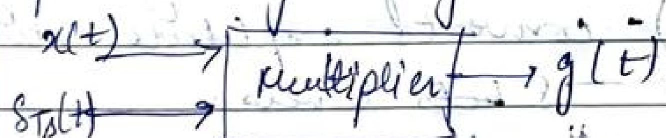


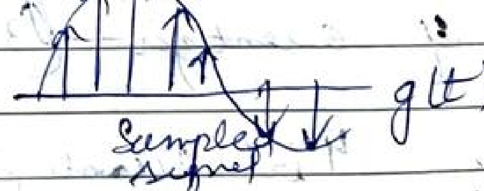
$\omega = 2\pi fm$
↳ Angular frequency.



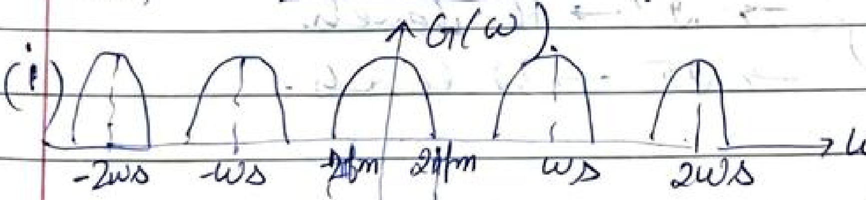
Sampling Time ⇒ $Ts = \frac{1}{fs}$

$\delta_{Ts}(t)$ → Impulse train consists of unit impulses repeating periodically every Ts seconds.



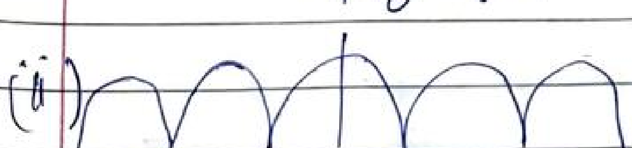


(i)



$fs > 2fm$ → Reconstructed. spectrum of sample signal without noise or overlapping

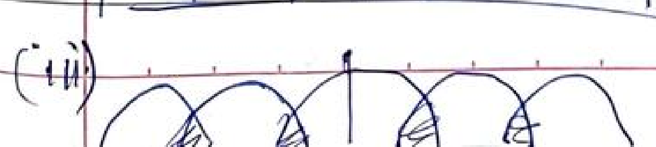
(ii)



$fs = 2fm$ → touched

undersampled signal
↳ overlapping
$fs < 2fm$.

(iii)



Noise occurs

28 Jan 2025

1st day lecture after this

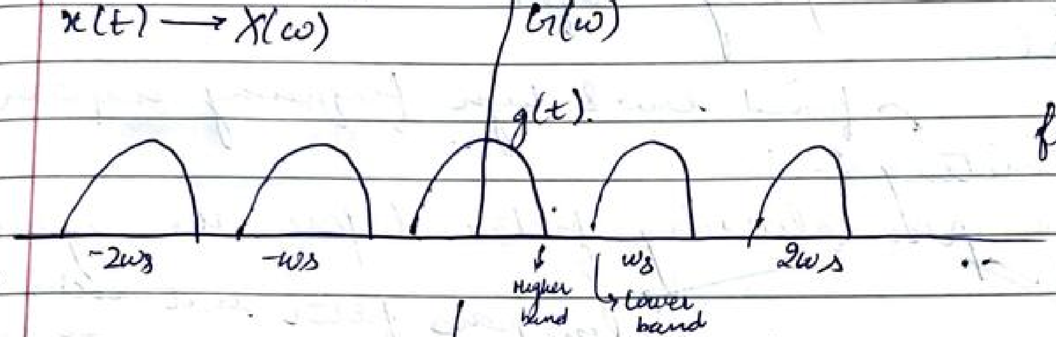
Definition:- Nyquist rate defines the rate of sampling

Sampling rate $\geq 2f_m$.

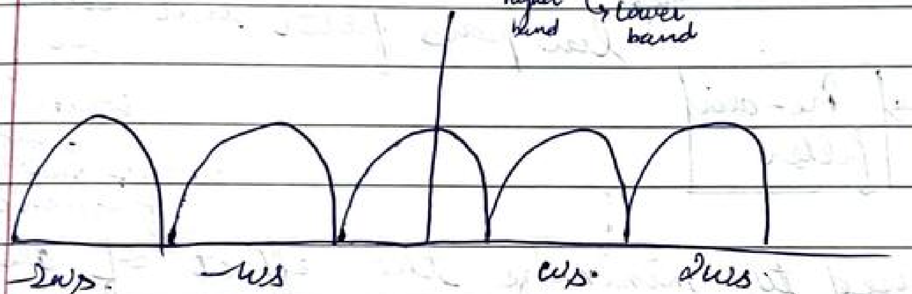
$$f_s \geq 2f_m$$

$$f_s = 2f_m$$

$x(t) \rightarrow X(\omega)$



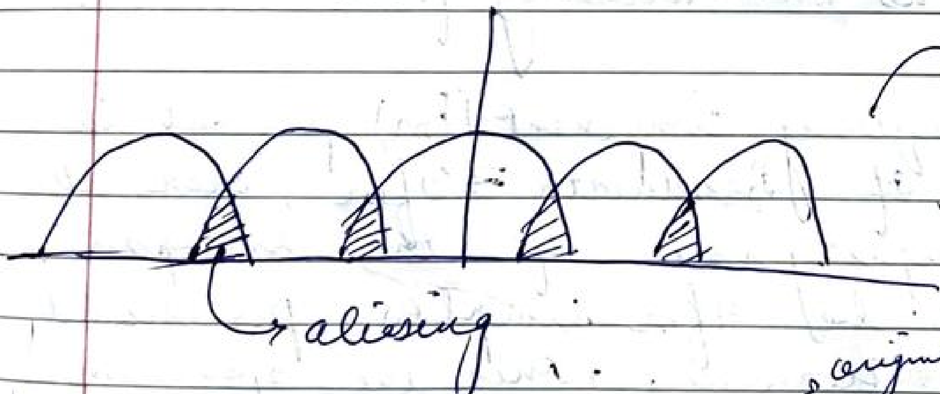
$f_s \geq 2f_m$.



$f_s = 2f_m$

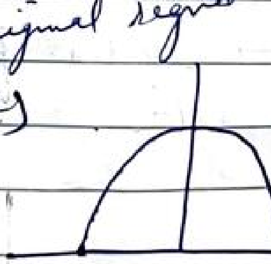
undersampling

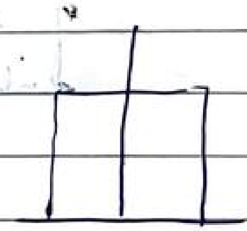
$f_s < 2f_m$.



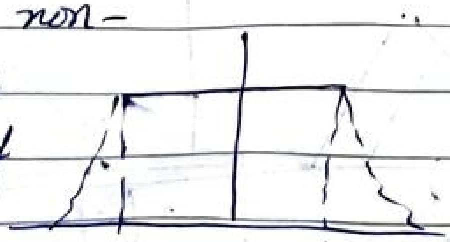
Case 1: If we have to reconstruct $x(t)$ (original signal) from $g(t)$ (sampled signal) there should not be any overlap between successive cycles. These non-overlapping repetitions are possible only if

Case 2: $f_s \geq 2f_m$. For non-overlapping repetitions





ideal response of filter



practical response of ideal filter

but touching each other with successive samples

$f_s = 2f_m$.

Case 3: when successive cycles of sampled spectrum will overlap each other, original spectrum cannot be expected.

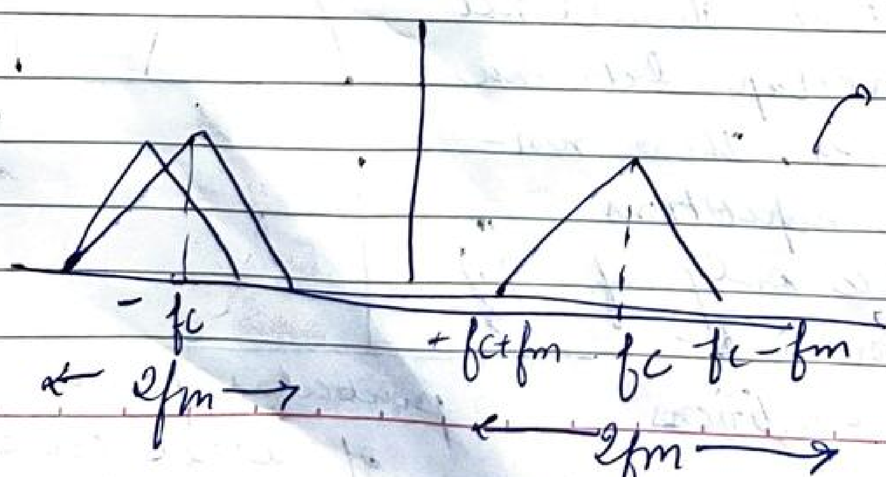
→ Nyquist Interval :→ $T_s = \frac{1}{2f_m}$
OR maximum sampling interval.

→ Bandlimiting :- (fixed low & high frequency components)
we use pre-aliasing filter before the signal is sampled
→ low pass filter that blocks all the frequencies which are above maximum frequency.

$x(t) \rightarrow$ Pre-alias filter

→ It is used to minimize the effect of aliasing.
→ also known as anti-aliasing filter

— Maximum frequency component (f_m) present in the signal. If Bandwidth → $2f_m$, then the minimum sampling rate of the bandpass signal must be $4f_m$ samples per second.
If f_c → center frequency, then the spectrum will be like.



→ spectrum of an arbitrary band pass signal

The frequency present in the band pass signal is $f_c + f_m$ to $f_c - f_m$. Highest frequency present in the bandpass signal is $f_c + f_m$. where center frequency is chosen greater than f_m. ($f_c > f_m$)

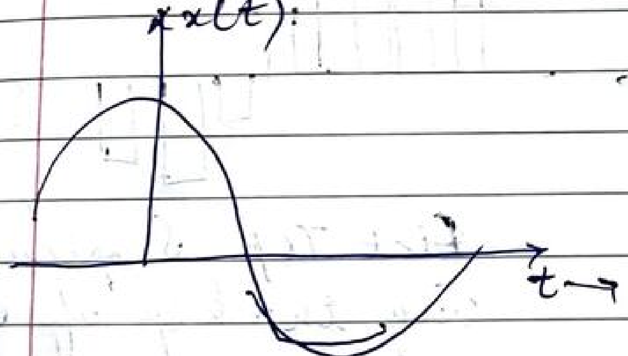
Sampling Techniques:-

Instantaneous sampling or ideal sampling or impulse sampling — Ideal sampling method :

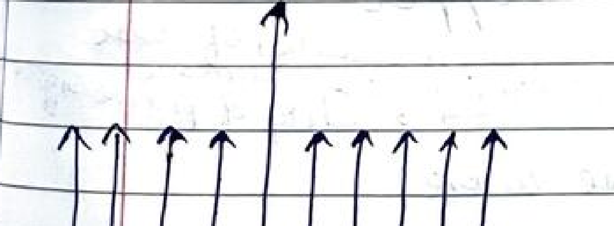
Natural Sampling

Flat top sampling

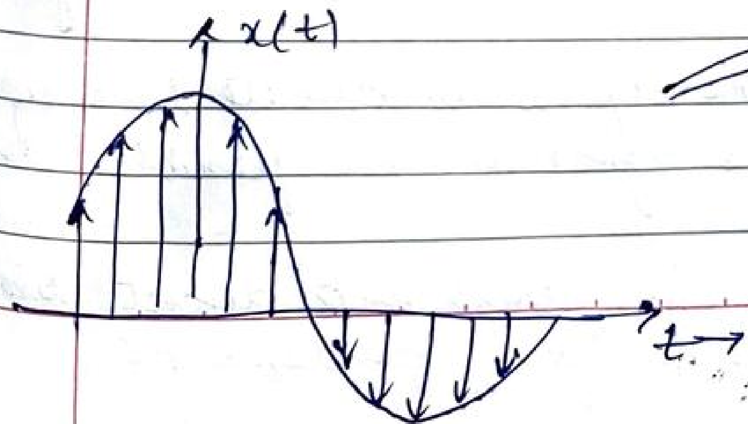
Natural Sampling, Flat top sampling — Practical sampling methods



In ideal sampling, the sampling function is a train of impulses, the input continuous time signal $x(t)$ is passed through a multiplier and the circuit that produces ideal sampling is called switching sampler.



⇒ Instanteous sampling



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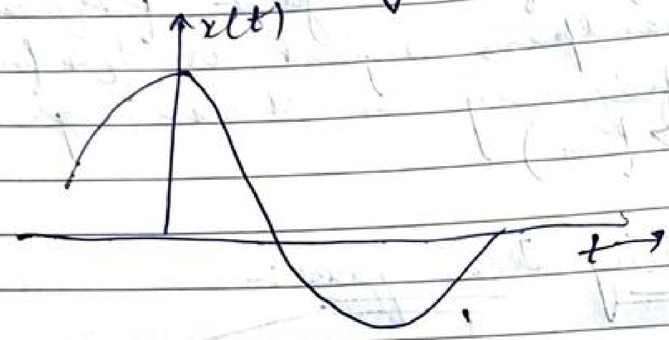
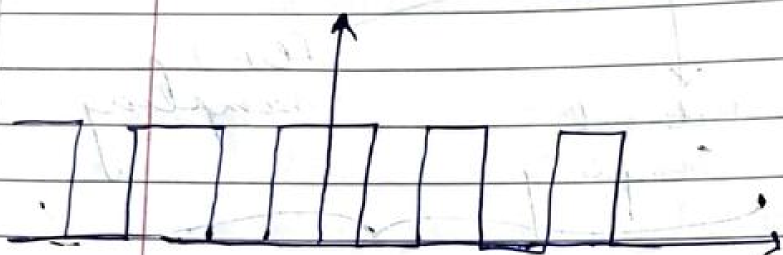
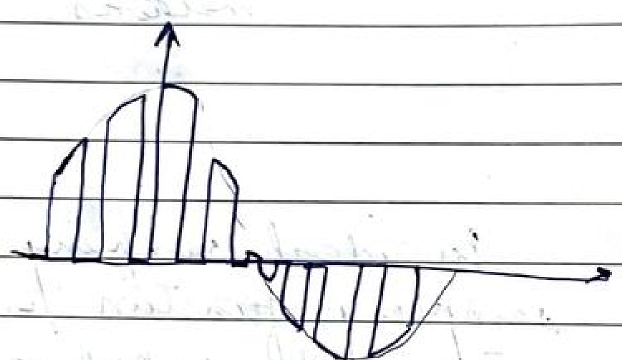
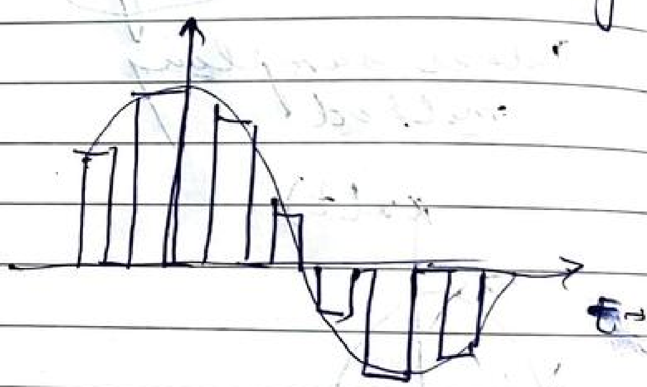
→ Natural Sampling:-





Flat-Top Sampling





Natural Sampling

Flat Top Sampling with the help of clippers

27/1/25

→ Analog Data → Digital Processing → reduction of noise
→ Storage is more
→ quality of picture is go

Signal → Discrete → Pulse wave.

→ To convert continuous time to discrete time by fourier transform

→ Shift from continuous time signal to discrete time sign

- Processing of discrete time signal is more flexible and preferable.
- Conversion of continuous time signal into discrete time signal can be achieved using a fundamental,

3/2/25 Pulse Modulation — Analog Pulse modulation (position, amplitude, width)
— Digital Pulse modulation (bits, words)

digital

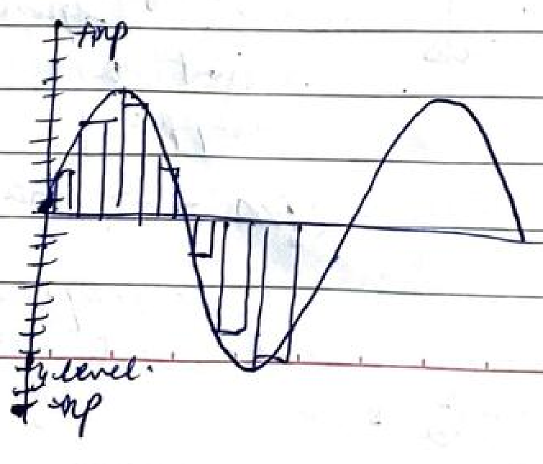
→ waveform coding technique/method is the conversion from analog signal to digital signal in bits using ADC where PAM signal after sampling can be converted into digital data and the transmission of digital data using transmission channel or transmission path,

→ Quantization :- is used to generate a fixed level or amplitude in which the sampled signal is approximated or rounded off to the nearest level.

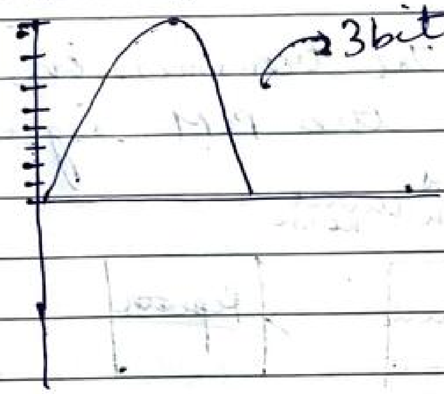
$\boxed{\text{PAM}}$ signal → digital bits

$\boxed{\text{Transmitter}}$ → $\boxed{\text{channel/media}}$ → $\boxed{\text{Receiver}}$ } 3 components

→ No. of levels are discrete set of levels according to the binary relation of the quantizer (
If Quantizer → (3 bit) - converts amplitude into 3 b
2^3 → 8 ~~bits~~ levels.
4 bit Quantizer → 2^4 → 16 levels.
5 bit " → 2^5 → 32 levels.



→ In quantization, the total amplitude range which the signal may occur is divided into a number of standard levels. Amplitude of $x(t)$ signal lies b/w $-mp$ to $+mp$. which is partitioned into 'L' no. of intervals. Each interval of magnitude $\Delta S = \frac{2mp}{L}$. Each level is quantized into its nearest level. The difference b/w the sampled signal & quantized signal is called quantization error.



→ To minimise the quantization error, the no. of levels can be increased while using high bit quantizer

→ With the increase of no. of levels, there are more bandwidth requirements to transmit the digital data.

→ Example of Pure digital Waveform coding technique :-

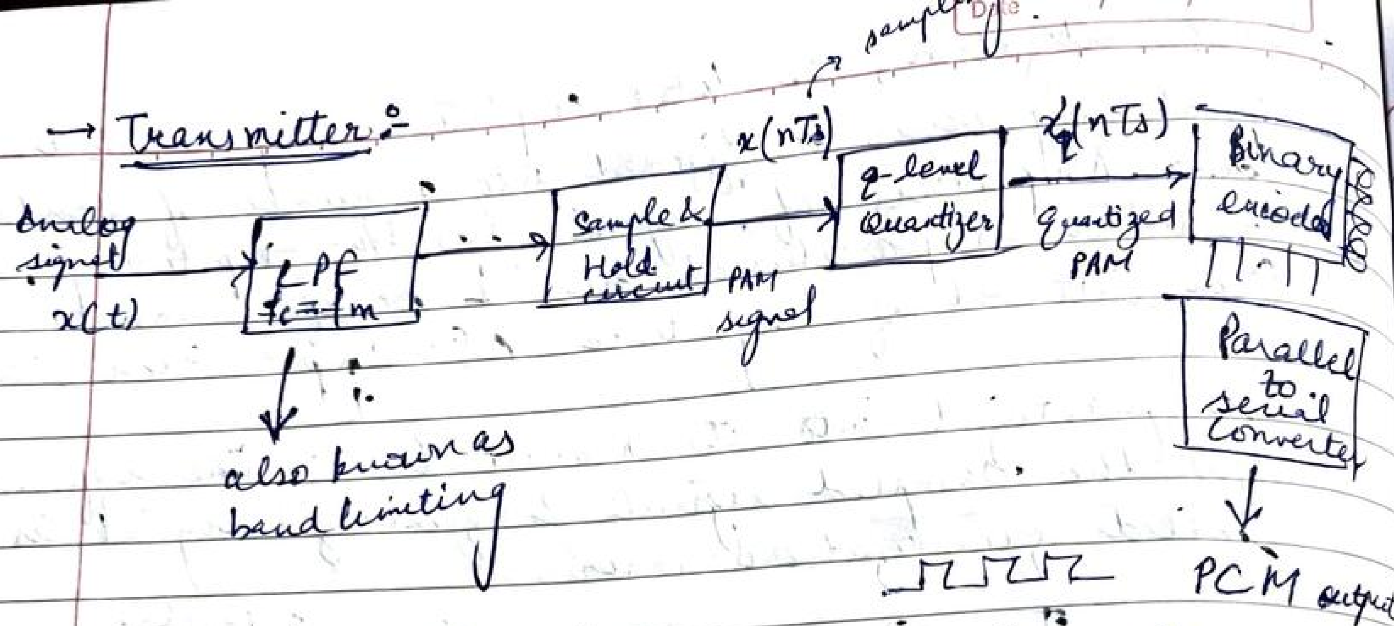
→ Pulse Code Modulation (PCM) :- PCM system consists a transmitter, transmission path & receiver. Transmitter consists sampling, quantizing & encoding operations.

Sampling $\xrightarrow{\text{PAM Signal}}$ Quantizing → encoding

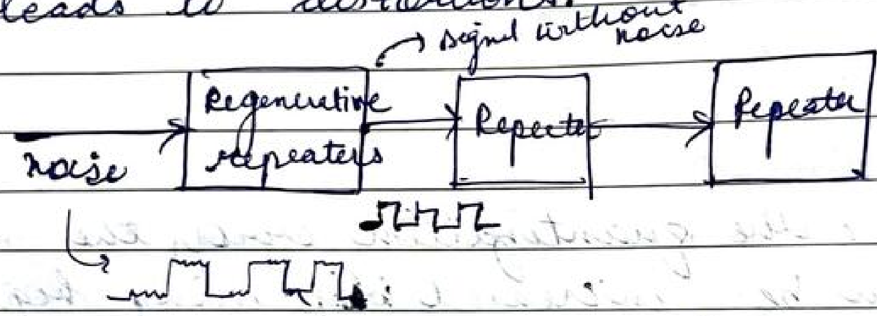
5.4 → 5

5.8 → 6

→ Transmitter:-



→ Transmission path:- The transmission channel introduce noise to the PCM signal that leads to distortions.



Repeaters receive the signal, suppress the noise, enhance the signal level then retransmit it.

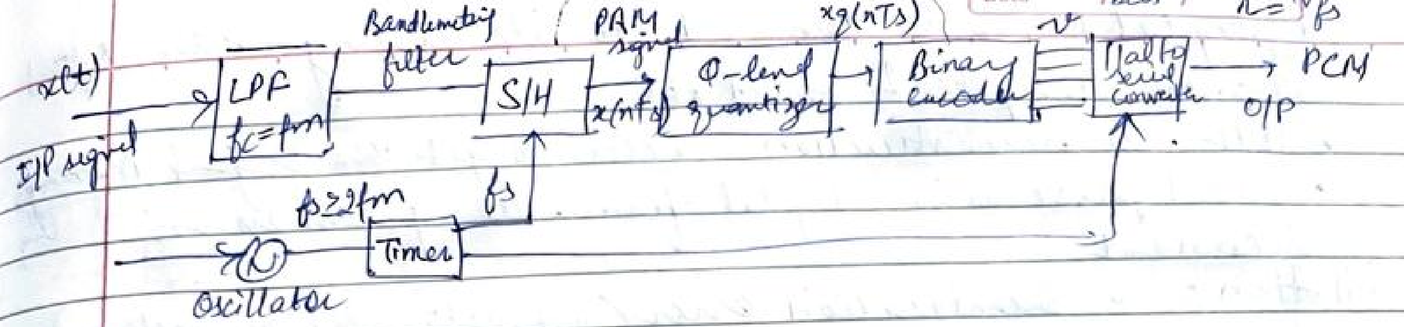
→ Receiver:-

|2|25

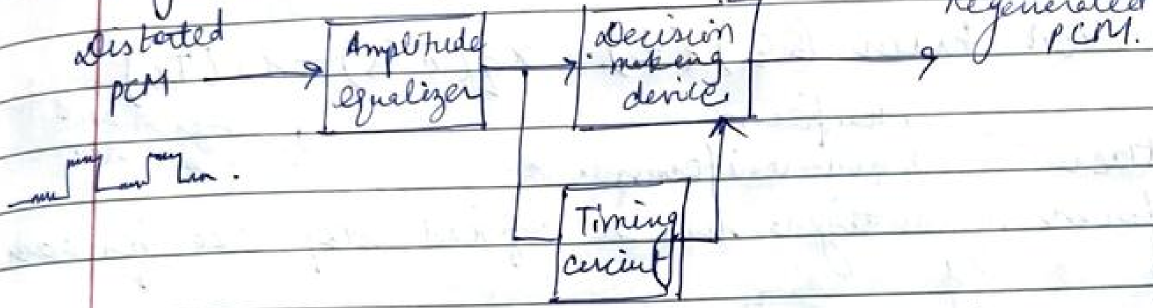
→ PCM Generator/Transmitter:-

The O/P of PCM transmitter or generator will be coded digital form or digital pulses with constant amplitude & width, position

AOC .

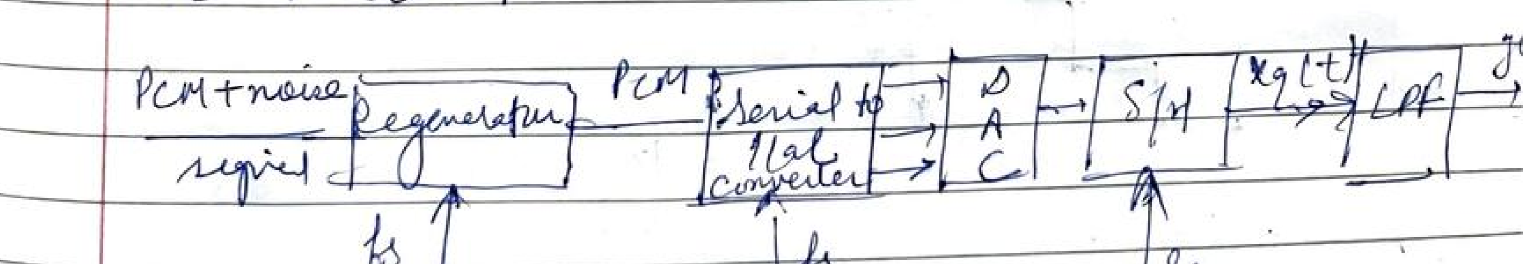


→ Regenerative Repeaters:-



During the transmission of PCM signal the ability to control the effects of noise & distortion can be achieved through regenerative repeaters which perfoms i) Equalization: it shapes the distorted PCM waveform to complensate for the effects of amplitude & phase distortions. (ii) Timing ckt produces a periodic pulse train which is derived from input PCM pulses. (iii) Decision Making use this pulses for sampling the equalized pulses where the SNR (signal to noise ratio) is maximum. It decides whether the PAM wave at its input has 0 or 1 while comparing with reference or threshold level.

→ PCM Receiver:-



• Regenerator reshapes the pulses & ~~removes the noise~~

Date / /

- Digital pulse is converted to its analog value ckt with sample & hold ckt
- LPF is reconstruction filter to get the original message
- PCM pulse is in digital form. Its parameters are constant.

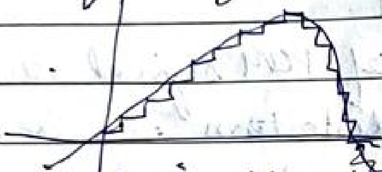
Limitation :- • Quantization error (estimates the parameters)
• Better than Pulse Modulator.

→ Quantization Error (QE) $(\varepsilon) = x_q(nT_s) - x_n(T_s)$

Quantizer is used for quantization.

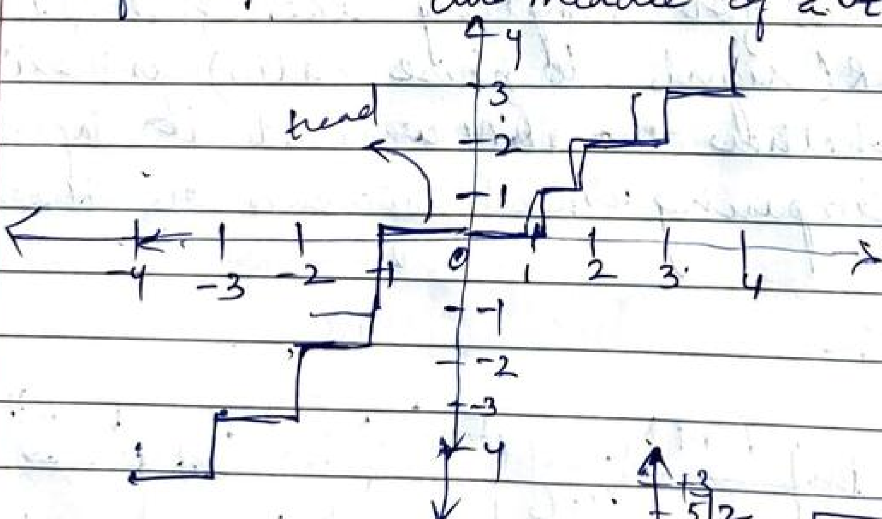
Quantizer → uniform
→ non-uniform :

1. A uniform quantizer has a fixed step size for each instant

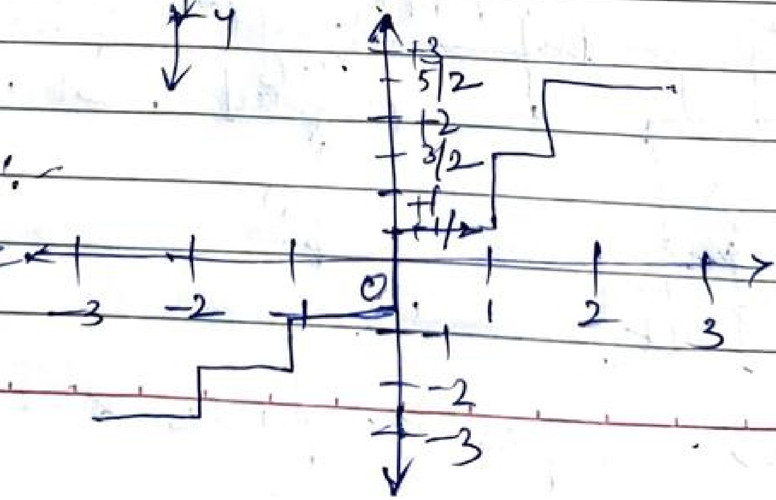


2. It compares discrete time input with its fixed digital level. It assigns anyone of the digital level which results in min distortion or error.

Uniform : → Symmetric mid rise
→ Symmetric mid ~~trad~~ type : origin lies in the middle of a tread of staircase signal



Symmetric mid rise :-



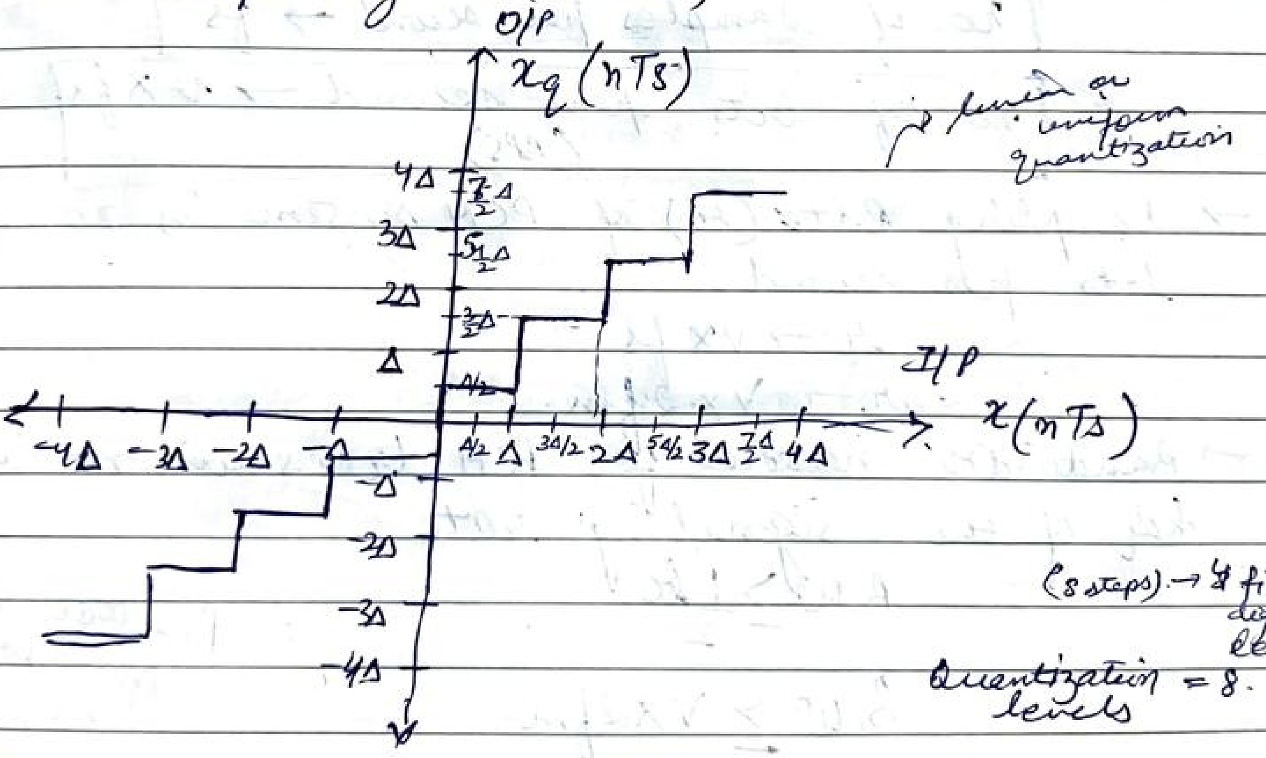
Mid Rise :- Origin lies in the middle of a rising part of a staircase.

Both type of quantizer are symmetric about the origin.

10/2/25

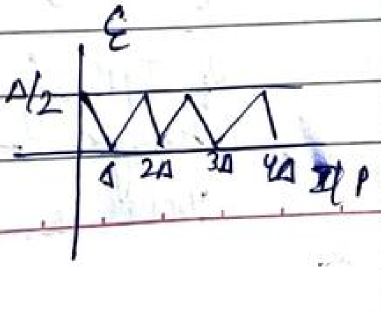
2) Non-uniform → step size may vary or adaptive

For uniform quantizer, mid rise type waveform with step size Δ (delta)



Assume that input $x(nTs)$ varies from -4Δ to 4Δ with step size Δ. Fixed digital levels are available at $\pm\frac{\Delta}{2}$, $\pm\frac{3}{2}\Delta$, $\pm\frac{5}{2}\Delta$, $\pm\frac{7}{2}\Delta$

→ Maximum Quantization error :- (in PCM) $\pm\frac{\Delta}{2}$



$$Q\mathcal{E} \Rightarrow \mathcal{E} = x_q(nT_s) - x(nT_s)$$

$$\mathcal{E}_{max} = \left|\frac{\Delta}{2}\right|$$

Date / /

Uniform quantizer with incorrect quantization characteristics. If the step size varied according to the input signal to reduce the quantization error, non-uniform quantization can be used.

→ Bandwidth of PCM system?
If the quantizer uses v no. of bits then the no. of quantization levels are 2^v

No. of quantization levels $= 2^v$
Each sample is converted in v binary bits so the $\boxed{\text{no. of bits per sample} \rightarrow v}$

$\boxed{\text{no. of samples per second} \rightarrow f_s}$

$\boxed{\text{no. of bits per second (BPS)} \rightarrow v \times f_s}$

→ Sampling Rate (r) of PCM system is no. of bits per second.

$r \rightarrow v \times f_s$

$r \rightarrow v \times 2f_m$.

→ Bandwidth needed for PCM transmission is half of the signaling rate

$B.W \geq \frac{1}{2} r$.

$B.W \geq v \times 2f_m$

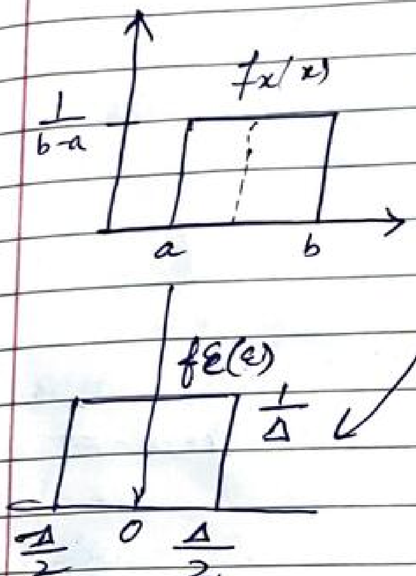
Practice numericals from book.

→ Quantization error for linear or uniform quantization :→

If I/P signal $x(nT_s)$ is a continuous amplitude in range $-x_{max}$ to x_{max}.
Total amplitude range :→ $x_{max} - (-x_{max}) \Rightarrow 2x_{max}$

If this total amplitude range is divided into q levels. Δ ??

$$\Delta = \frac{2x_{max}}{q}$$



If signal $x(t)$ is normalised to minimum and maximum value to 1. If step size is sufficiently small, then quantization error will be uniformly distributed random variable. maximum error will be

$$\varepsilon_{max} \rightarrow \left|\frac{\Delta}{2}\right|$$

PDF → Probability distribution function for a uniformly distributed random variable b/w range a & b

$$f_x = \begin{cases} 0 & \text{for } x \le a \\ \frac{1}{b-a} & a \le x \le b \\ 0 & \text{for } x > b \end{cases}$$

PDF (Probability distribution function f_ε for quantization error ε)

$$= \begin{cases} 0 & \varepsilon < -\frac{\Delta}{2} \\ \frac{1}{\Delta} & -\frac{\Delta}{2} \le \varepsilon \le \frac{\Delta}{2} \\ 0 & \varepsilon > \frac{\Delta}{2} \end{cases}$$

→ Quantization error has 0 average value bcoz it's uniform distribution function.

SNR (Signal to Noise Ratio) = $\frac{\text{Signal Power}}{\text{Quantization Noise Power}}$

(Normalised) (Normalised)

Noise Power (P) = $\frac{V_{noise}^2}{R}$

→ mean square value of random variable x is

$$\bar{x}^2 = E(X^2) = \int_{-\infty}^{\infty} x^2 f_x(x)\, dx.$$

$$\Rightarrow \int_{-\Delta/2}^{\Delta/2} \epsilon^2 f_\epsilon(\epsilon)\, d\epsilon.$$

$$\Rightarrow \int_{-\Delta/2}^{\Delta/2} \epsilon^2 \frac{1}{\Delta}\, d\epsilon.$$

$$\Rightarrow \frac{1}{\Delta}\left[\frac{\epsilon^3}{3}\right]$$

$$\Rightarrow \cancel{\frac{1}{3\Delta}\left[\frac{\Delta^3}{8}\right]} \Rightarrow \frac{\Delta^2}{12}$$

$$E(\epsilon^2) = \frac{\Delta^2}{12}$$

load resistance

$$v^2_{noise} = \frac{\Delta^2}{12}$$

If R = 1 Ω for normalised quantized power.

$$P = v^2_{noise}$$

$$SNR \Rightarrow \frac{P}{\Delta^2/12}$$

If $q \rightarrow$ quantization level, then step size

$$q \text{ levels} \Rightarrow 2^v$$

Step size

$$\Delta = \frac{2x_{max}}{q}$$

$$\Delta = \frac{2x_{max}}{2^v}$$

Imp!!
PCM block diagram
derivation levels
quantization
Signal to Noise

$$S/N \rightarrow SNR = \frac{3P}{x^2_{max}} \cdot 2^{2v}$$

signal to noise power ratio of quantizer, increases exponentially, if input is normalised then $x_{max} = 1$

$$S/N \Rightarrow 3P \cdot 2^{2v}$$

Numerical

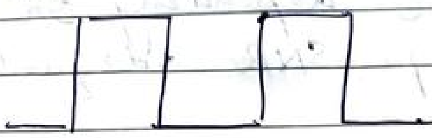
$$(S/N)_{dB} \Rightarrow (4.8 + 6v)\, dB$$ → for PCM system

→ signal Power P is normalised if $\boxed{P \leq 1}$

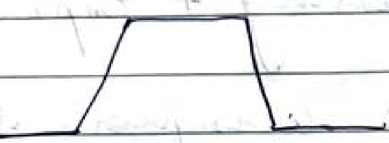
$$(S/N)_{dB} < (4.8 + 6v)\,dB.$$

11/2/25 **Noise effects in PCM**

Ideally :-



Practically:-



PCM is still uneffective as it doesn't contain any information of width & position.

→ **Robust Quantizer** (Non-uniform Quantization) //non-linear

4 bit Quantizer → 2^4 levels
⇒ 16 levels.

$$\Delta = \frac{2x_{max}}{q} \Rightarrow \frac{2}{16} \Rightarrow \frac{1}{8}.$$

$$\varepsilon_{max} = \left|\frac{\Delta}{2}\right| \Rightarrow \frac{1}{16}.$$

Quantization error wrt to full voltage scale
Amp 16 → 7%
Amp 8 ⇒ 12%
Amp 4 ⇒ 25%
Amp 3 : 35%
2 ⇒ 50%

Quantization error is $\frac{1}{16}$ part of full voltage range.

For full range voltage, if full range voltage is 16 V. maximum error is , volt i.e. 6 %. For the low signal amplitudes 2 or 3 Volt, ε_{max} of 1 volts is quite high like 50% or 30%. This occurs due to uniform quantization. Signal to Noise ratio must be constant over a wide range of I/P power levels. A quantizer that satisfies these requirements are called robust quantizer.

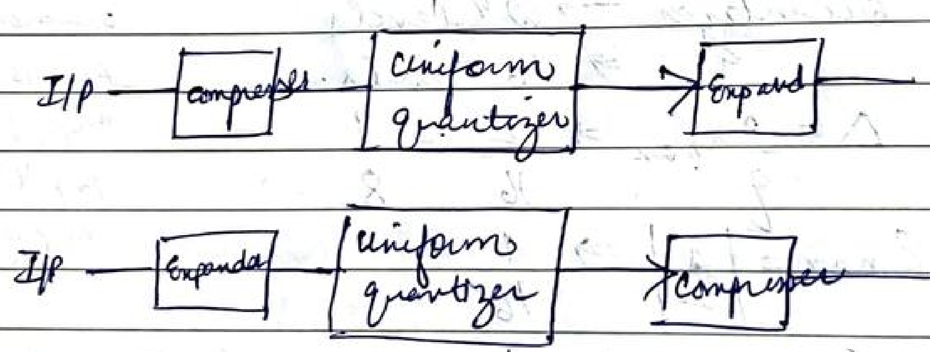
Low power signals P<<1, noise can be minimised by keeping step-size small. In non-uniform quantization

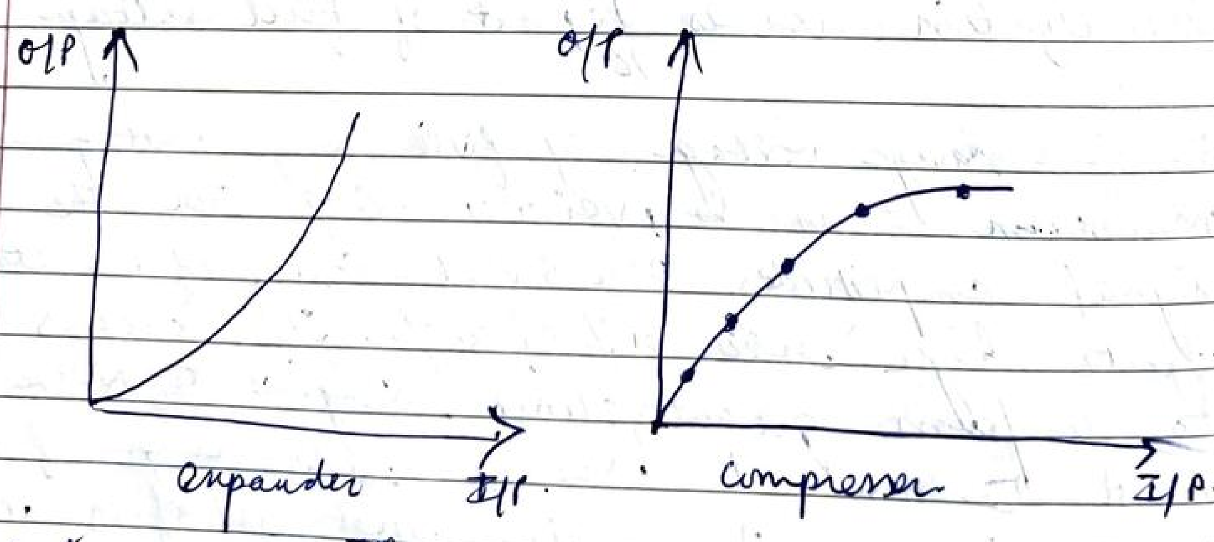
- its robust in nature
- characteristics are non-linear
- step size variable depending on input signal amplitude to keep SNR adequately high.

* Companding :- Compressing + Expanding

→ Companding is used to improve the SNR of weak signal.

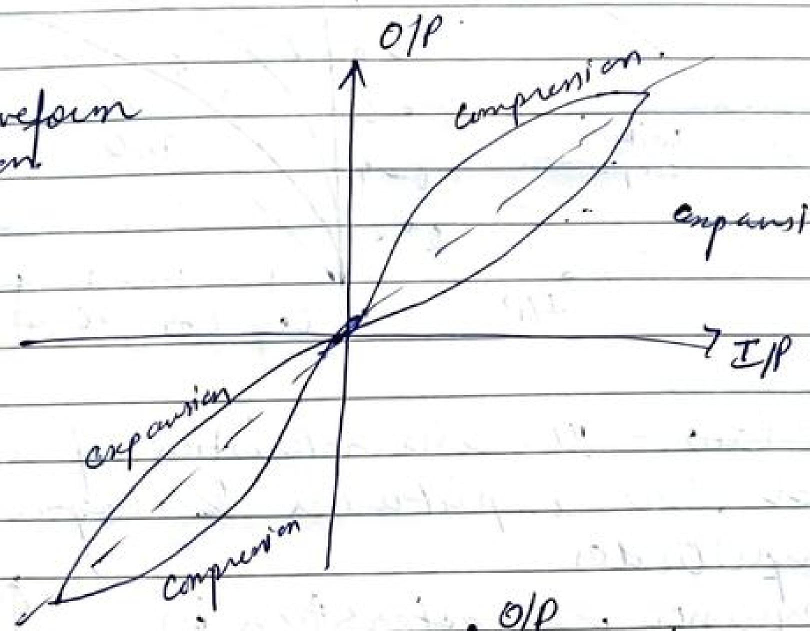
→ weak signals are amplified & strong signals are attenuated before applying to a uniform quantizer.



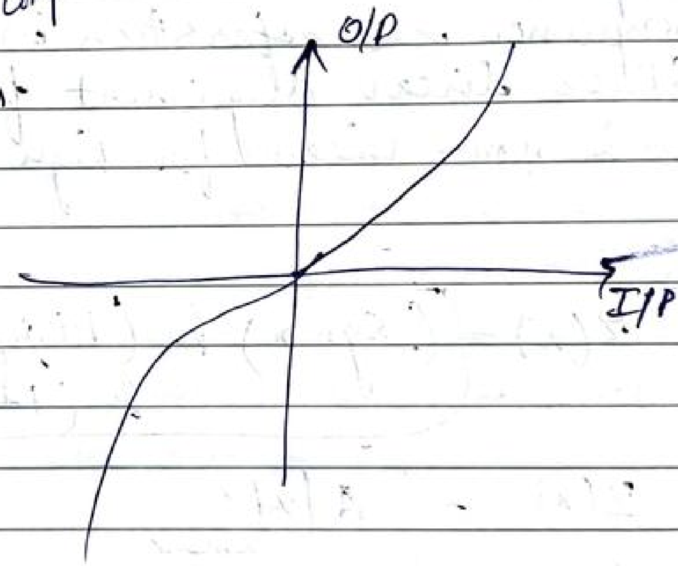


→ gain of weak I/P signals is high but for strong signals it is done in case of compression. weak signals artificially boosted to improve SNR. Compression at transmission & expansion at receiver is known as companding.

Companding waveform for compression



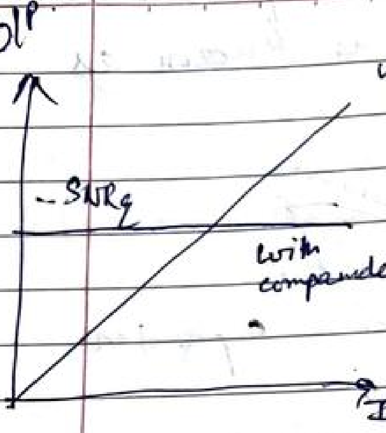
companding waveform for expansion.

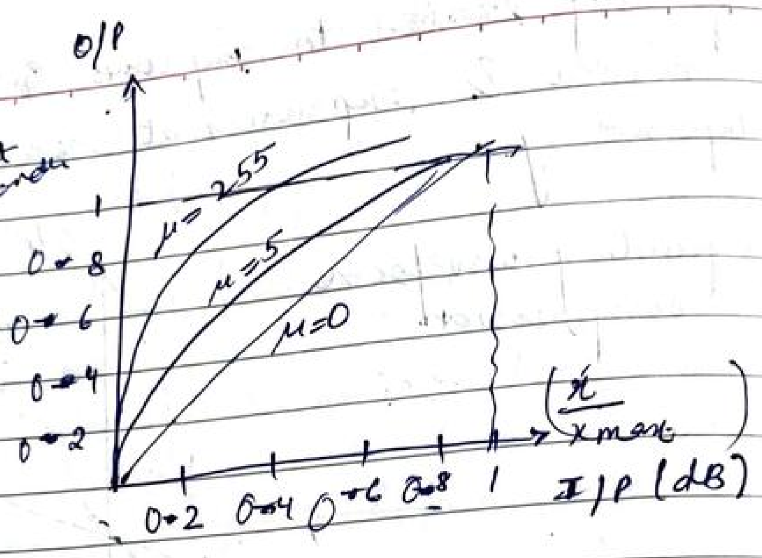


→ Compander characteristics due to inverse nature of compresser & expander, the overall characteristics of compander is straight line, Hence, all the boosted signals are brought back to their original amplitudes.

→ Companding can be done using μ-law
A-law

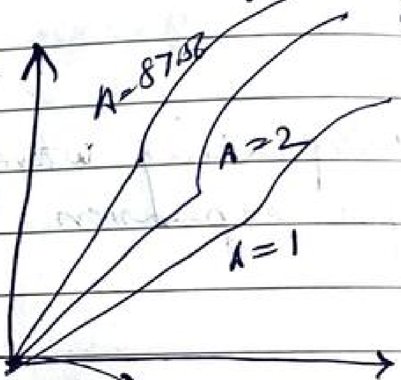
→ Compresser characteristics is continuous, linear for small input values & logarithmic for higher values.





~~A law~~ :- The characteristics of compander linear for low amplitudes & logarithmic for higher amplitudes

A law - Compresser characterstics is P-SPICE linear alignment for low & non-linear for high



$$\boxed{z(x) = (\text{Sgn } x) \frac{\ln(1+\mu(x/x_{max}))}{\ln(1+\mu)}}$$

$$\frac{z(x)}{x_{max}} = \begin{cases} \dfrac{A|x|/x_{max}}{1+\log_e A} \\[2ex] \dfrac{1+\log A|x|/x_{max}}{1+\log_e A} \end{cases}$$

$\frac{|x|}{x_{max}}$ lies b/w 0 & 1

$0 < \frac{|x|}{x_{max}} < 1$

What is PCM
PCM advantages
* Disadvantages
* applications

Advantages → * used for telephony systems
* storage
* noise effects are more less

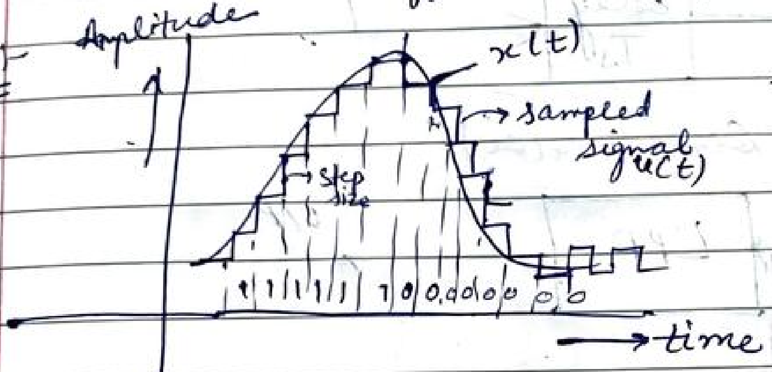
Disadvantages/limitations → not good for low amplitude SNR ↓
→ Bandwidth requirement & higher data rate
→ due to sampling companding complexity increases

12/2/25 **Delta Modulation** → used to overcome the limitations of PCM system.

- Bandwidth limitation overcome. → multiple bits per sample.

In Delta modulation, only single bit per sample is transmitted (either 1 or 0). Present sample value is compared with previous sampled value. The difference is transmitted.

eg:-



Step size is fixed.
Δ is fixed

Input signal $x(t)$ is approximated to step signal by delta modulator then difference between $x(t)$ & stair case approximated signal is compared b/w $+\Delta$ & $-\Delta$. If $+\Delta$ the $x(t) > u(t)$ increase the step by one size otherwise decrease the step by one size $-\Delta$ (diff is negative).

$+\Delta \rightarrow$ transmit binary 1

$-\Delta \rightarrow$ transmit binary 0.

→ If error at present sample $e(nT_s) \rightarrow x(nT_s) - \hat{x}(nT_s)$

$\hat{x}(nT_s) \rightarrow$ last sample / previous sample approximation of stair case waveform.

→ If $u(nT_s)$ is present sample approximation, then $u((n-1)T_s) \Rightarrow \hat{x}(nT_s)$

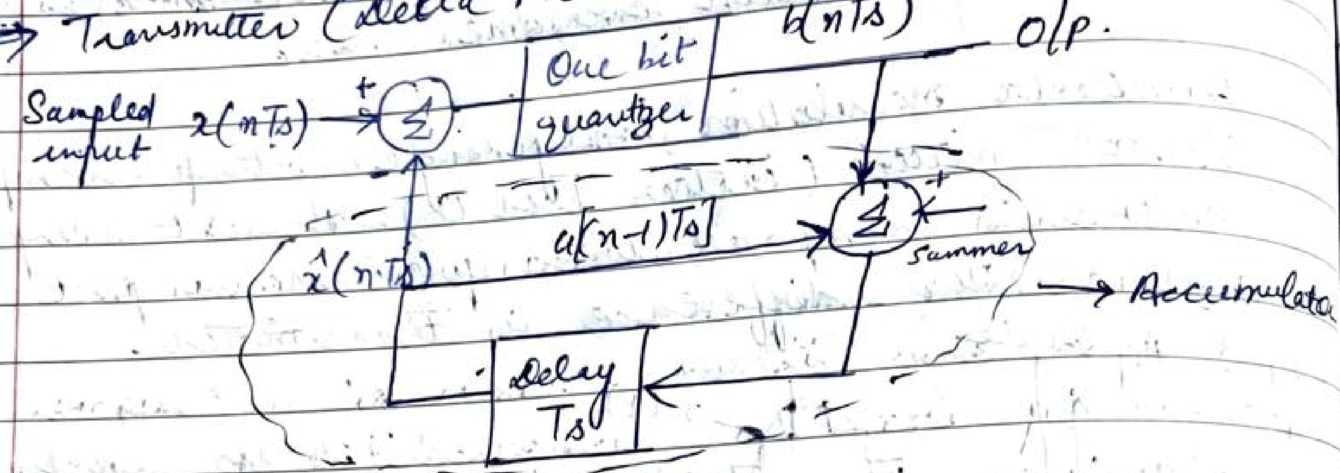
→ signal $b(nT_s) \Rightarrow \Delta \, \mathrm{Sgn}\left[e(nT_s)\right]$ depending on the sign of error i.e. $e(nT_s)$, the sign of step size Δ is decided.

$$b(nT_s) \Rightarrow \begin{cases} +\Delta \rightarrow x(nT_s) \geq \hat{x}(nT_s) = 1 \\ -\Delta \rightarrow x(nT_s) < \hat{x}(nT_s) = 0 \end{cases}$$

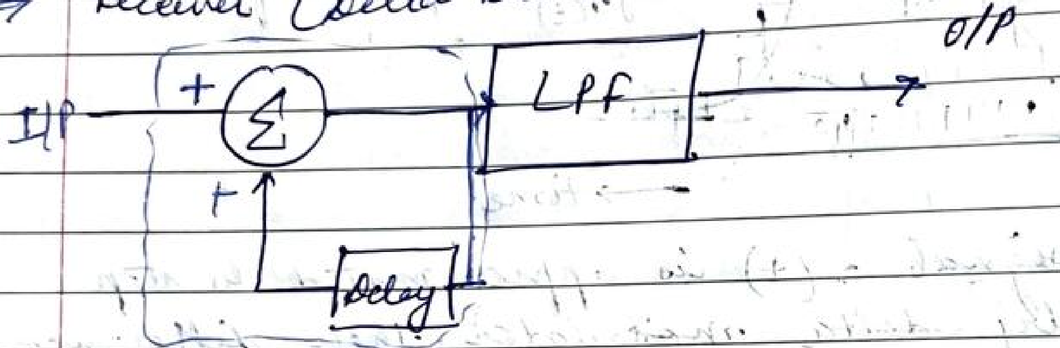
Binary Signal

→ Delta Modulator & Demodulator Circuit

⇒ Transmitter (Delta Modulator)



⇒ Receiver (Delta Demodulator)



The summer in the accumulator adds quantizer O/P $\pm\Delta$ with the previous sample approximation. Previous sample approximation is restored with the delay of 1 sample period (T_s) depending on the sign of error at present sample. 1 bit quantizer generates O/P $+\Delta$ or $-\Delta$.

At the receiver accumulator generates the staircase O/P and delayed by 1 sampling period then added to the input signal. If I/P is binary 1, it will

LPF is used to smoothen the staircase signal to reconstruct original message

→ used in telephony system → coding techniques

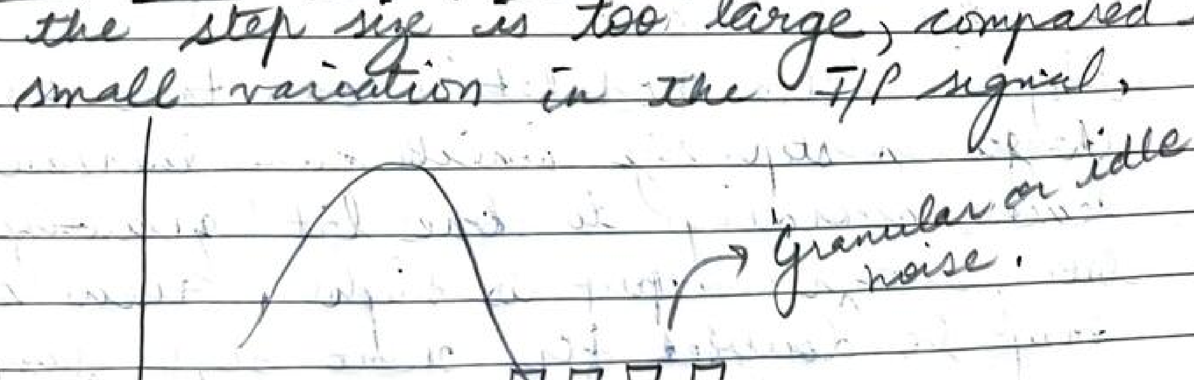
→ circuitry is simple than PCM system
→ Bcoz single bit is used, neither more bandwidth is required nor sampling rate.

Drawback :- Due to the fixed step size in delta modulation slope overload error/ distortion and granular/noise/error/ distortion / Idle ~~Idle~~.

* Slope overload error:-
Due to large dynamic range of I/P signals. Rate of Rise in amplitude is so high that staircase signal cannot approximate it.

step size Δ becomes too small for sampled or approximated value to follow input signal. To overcome, adaptive delta modulation is used.

* Granular/ noise /error/distortion:-
When the step size is too large, compared to the small variation in the I/P signal.



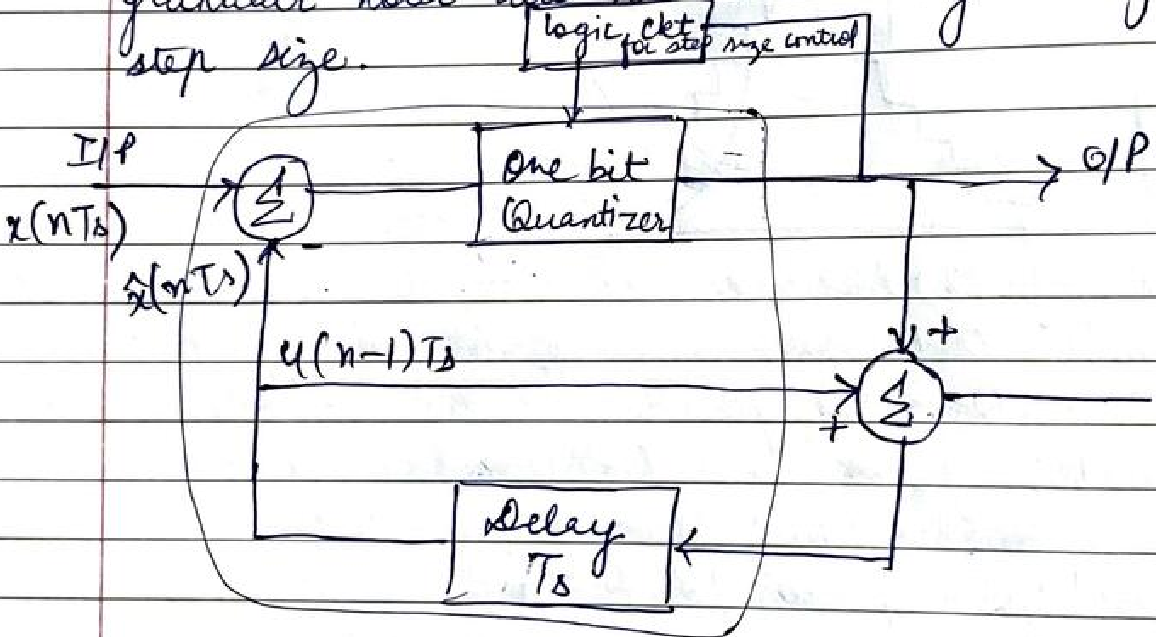
Sanjay Sharma → Book

18/2/25

To overcome the drawbacks of delta modulation, adaptive delta modulator or continuous variable slope delta modulation (CVSDM)

→ <u>Adaptive Delta Modulation</u>:-

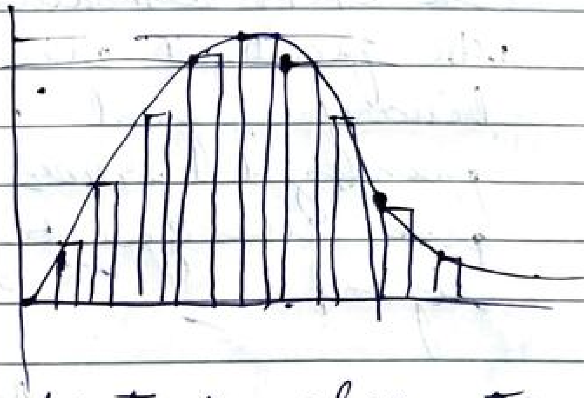
→ Adaptive Delta Modulation, step size can be increased or decreased from the previous response. Whenever consecutive ones are transmitted, step size can be increased & vice versa. granular noise can be avoided by using small step size.



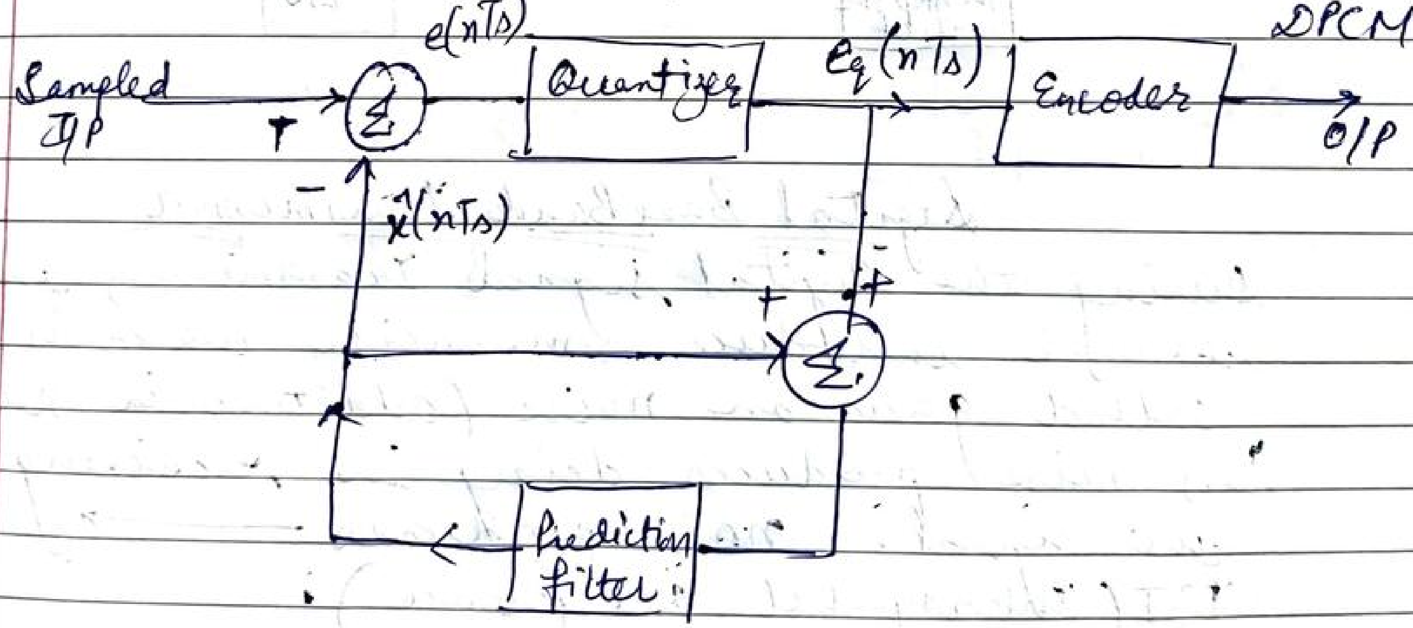
Adaptive Delta Modulation can take continuous changes in step size which can increase or decrease according to One bit quantizer. If one bit quantizer output is high, then step size may be doubled for next step sample.

→ DPCM (Differential Pulse Code Modulation)

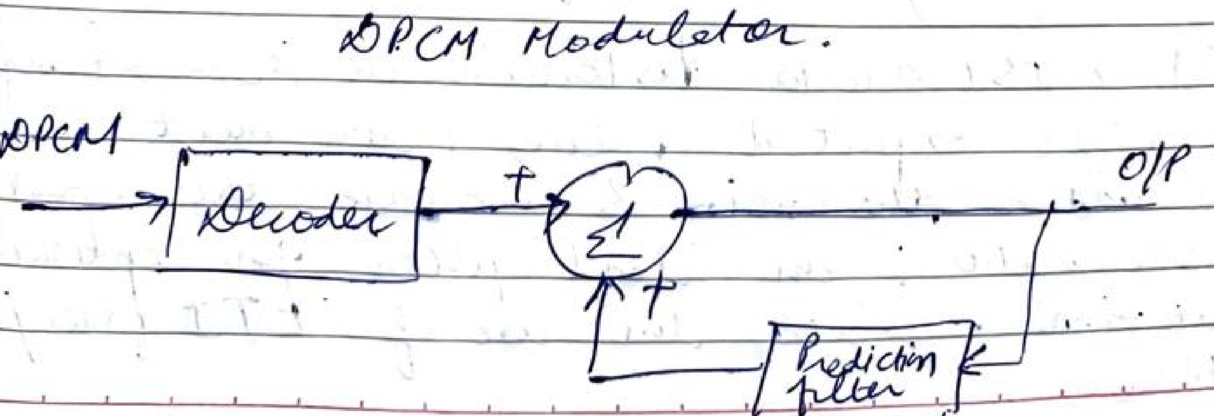
In information signal, during the transmission, there may be chances of same information in consecutive samples.



Samples of signals are highly correlated with each other because any signal does not changes fastly. Adjacent samples may carry the same information called Redundant samples to enhance the overall bit rate this redundancy can be reduced using differential PCM in which a prediction filter is used.



DPCM Modulator.



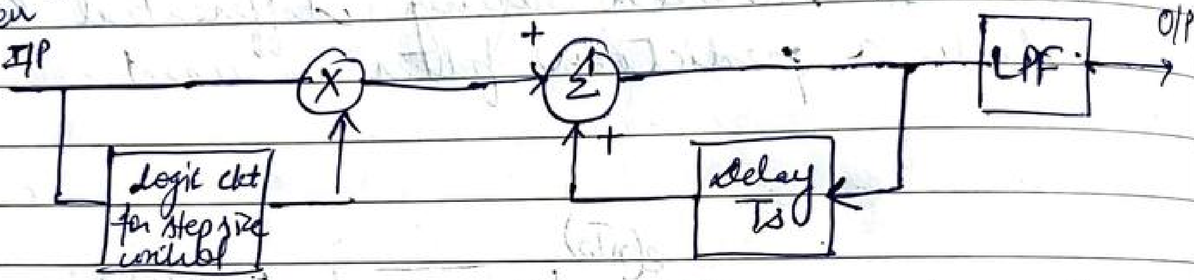
DPCM Demodulator

Date / /

In Modulator, present sample is predicted from the past samples. Prediction is close to actual sample value.
In DPCM Demodulator, decoder first reconstruct the quantized error signal from incoming binary signal. Prediction filter O/P & quantized error signals are summed up to give the quantized version of the original signal.

Adaptive Demodulator circuit

Comparison
Application
Drawback & their overcome.



Digital BaseBand Transmission

During the Digital Signal Transmission, the channel introduce some noise which is called gaussian noise (additive in nature). This noise produces delay in receiving the signal. noise produces ⟶ ISI (Intersymbol Interference).

ISI :→ ISI due to distortion in received signal from the expected value. Care must be taken to reduce the noise & increase the signal to noise ratio. An ideal filter for optimum detection involves the use of LTI filter / matched Filter.

LTI → Linear Time Invariant.

→ Line Coding Formats :-

24/2/25 To avoid ISI, Line coding is used.

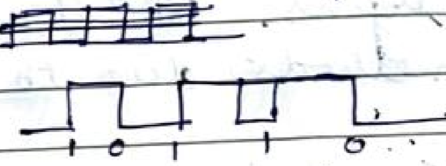
Line coding is pulse waveform representation of digital data received from source coding techniques / channel coding. ~~This comes from~~ There are two types of waveform representation

RZ → Return to zero

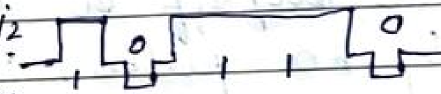
NRZ → Non-Return to zero

In digital transmissions, output of multiplexer is coded into pulses or waveforms for binary data.

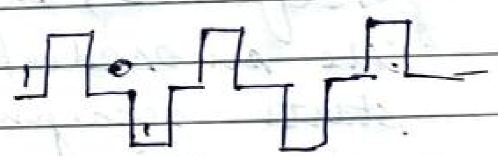
half-width pulses RZ → On-off RZ



→ Polar RZ



→ Bipolar RZ / → Alternate Mark Inversion.



alternate 1's in alternate polarities

→ Pseudoternary Representation.

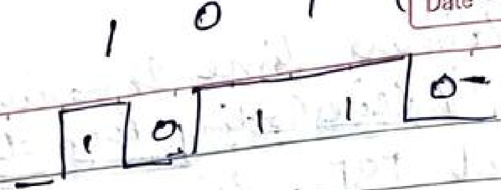
RZ → non transparent method

Definition :-

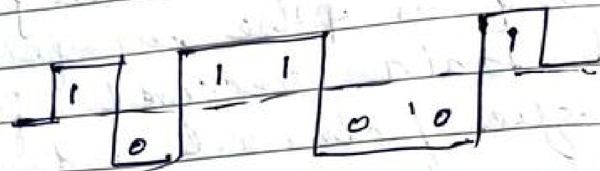
In AMI technique, pulse representing consecutive one's between alternate in sign if an error is made in the detection of pulses, the received pulse sequence will violate the AMI rules which can be detected easily. Best suited for error detection.

→ NRZ (Non Return to Zero) full-width pulses

→ On-Off NRZ :-



→ Polar NRZ :-



→ In NRZ method, half width pulses or full ~~bit~~ width pulses are used. The pulse amplitude value have constant amplitude. It donot have a chance to go to zero before the next pulse begins. NRZ coding schemes are best efficient methods due to transparent technique.

→ In On/Off or Bipolar RZ, a zero is transmitted by no pulse. If there are too many zeroes in a sequence, there is no signal at input & the sinusoidal output of resonant circuit starts decaying causing errors in the timing information. Transparent line codes i.e. NRZ line codes in which the pattern does not effect the accuracy of the timing information

→ Unipolar RZ & NRZ formats :-

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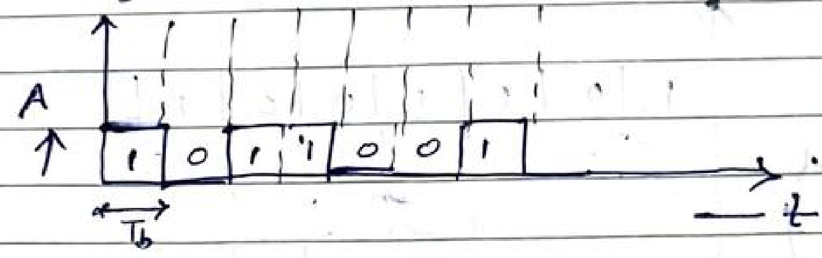
Unipolar RZ waveform has a single polarity & On-off method waveform has a zero value for 0 transmission & A volts for 1. A volts is present for $T_b/2$ and remaining half waveform for return to zero value.

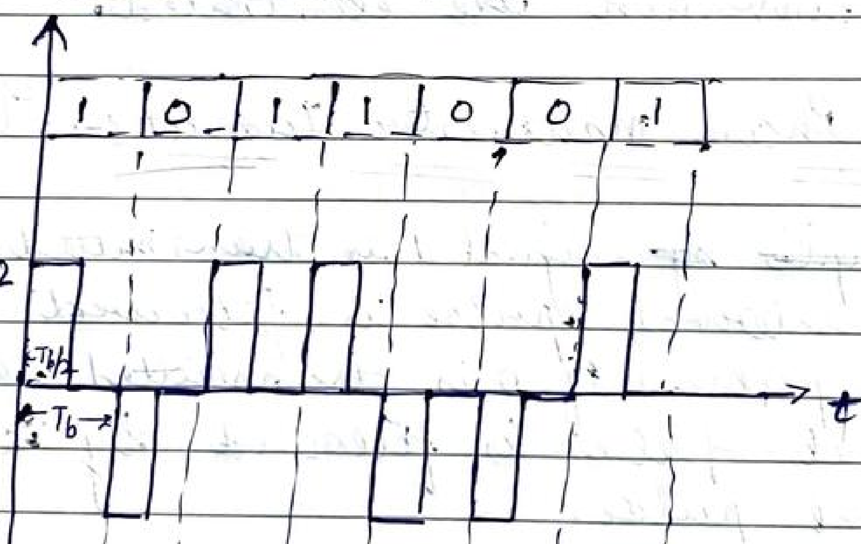
Unipolar NRZ



unipolar NRZ formatting does not have a separation. We require synchronization circuit.

→ Polar RZ & NRZ formats:-

Polar RZ.



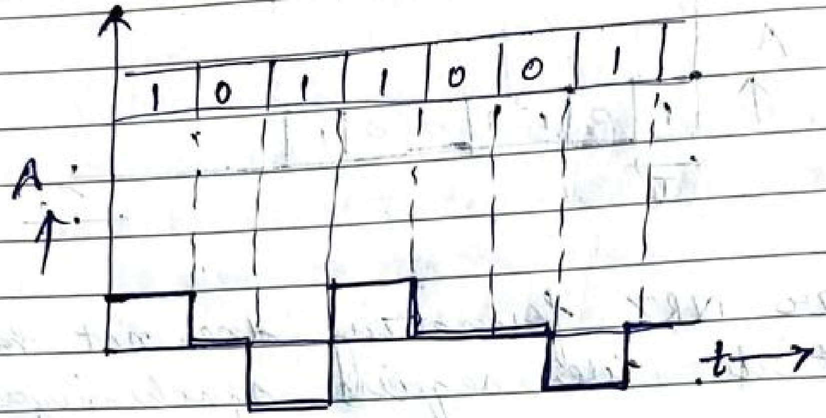
Good!

Polar NRZ.

Pulse is transmitted only on half duration in Polar RZ. whereas in NRZ format, 1 is a

positive polarity and 0 is a negative polarity.
In NRZ format, pulse is transmitted for full duration.

→ Bipolar NRZ (AMI) :→

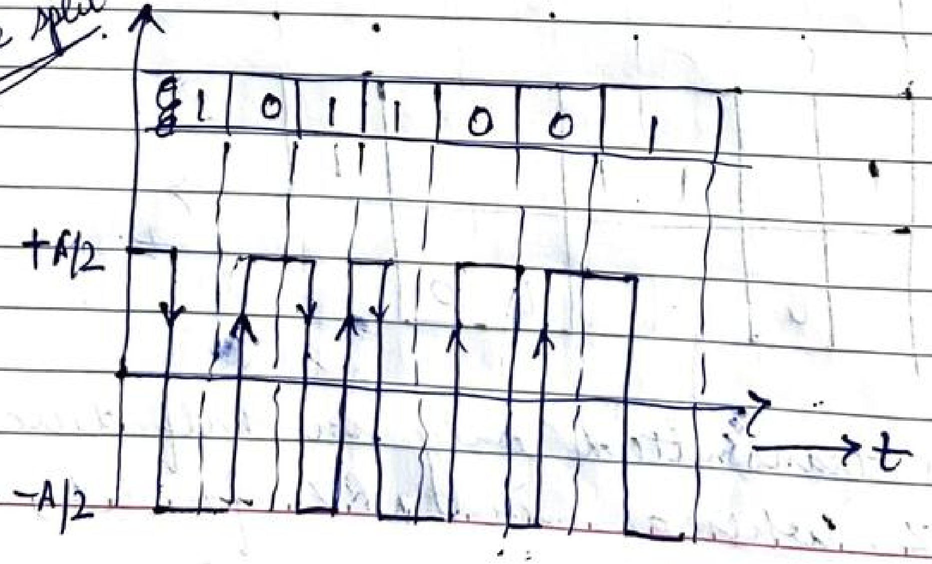


In AMI technique, ambiguities due to transmission sign inversion are eliminated.

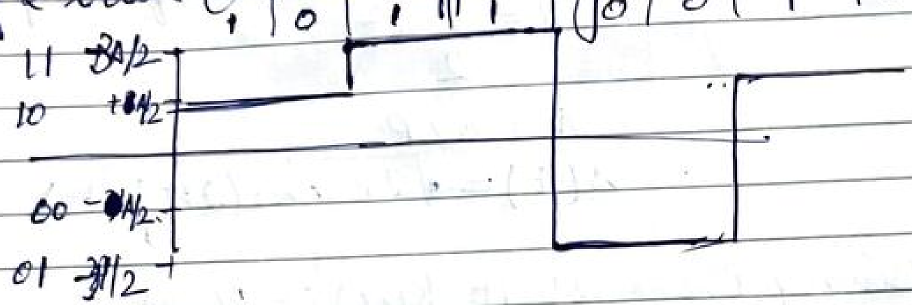
→ Split Phase Manchester Format :- efficient method than all.
(NRZ)

If ~~simple~~ ~~sig~~ signal 1 is transmitted then a +ve half interval pulse is followed by a −ve half pulse. If 0 is transmitted then −ve half interval pulse is followed by a +ve half interval pulse.

180° phase split



→ Polar Quaternary Format: To reduce the signaling rate, the message bits are grouped in the block of 2 bits. (increases efficiency)



00 combination means $-3A/2$.
01 " " $-A/2$.
10 " " $A/2$
11 " " $3A/2$

→ For two message bits only one pulse is transmitted with duration of $2T_b$ (which reduces the signaling rate.

→ For efficient or optimum transmission systems, transmission bandwidth must be as small as possible. Power efficiency, transmitted power should be as small as possible. Error detection & correction capability should be large. Power spectral density should be minimum, adequate timing contents must be present & transparency must be there.

→ characteristics of line coding

RZ < NRZ < Split Manchester < Quaternary Format

25/2/25 Digital Modulation Techniques:-

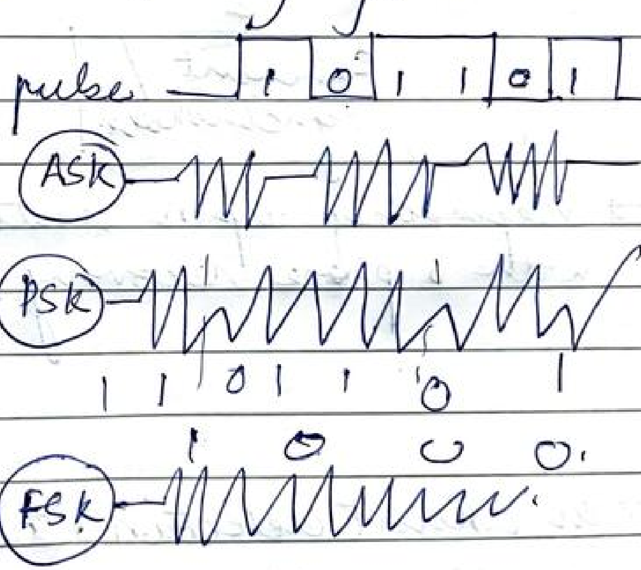
Baseband signal is digital waveform / Binary superimposed on carrier signal to give digital modulation techniques.

TYPES:- 1.) ASK (Amplitude Shift keying) / On-Off keying method

2.) PSK (Phase Shift Keying) Method.

3.) FSK (Frequency Shift keying) Method.

1.) For every 1, presence of pulse
0, no pulse



→ In PSK and FSK, it consists constant amplitude envelope, so the effects of non-linearities.
↳ noise interference is minimum and preffered over ASK.

* More than 1 Bit, then is called M-array modulation
  - → When transmitting 2 or more bit simultaneously instead of single bit transmission.

* Characteristics of Modulation Techniques for optimum Use :-

1.→ Maximum Data Rate
2.→ Minimum probability of symbol error.
3.→ Minimum transmitted Power
4.→ Maximum Channel Bandwidth.
5.→ Maximum Resistance to Interference signal.
6.→ Minimum ckt. complexity.

* Single Bit Transmission & M-array
  - → This is best as low-resistance.

* Digital Modulation Techniques
  - Coherent / Synchronous
  - Non-coherent

→ Depending upon whether the receiver is equipped with a phase recovery ckt or not

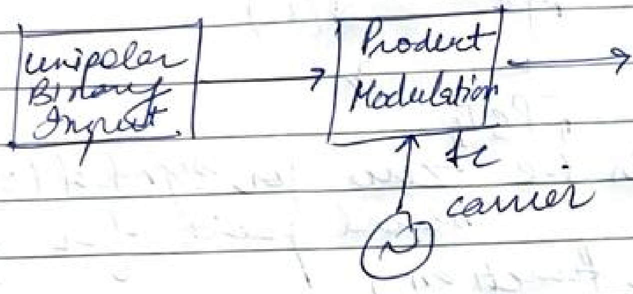
PLL – Phase Locked Loop

1.→ In coherent detection, the local carrier generated from oscillator at receiver is phase lock with carrier at transmitter.
2.→ Phase recovery circuit ensures that the oscillator is synchronous in frequency & phase with the transmitter.

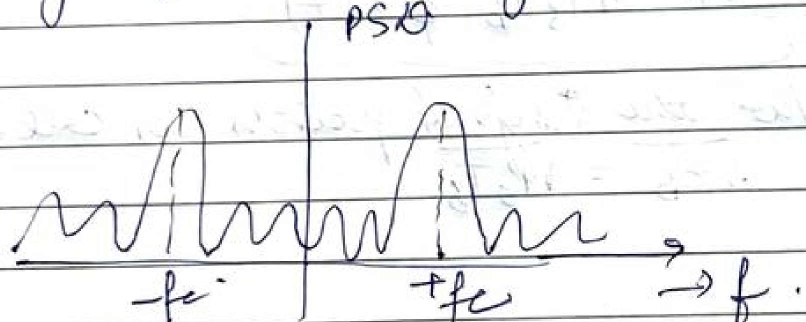
(2) Non-coherent doesnt have recovery ckt & Phase Lock loops.

(K) ASK COHERENTS:-

ON/OFF KEYING METHOD :-

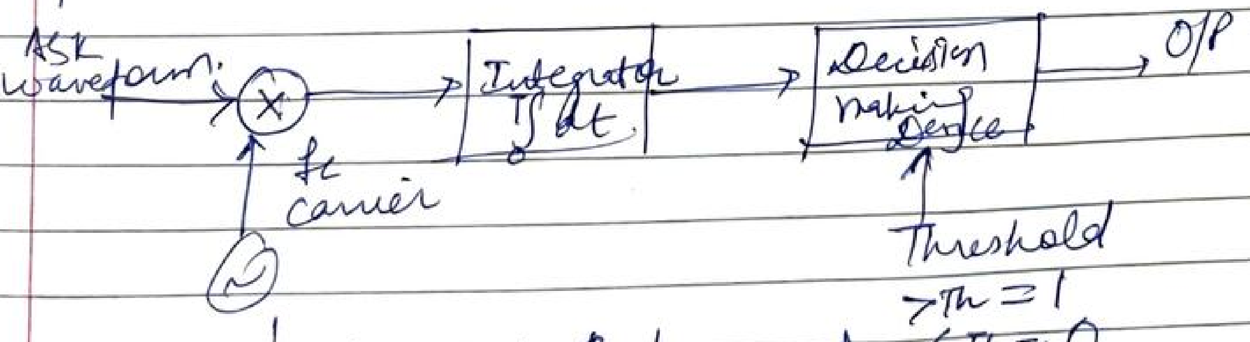


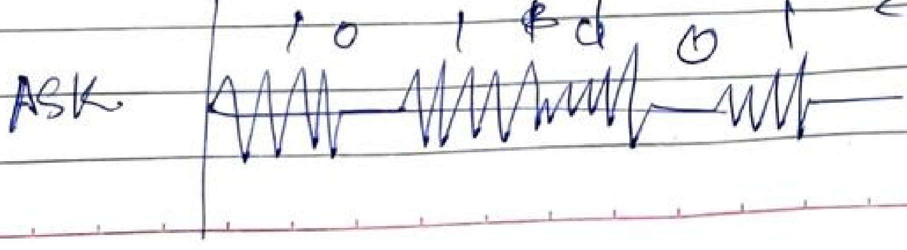
* ASK is only 1 unit energy carrier switched ON/OFF depending upon I/P binary sequence.



* ASK signal has shift of basement spectrum of $\pm fc$.
* Two impulses occurs at $\pm fc$.

Ideal spectrum of ASK has infinite bandwidth but practically bandwidth is defined as bandwidth practical.





→ Signal space Diagram of ASK :-

$$\begin{cases} s(t) = (\sqrt{2P_s})\cos 2\pi f_c t & \text{symbol (1)} \\ s(t) = 0 & \text{symbol (0)} \end{cases}$$

ASK have 2 points i.e zero for symbol (0)
second point for symbol (1)

If $\phi(t)$ is carrier function,
$s(t)$ can be represented as $(\sqrt{P_s T_b}\ \sqrt{2/T_b}\cos 2\pi f_c t)$ ϕ_1

$$\boxed{s(t) = \sqrt{P_s T_b}\ \phi_1(t)}$$

Distance b/w the 2 signal points is called E_b.

$$\sqrt{E_b} = \sqrt{P_s T_b}$$

1 Day Work After this

27/2/25 ASK/OOK

$A \rightarrow$ Peak value of amplitude

$$P = \frac{1}{2}A^2$$

$$A = \sqrt{2P}$$

$$s(t) = \sqrt{2P}\cos(2\pi f_c t)$$

→ BPSK (Binary Phase Shift Key) :- There is 180° phase shift b/w the transmission of two signals

When binary 1 is transmitted :- $s_1(t) = \sqrt{2P}\cos(2\pi f_c t)$

" " 0 " " :- $s_2(t) = \sqrt{2P}\cos(2\pi f_c t + \pi)$

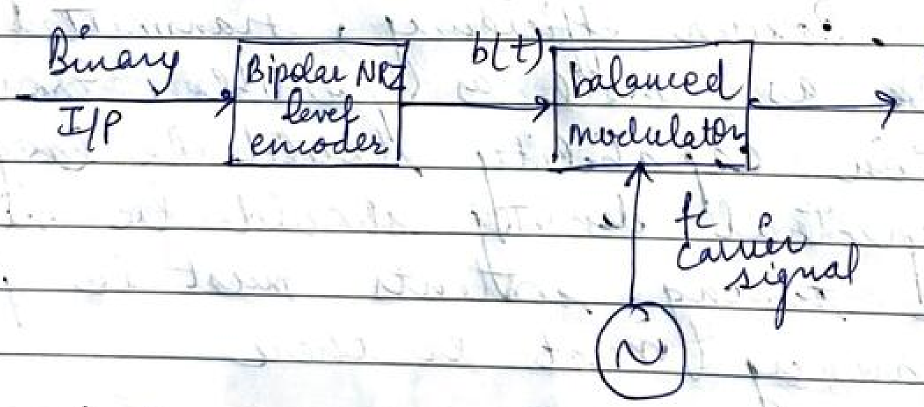
$$\Rightarrow -\sqrt{2P}\cos(2\pi f_c t)$$

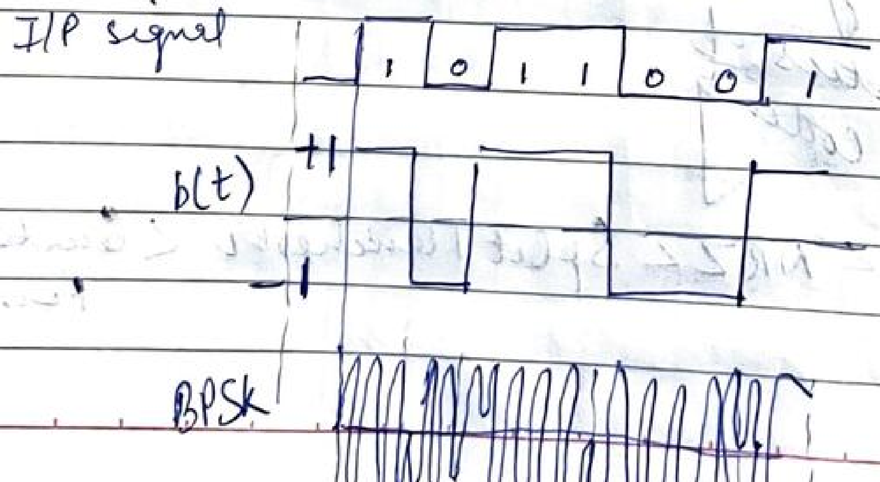
$$s(t) = -\sqrt{2P}\cos(2\pi f_c t)$$

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t) \quad \text{where } b(t) = \pm 1$$

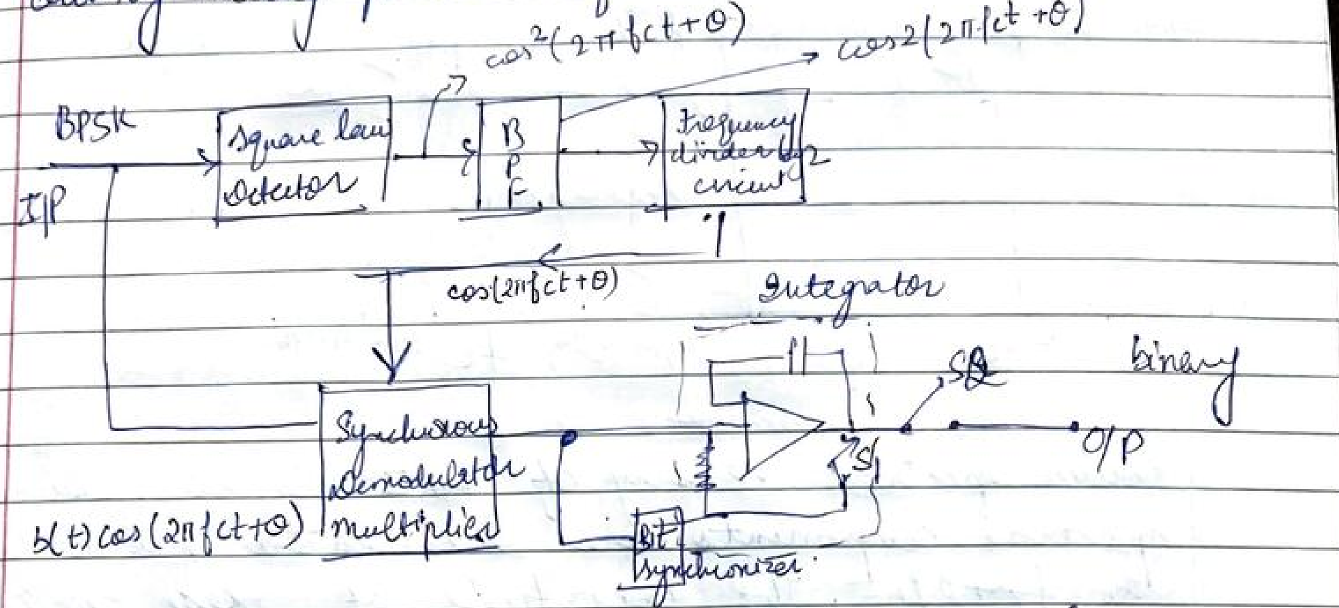
$b(t)$ is a non-return ~~to~~ zero (NRZ) bipolar signal

→ **BPSK Modulator :-**





→ **BPSK Detection :-** Coherent BPSK Detection can be carried out by using phase shifters :



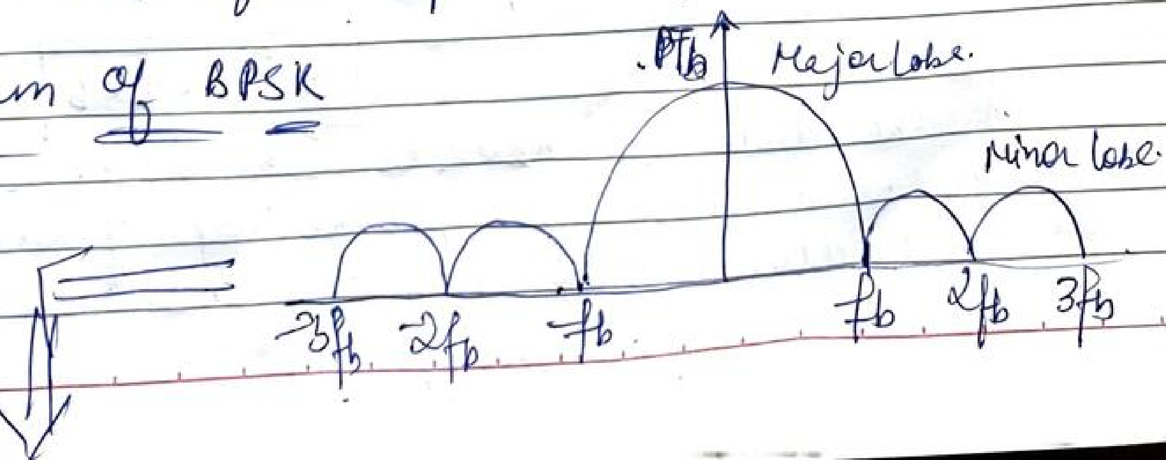
A fixed phase shift in the transmitted signal θ

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t+\theta)$$

Band Pass Filter (BPF) removes DC level. Integrator integrates the signal over one bit period. Bit Synchronizer takes square of starting & ending times of a bit. At the end of bit duration, the bit synchronizer closes S_2 temporarily. This connects the O/P of integrator to decision device. Bit Synchronizer then open switch S_2 & Switch S_1 is closed. This resets the integrator voltage to 0.

→ Phase changes occurs in the carrier only at zero crossing. BPSK has a full cycles of sinusodial Carrier. Then if its spectrum of BPSK ;

→ Spectrum of BPSK



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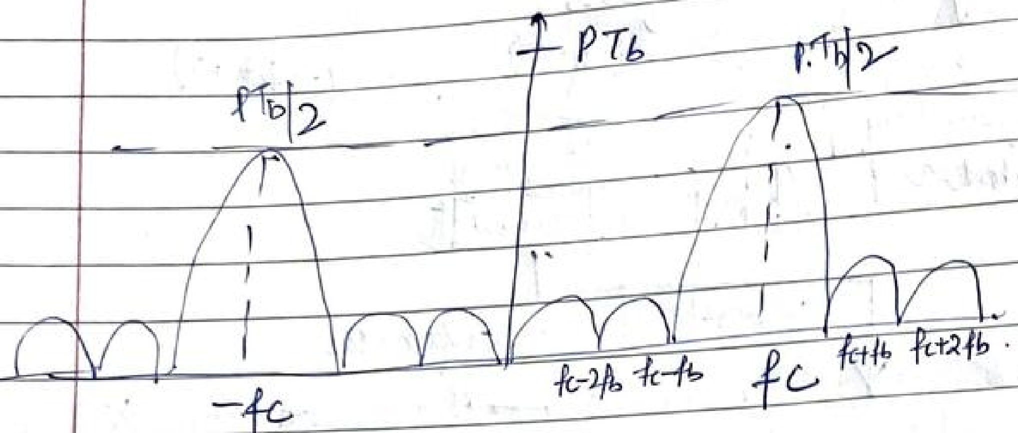
Peak value of carrier is PT_b.



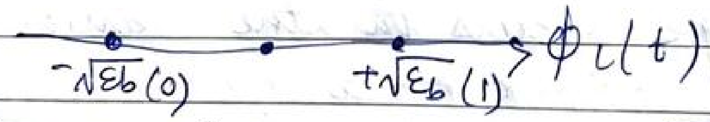
Power spectral density of NRZ signal shows that spectral components are translated from $f_c + f_b$ to $f_c + 2f_b$. The magnitude of these components is divided by half. The main lobe ranges from $-f_b$ to $+f_b$. Two lobes one at f_c & other at $-f_c$.

$$f_b = \frac{1}{T_b}$$

Bit energy $\varepsilon_b = PT_b$

Power X Bit duration

Signal space diagram for BPSK shows two points $-\varepsilon_b(0)$ & $+\varepsilon_b(1)$.



Separation b/w two points $d = \sqrt{\varepsilon_b} - (-\sqrt{\varepsilon_b})$

$$\boxed{d = 2\sqrt{\varepsilon_b}}$$

As the distance increases, the isolation b/w the symbols is more. The probability of error reduces.

Bandwidth of BPSK is $2f_b$.

$$\boxed{BW = 2f_b}$$

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→ BPSK is more prefferable than ASK.

⇒ Limitations of BPSK :- 1.) The squared signal remains same as before. Hence, the recovered carrier signal is unchanged even if I/P signal has changed its sign.