MD IFTAKHAR KABIR SAKUR

25th BATCH

COMPUTER AND COMMUNICATION ENGINEERING

International Islamic University Chittagong

Page NO.

1. 1 to 35 – Mid Term
2. 36 to 69- Final Term
3. 70 to 89 – Class Lecture

COURSE CODE: CCE-3611

COURSE TITLE: Digital Communication

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Digital Communication

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CCE-3611

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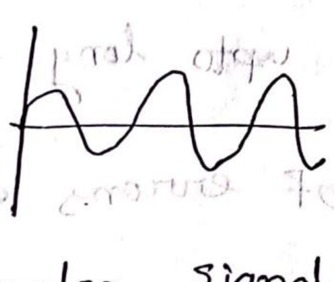
(Digital Communication)

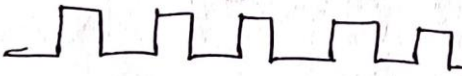
The necessity of Digitization

In conventional methods of Communication we used analog signals for long distance communications. Which is the reason for signal losses ~~suset~~ such as distortion, noise interference & other losses such as security breach.

To overcome this problem we used digitization method for signal. Which is consist of 0 & 1. Which means voltage low & voltage HIGH respectively.



(Analog Signal)



(Digital Signal)

Elements of Digital Communication :-

The elements which form a digital communication system :

Input Signal → Input Transducer → Source Encoder → Channel Encoder → Digital Modulator

- Input Transducer: (Takes physical input and converts into electrical signal)
- Source Encoder: (Redundant bits বাদ দিয়ে, data compress করে file আকারে ছোট করে)
- Channel Encoder: (Does coding for error correction. Noise এর কারণে signal distorted হতে পারে। তাই এটার জন্য extra bit Add করায়)
- Digital Modulator: (Carrier Signal add হয়)

↓ (medium through which we send signal)

Channel

↓

Digital Demodulator → Channel Decoder → Source Decoder → Output Transducer → Output Signal

- Output signal: The output of the signal
- Output Transducer: It converts electrical energy back into its original form.
- Source Decoder: Converts binary data back to the wave form.
- Channel Decoder: Removes parity bits from binary data. Doesn't affect the signal quality & transmit the data securely.
- Digital Demodulator: (Demodulate হয় এবং Source signal Carrier Signal হতে আলাদা হয়।)

Advantages of Digital Communication

- → Fast, more accurate & more reliable than analog communication.
- → Can be quickly transmitted upto long distances.
- → The detection & correction of errors is easy.
- → Allows easy removal of noise, cross-talk or any interference in the signal.
- → Not that expensive for advance tech.
- → Transmission speed of signal is HIGH.
- → Facilitates video & audio conferencing, Allow quick meetings & discussion with several people.

Disadvantages of Digital Communication :-

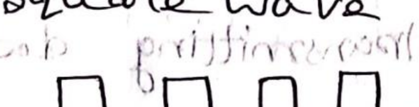
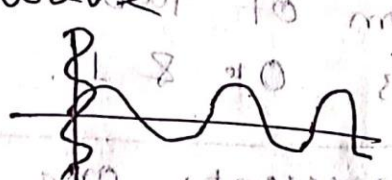
=> **High power Consumption :-**
Consumes high power due to the requirement of greater number of components, high bandwidth and High transmission speed.

=> **High transmission bandwidth :-**
Requires HIGH transmission bandwidth to transmit the signals at HIGH speed.

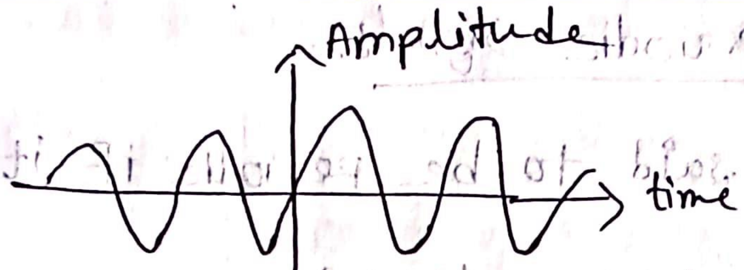
=> **HIGH power loss :-**
The power loss in here is higher than analog communication due to the high processing speed and hardware components.

Digital vs. Analog

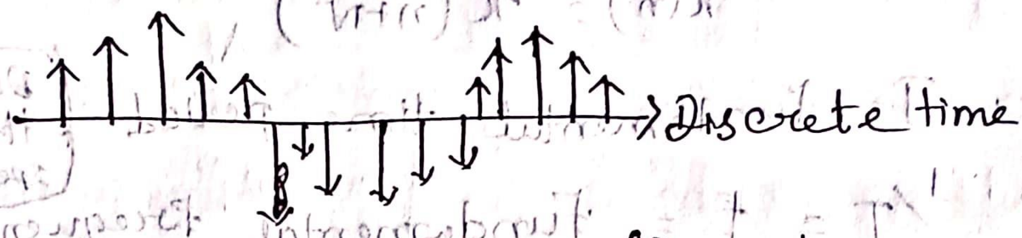
Digital Communication	Analog Communication
① uses digital signals with discrete values for transmitting data represented in the form of two binary digits 0 & 1.	① uses analog signals for transmitting data.
② It represents one bit at a time.	② It represents continuous values at a time.
③ The noise immunity is Good.	③ It is poor in noise immunity.

Digital Communication	Analog Communication
(4) Error probability is Low.	(4) Error probability is HIGH
(5) The digital Com. system uses an encoder & decoder to convert the information into bits & vice-versa	(5) It can't
(6) More flexible	(6) Less flexible
(7) HIGH Cost	(7) Low Cost
(8) power Consumption is Low.	(8) power Consumption is HIGH.
(9) Data transmission is more accurate	(9) Less accurate
(10) Square wave 	(11) ~~Better~~ Represented by sine wave or cosine wave 
(11) Example:- Clock signals	(11) Example:- Audio signals, speech signals, sound waves

Continuous Time and Discrete Time Signals:-

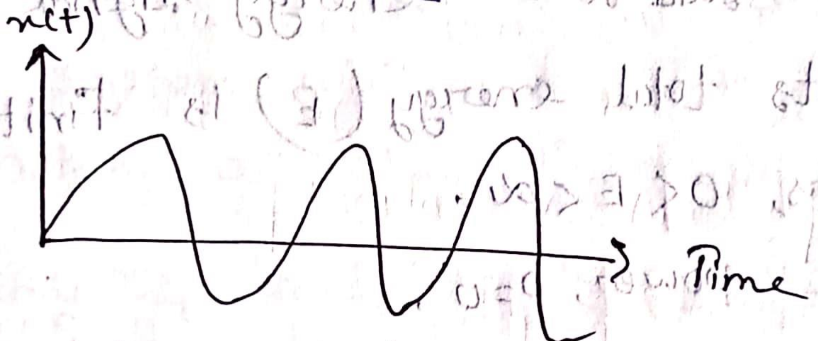


(Continuous time)



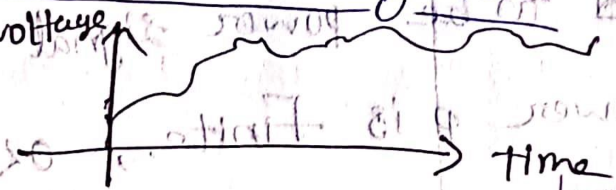
(Discrete

Deterministic Signals:-



=) A signal is deterministic if there is no uncertainty with respect to its value at any instant of time.
Can be defined by a mathematical formula!

Non deterministic Signals:-



These signals are random in nature. It can't be described by any mathematical formula. They are modelled in probabilistic term.

periodic & Aperiodic signals:-

A signal is said to be periodic if it satisfies the condition $x(t) = x(t+T)$ or

$$x(n) = x(n+N)$$

where, T = Fundamental time period

$1/T = f$ = Fundamental frequency.

(Does not repeat itself after a specific interval of time)

Energy Signal:-

A signal is said to be energy signal if & only if its total energy (E) is finite

That means, $0 < E < \infty$.

The average power, $P = 0$.

Example:- The Non-periodic signal:-

$$\text{Energy, } E = \int_{-\infty}^{\infty} x^2(t)\,dt$$

power Signal:-

A signal is said to be power signal if its average power P is finite, $0 < P < \infty$.

For a power signal, the total Energy $E = \infty$

Example: The periodic signal.

power $P = \lim_{T\to\infty} \frac{1}{2T} \int_{-T}^{T} x^2(t)\, dt$

Spectral Density

Spectral:- A Group of set up Frequencies.

So, The spectral Density of signal stands for the distribution of the signal's energy or power in the frequency domain.

Importance:- When considering Filtering in Communication system while evaluating the signal & noise at the Filter output.

* Energy spectral density (ESD)

or power spectral Density (PSD) is used in the evaluation.

~~power~~ Energy Spectral Density

It defined as the distroibution of energy of the signal per unit bandwidth.

It is denoted by $\psi(f)$

$$\psi(F) = \frac{\text{Energy}}{\text{Unit Band width}} = \frac{E}{\Delta F}$$

$$\cancel{\psi(F) = |x(F)|^2 = |x(\omega)|^2}$$

According to parseval's theorem, the energy of $x(t)$:-

$$E_x = \int_{-\infty}^{\infty} x^2(t)\,dt$$

$$= \int_{-\infty}^{\infty} |x(F)|^2\,dF$$

Therefore: $E_x = \int_{-\infty}^{\infty} \psi_x(F)\cdot dF$ और

So, the energy spectral Density is symmetrical in Frequency about origin & total energy of the signal, can be expressed as:-

$$E_x = 2\int_{0}^{\infty} \psi(F)\,dF$$

If it is autocorrelation for an energy signal $\mathcal{E}$ (ESD) from a Fourier Transform :-

autocorrelation $R_{xx}(\tau) \xleftrightarrow{FT} \psi(F)$ or, $\Psi(\omega)$

$\boxed{\text{power Density Spectral (PSD)}}$

Sign PSD Function $G_x(F)$ of the periodic signal $x(t)$ is a real, even & nonnegative function of frequency that gives the distribution of the power of $x(t)$ in the frequency domain.

→ PSD is represented as:

$$G_x(F) = \sum |C_n|^2 \delta(F - nF_0)$$

→ Whereas the average power of a periodic signal $x(t)$ is represented as:-

$$P_x = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} x^2(t)\, dt = \sum_{n=-\infty}^{\infty} |C_n|^2$$

→ Using PSD, the average normalized power of a real-valued signal is represented as:-

$$P_x = \int_{-\infty}^{\infty} G_x(F)\, dF = 2\int_{0}^{\infty} G_x(F)\, dF$$

Signal Transmission Through Linear System

⇒ The response of the system for an impulse is called as impulse response in the system.

This is represented as $h(t)$ or, $h(p)$

$$x(t) \rightarrow \boxed{h(t)} \rightarrow y(t)$$

So, $y(t) = h(t)$ when $x(t) = \delta(t)$

$$y(\omega) = x(\omega) \cdot H(\omega)$$

$$\therefore H(\omega) = \frac{y(\omega)}{x(\omega)}$$

So, impulse response, $F^{-1}[H(\omega)] = h(t)$

It is a transfer function of a system.

~~Example~~ Question:- A transfer function of a system is give so what may be the impulse respons?

Ans :- $H(S) \Leftarrow$ Will be given

$$L^{-1}[H(S)] = h(t)$$

Or, $F^{-1}[H(\omega)] = h(t) \Rightarrow$ Solve formula

Transfer function of LTI System :-

LTI :- Linear Time Invariant

The frequency-domain output signal $y(F)$ is obtained by taking the Fourier transform.

$$\boxed{y(F) = X(F) \cdot H(F)}$$

* Frequency transfer function or the frequency response is defined as :-

$$H(F) = \frac{y(F)}{x(F)}$$

$$H(F) = |H(F)| e^{j\theta(F)}$$

* The phase response is defined as :-

$$\theta(F) = \tan^{-1} \frac{Im\{H(F)\}}{Re\{H(F)\}}$$

Understanding:-

Linear System + Time Invariant System

→ LTI Systems.

$\delta(t)$ → Impulse ~~response~~

$\boxed{x(t)} \longrightarrow \boxed{\text{LTI System}} \longrightarrow \boxed{y(t)} = h(t)$

I/P (under $x(t)$), O/P (over $y(t)$), Impulse response (under $h(t)$)

Transfer Function is defined as the ratio of Laplace transform of O/P to the Laplace transform of input when all the initial conditions are assumed to be zero.

Distortionless Transmission

→ Must have some time delay and different amplitude than the input. There is no distortion. Must have the same shape as input.

For ideal distortion less transmission:-

→ Output signal in time domain:- $y(t) = Kx(t - t_0)$

→ Output signal in frequency domain:-

$$y(F) = KX(F)e^{-j2\pi F t_0}$$

→ System Transfer Function:-

$$H(F) = Ke^{-j2\pi F t_0}$$

Autocorrelation

Correlation is a process of matching

→ Matching of a signal with a delayed version of itself.

Autocorrelation of an Energy signal:

Energy signal $x(t)$:-

$$R_x(\tau) = \int_{-\infty}^{\infty} x(t) \cdot x(t+\tau)\, dt \qquad [-\infty < \tau < \infty]$$

→ τ unit time এ signal এর copy shift হয় তার পরিমাপ, $R_x(\tau)$

→ $R_x(\tau)$ time এর কোনো function নয়,

→ variable τ → parameter search, scan এর কাজ করে।

→ এই function টি τ তে waveform এর shifted copy.

properties:-

① $R_x(\tau) = R_x(-\tau)$ → [Symmetrical in τ about zero]

② $R_x(\tau) \leq R_x(0)$ for all τ [Maximum value origin এ হয়]

③ $R_x(\tau) \leftrightarrow \psi_x(f)$ [Autocorrelation এবং ESD Fourier Transform pair তৈরি করে]

④ $R_x(0) = \int_{-\infty}^{\infty} x^2(t)\, dt$ [value at the origin হবে signal এর energy]

Autocorrelation of a periodic (power) signal:-

Function of a real valued power signal $x(t)$ is defined as:-

$$R_x(T) = \lim_{T\to\infty} \frac{1}{T} \int_{-T/2}^{T/2} x(t)\, x(t+T)\, dt \quad [-\infty < T < \infty]$$

when, $x(t)$ = periodic with T_0 then,

$$R_x(T) = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} x(t) \cdot x(t+T)\, dt \quad [-\infty < T < \infty]$$

⇒ properties:-

(i) $R_x(T) = R_x(-T)$ [Symmetrical in T about zero]

(ii) $R_x(T) \leq R_x(0)$ For all T

[Maximum value occurs at the origin]

(iii) $R_x(T) \leftrightarrow G_x(F)$ [Autocorrelation আর PSD Fourier Transform pair তৈরি করে]

(iv) $R_x(0) = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} x^2(t)\, dt$ [Origin এর value = The average power of the signal]

Random process

→ Random process $\{X(A, t) \Rightarrow$ two variables

↓ ↳ Time

Event A

→ Each of the sample function can be ~~generated~~ regarded as the output of a different noise generator.

Example:-

Specific event → A_j → $X(A_j, t_j)$

Time → t_j

but, $t_j = t_K$

so, $X(A_j, t_K)$

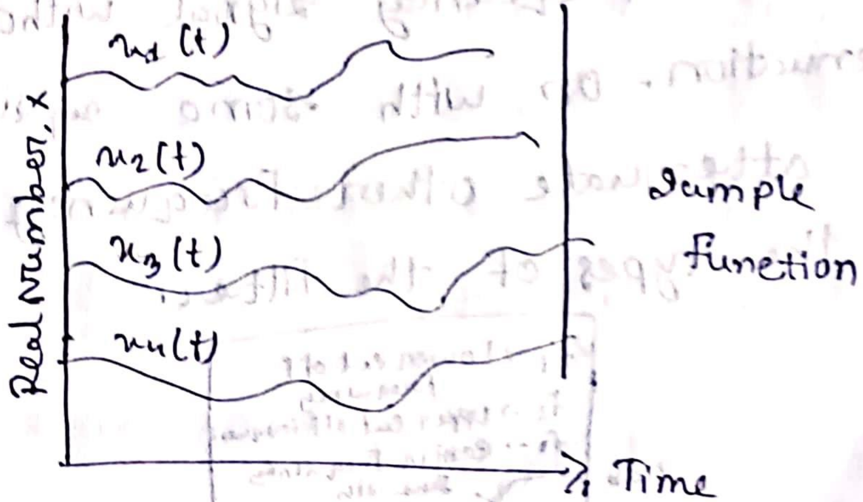


Fig: - Random Noise process

Principle Features of PSD Functions:-

1] $G_x(f) \geq 0$ [Always real valued]

2] $G_x(F) = G_x(-F)$ [For $x(t)$ real-valued]

3] $G_x(F) \longleftrightarrow R_x(\tau)$ [PSD & a auto correlation form a Fourier Transform pair]

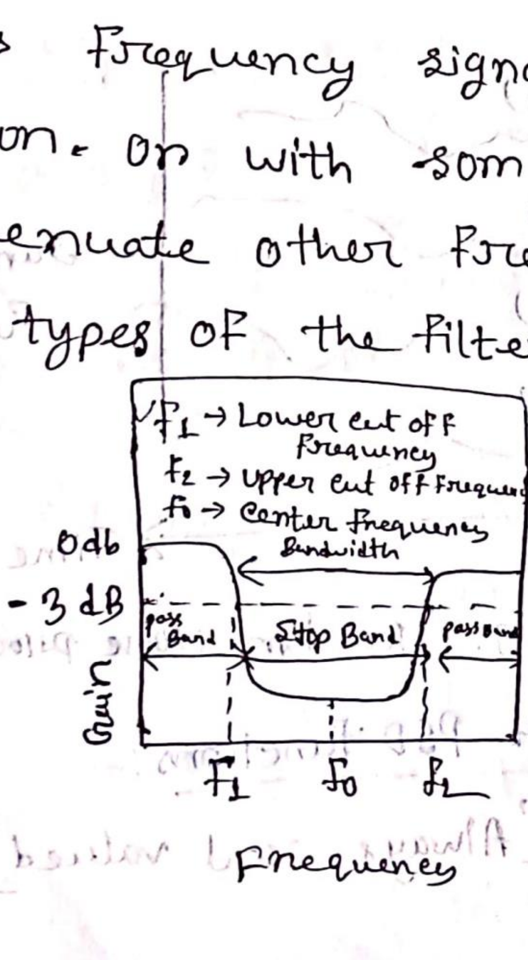
4] $P_x = \int_{-\infty}^{\infty} G_x(F) \cdot dF$ [Realtion of Power & PSD]

Filters

→ Also known as a Frequency selective ~~signal~~ circuit.

→ Used for filtering out some of the input signals on the basis of their frequencies.

→ It passes frequency signal without attenuation or with some amplification And attenuate other frequency depending on the types of the filter.



Passband :-

→ Attenuation ছাড়া যে frequency গুলো filter এর মধ্যে যায়।

→ The passband may have some gain depending on the configuration of the circuit.

Stopband :-

→ Frequencies of the input that are blocked or attenuated in the filter.

→ $-3dB$ gain is considered for the first order filter

→ 2nd order filter has $-6dB$ gain which decreases with the order of the filter.

Cutoff Frequency :-

The passband & stopband are distinguished from each other by the cutoff frequency or corner frequency.

→ The output signal's frequency is 70.7% of the input signal's voltage. It is also known as $-3db$ frequency. As $-3db$ represents half power of the input signal.

Lower Cutoff Frequency:-

→ The gain of the Filter is -3db and it is denoted by F_1

⇒ Bandpass Filter এ এই point এর নিচের Frequency pass করলেও Band stop Filter block করে দেয়।

⇒ ## Upper cutoff frequency:-

At which the output power is reduced by $\frac{1}{2}$ of the input ~~signal~~ power. It is denoted by F_2.

Bandpass Filter does not allow Frequency after this point.

Center Frequency, F_0 :-

Frequency lies at the center of the passband or stopband is called Center Frequency. It lies between lower & upper cutoff Frequency.

$$F_0 = (F_1 + F_2)/2$$

Bandwidth:-

→ Frequencies that are passed without any attenuation. or the Frequencies that are attenuated.

→ Frequencies before (low pass) & after (High pass) cutoff is called bandwidth.

$$B = f_2 - f_1$$

Based on the construction

Two types:-

1) **Passive Filters:-**

→ Made up of passive components like resistors, capacitors, inductors.

→ No need of any external source of energy.

→ No voltage gain.

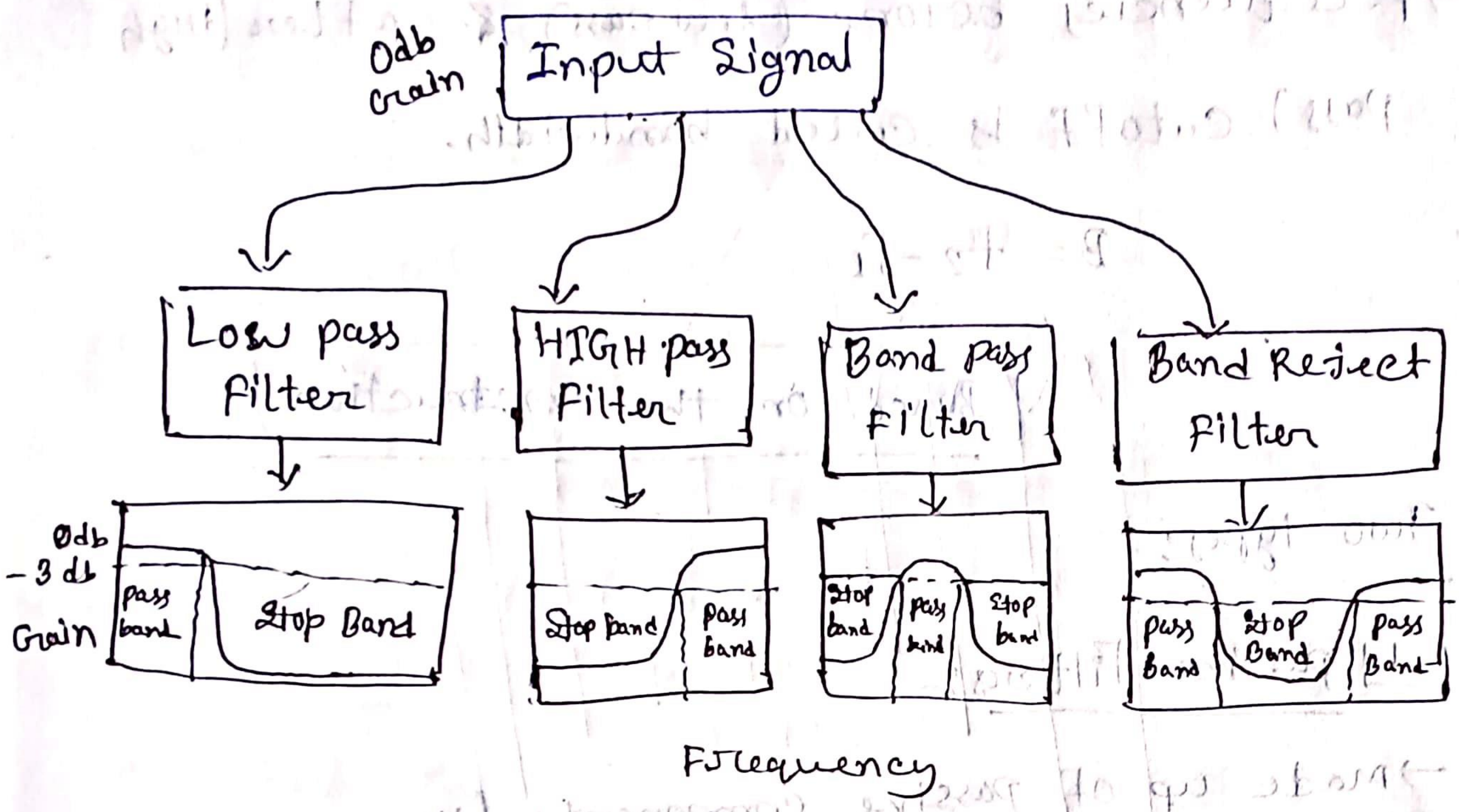
→ Output voltage less than Input voltage.

2) **Active Filters:-**

→ Made up of active components, operational amplifiers, transistors etc.

→ They need external source of power.

→ Gives high voltage gain, used to amplify weak input signals.



Low pass Filter:

→ Allow low-frequency without any Attenuation.

→ Reject High-frequency signals.

→ Have reactive component, varies with input frequency.

→ Either the voltage gets decreased or increased for these components.

→ frequency < cutoff frequency [passed]

→ frequency > cutoff frequency [blocked]

High pass Filter:-

→ High frequency pass without attenuation

→ Frequency < cutoff frequency [blocked]

→ Frequency > cutoff frequency [passed]

Band pass Filter:-

⇒ Allow a specific band of frequencies

⇒ Blocks any other frequencies higher or lower than passband

⇒ Have two cutoff frequencies → lower, → upper

⇒ Known ~~pass~~ bandpass frequencies [passed]

⇒ combined of low & High pass filter together will provide a bandpass filter.

→ Both low & High frequency [Blocked]

Band Reject Filter:-

⇒ Allows both low & High frequency

⇒ Reject fined band frequency

→ Have lower & Higher cutoff frequencies.

→ Any signal in between lower & Higher will be rejected.

The Bandwidth Dilemma

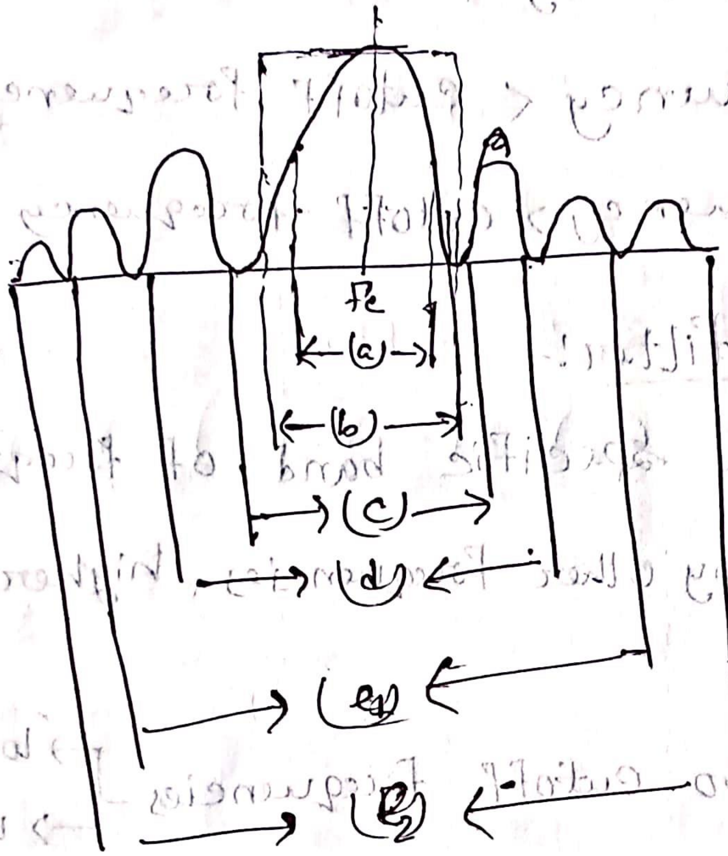


Fig :- TBD

(a) → Half power

(b) → Noise equivalent

(c) → Null to Null

(d) → 99% power

(e1) → Bounded PSD 35dB

(e2) → Bounded PSD 50 dB

Sampling

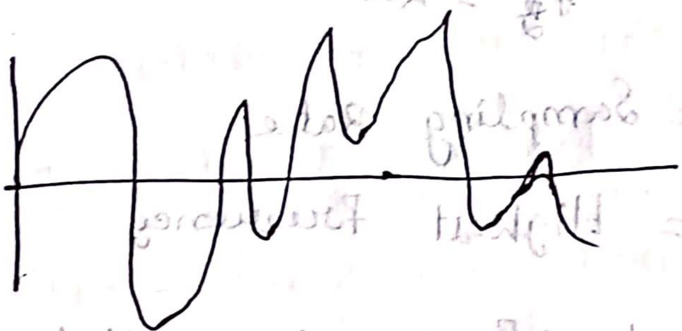
⇒ Sampling : The process of measuring the instantaneous values of Continuous-time signal in a discrete form.

→ A piece of data taken from whole data.

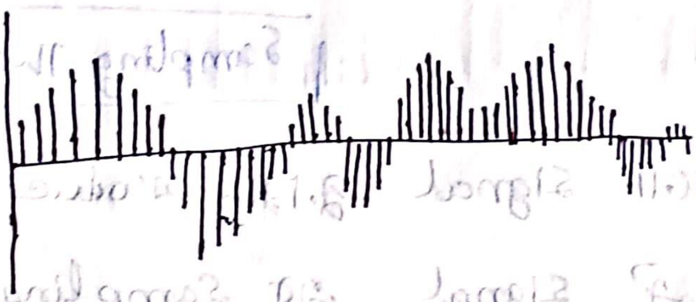
→ Source generated → Digitalized having 1s & 0s.

→ If low → 0
High → 1

Continuos :-



Digitalized :-



Sampling Rate :-

Sampling Frequency = $1/T_s = f_s$

⇒ Sampling Frequency can be called as Sampling Rate.

⇒ Sampling Rate denotes the number of samples taken per second.

⇒ The sampling should be like the data in the message signal should neither be lost nor it should get over-lapped

Nyquist Rate:–

The sampling Rate should be twice the highest Frequency.

$$F_s = 2W$$

F_s = Sampling Rate

W = Highest Frequency

→ The rate of sampling called as Nyquist rate

→

$\boxed{\text{Sampling Theorem}}$

⇒ কোনো signal পুনরায় produce করা যেতে পারে যদি ঐ signal এর Sampling Rate F_s হয় F_s হবে সর্বোচ্চ Frequency W এর দ্বিগুণ।

⇒ Band limited signal non-zero between $-W$ & W Hertz.

$$x(F) = 0 \text{ for } |F| > W$$

=) If a signal is sampled above Nyquist rate then a signal can be recovered.

=) If ~~is~~ it is lower can't be recovered.

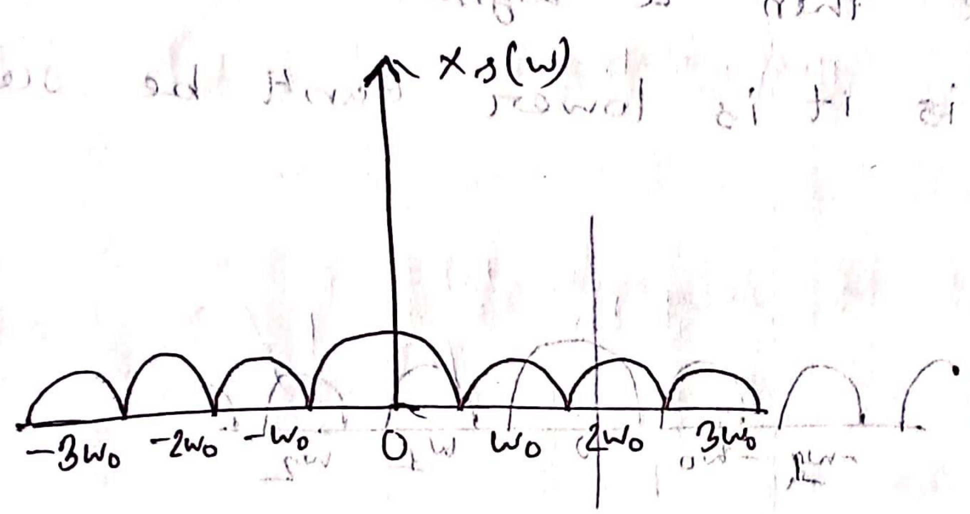
=) If we do Fourier transform of the signal $x_s(t)$ here, the information is reproduced without any loss. There is no mixing up & hence recover is possible.

$$X_s(W) = \frac{1}{T_0} \sum X(w - nw_0)$$

T_s = Sampling period

$$w_0 = \frac{2\pi}{T_s}$$

☐ When we do the sampling with frequency 2w :-

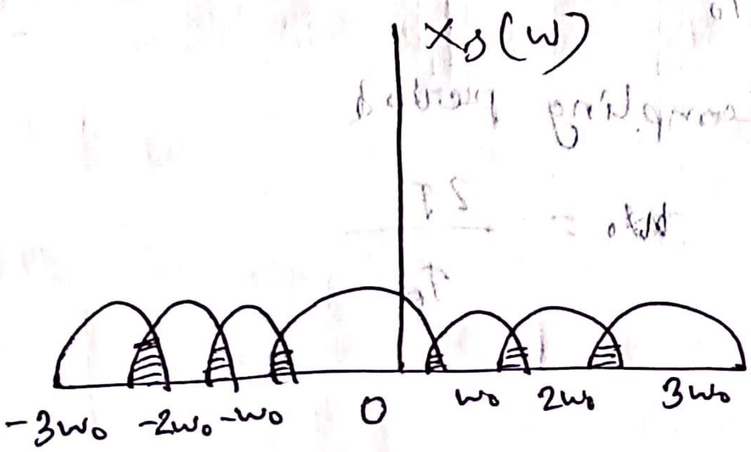


fs = Sampling frequency

w = Highest frequency

=) As it is seen the frequency is replaced without any lose.

☐ But if $f_s < 2w$

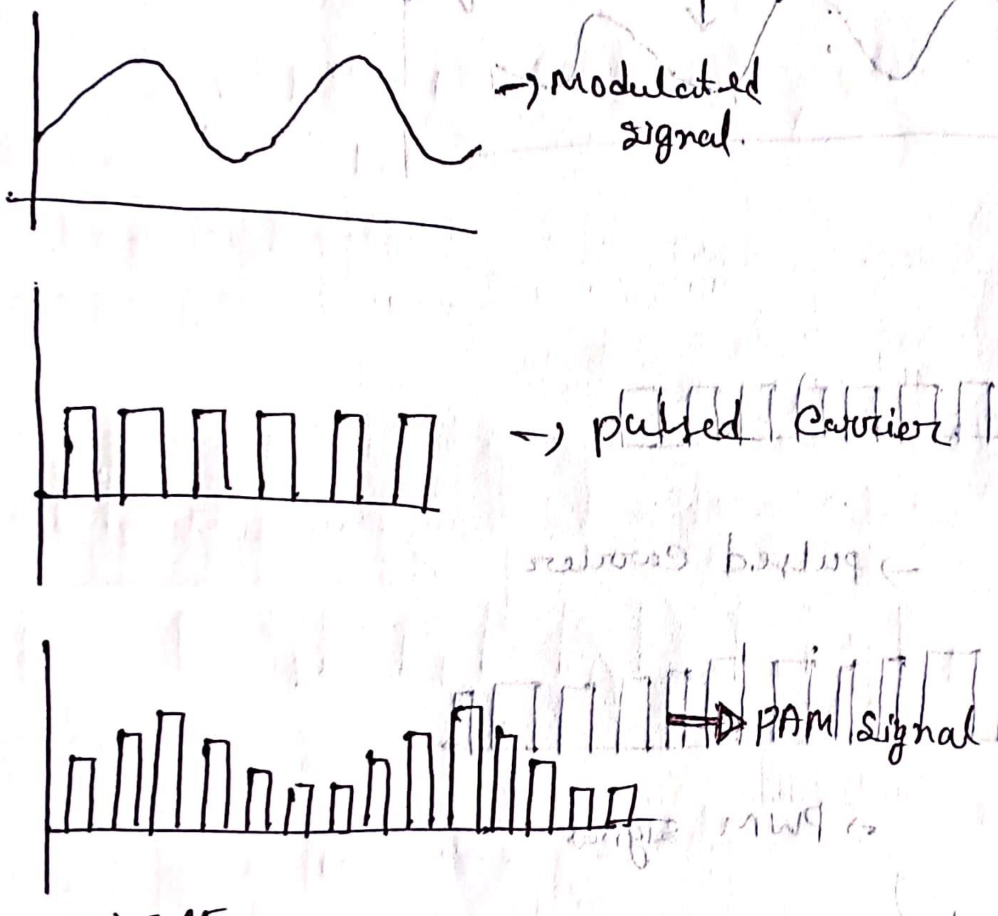


|pulse Modulation|

PAM, PWM, PPM

PAM:- Pulse Amplitude Modulation.

⇒ Carrier signal gets changed according to the amplitude of the message signal.

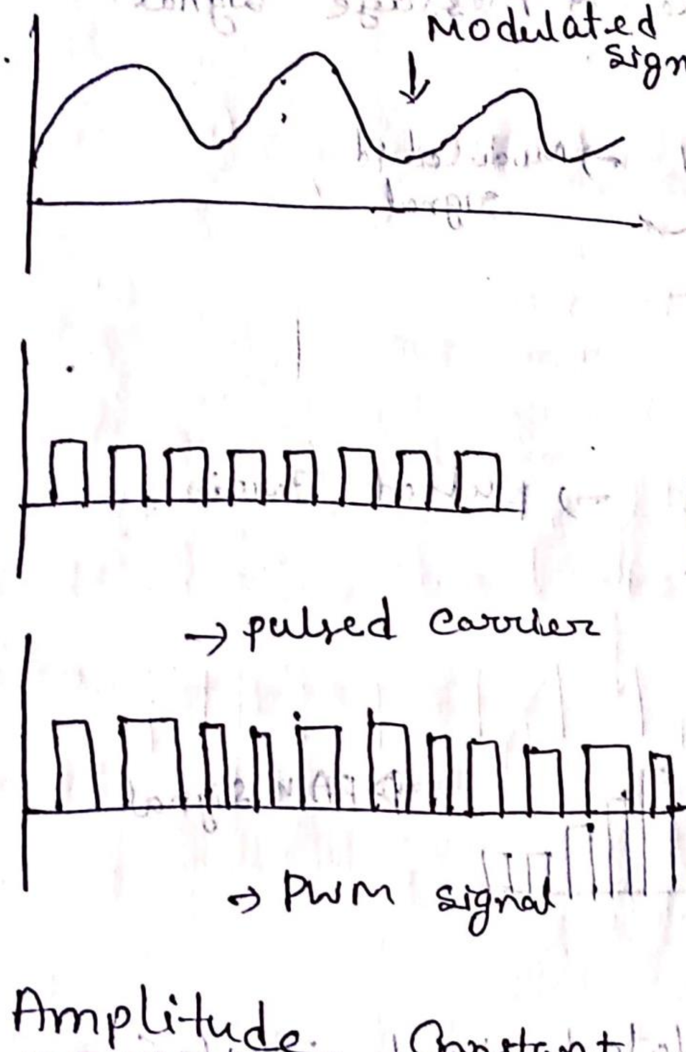


=) pulse (→ carrier এর) vary করে Modulated signal উপর নির্ভর করে।

→ It pulse train rather than continuous wave signal।

PWM:- (Pulse width Modulation)

=) we change the width of the pulse here according to the pulse width Modulation signal.



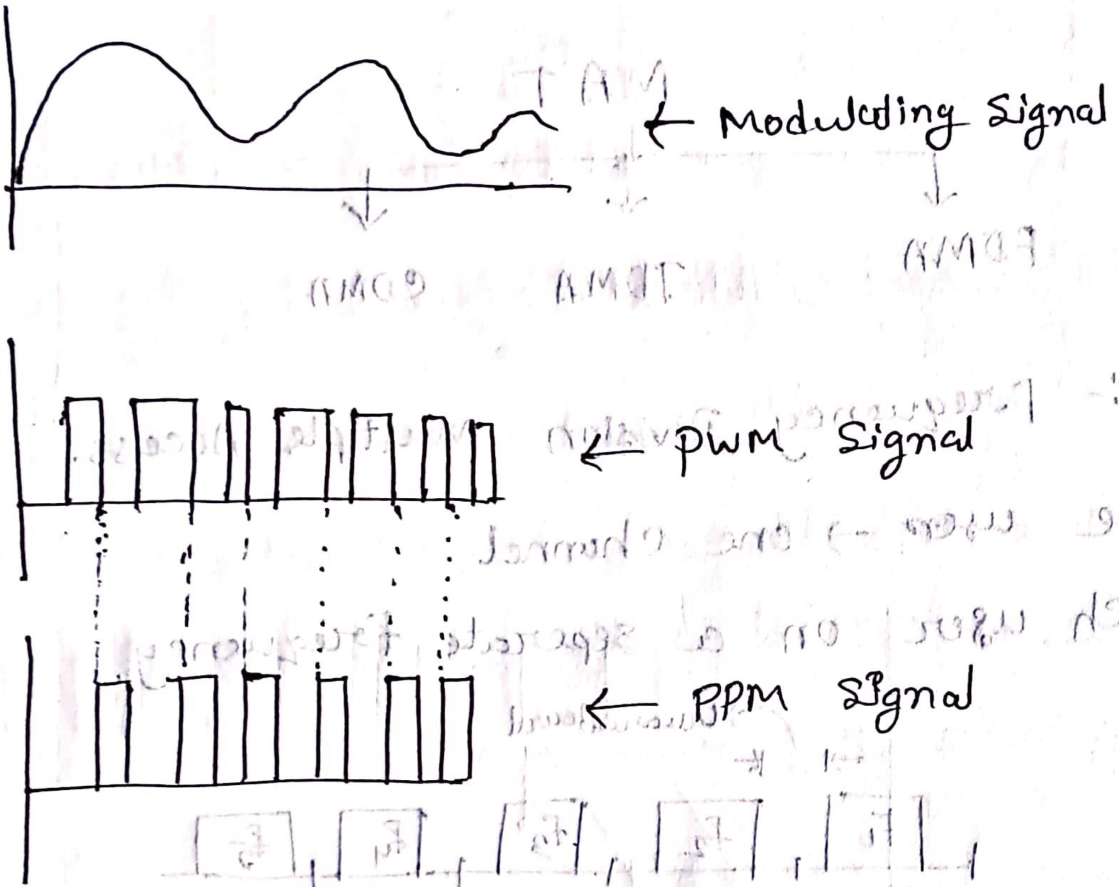
=) Amplitude Constant

=) width varying

=) By the variation in the width, the frequency of the pulses in the PWM shows variation.

PPM:- (pulse position Modulation)

→ Technique where position of the pulses is changed ~~with~~ by modulating signal



Multiple Access Technique

=> Techniques that,

=> Allows many users to share simultaneously a finite number radio spectrum.

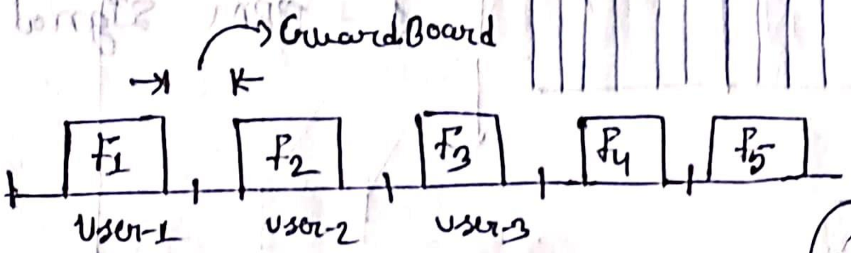
MAT

- FDMA
- TDMA
- CDMA

FDMA :- Frequency Division Multiple Access.

-> One user -> one channel

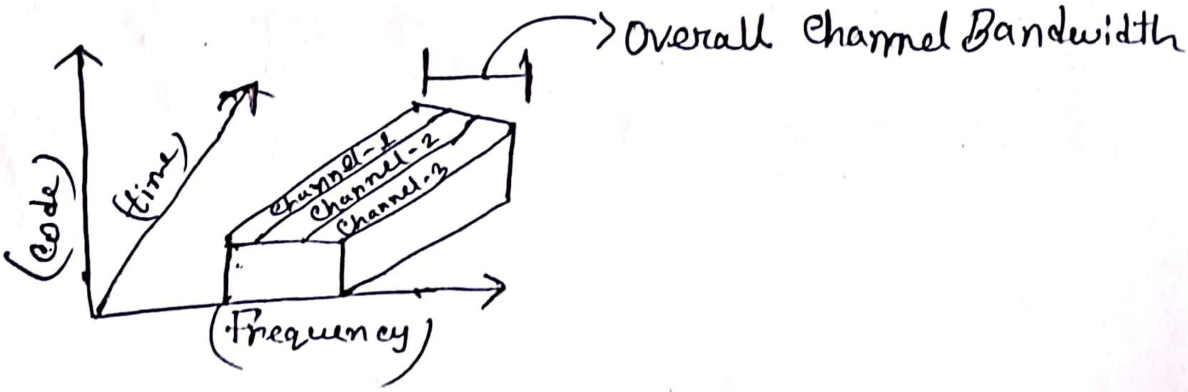
-> Each user on a seperate frequency.



10 MHz to 15 MHz

=> No timing problem, requires high performing filters.

=> If not use -> The channel is left idle.



⇒ Requires RF Filtering to minimize adjacent Channel interference

⇒ Efficient only for small number of station.

TDMA

⇒ Time Division Multiple Access

⇒ Used by a user for a fixed amount of time.

⇒ Systems digital in nature

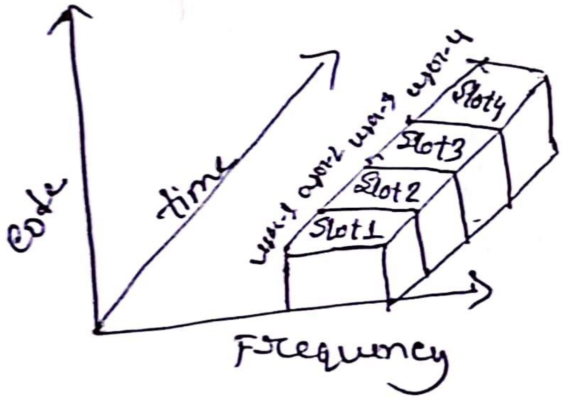
⇒ There must be synchronization so clock is used

⇒ Tx & Rx must have to aware of this clock

⇒ Single carrier frequency with multiple users

⇒ Demand এর উপর নির্ভর করে slot assigned.

⇒ Multipath, Distortions & each user has predetermined time slot.



CDMA

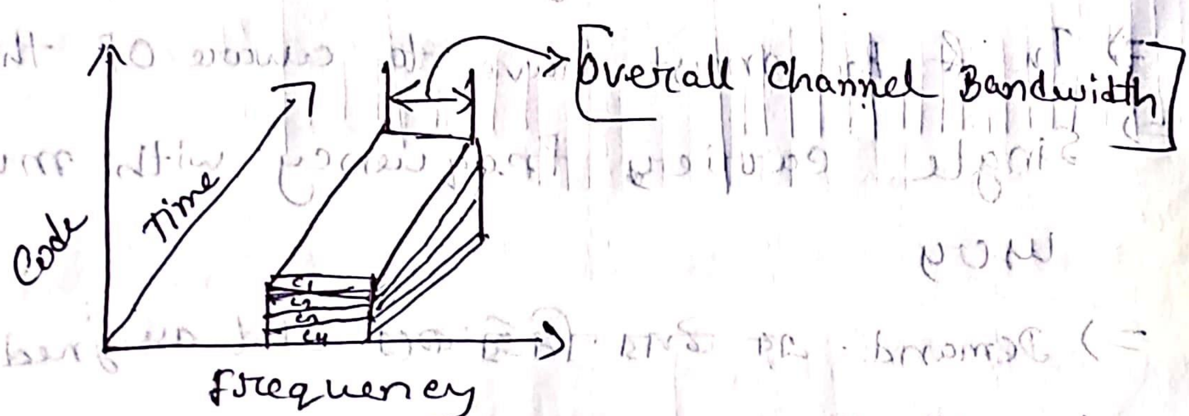
=> Code Division Multiple Access

=> No restriction on time and frequency.

=> Users are not separated by time & frequency slot rather than by code.

=> Receives the signal → Decode it → Recovers the original data

=> The coded data signal must be higher than the bandwidth of the original data signal.

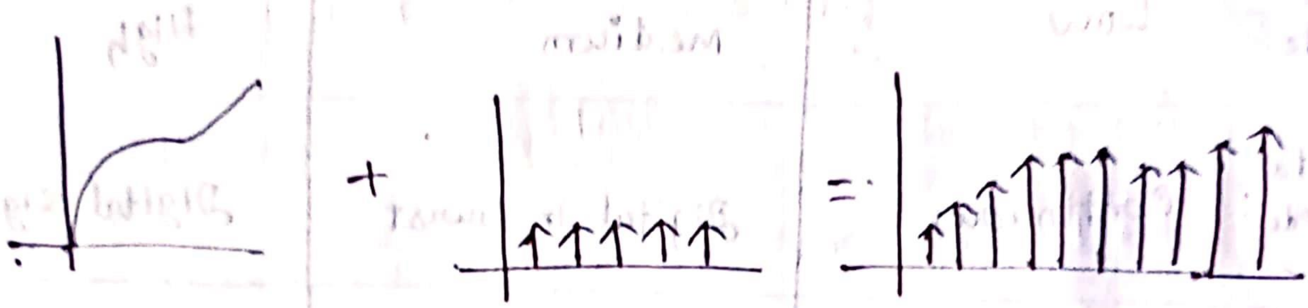


Comparison

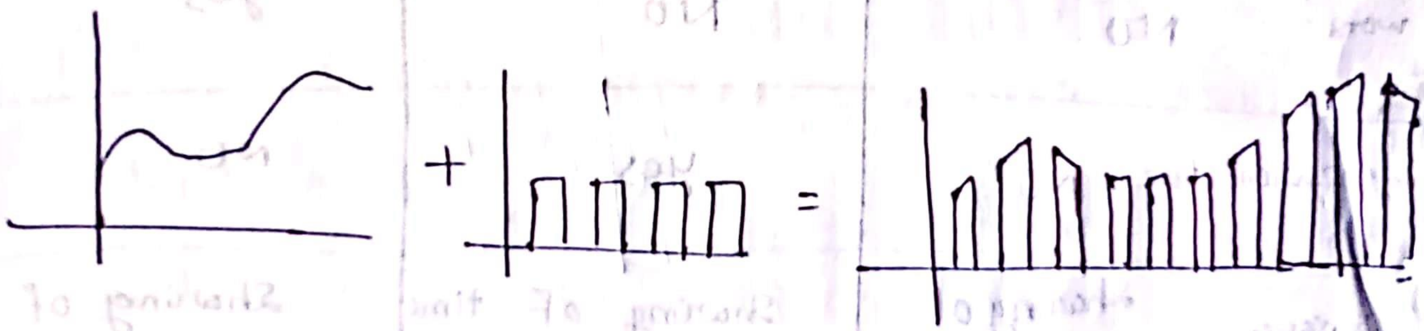
	FDMA	TDMA	CDMA
Data Rate :-	Low	Medium	High
Data Mode :	Continuous	Digital in burst	Digital signal
Cost :-	High	Low	Installation High operation Low
Code word :-	NO	NO	yes
Synchronization :-	NO	yes	NO
Technique :-	Sharing of Overall bandwidth	Sharing of time	Sharing of time & frequency both

Sampling

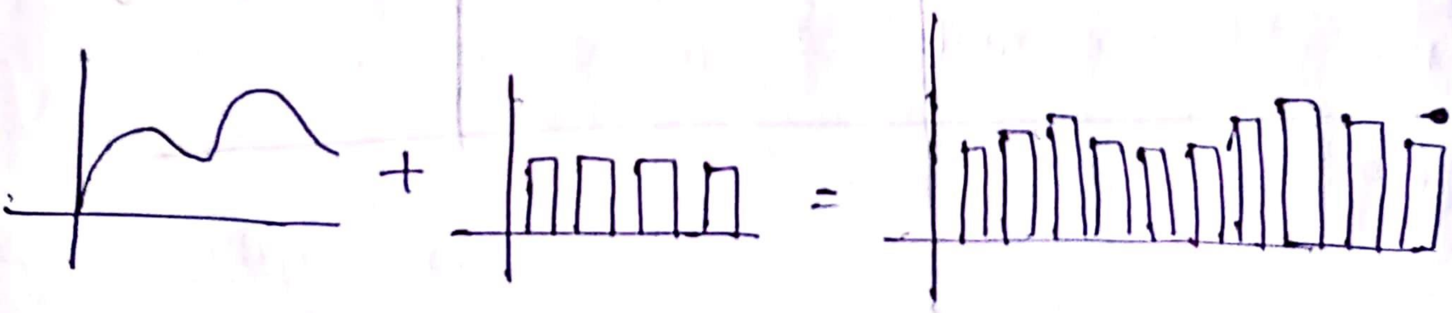
Ideal Sampling:-

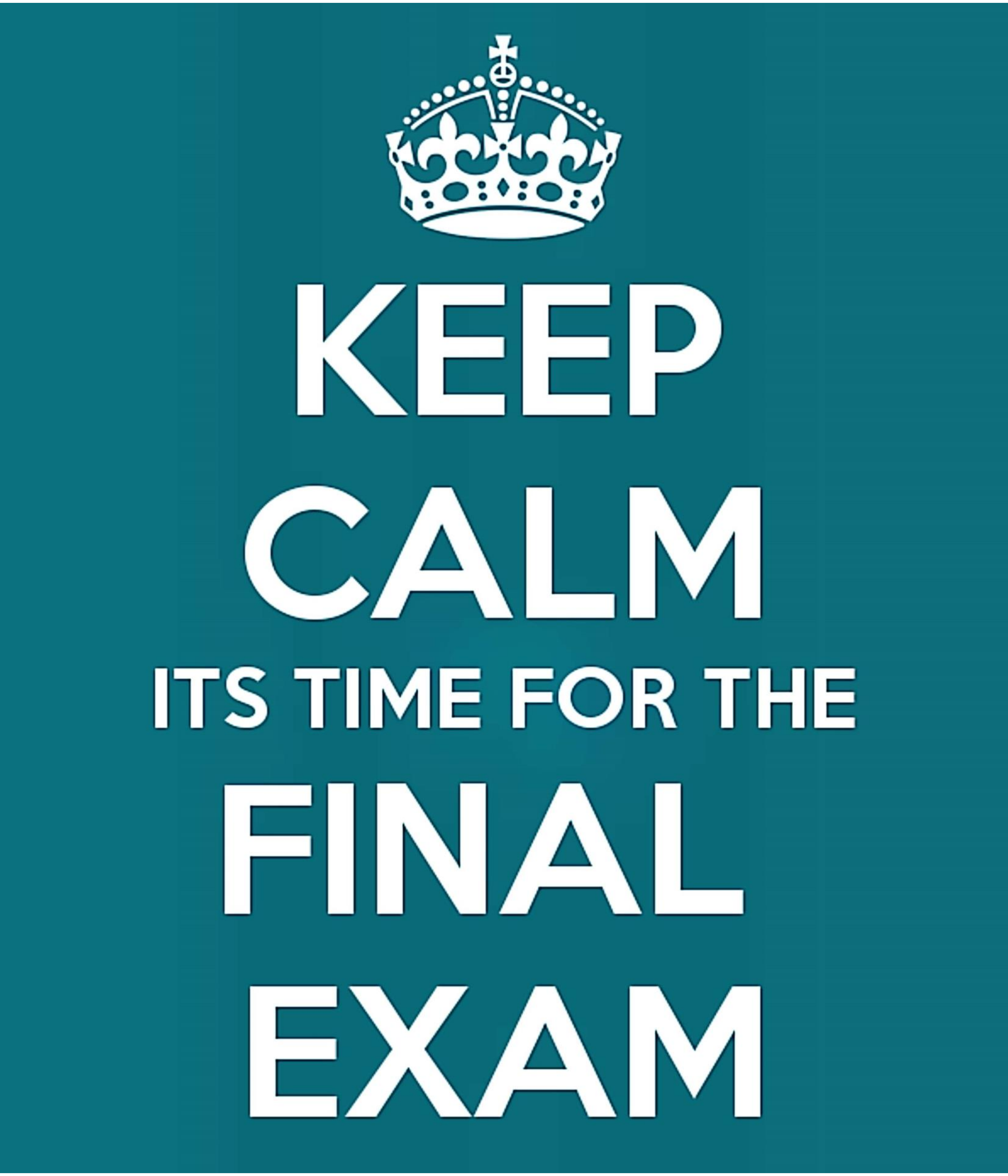


Natural Sampling:-



~~Top~~ Flat - Top:-




KEEP
CALM
ITS TIME FOR THE
FINAL
EXAM

DCOM
(Final) part-02

DM Receiver/Delta Demodulator :-

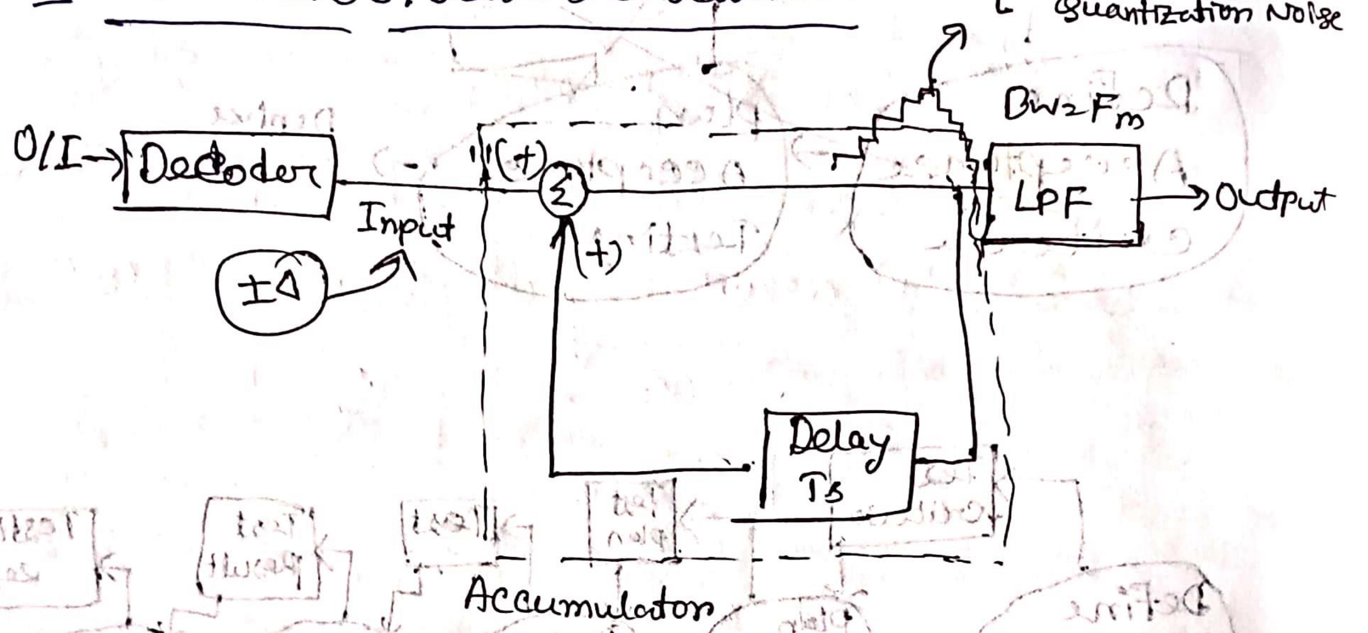


Fig:- DM receiver/Delta Demodulator

Advantage :-

(1) Low signaling rate

(2) Low Bw

(3) Less complicated to implemented than PCM.

Disadvantages :-

(1) Distortion (Slop overload & Granular)

Quantization Noise in DM Signal:-

→ Slope overload distortion

→ Granular Noise

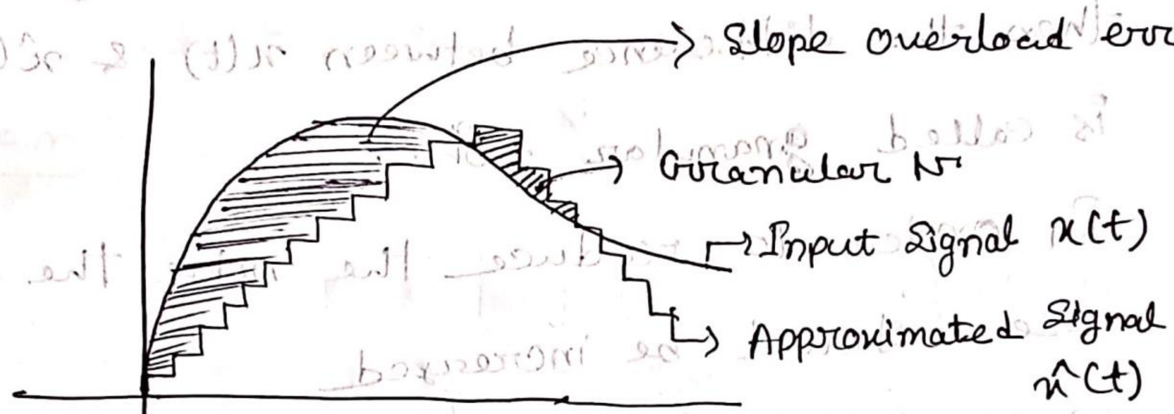


Fig:- Quantization Noise

(i) Slope overload distortion:

IF slope of $x(t)$ is much higher than $\hat{x}(t)$ over a long duration then $\hat{x}(t)$ can't follow $x(t)$. Difference between $x(t)$ & $\hat{x}(t)$ is called slope overload error.

✱ why slope of staircase $[\hat{x}(t)]$ is less ?

⇒ Because step size is small. So, to reduce error, increase step size or increase frequency.

$$Slope = \frac{\Delta}{T_s} = \Delta \cdot F_s$$

Granular Noise:-

When input signal $x(t)$ is relatively constant in amplitude but $\hat{x}(t)$ is bouncing up-down then the difference between $x(t)$ & $\hat{x}(t)$ is called granular noise.

In order to reduce the noise the step size should be increased.

Problem & its solve:-

→ If step size is increased the granular noise will be increased too.

→ If step size is decreased then the ~~so~~ slope overload distortion will be increased.

And in Delta Modulation step size is not a variable.

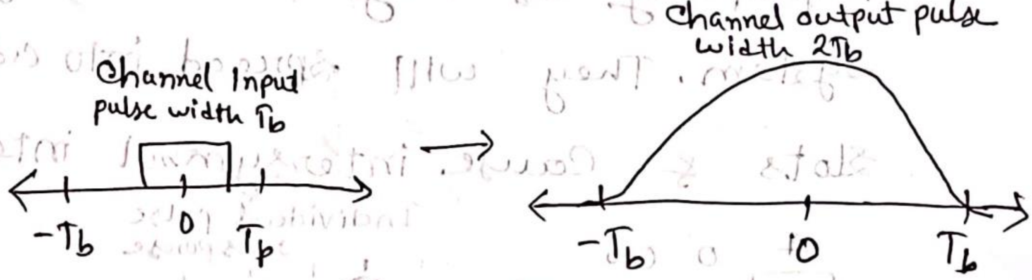
→ So a system is adapted it is called Adaptive Delta Modulation (ADM).

Intersymbol Interference (ISI)

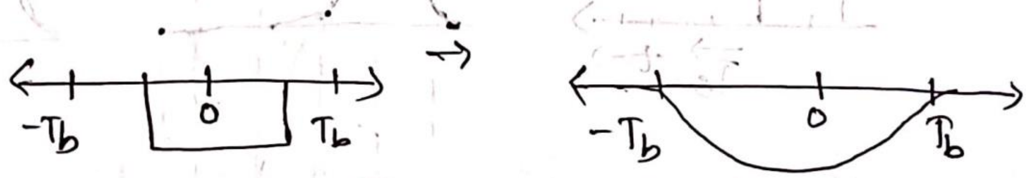
ISI occurs when a pulse spreads out in such a away that it interferes with adjacent pulses at the sample instant.

Example:- অর্থাৎ $1T_b$ input $2T_b$ output হবে।

Data-1:

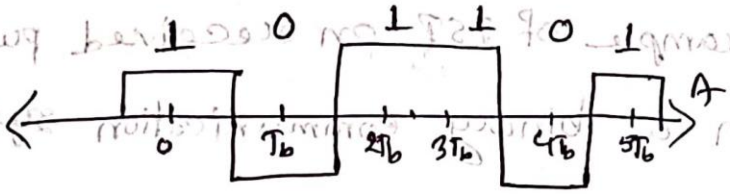


Data-0:-



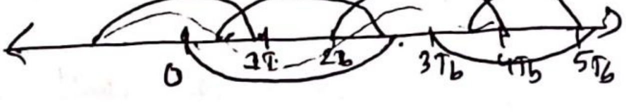
For the input data stream:-

1 0 1 1 0 1

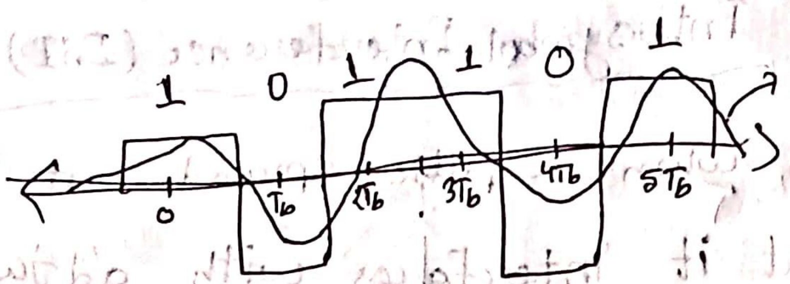


(0 এর জন্য নিচে Pulse)

Output is the superposition of each bit:-



$2T_b$ হবে সবগুলোর



For example:-
If the rectangular multilevel pulses are filled improperly as they pass through a communication system. They will spread into adjacent time slots & cause intersymbol interference.

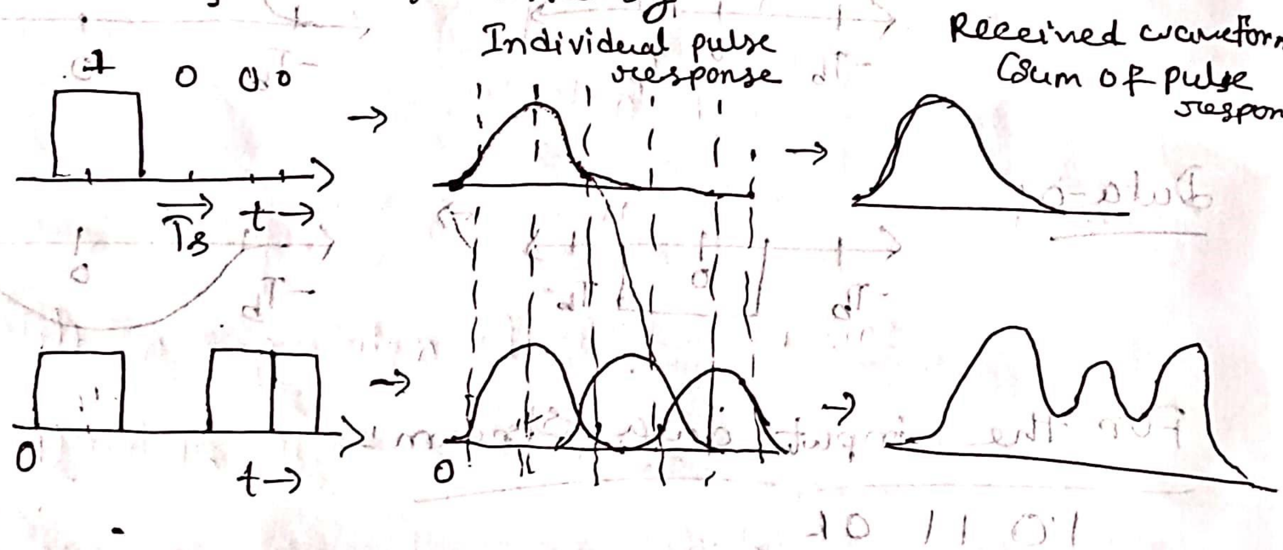


Fig:- Example of ISI on received pulses in a binary communication system

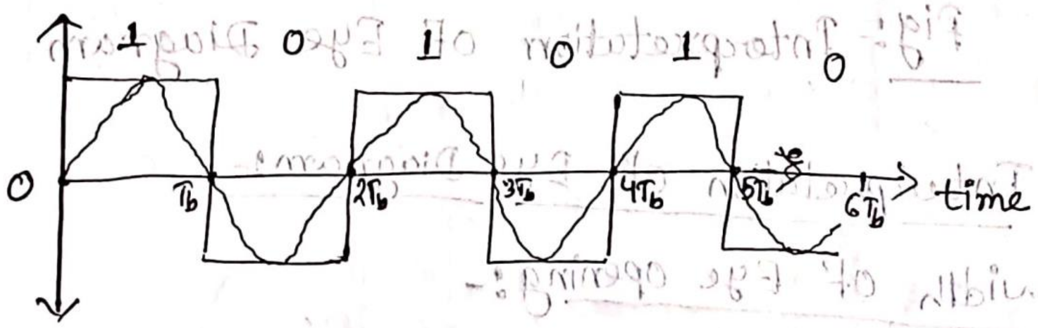
EYE Diagram

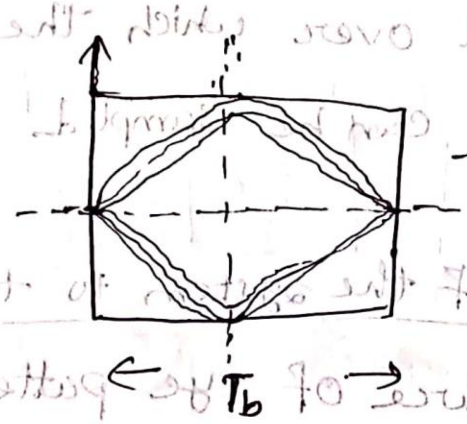
→ Summarises the effect of ISI by showing the responses of 0's & 1's.

→ Eye Diagram is generated by overlaying the plots of the received signal for every symbol time.

→ Looks like eye. So it is eye diagram.

How to get Eye Diagram:-





→ This diagram looks like an eye

⊛ Eye opening shows ISI.

⊛ Higher the ~~eye~~ opening, lower the ISI.

Eye Diagram Display:-

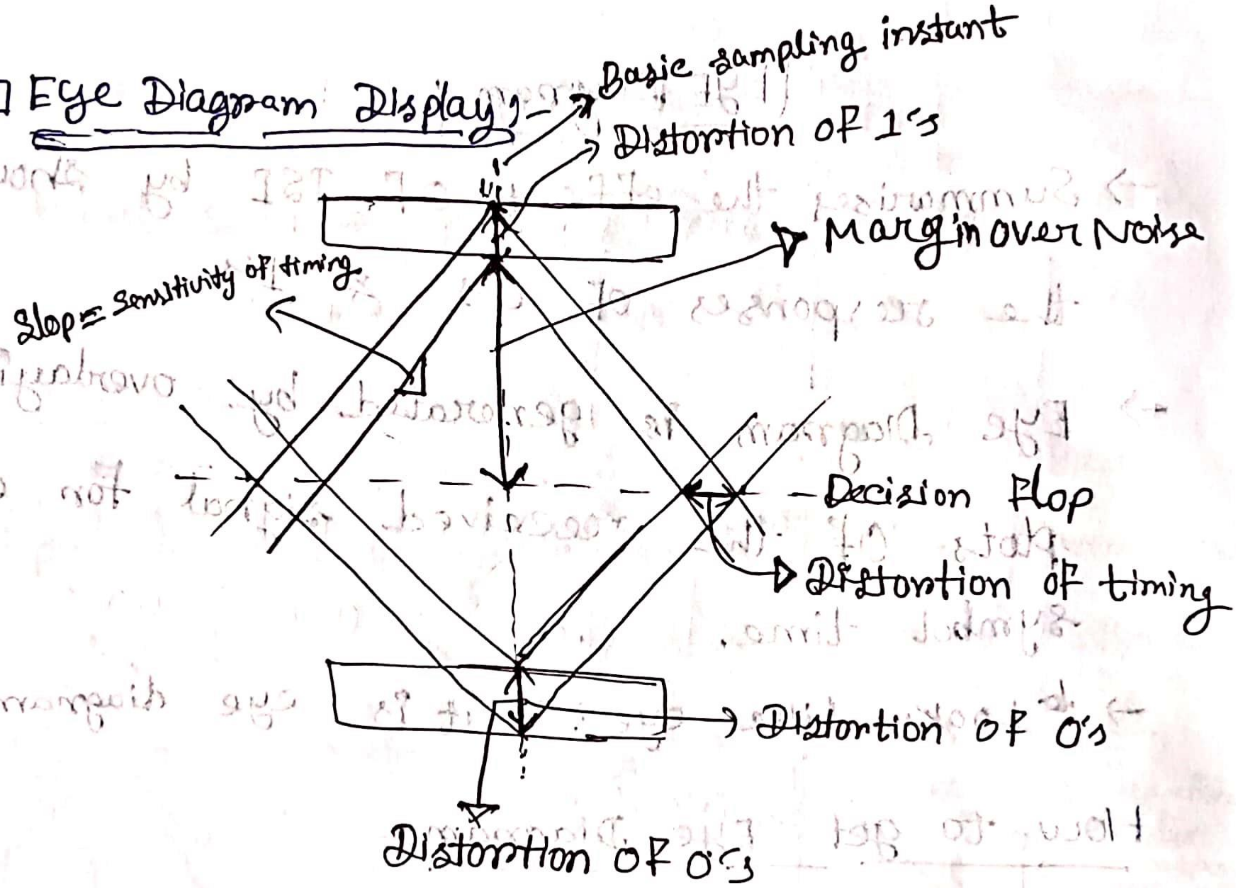


Fig:- Interpretation of Eye Diagram

Interpretation of Eye Diagram:-

(1) Width of Eye opening:-

Time interval over which the received distorted signal can be sampled with minimum errors.

(2) Sensitivity of the system to timing:-

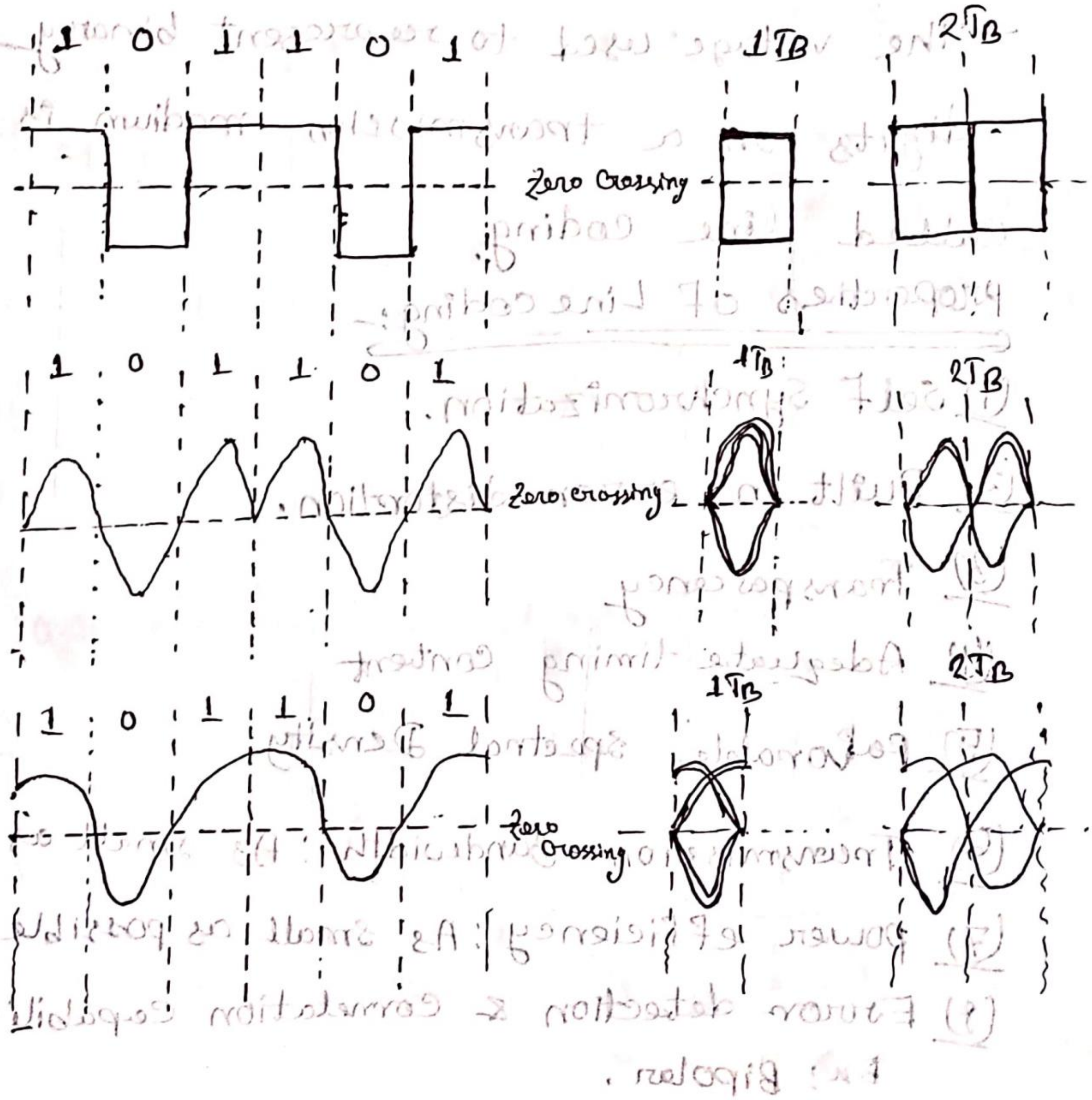
Rate of closure of eye pattern when the sampling time is varied.

(3) Margin over Noise:-

Height of eye opening for specified sampling time.

(4) Draw the eye pattern form:- (Question 2019(a), "Draw the eye pattern for the given ....")

(a) 101101



Line coding

(*) What is Line Coding? Write the properties of line coding?

=> Special coding system choose to allow transmission to take place in a Communication system.

-> The voltage used to represent binary digits on a transmission medium is called line coding.

properties of Line coding:-

(1) Self Synchronization.

(2) Built in error distortion.

(3) Transparency

(4) Adequate timing content

(5) Fatorable spectral Density

(6) Transmission Bandwidth : As small as possible.

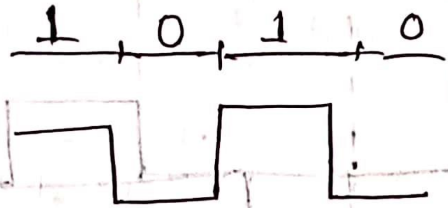
(7) power efficiency : As small as possible

(8) Error detection & correlation capability

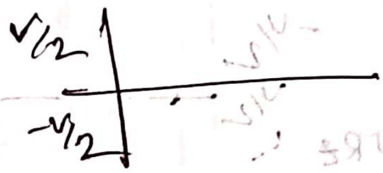
Ex: Bipolar.

RZ:- Return to zero (1 TB period where 0 is returned

NRZ: Full TB period এর জন্য 1 থাকবে। বা Half TB 1 Half TB 0 (RZ) এরকম থাকবে না। যা RZ এ থাকে।



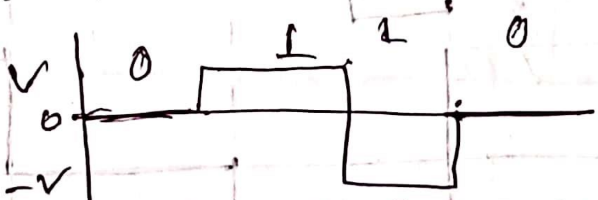
Unipolar, polar



Polar → $+V/2$ & $-V/2$ ⇒ অর্থাৎ, 0 এর জন্য নেগাটিভ থাকবে ($-V/2$)

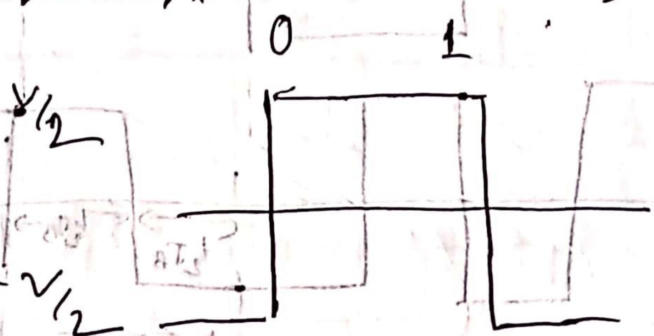
Unipolar → V & 0

Bipolar → V, 0, -V



(0 এর জন্য No sequence, 1 এর জন্য sequence)

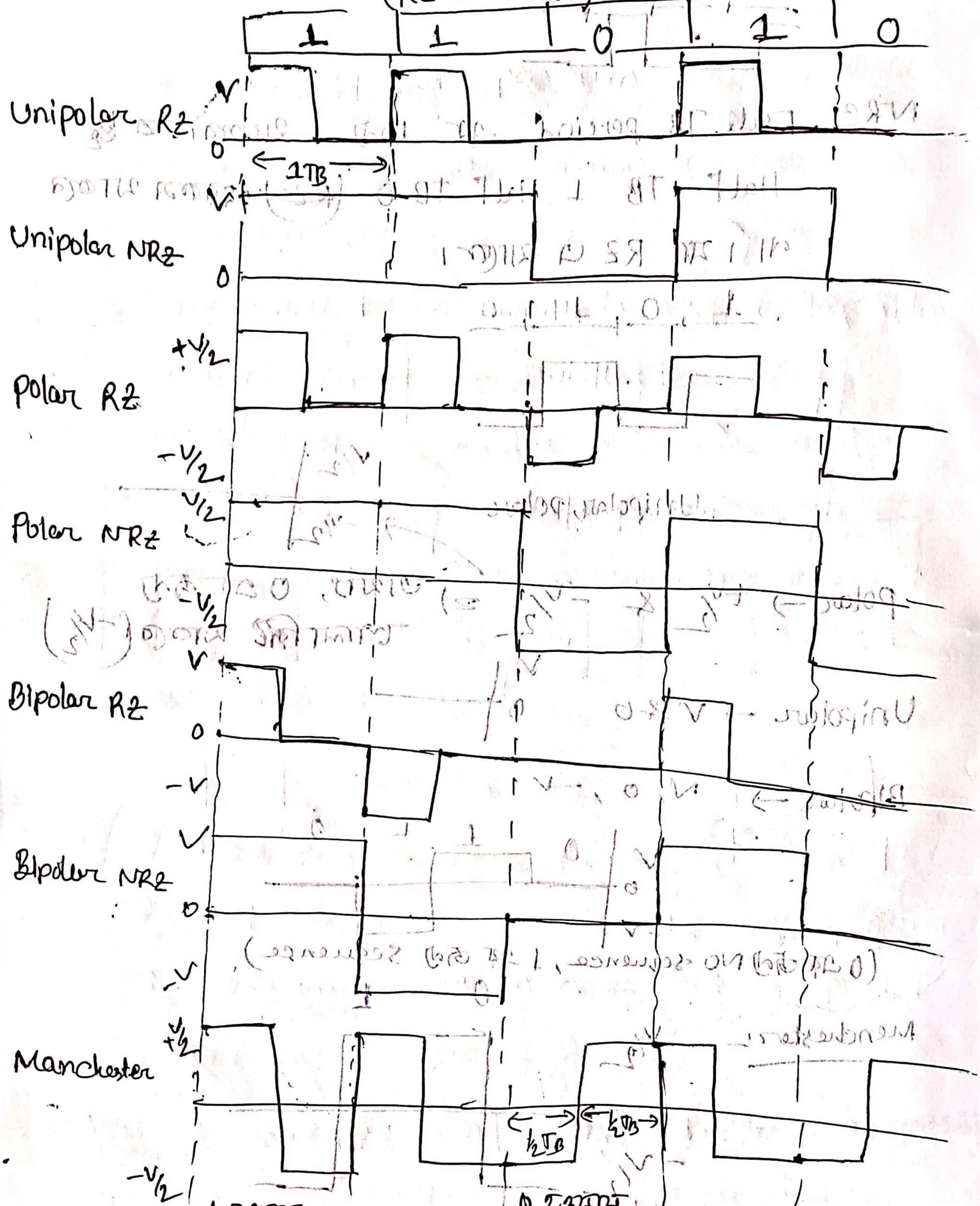
Manchester:-



যদি 0 থাকে তবে $-V/2$ to $V/2$ তে যাবে

1 " " $V/2$ to $-V/2$ তে যাবে $\frac{1}{2}$ TB period

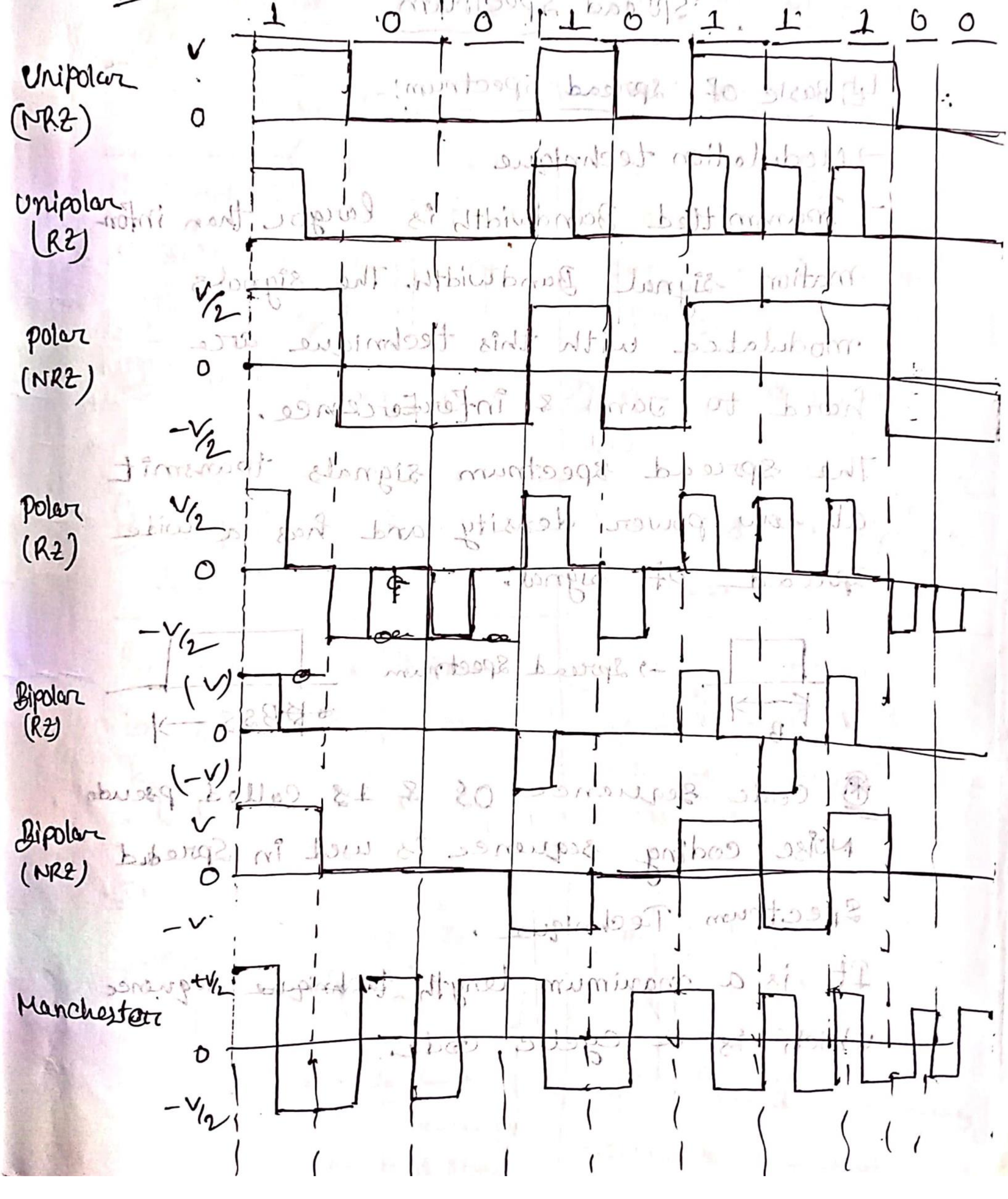
Unipolar, polar, Bipolar, Manchester (RZ & NRZ)



1 এ প্রথম
+V/2 to -V/2
@ 1/2 TB period

0 এ প্রথম
-V/2 to V/2
@ 1/2 TB period

Ex-02 Binary Data



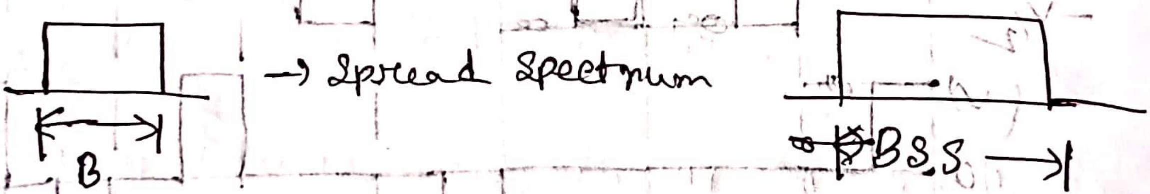
Spread Spectrum

Basic of spread spectrum:-

→ Modulation technique

→ Transmitted Bandwidth is larger than information signal Bandwidth. The signals modulated with this technique are hard to Jam & interference.

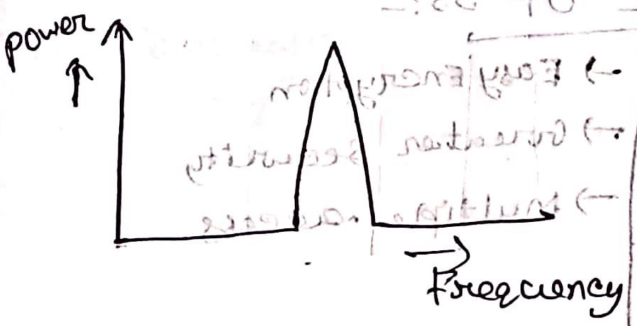
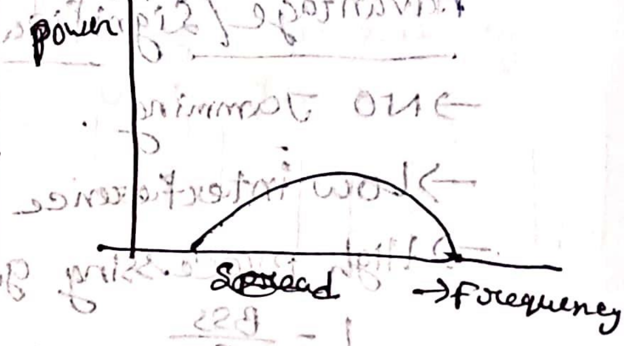
The spread spectrum signals transmit at low power density and has a wide spread of signal.



Code sequence 0s & 1s called pseudo Noise coding sequence is used in Spread Spectrum Technique.

It is a maximum length ~~technique~~ sequence which is a cyclic code.

Narrow Band vs Spread-Spectrum Signals:-

Narrow Band	Spread-Spectrum Signals
① Band of signals occupy a narrow range of frequencies.	① Band of signals occupy a wide range of frequencies
② power density is high.	② power density is low.
③ Spread of energy is low.	③ Spread of energy is wide.
④ Though the signal is good but it can be jammed.	④ The spectrum signals are highly resistant to interference or jamming.
⑤  → Narrow band signal	⑤  → Spread spectrum signal

Block Diagram of Spread Spectrum:-

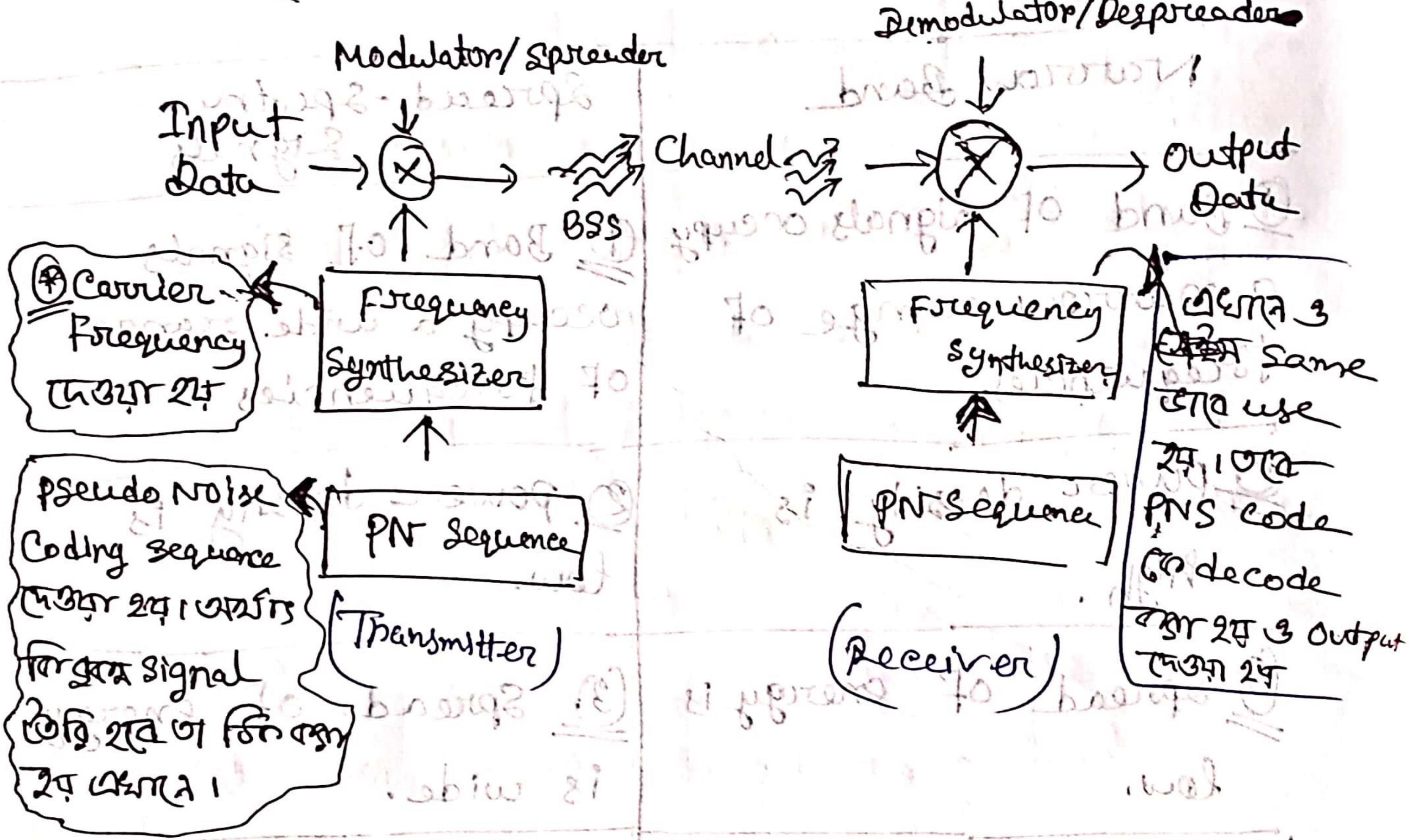


Fig:- Spread Spectrum

⇒ এভাবে Double PNS দেওয়া হয় Unauthorize Or third party interruption যেতো না হয়। অর্থাৎ for security purpose

Advantage/Significance of SS:-

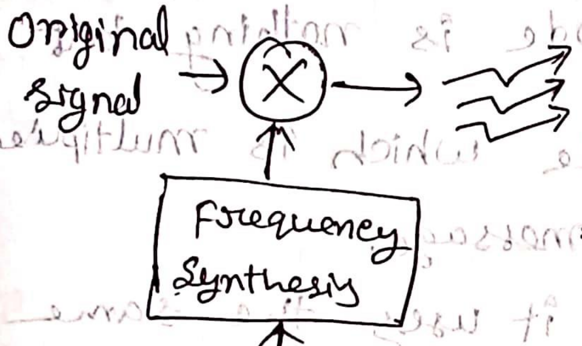
→ No Jamming.
→ Low interference
→ High processing gain

$$L = \frac{BSS}{B}$$

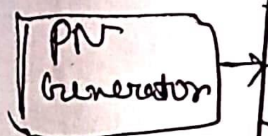
→ Easy Encryption
→ Greater Security
→ Multiple access.

Frequency Hopping Spread Spectrum(FHSS):-

→ Mode to change frequency than one to another in a specified time interval, which called frequency hopping.

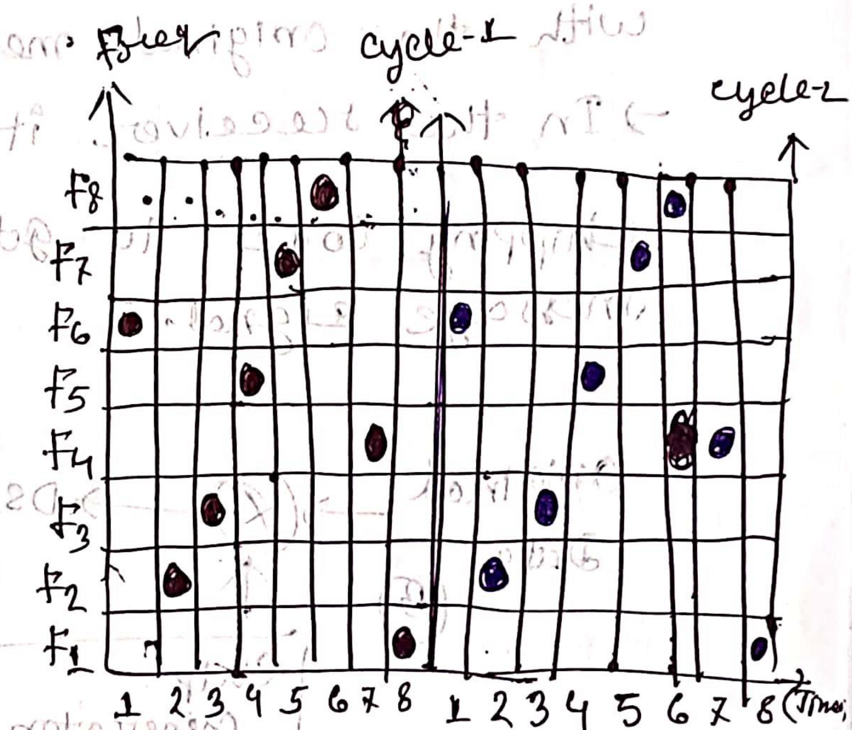


Code	Carrier
000	f_1
001	f_2
010	f_3
011	f_4
100	f_5
101	f_6
110	f_7
111	f_8



3 bit code

Total Combination $= 2^3 = 8$



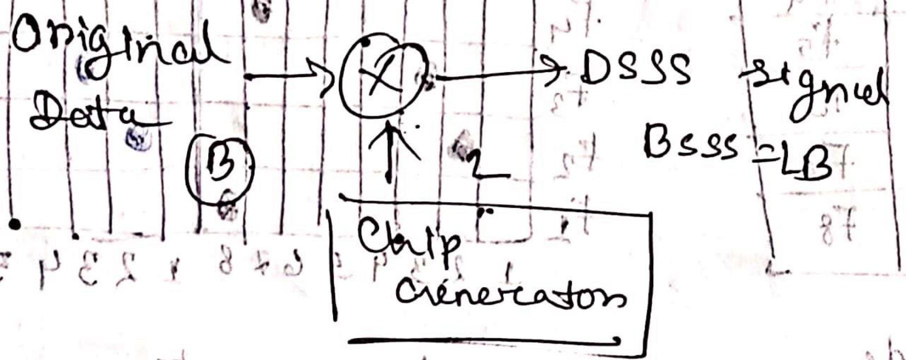
→ Changing frequency with time

Direct Sequence Spread Spectrum (DSSS):-

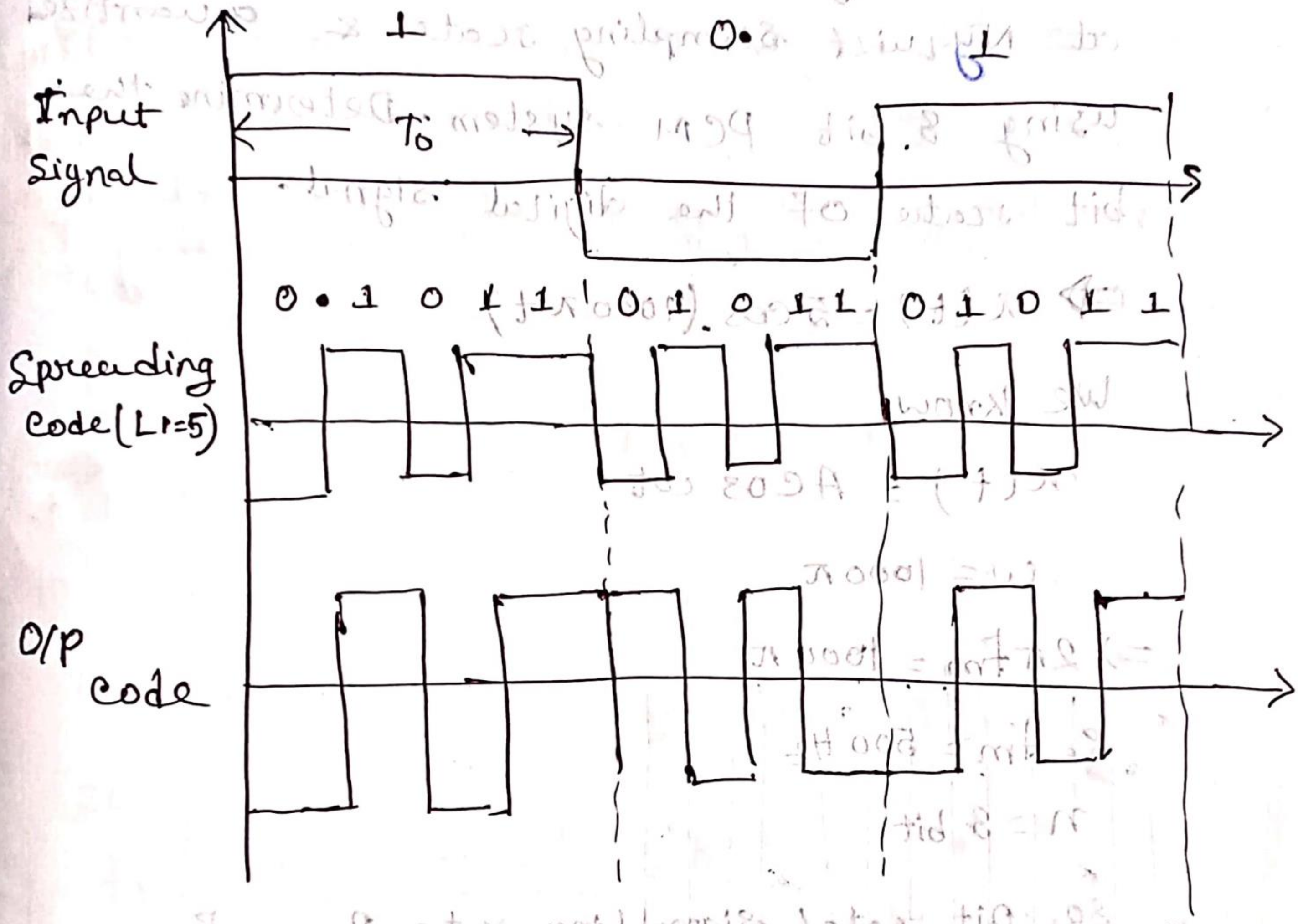
→ Using DSSS when someone sends data, each & every bit of the user data is multiplied by a secret code called as chipping code.

This ~~cod~~ chipping code is nothing but the spreading code which is multiplied with the original message.

→ In the receiver it uses the same chipping code to get the actual message signal.



Waveform of DSSS :- Spreading code = 01011



T_0 = Bit Time period

T_c = Chip Time period

Math

$\omega = 2\pi f_m$

Ex-01: A signal $x(t) = 5\cos(1000\pi t)$ is sampled at Nyquist sampling rate & quantized using 8 bit PCM system. Determine the bit rate of the digital signal.

$\Rightarrow$ $x(t) = 5\cos(1000\pi t)$

We know,

$x(t) = A\cos\omega t$

$\omega = 1000\pi$

$\Rightarrow 2\pi f_m = 1000\pi$

$\therefore f_m = 500\ Hz$

$n = 8$ bit

So, Bit rate/ signalling rate, $R_b = nF_s$

$= n \times 2f_m$

$= 8 \times 2 \times 500$

$= 8000$ bit/s.

(Ans)

[2] A PCM-TDM system multiplexes 10 band limited voice channel (300-3400 Hz) & uses a $\boxed{256 \text{ level}}$ quantizers. If the signal is sampled at a rate of $17\frac{11}{17}\%$ higher than the Nyquist rate then what'll be the max energy bandwidth of the transmission channel?

⟹ $L = 256 = 2^8$

$\therefore n = 8$

we know, $R_b = n\boxed{F_s}$

so,

$F_s = 2F_m + 17\frac{11}{17}\%$ of $2F_m$

$= (2\times 3400) + 17\frac{11}{17}\% (2\times 3400)$

$\cancel{= 6800 + 748}$

$\cancel{= 7548 \text{ Hz}}$

$= 6800 + 17.65\% \times 6800$

$= 6800 + 0.1765 \times 6800$

$= 6800 + 1200$

$= 8000 = 8 \text{ KHz}$

Maximum energy bandwidth of one channel $= 8\times8 = 64$ Kbit/s

" " " " 10 " $= 64\times10$ Kbit/s

$= 640$ Kbit/s

03 What should be the minimum sampling frequency to successfully reconstruct the signal $x(t) = \sin(100\pi t) + \sin(200\pi t)$? If 4096 levels are used to encode the sampled signal then what will be the bit rate of the system?

⇒ Here,

$\sin\omega t = \sin 200\pi t$

$\omega = 200\pi$

$\therefore 2\pi f_m = 200\pi$

$\Rightarrow f_m = 100$ Hz

Here, $L = 4096 = 2^{12}$

$\therefore n = 12$

We know,

$F_s = 2.f_m = 2\times100 = 200$ Hz

$R_b = nF_s = 12\times200 = 2400$ bit/s

4) Three analog signals having bandwidths of 1200 Hz, 600 Hz & 600 Hz are sampled at their Nyquist rate, encoded with 12 bit words, and time division multiplexed. Find the bit rate for the multipled signal.

~~Bandwidth of multiplexed signal, f_m~~

⇒ We know,

bit rate, $R_b = nF_s$

$n = 12$ bit words

$F_s = 2f_m =$

Here, $f_m = (1200 + 600 + 600)$ Hz

$= 2400$ Hz

$\therefore 2f_m = 2 \times 2400 = 4800$ Hz

$R_b = nF_s = 12 \times 4800$

$= 57600$ bit/s.

5) Two analog signals of $\boxed{200\ KHz}$ & $\boxed{300\ KHz}$ are sampled at their Nyquist rates, encoded using PCM & $\boxed{512\ level}$ binary code & multiplexed. Find the output bit rate

⇒ $L = 512 = 2^9$

$\therefore n = 9$

$f_m = (200+300) = 500\ KHz$

$\therefore f_s = 2f_m = 1000\ KHz$

$R_b = n f_s$

$= 9 \times 1000$

$= 9000$ kbit/s.

Q) In PCM, total number of quantization level is 256, highest frequency is 4KHz and sampling frequency is 12.5% of Nyquist frequency. Find the bit rate.

⇒ We know,

$R_b = nf_s$

Here, $L = 256 = 2^8$

$\therefore n = 8$

So,

Frequency, $f_m = 4KH$

$\therefore f_s = 2f_m + 12.5\%$ of $2f_m$

$= 2 \times 4 + 12.5\% \, (2 \times 4)$

$= 8 + 1$

$= 9 \, KHz$

$\therefore R_b = 8 \times 9 = 72$ kbits/s.

(Ans)

(7) An Analog signal having a bandwidth of 8KHz is converted to PCM signal using sampling at a rate 50% above Nyquist rate. The samples are quantized into 1024 levels. Determined the bit rate of the system.

⇒ We know,

Bit rate, $R_b = nf_s$

$L = 1024 = 2^{10}$

$n = 10$

So, $f_m = 8$ KHz

We know,

$f_s = 2f_m + 50\%$ of $2f_m$

$= (2 \times 8) + 50\% (2 \times 8)$

$= 16 + 8$

$= 24$ K~~Hz~~/s

$\therefore R_b = nf_s = 10 \times 24 = 240$ Kbit/s.

(Ans)

8] Two analog signals, having bandwidth 500 Hz & 200 Hz, are sampled at 20% higher than Nyquist rates, encoded with 8 bit words and time division multiplexed. Find the bit rate of the multiplexed signal.

$\Rightarrow$ We know,

The bit rate, $R_b = n F_s$.

$n = 8$

$F_m = 500 + 200 = 700$ Hz

$F_s = 2F_m + 20\%$ of $2F_m$

$= \cancel{700 + 20\% \times 700} = 1400 + 20\% \times 1400$

$= 1680$

$= \cancel{700 + 140}$

$\cancel{= 840}$

$\therefore R_b = \cancel{8 \times 840} = 8 \times 1680$

$= 13440$ kbit/s

(Ans)

9) A signal has two frequency component of 3KHz & 4KHz. If the sampling frequency is 25% higher than Nyquist rate and it is coded into 512 level to make PCM signal, what is the bit rate of the signal?

⇒ We know,

bit rate $R_b = n \cdot f_s$

Here, $L = 512 = 2^9$

$\therefore n = 9$

$f_m = \cancel{(3+4) = 7\ KHz}$ 4 KHz

$f_s = 2f_m + 25\%$ of $2f_m$

$= (2 \times 4) + 25\%$ of (2×4)

$= \cancel{(14 + 3.5)}$ 8 + 2

$= \cancel{17.5\ Kbit/s}$, 10 KHz

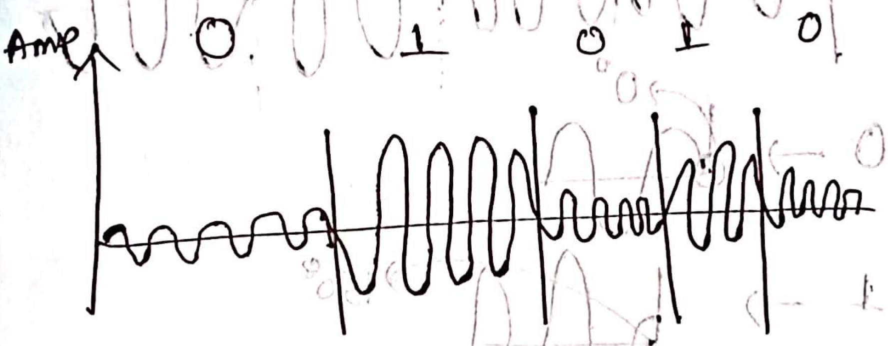
$\therefore R_b = 9 \times 10 = 90$ Kbit/s.

(Ans)

ASK, PSK, FSK, DPSK, QPSK

ASK (Amplitude Shift King):-

The strength of the carrier signal is varied to represent binary 1 or 0. Frequency & Phase remains constant.

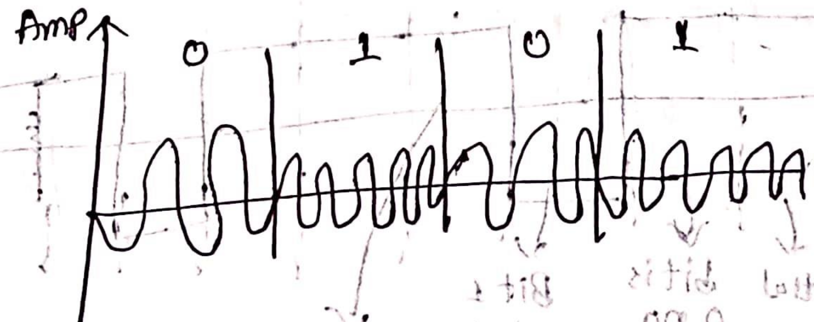


0 → Low voltage

1 → High voltage

FSK (Frequency Shift King):-

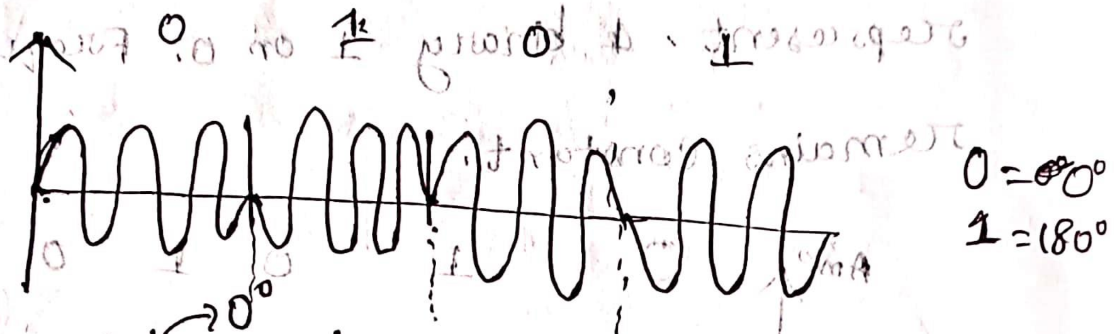
Frequency of the carrier signal is varied to represent binary 1 & 0. Amplitude & phase constant. (Avoide Noise)

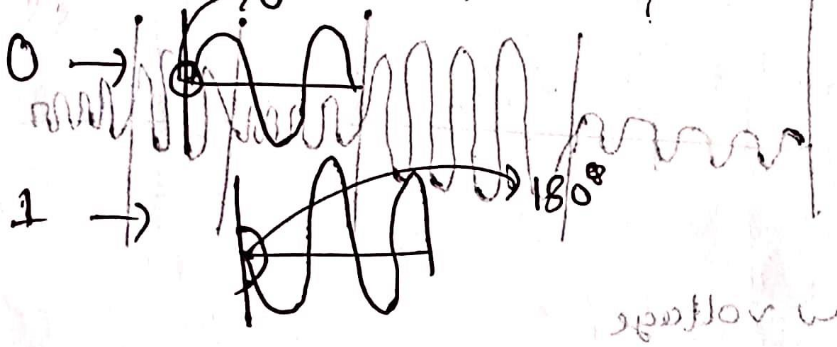


0 → এর জন্য Frequency কম

1 → " " " বেশি

PSK → phase is varied, Amplitude Frequency is constant (0°, 90°, 180°)

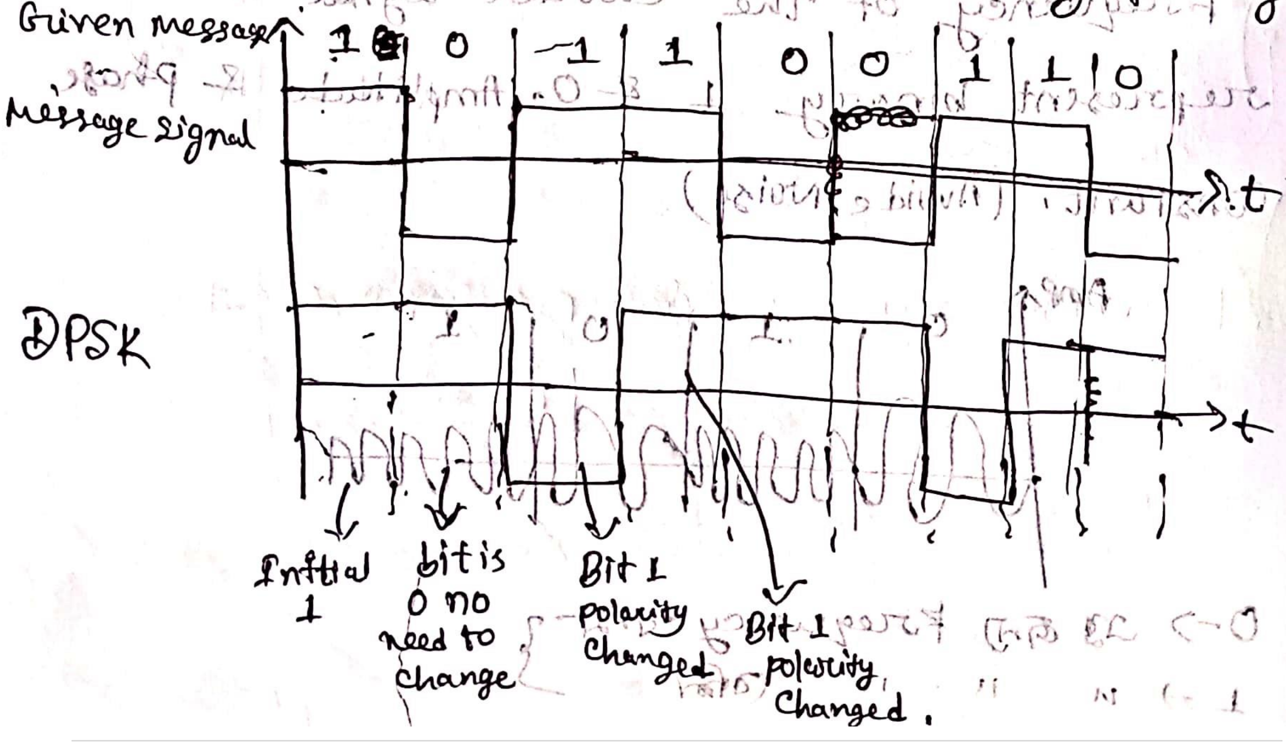




DPSK (Differential Phase Shift King):

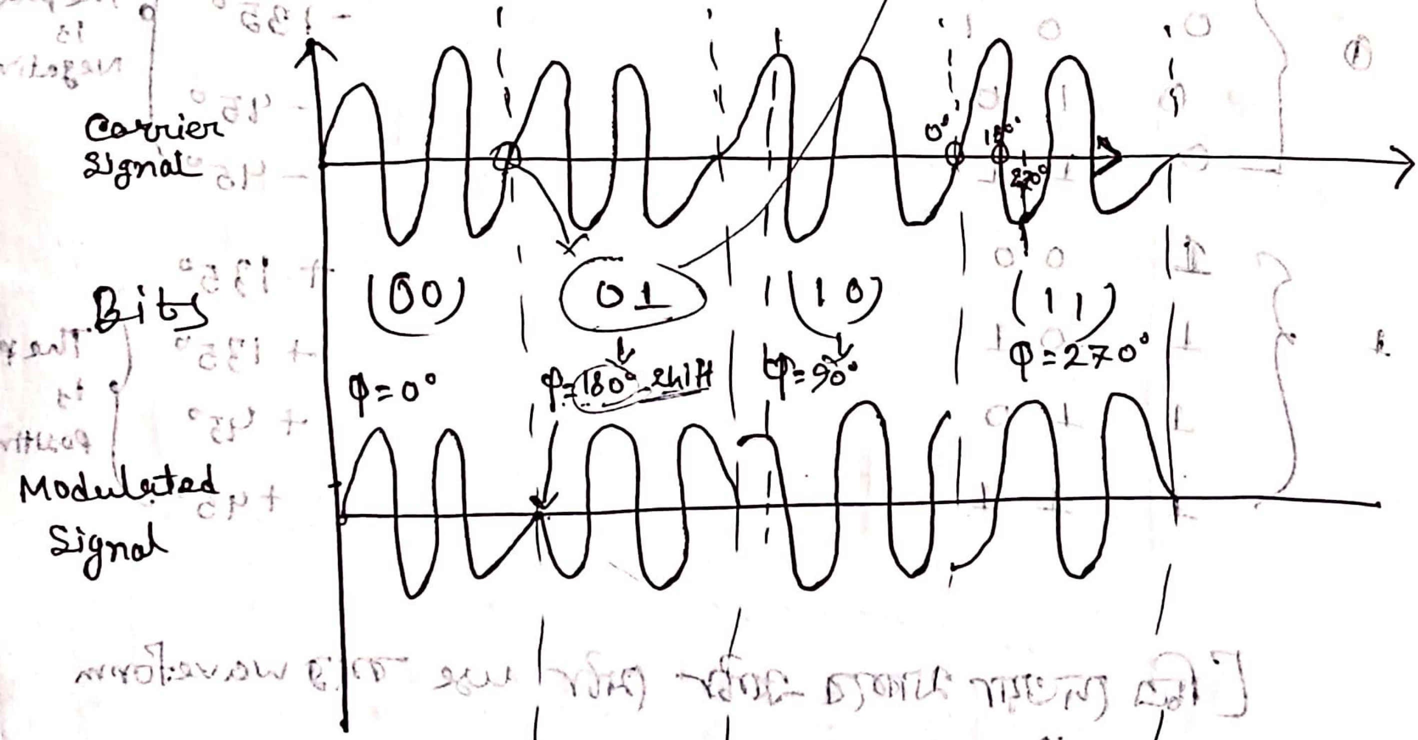
→ IF next data is 1, then change polarity of o/p

→ IF next data is 0, then don't change polarity of o/p.



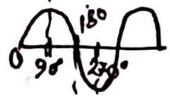
QPSK:- A form of PSK where two bits are modulated at once. It selects 4 possible carrier phase shift 0°, 90°, 180°, 270°.

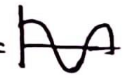
bits	→ ϕ
00	→ 0°
01	→ 180°
10	→ 90°
11	→ 270°



How to draw?

=> Bit যেভাবে আসবে (00, 01, 10, 11) sequence এ তাহলে degree count হবে



0° =  90° = 

F) ~~QPSK~~ 8-QAM:-

ASK + PSK = QAM

8 QAM — Bits per symbol → 3

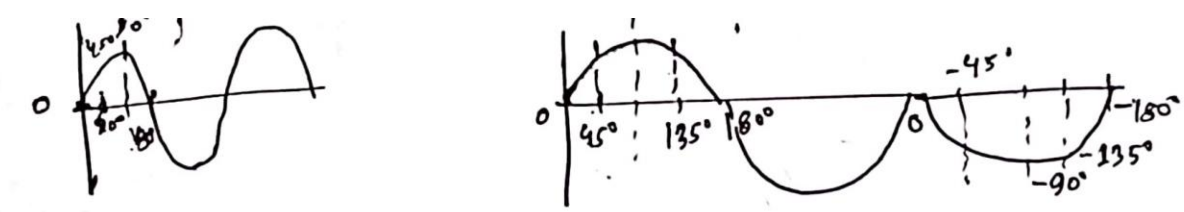
$2^3 = 8$

⇒ 3 bits are combined together to form a symbol (tribits).

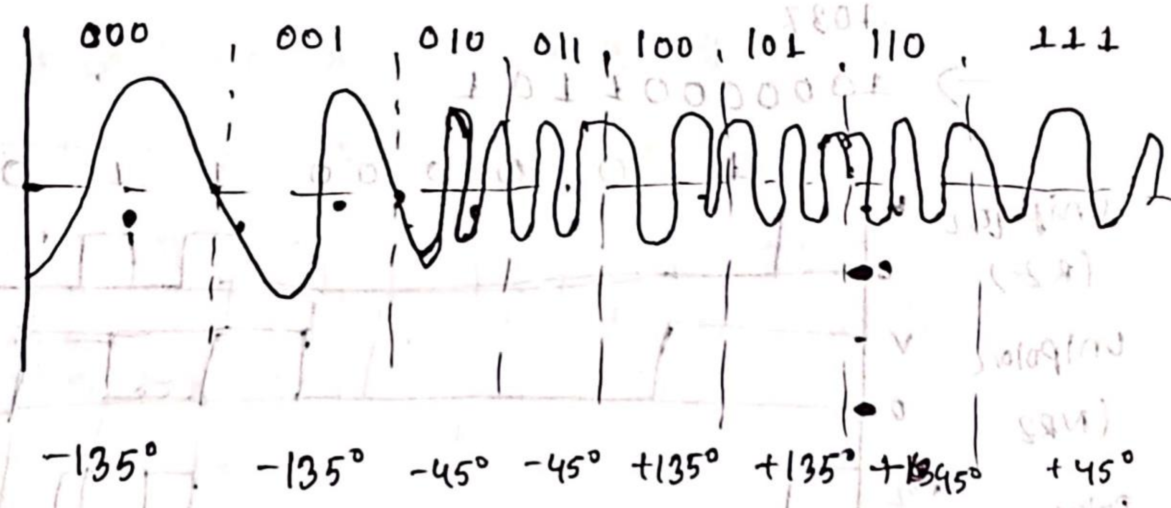
Quadrature

Q	I	C	Phase	
0	0	0	− 135°	The phase is Negative
0	0	1	− 135°	
0	1	0	− 45°	
0	1	1	− 45°	
1	0	0	+ 135°	The phase is Positive
1	0	1	+ 135°	
1	1	0	+ 45°	
1	1	1	+ 45°	

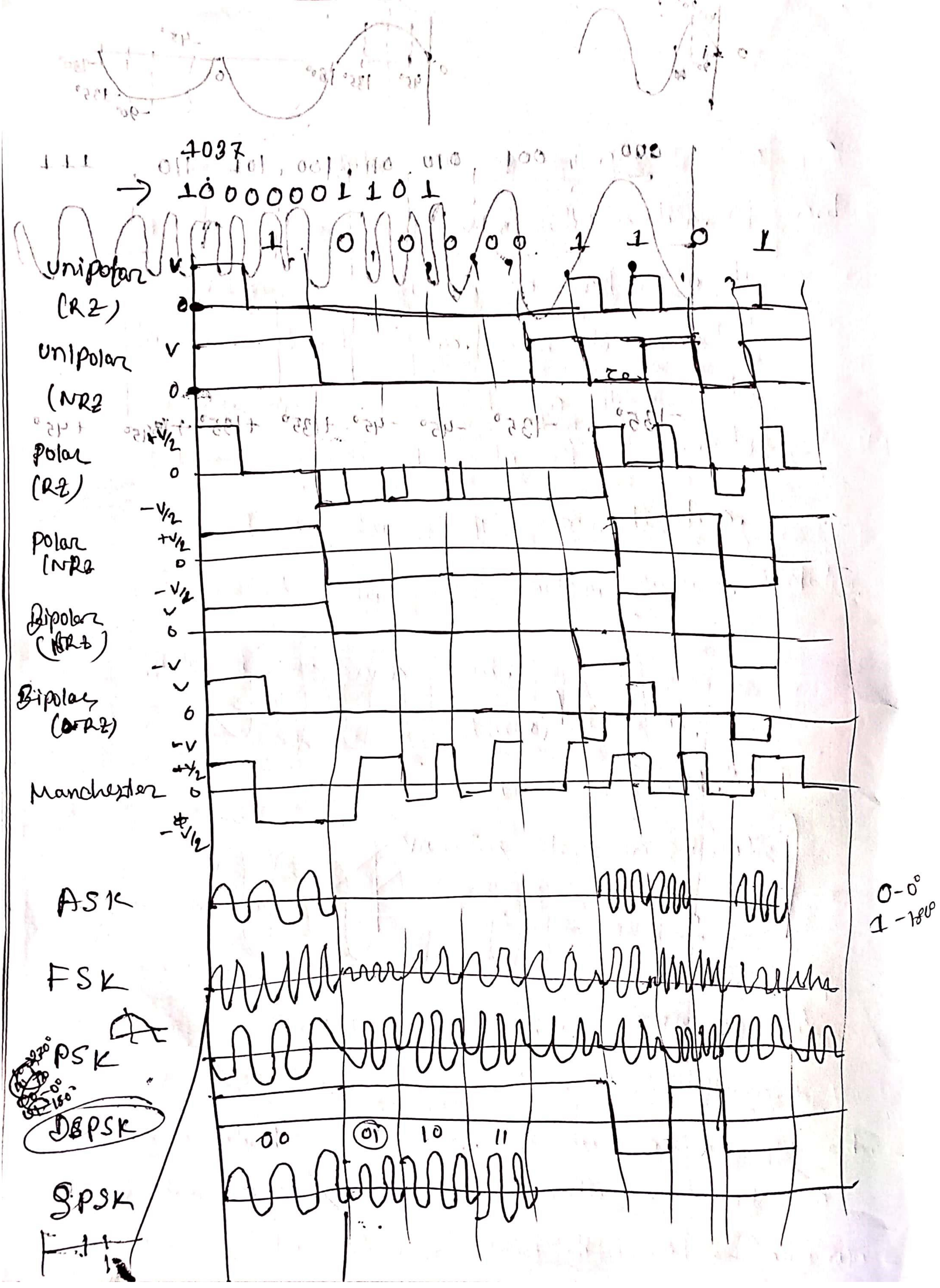
[চিত্র দেওয়া থাকবে একটা সেটা use করে waveform আঁকতে হবে]



0
45° 90°
180°
0
45°
135°
180°
0
-45°
-90°
-135°
-180°



000
001
010
011
100
101
110
111
-135°
-135°
-45°
-45°
+135°
+135°
+45°
+45°


1037
→ 10000001101
1 0 0 0 0 0 0 1 1 0 1
Unipolar (RZ)
Unipolar (NRZ)
Polar (RZ)
Polar (NRZ)
Bipolar
Bipolar
Manchester
ASK
FSK
PSK
DBPSK
QPSK
00
01
10
11
0-0°
1-180°

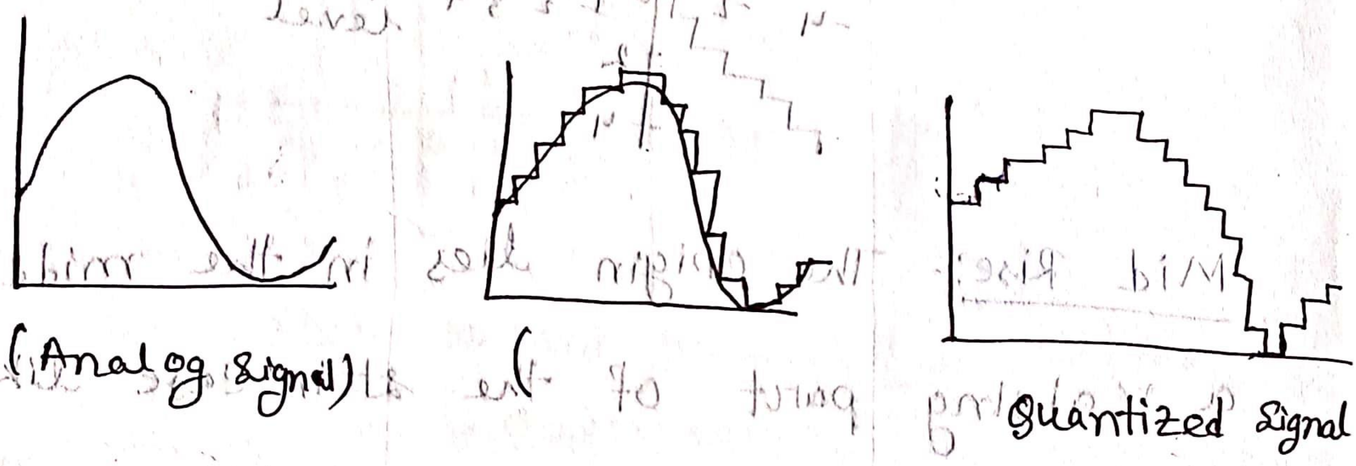

LECTURE

FINAL

Math

(*) Signal $x(t) = 5\cos(1000\pi t)$, sampled at Nyquist sampling rate & quantized using 8 bit PCM system. Determine the bit rate of the digital signal.

Quantization



Representation Levels:- The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels.

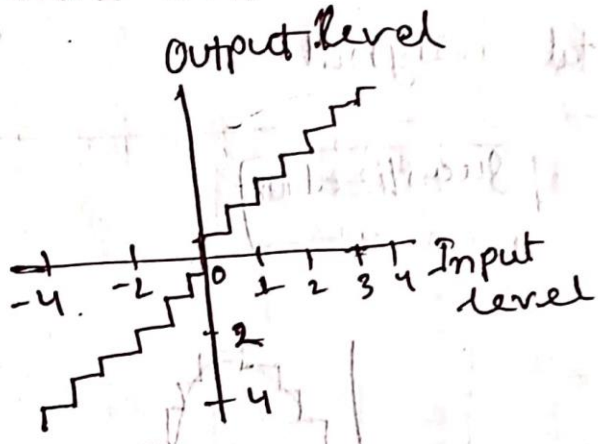
Quantum / Step-Size:- The spacing between the two adjacent ~~repreprepres~~ representation levels is called a quantum or step size

Types of quantization:-

2 types :- Uniform! - quantization level are uniform spaced
NonUniform:- quantization levels are unequ...

1) Uniform or Mid Rise quantization:-

The type of quantization in which the quantization levels are uniformly spaced is called uniform quantization.



Mid Rise:- The origin lies in the middle of a raising part of the stair case like graph. The quantization levels in this type are even in number.

~~Non uniform~~/ Mid tread Quantization:-

The type of quantization in which the quantization levels are uniformly spaced is termed as a Uniform quantization.

The type of quantization in which the quantization levels are unequal & mostly the relation between them is logarithmic, is termed as Non-Uniform quantization.

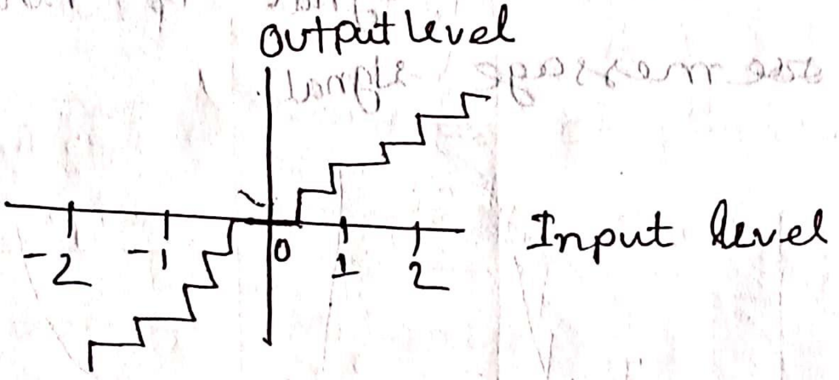


Fig:- Mid-tread type uniform quantization

Mid Trade:-

The type is so called because the origin lies in the middle of a tread of the stair case like graph. The quantization levels in this type are odd in number.

Both mid-rise & mid-trade type of uniform quantizers are symmetric about the origin.

Quantization Error

The difference in the input value and the quanti value of the signal is known as the quantization error.

If The quantization signal = $m_q(t)$

Message signal = $m(t)$

So, Quantization error, $q_e = m(t) - m_q(t)$

That means received signal is not equal to the ~~sse~~ message signal.

Companding

Compressing + Expanding = Companding

That means it does both.

It's non-linear technique या PCM এ use হয়

Noise reduce করতে।

That means, it compresses the data at the transmitter & expands the same data at the receiver.

There are two techniques of this :-

(1) A - Law Companding Technique:-

It is used when Mid-rise.

The techniques:-

1) Uniform quantization is achieved at A = 1, where the characteristic curve is linear and no compression is done.

2) A-law has mid-rise at the origin. যাতে ও থাকে Non-Zero value দেয়।

3) Used for PCM telephone system.

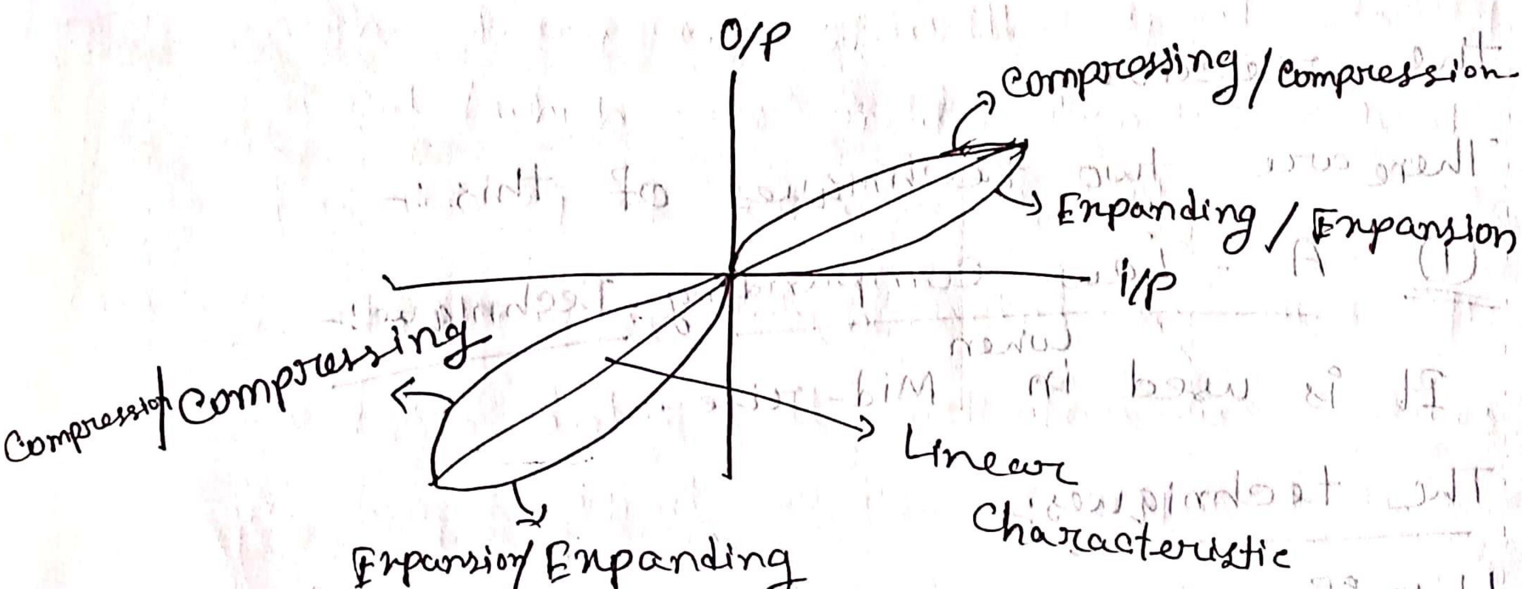
μ-law Companding Technique:-

⇒ 1| Uniform quantization is achieved at $\mu = 0$, where the characteristic curve is linear and no compression is done.

2| μ - law mid-tread at the origin.

যদি এটার value 0.

3| It is used in speech & music signal.

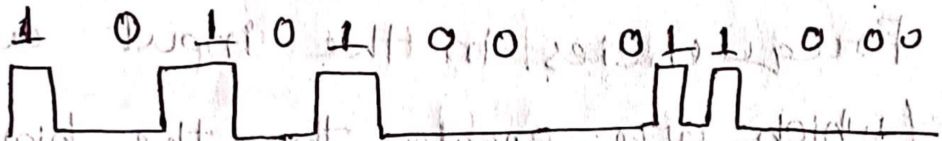


Pulse Code Modulation (PCM)

Converts an analog input signal to the digital signal, which is a combination of the binary sequence created from the binary digits 0 & 1.

PCM wave is series of digits.

Electrical representation of PCM:-



(a)

0 = Indicates absence of pulse

1 = " presence of "

Basic Elements of PCM System:-

The transmitter section of PCM circuit consists of sampling, quantizing & Encoding.

In the receiver section the regeneration of impaired signal happens. Most important parts are Quantizer, decoder, Reconstruction Filter.

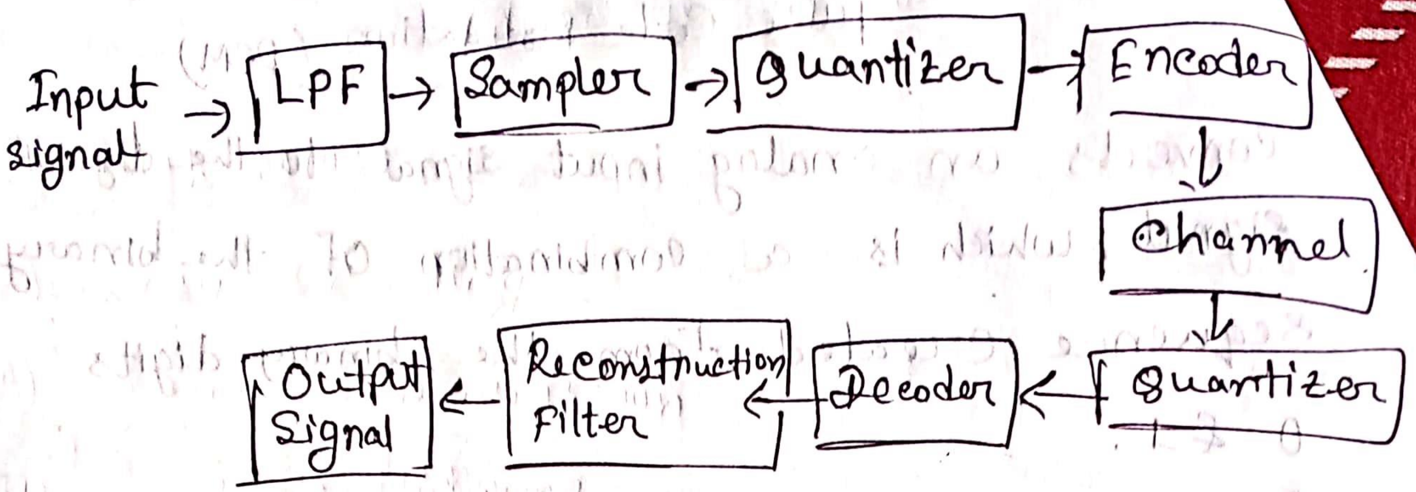


Fig: Basic element of PCM system.

LPF:- Low pass filter. It eliminates all kinds of high frequencies in the input analog signal (which are greater than the highest frequency of message signal.

Sampler:- Technique which helps to collect the sample data at instantaneous values of message signal. So to reconstruct the original message signal the sampling rate must be greater than twice the highest frequency component w.

Quantizer:- Reduce the excessive bits. It reduces the redundant bits and compressed the value.

Encoder:- It does the digitization. It ~~digi~~ designates each quantized level by a binary code

ommunication Channel:-

A medium between transmitter & receiver. Transmits a pcm signal from the transmitter to receiver.

Also includes a repeater that can regenerated the signal, improve signal strength & reduce noise.

Regenerative Repeater:-

It increases the signal strength. ~~& reduce noise effects~~. It compensate the signal loss & reconstruct the signal & also increase its strength.

Decoder:- It ~~rege~~ reconstruct pulse coded waveform to reproduce the original signal

Reconstruction Filter:-

After digital to analog conversion, a low-pass-filter is employed, which is called as the reconstruction filter to get back the original signal

PCM circuit digitizes the given analog signal, codes it then samples it. & then transmit it ~~to~~ in an analog signal. The whole process is repeated in reverse way to get the original signal

Advantages of PCM

① High Noise

② Easy Encoding

③ Secure Transmission

④ Easy Multiplexing

⑤ High efficiency

⑥ Use of repeaters

⑦ Data storage (The digital data can be stored easily)

Disadvantage of PCM:-

① Complex process

② Large Bandwidth

③ Quantization Noise:

Bandwidth of PCM Signal

Let the quantizer uses N ~~bits~~ number of binary digits to represent each level.

Number of levels that can be represented by N digit will be L.

$$L = 2^n$$

$N = 3$ then total number of levels will be

$$L = 2^3 = 8$$

* Number bits per second is called signalling rate of PCM which is denoted by r.

$\therefore r = N f_s$ bits/sec $\uparrow N$

* Bandwidth needed for PCM transmission will be half of the signalling rate.

$\therefore$ Transmission bandwidth of PCM :-

$$B_T \geq \frac{1}{2} r$$

$$\Rightarrow B_T \geq \frac{1}{2} N f_s$$

$$\Rightarrow B_T \geq \frac{1}{2} N 2 \cdot W$$

$$\therefore B_T \geq N W$$

Differential pulse coded Modulation

DPCM is a signal encoder that uses the baseline of pulse code Modulation (PCM). কিন্তু আরও কিছু functionalities ও add করে signal এর sampling এর উপর predict করে।

→ Input Analog or digital দুটোই হতে পারে। যদি Input continuous Time Analog Signal হয় তবে এটা প্রথমে sampled হতে হবে যাতে DPCM encoder এর Input discrete Time signal হয়।

→ DPCM receiver Decoder & prediction filter দ্বারা গঠিত।

→ Noise না থাকলে Encoded receiver input, encoded transmitter output এর সমান হবে।

Why used? (দুইটা bit এর difference নয়)

→ Reduce redundancy of digital signal

→ If redundancy is reduced then then the overall bitrate will decrease & number of bits required to transmit one sample will also reduce.

→ DPCM works on the principle of prediction.

→ ADPCM is a variant of DPCM that varies e size of quantization step, to allow further reduction of the required data bandwidth. For a given signal-to-noise ratio.

☐ ~~Why~~ ~~NOT~~ PCM?

=) Special circuit use হয় → predictor (or prediction filter)

→ To predict values of future samples of $x(t)$

Example:-

	V_L				V_H
$x(nT_s)$	0	1	3	6	10
		(1-0)	(3-1)	(6-3)	(10-6)
$e(nT_s)$ Difference Signal		1	2	3	4

Step Size = $\frac{V_H - V_L}{9}$

In case 1

10 − 0 = 10

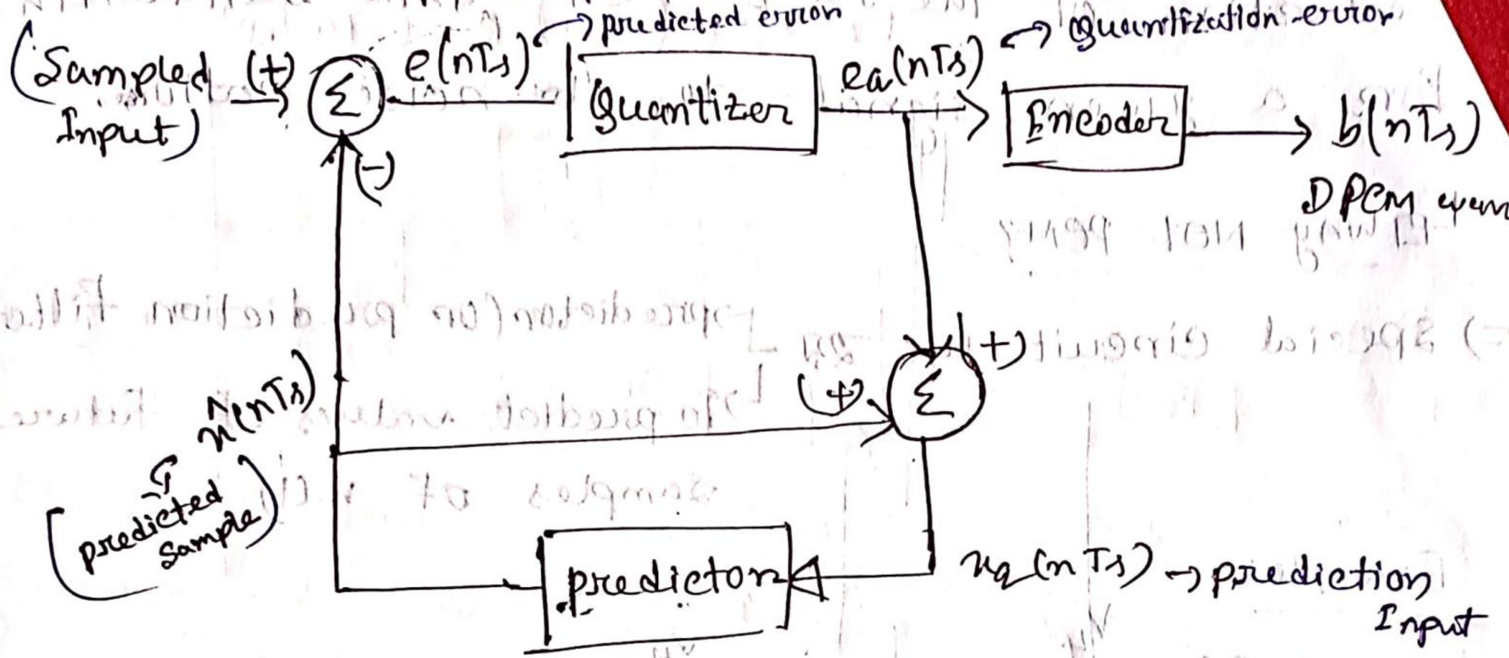
Case - 2

4 − 1 = 3

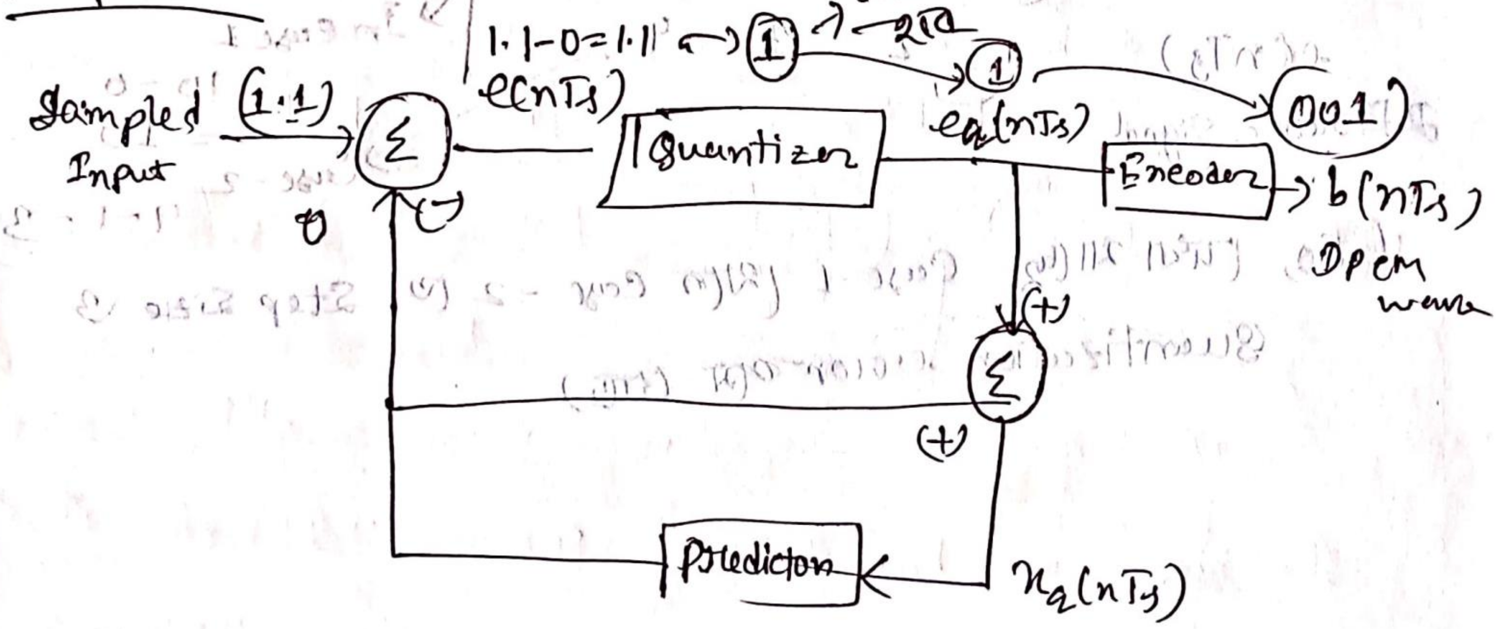
So, দেখা যাচ্ছে Case-1 থেকে Case-2 তে Step size ও Quantization error কম হচ্ছে)

DPCM Transmitter

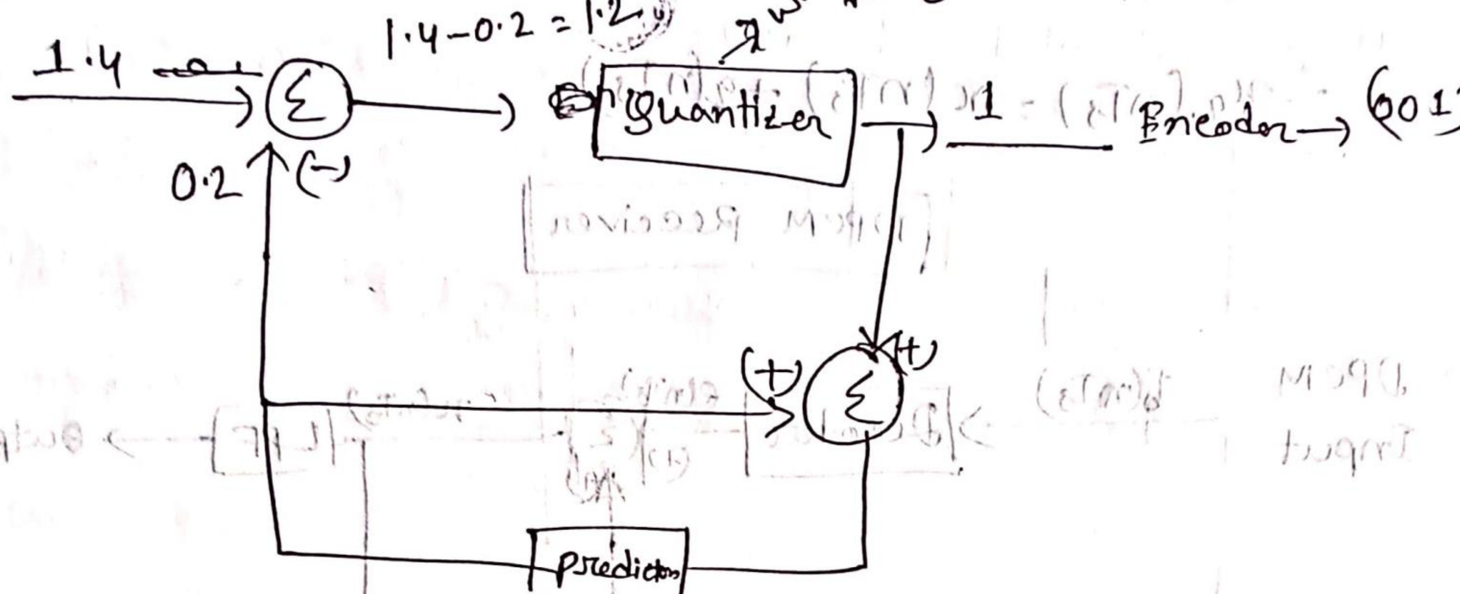
$x(nT_s) \rightarrow$ Sampled input to DPCM transmitter



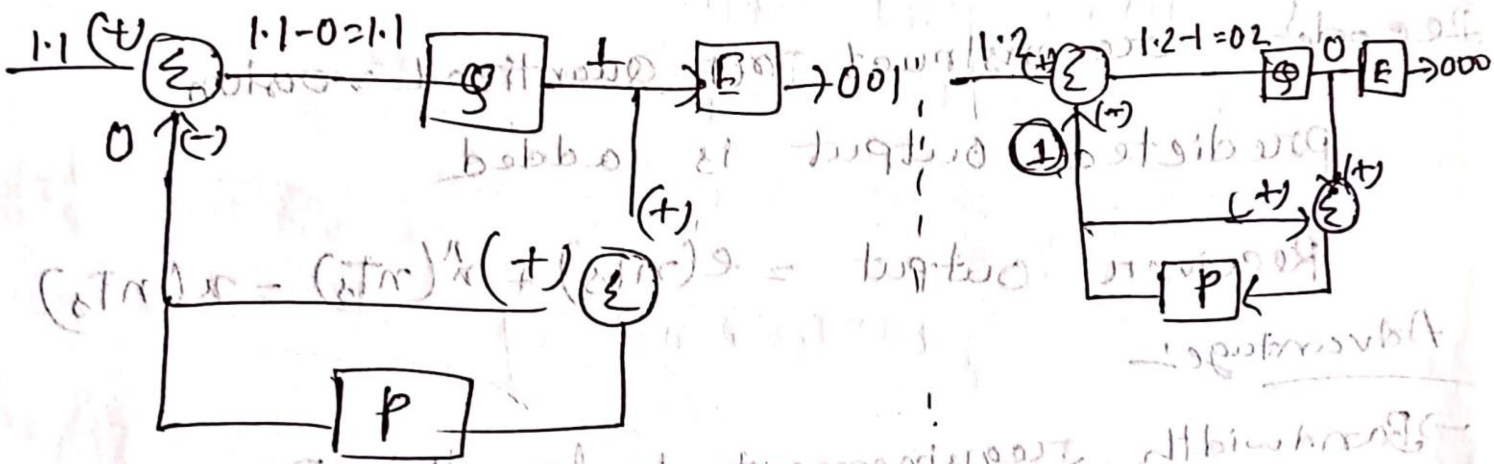
Example-11



Example-2



Example-3



$$\boxed{001000}$$

$$e(nT_s) = \{x(nT_s) - \hat{x}(nT_s)\}$$

$$e_q(nT_s) = \{e(nT_s) + q(nT_s)\}$$ --- Quantization output

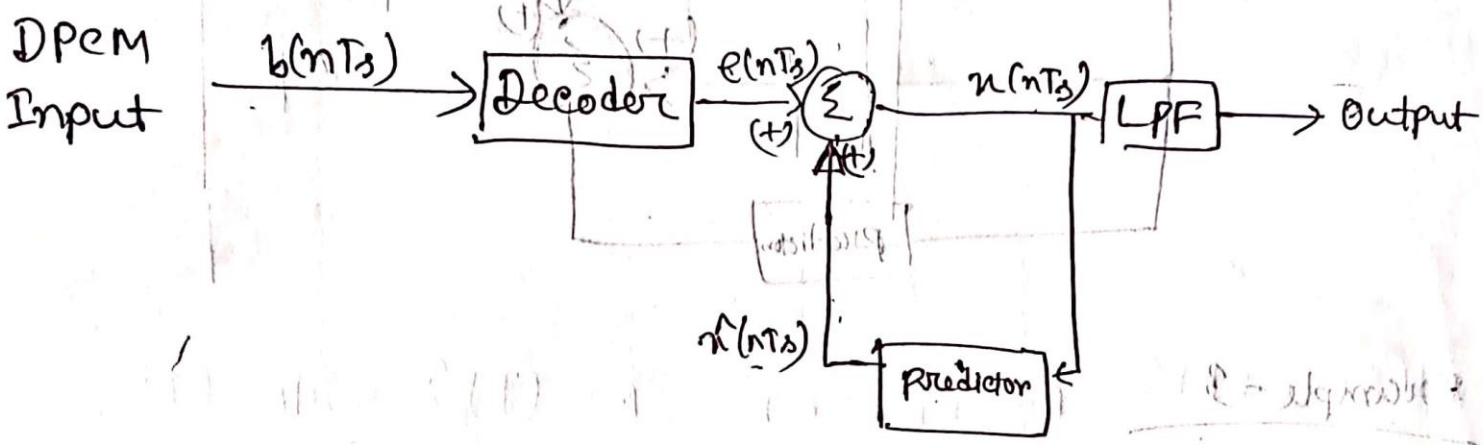
$$x_q(nT_s) = e_q(nT_s) + \hat{x}(nT_s)$$ --- output of adder

$$= e(nT_s) + q(nT_s) + \hat{x}(nT_s)$$

$= e(nT_s) + q(nT_s) + x(nT_s) - e(nT_s) -$

$\therefore x_q(nT_s) = x(nT_s) + q(nT_s)$

$\boxed{\text{DPCM Receiver}}$



Decoder reconstruct ~~ms~~ quantized version

predicted output is added

Receiver output $= e(nT_s) + \hat{x}(nT_s) = x(nT_s)$

Advantage:-

→ Bandwidth requirement is less than PCM

→ Quantization error is reduced because of predictor

→ Bits are reduced.

Application:-

→ Speech

→ Image

→ Audio Compression

→ Suitable for real time Application

Delta Modulation:-

PCM - এ N digit number of binary per quantized sample এ transmit হয়।

যার কারণে Bandwidth বেশি লাগে। এটা থেকে মুক্তি পেতেই Delta Modulation use করা হয়।

Concept:-

এখানে 1 bit per sample instead of n bit

$$Bw = nF_s$$

$$= 1 \cdot F_s \Rightarrow F_s$$

$x(t)$ এর সাথে $\hat{x}(t)$ এর Comparison হয় আর সেটার result টাই transmit হয়।

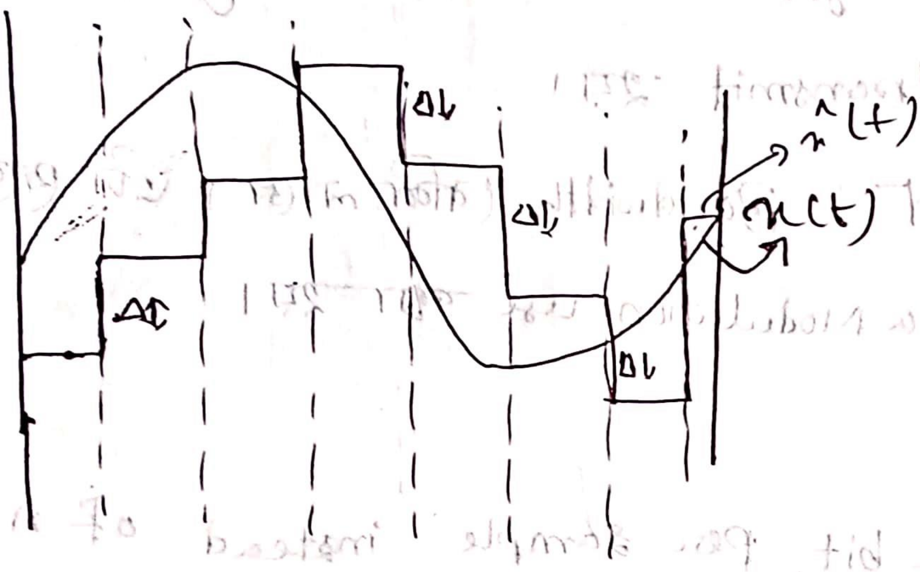
~~$x(t) > \hat{x}(t)$~~

$x(t) > \hat{x}(t) \rightarrow$ Increase $\hat{x}(t)$ by 'Δ' $\rightarrow 1$

$x(t) < \hat{x}(t) \rightarrow$ Decrease $\hat{x}(t)$ by 'Δ' $\rightarrow 0$

D.M output = 1 হয় যদি staircase signal +Δ হয়

D.M output = 0 হবে '' '' '' −Δ হয়।



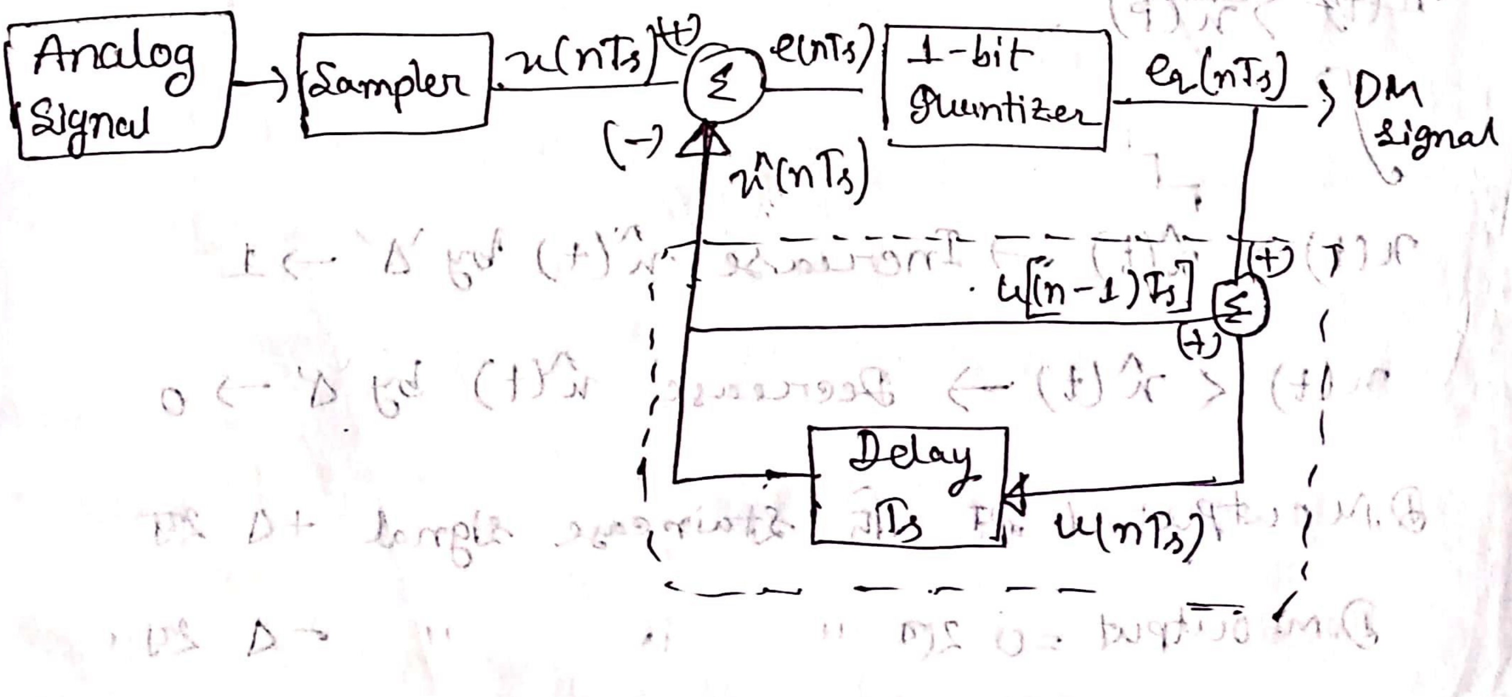
মুহূর্তে $x(t)$ যদি $\hat{x}(t)$ এর উপরে থাকে তবে $\hat{x}(t)$ প্রতিবারে

Δ পরিমান উপরে উঠবে।

আর যদি $\hat{x}(t)$ নিচে থাকে।

⊛ কম Bit দিয়ে same info পাবো এটাই DM এর কাজ।

DM Transmitter



Working:-

(1) $x(nT_s)$ and staircase approximated $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

$x(nT_s)$ ↑ present sample value of $(x(t))$

$\hat{x}(nT_s)$ ↑ Approximated value $\hat{x}(nT_s)$

IF ~~$\hat{x}(nT)$~~ $x(nT_s) > \hat{x}(nT_s) \rightarrow$ error positive $\rightarrow +\Delta$

$x(nT_s) < \hat{x}(nT_s) \rightarrow$ error Negative $\rightarrow -\Delta$

Depending upon sign of $e(nT_s)$, 1 bit quantizer generates $+\Delta$ or $-\Delta$ (1 or 0)