Digital Communication

→ Digital communication is due to noise & doesn't affect much with modulation

Continuous Signal
↓
Sampling
↓
Quantization
↓
Encoder
↓
Digital Signal

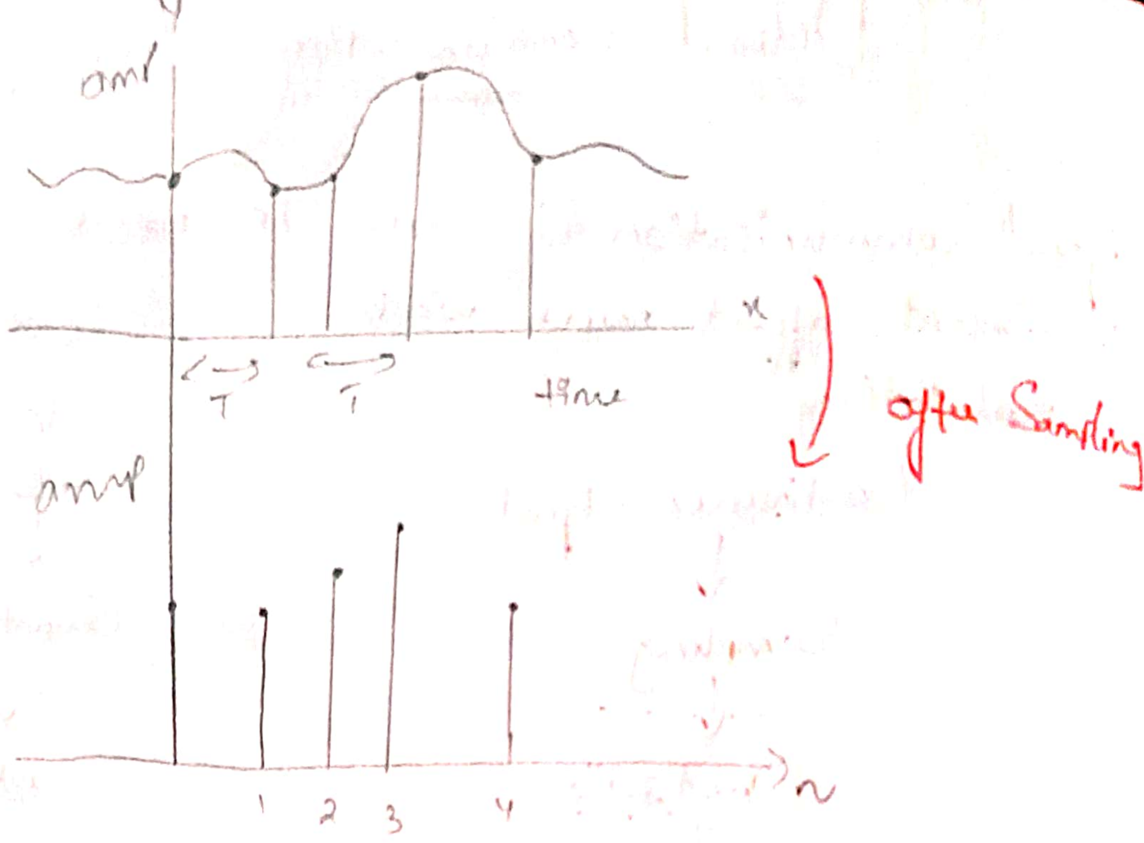
Cont Signal
↓
Sampling
↓
Quantization
↓
Encoder
↓
Digital Signal

Output of Sampling is a discrete Signal

Sampling :-
Store the magnitude at or after a Particular interval of time (Regular interval of time)

→ magnitude of a Signal is Stored at a Regular interval of time

I/P = Cont Signal
O/P = discrete signal



Sampling

=> Difference

→ (In descre Signal Sample can take any Value
& In descrete Signal magnitude can be anything)

→ In digital Signal it has Pre defined levels
(choose any one)

Ex:- If we take 4 bits of data then the Possible Combinations are 16 - So the amplitude of a digital Signal Can be choose from the one of the 16 levels

Sampling Theorem

Sampling Theorem States that in order to Construct the Signal without loosing any information. The message signal must be Sampled at Nyquist Rate.

Nyquist Rate $= f_s$ $\boxed{f_s > 2f_m}$

Nyquist interval $= \frac{1}{f_s} = T_s$ $\frac{1}{f_s} = T_s$

$f_s \rightarrow$ Sampling freq / Nyquist Rate

$f_m \rightarrow$ frequency of msg Signal

Q) Determine Sampling frequency of $40 \sin(200\pi t)$

$A_m \sin(2\pi f t)$

$f = \frac{\omega}{2\pi}$

$40 \sin(200\pi t) =$

$A_m \sin(2\pi f t)$

$40 \sin(200\pi t)$

$A_m \sin(2\pi f t)$

$A_m = 40$, $f_m = 100$

as Per the Condition $f_s > 2f_m$. We have to choose the f_s Value greater than 200.

(i) Determine sampling frequency of

$30 \sin(100\pi t) + 60 \sin(300\pi t)$

$f = 50$ $\qquad f = 150$

so $f_m = 150$

(greater one)

Nyquist rate $= 300$

Nyquist interval $= 1/300$

$f_s > 300$

$\sin^2\theta = \frac{1-\cos 2\theta}{2}$

(ii) Determine sampling frequency of $30 \sin^2(100\pi t)$

$30 \sin^2(100\pi t) = 30(1-\cos 200\pi t)/2$

$= 30 - 30(\cos 200\pi t)/2$

$f_m = 100$

$f_s > 200$

$\underline{\underline{f_s = 200}}$

Determine sampling frequency of

(TU 2082) $30 \sin(1000\pi t) \cdot 60 \sin(3000\pi t)$

$\sin A \sin B = \frac{1}{2}[\cos(A-B) - \cos(A+B)]$

$= \frac{1}{2}[\cos(1000-3000)\pi t - \cos(1000+3000)\pi t]$

$= \frac{1}{2}[\cos(-2000)\pi t - \cos 4000\pi t]$

$= \frac{1}{2}[\cos 2000\pi t - \cos 4000\pi t]$

freq of $\cos 2000\pi t = 1000$

freq of $\cos 4000\pi t = 2000$

$f_m = 2000$

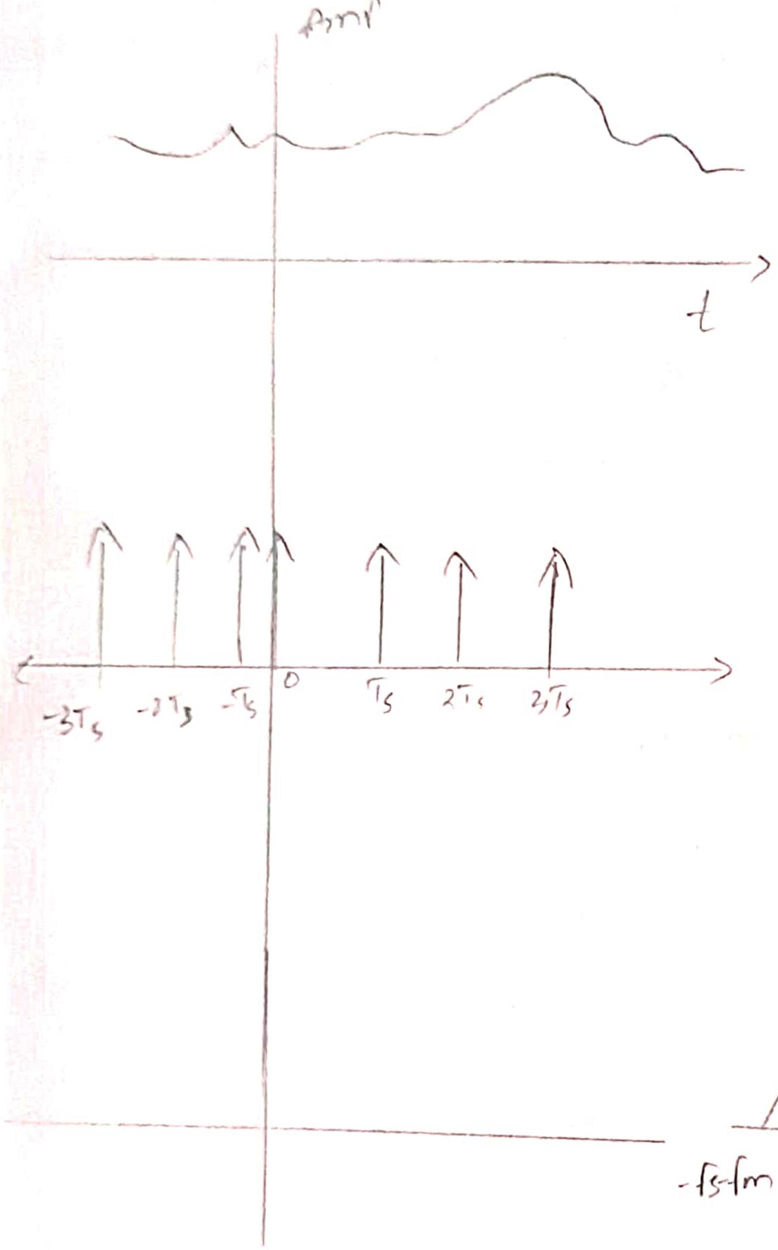
$f_s = 2f_m$

$f_s = 2 \times 2000$

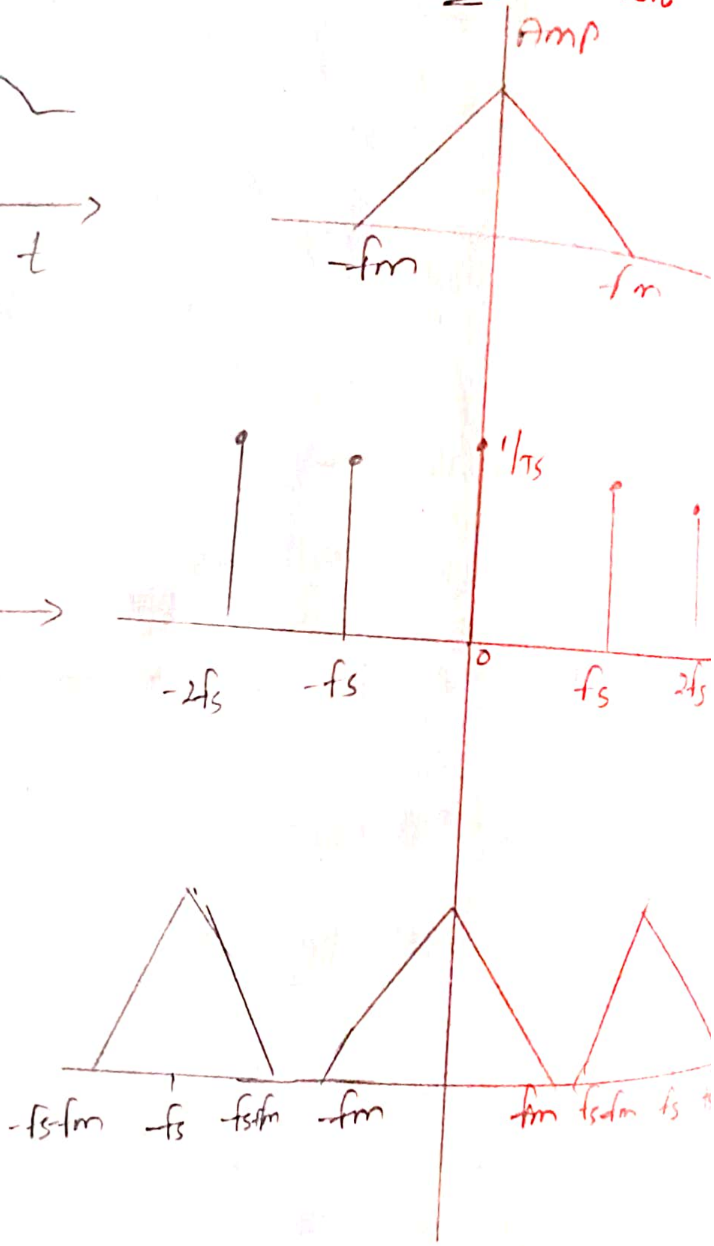
$\underline{\underline{4000}}$

Proof of Sampling Theorem

Time Domain



Frequency domain

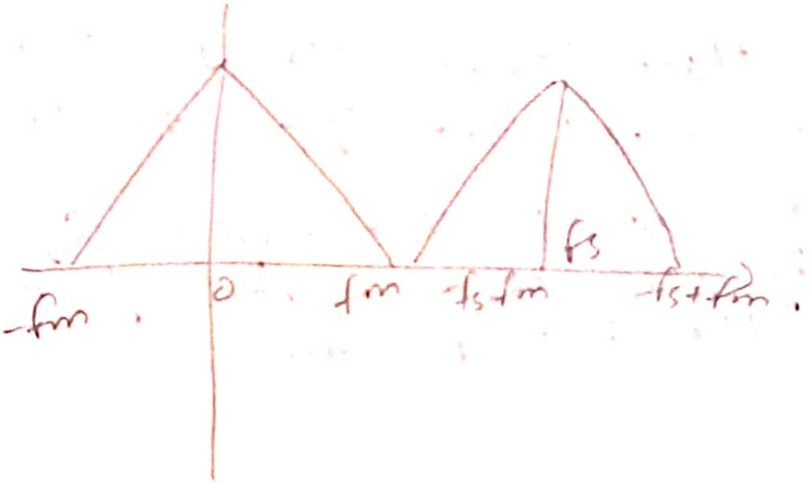


If we multiply the signal in the time domain then the signal should be convolved in the frequency domain

=) At every instant of frequency we have to convolve the signal

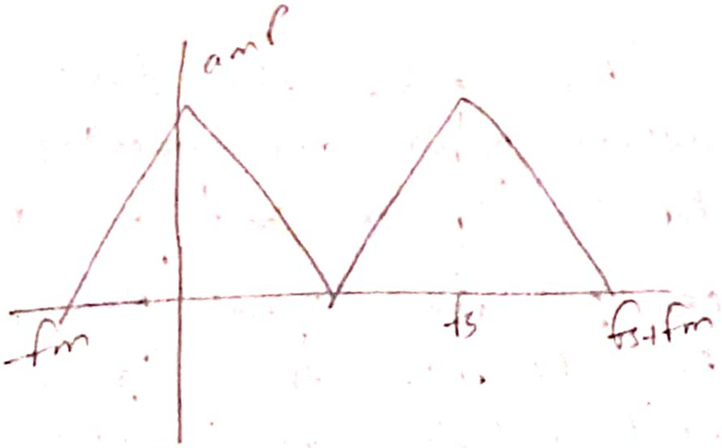
-) According to the condition $f_s > 2f_m$ we have 3 cases:-

① Case 1 :- $f_s > 2f_m$ (oversampling)



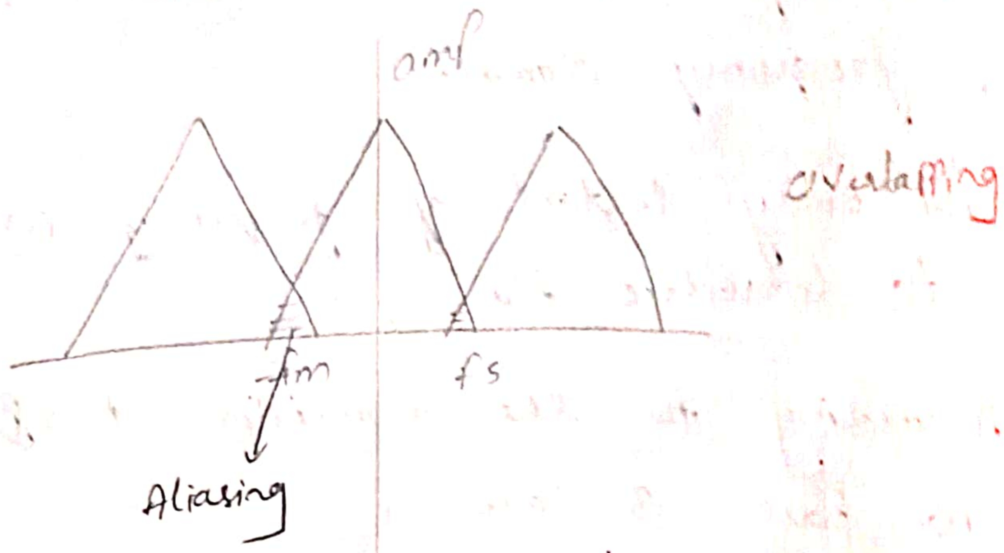
Small gap b/w 2 signals

Case 2 :- $f_s = 2f_m$ (critical sampling)



no space

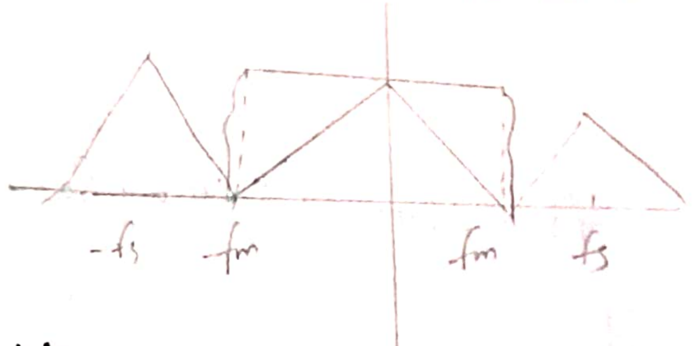
Case 3:- $f_s < 2f_m$ (undersampling)



Aliasing occurs when $f_s < 2f_m$

from the above three cases the 1st case is very easy to extract the signal at the receiver

-> In real time scenario, the original signal cannot be extracted if we use the ideal filter for the signal in the case $f_s = 2f_m$

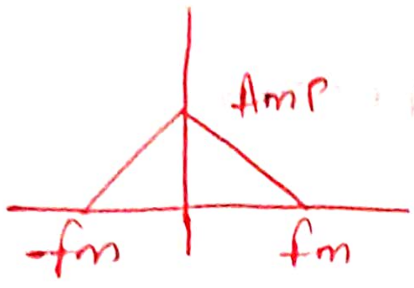


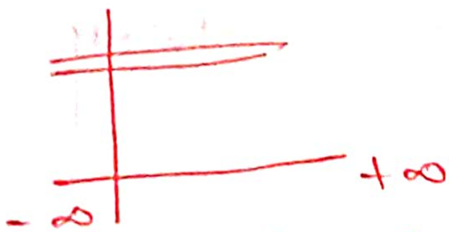
-> while we are considering the dotted line the small part of other signal are getting into the ideal filter space so we can't get original signal

-> If we consider 1st case signal because of the space present in between the signals/waves the ideal filter doesn't entering as combining with other copies - so we can easily extract the original signal.

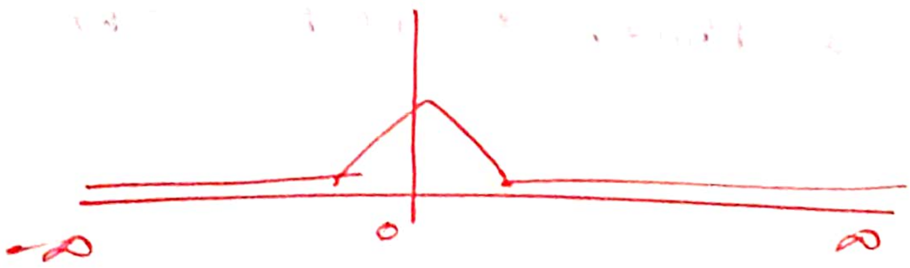
anti - Aliasing filter:-

whenever any signal is generated
let us consider the signal





If we add noise to the above signal then



If the above signal is not able to do the sampling because it ranges from -∞ to ∞. So for sampling we have to apply the low pass filter. when we apply the LPF then the signal will get into some amount of range. then we can do sampling of a signal

Sampling

Over Sampling — $f_s > 2f_m$

Critical Sampling — $f_s = 2f_m$

under Sampling — $f_s < 2f_m$

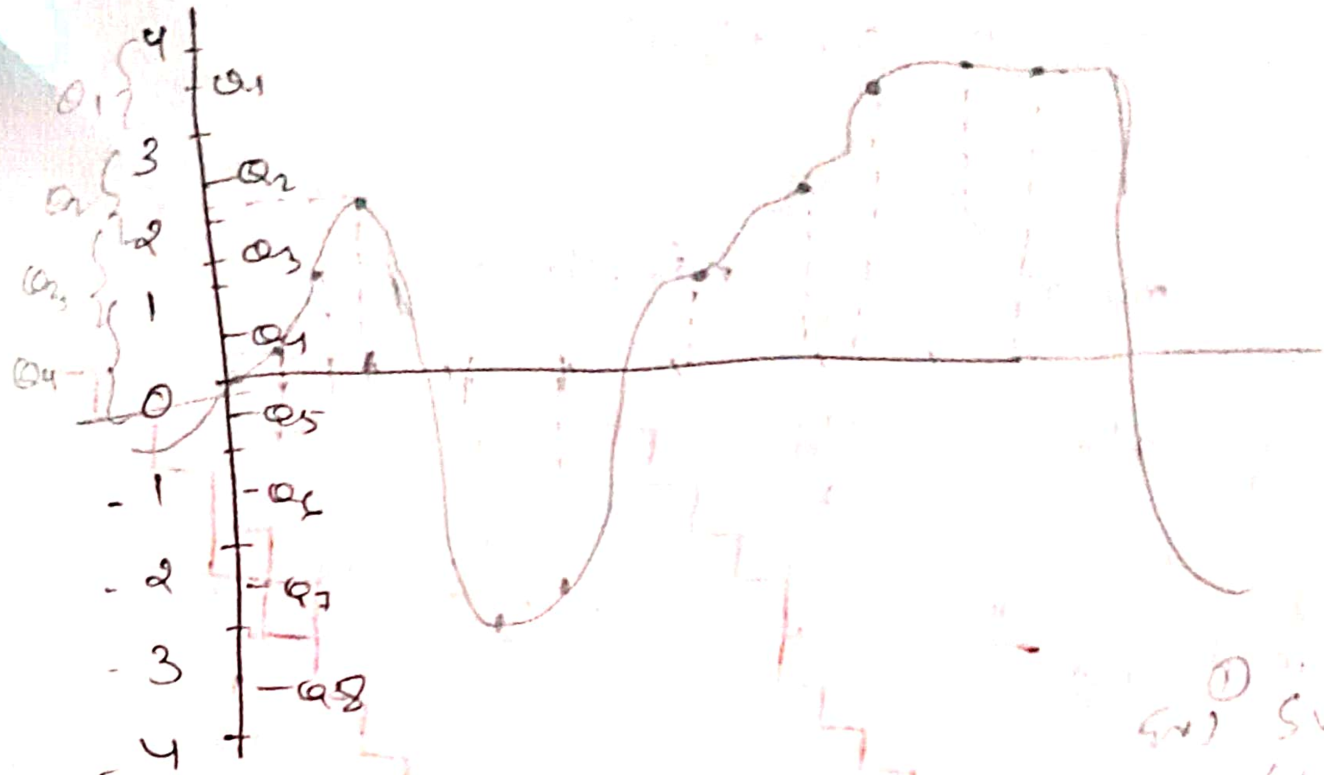
Quantization

Quantization is a Process in which the

S

Quantization are of 2 types
→ uniform
→ Non-uniform

→ Range of the signal divided in equal Parts for uniform Quantization
the centre of each Part called quantised level
→ from the graph
If we consider the 1st Part (ranges from 0-1) Quantized level Q_4 the Samples which are mapped over the range of Q_4 they all have Same unique code.



① SV = 0.2 } Q4
QV = 0.5

QE = SV − QV
= −0.3

Sample Value = 0.2

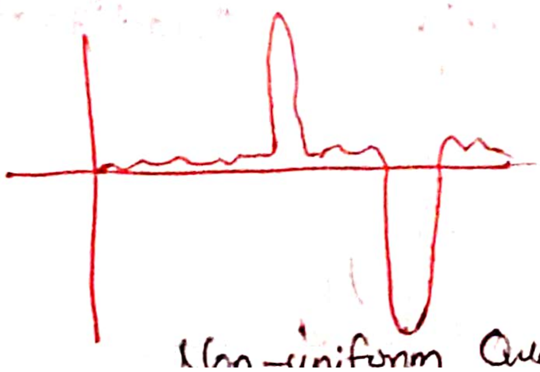
Quantized Value = 0.5

In digital signal representation for any sample point it must be mapped to its corresponding Quantized level

$$QE = SV - QV$$
$$= 0.2 - 0.5$$
$$= -0.3$$

-> This value must be low value (as small as possible)

Use uniform Quantization

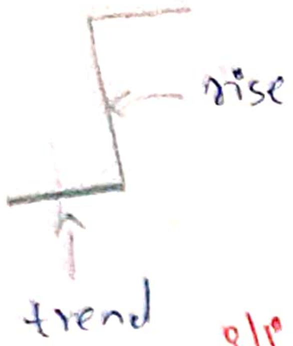


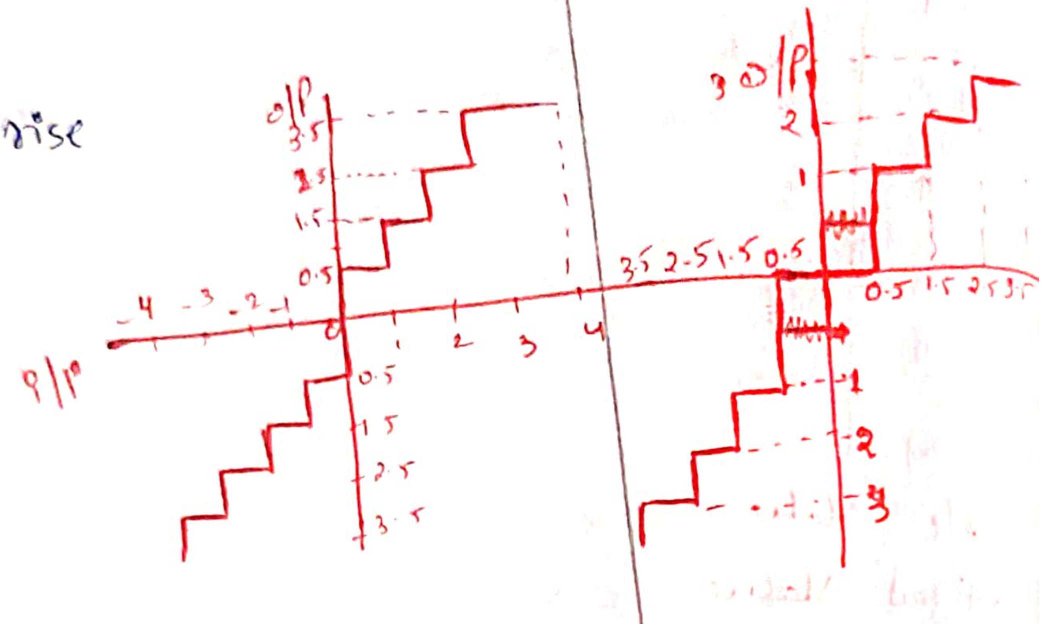
Non-uniform Quantization

uniform Quantization

mid-rise

mid-trend





mid-rise

-> the origin is at middle of ris

mid-trend

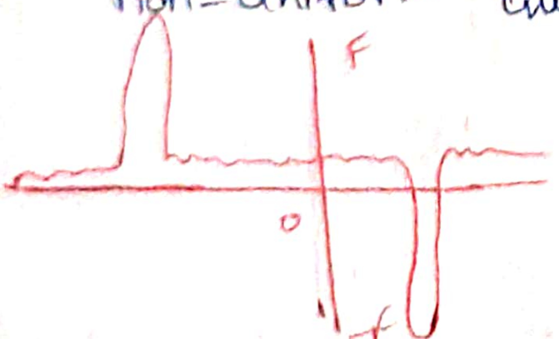
-> the origin is at the middle of trend

-> Non-uniform Quantization

-> the range of the signal will not be divided equally as per the bits.

-> Depending on the signal we can go with non-uniform Quantization.

-> If the majority of the signal is distributed over a particular range. then we can go with the non-uniform Quantization



let no. of bits per Sample = n

No. of Quantization levels = 2^n

Quantization error = SV − QV

length of Segment = Δ = $\frac{\text{Range of the Signal}}{2^n} = \frac{V_{max} - V_{min}}{2^n}$

maximum voltage (V_{max}), minimum voltage (V_{min})

or

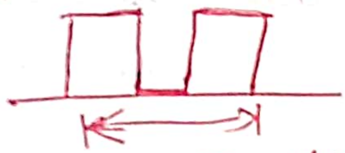
Step size

or

length of Quatization level...

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

Bit duration "T_b"



$T_S = 1/f_s$

(T_b) Bit duration = $\frac{T_s}{n}$

$T_b = \frac{T_s}{n}$

f_s - Samples per Second

n - bits per Sample

$T_s = \frac{1}{f_s}$ $\quad$ T_s = Sampling time

→ Bit rate (R_b) = $\frac{\text{bits}}{\text{Sample}} \times \frac{\text{Sample}}{\text{sec}} = nf_s$

→ Bandwidth :- Space required to transmitt the Signal

→ max Bandwidth = R_b

min Bandwidth = $R_b/2$

→ Bit rate = no. of bits per Second = R_b

$= n \times f_s$

$R_b = nf_s$

Q) a message signal of $10\sin 2\pi 10^3 t$ is transmitted through 5bit ~~Quantization~~ ~~ler~~ system find all the parameters of this system.

1) $\rightarrow n = 5$

$\rightarrow$ no. of Quantization levels $= 2^n = 32$

$\rightarrow$ length of segment $= \Delta = \frac{10+10}{2^n} = \frac{20}{2^n} = \frac{20}{32} = 0.625$

$\rightarrow$ Bit duration

$T_b = \frac{T_s}{n}$ $\quad \because T_s = \frac{1}{f_s}$

$f_m = 1000$, $f_s > 2f_m$

$f_s > 2000$

we can take critical sample frequency $f_s = 2000$

$T_s = \frac{1}{f_s} = \frac{1}{2000}$

$T_b = \frac{1}{2000} \times \frac{1}{5} = \frac{1}{10000}$

$\boxed{T_b = 10 \times 10^{-3}}$

Q2 a message signal of $4\sin 2\pi 10^2 t$ is a transmitted through 3bit system if the sample value is $2.3, 1.6, 3.9, 3.1, 2.8, -2.3, -3.6, -1.1$

determine quantization error, Quantization value and ~~incoded output~~ and determine all these parameters if sampling frequency is 150% above the niquisht rate

→ Bit rate $= nf_s = 5 \times 2000$

$R_b = 10,000$

$= 10 \times 10^3$

$\boxed{R_b = 10 \times 10^3}$ = maximum bandwidth

minimum bandwidth $= 5 \times 10^3 = \frac{R_b}{2}$

Q.2
Sol

$f_m = 10^2 = 100$

$f_s > 2f_m$

$f_s > 200$

$Q.E = S.V - Q.V$

$N.R = f_s = \frac{150}{100} \times 200 = 300$

$f_s = 300 + 200$

$\boxed{f_s = 500}$

S.V	Q.V	QE	encoded o/p
2.3	2.5	-0.2	110
1.6	2.5	0.1	101
3.9	3.5	0.4	010
3.1	3.5	-0.4	111
2.8	2.5	0.3	110
-2.3	-2.5	0.2	001
-3.6	-3.5	-0.1	011
-1.1	-1.5	0.4	010

Q) A Sinusodal message signal is transmitted through a system such that maximum Quantization error should be 2% of peak to peak amplitude of the message signal. find minimum no. of bits required to represent each sample

Sol
max Q.E $= \pm \frac{\Delta}{2} < 2\%$ of Peak to Peak Voltage

$$= \pm \frac{\Delta}{2} < 2\% \; 2A_m$$

$$= \frac{\Delta}{2} < \frac{2}{100} 2A_m$$

$$= \cancel{\frac{1}{2}} \frac{2A_m}{2^n} < \frac{2}{100} 2A_m$$

$$\frac{1}{2^n} < \frac{1}{25}$$

$$25 < 2^n$$

apply log on both sides

$$\log(25) < \log(2^n)$$

$$= 1.39 < n \log(2)$$

$$= 1.39 < n\,(0.30)$$

$$4.63 < n$$

$$\boxed{n > 4.63}$$

Modulation

- Analog
  - Amp
    - AmP
    - DSBSC
    - SSBSC
  - Angle
    - FM
    - PM
- Digital
  - Base Band
    - Pulse analog mod
      - PAM
      - PPM
      - PWM
    - Pulse digital mod
      - PCM
      - DPM
      - DM
      - ADM
  - BandPass mod
    - 1-bit Signalling
      - ASK
      - PSK
      - FSK
    - M-ary Signaling
      - M-ASK
      - M-PSK
      - M-FSK

to reduce Quantization error $\Bigg\}$ decrease range / increase n

$$\downarrow [Q_e]_{max} = \pm \frac{\Delta}{2} \downarrow$$

$$\downarrow \Delta = \frac{\text{Range} \downarrow}{2^n \uparrow}$$

Signal to quantization noise power ratio (SQNR

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

$\because$ noise is unwanted entity

$$SQNR = \frac{P_s}{P_{Q_e}}$$

Power associated with message signal $= P_s$

Power associated with Quantization noise or error $= P_{Q_e}$

$\rightarrow$ let $m(t) = A_m \cos 2\pi f_m t$

$$P_s = \frac{\left(\frac{A_m}{\sqrt{2}}\right)^2}{R} = \frac{A_m^2}{2R} \text{ or } \frac{A_m^2}{2} \text{ when } R = 1\Omega$$

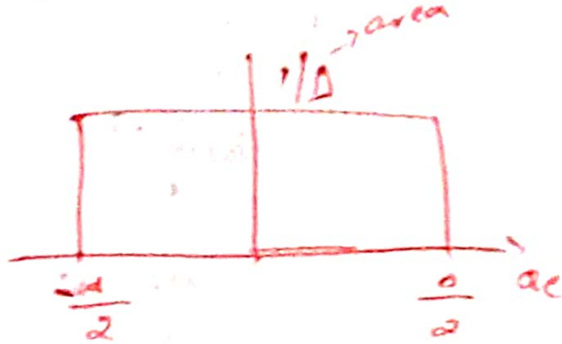
→ let $m(t) = A_m \cos 2\pi f_m t$

$P_{Q_e} = \text{Power}(Q_e)$ ← Random ~~var~~ variable

$= \text{MSQ}(Q_e)$

$= E(Q_e^2)$

$= \int Q_e^2 \, F_{Q_e}(Q_e) \, dQ_e$



$$\int_{-\Delta/2}^{\Delta/2} Q_e^2 \frac{1}{\Delta} \, dQ_e$$

$\cancel{\Delta = \frac{2A_M}{2^n}}$

$$\frac{1}{\Delta} \frac{Q_e^3}{3} \Bigg|_{-\Delta/2}^{\Delta/2}$$

$$= \frac{1}{3\Delta} \times \frac{\Delta^3}{8} + \frac{\Delta^3}{8} = \frac{1}{24\cancel{\Delta}} 2\Delta^{\cancel{3}2} = \frac{\Delta^2}{12} = \frac{1}{\cancel{12}_3} \frac{\cancel{4}A_m^2}{2^{2n}} = \frac{1}{3} \frac{A_m^2}{2^{2n}}$$

Signal to Quantization noise Power Ratio (SQNR)

$$SQNR = \frac{\text{Power of Signal}}{\text{Power associated with Quantization error}} = \frac{P_s}{P_{Qe}}$$

let $m(t) = A_m \cos 2\pi f_m t$.

$$P_s = \frac{V^2}{R} \qquad V = V_{rms}$$

$$= \frac{(V_{rms})^2}{R} \qquad V_{rms} = \frac{V_{max}}{\sqrt{2}}$$

$$= \frac{\left(\frac{A_m}{\sqrt{2}}\right)^2}{R} = \frac{A_m^2}{2R}$$

$$\boxed{P_s = \frac{A_m^2}{2R}}$$

* P_{Qe} is an undetermined entity
* We can consider Quantization error as a random Variable

$$P_{Qe} = \text{Power}(Q_e)$$

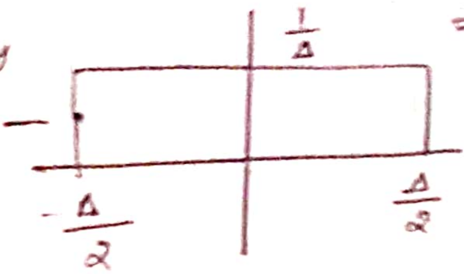
Q_e = random Variable

$$P_{Qe} = \underbrace{msQ}_{\text{mean Square}}(Q_e)$$

$$= E[Q_e^2]$$

$$P_{Qe} = \int_{-\infty}^{\infty} Q_e^2 \, f_{Q_e}(q_e) \, dq_e$$

Uniform Probability Distribution fn



$\Rightarrow$ Prob of PDF = 1

$1 = l \times \frac{\Delta}{2} \times \frac{\Delta}{2}$

$= l \times \frac{2\Delta}{2}$

$l = ...$

$= l = 1/\Delta$

$F_{Qe}(Qe)$ = Probability Distribution function

$$F_{Qe}(Qe) = \frac{1}{\Delta}$$

$$= \int_{-\Delta/2}^{\Delta/2} Q_e^2 \left(\frac{1}{\Delta}\right) dQe$$

$$= \frac{1}{\Delta} \int_{-\Delta/2}^{\Delta/2} Q_e^2 \, dQe$$

$$= \frac{1}{\Delta} \left[\frac{Q_e^3}{3} \right]_{-\Delta/2}^{\Delta/2} = \frac{1}{3\Delta} \left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8} \right]$$

$$= \frac{1}{3\Delta} \left(\frac{2\Delta^3}{8} \right)$$

$$P_{Qe} = \frac{\Delta^2}{12}$$

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

$$\Delta = \frac{\text{range of Signal}}{2^n}$$

$$\Delta = \frac{2A_m}{2^n}$$

$$P_{Qe} = \frac{\left(\frac{2A_m}{2^n}\right)^2}{12} = \frac{\frac{4A_m^2}{2^{2n}}}{12} = \frac{4A_m^2}{\frac{12 \cdot 2^{2n}}{3}}$$

$$P_{Qe} = \frac{A_m^2}{3 \cdot 2^{2n}}$$

$$SQNR = \frac{P_s}{P_{qe}} = \frac{\frac{A_m^2}{2R}}{\frac{A_m^2}{3 \cdot 2^{2n}}} = \frac{A_m^2}{2R} \times \frac{3 \cdot 2^{2n}}{A_m^2}$$

$$= \frac{3 \cdot 2^{2n}}{2R}$$

$$= \frac{3}{2} \cdot \frac{2^{2n}}{R} \quad \text{if } R = 1$$

$$= \frac{3}{2} \cdot 2^{2n}$$

* SQNR is high as much as possible

$$SQNR = \frac{3}{2} 2^{2n}$$

SQNR in decibals $= SQNR_{dB} = 10\log\left(\frac{3}{2} 2^{2n}\right)$

$$= 10\log_{10}\frac{3}{2} + 10\log_{10} 2^{2n}$$

$$= 10\log_{10}\frac{3}{2} + 20n\log_{10} 2$$

$$= 1.76 + 6.0n$$

$$\boxed{SQNR = 1.76 + 6.0n}$$

no. of bits	SQNR (1.76 to 6.0n)
1	7.76
2	13.96
3	19.82
4	25.84

difference b/w SQNR of each bit is 6dB

N bit —— (SQNR) x

N+3 —— x + 6×3 = x + 18

Compander :-

=) Compander is used in non-uniform quantization or distribution

-) In real-time scenario we don't know whether the signal is uniformly distributed or non-uniformly distributed. If the input signal is non-uniformly distributed then by using compander which is used for performing uniform quantization

-) In non-uniform distributed signal. the compander will increase the lower magnitude signal to certain level which will get to uniform distributed signal

i/p → Compressor → Uniform Quantization → Expander → o/p

Compressor :-

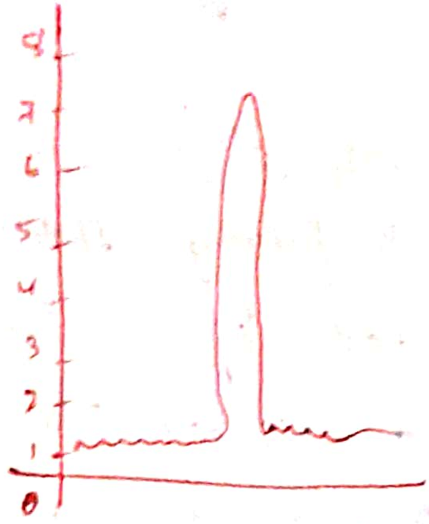
which is used for mapping the samples to the levels which we have

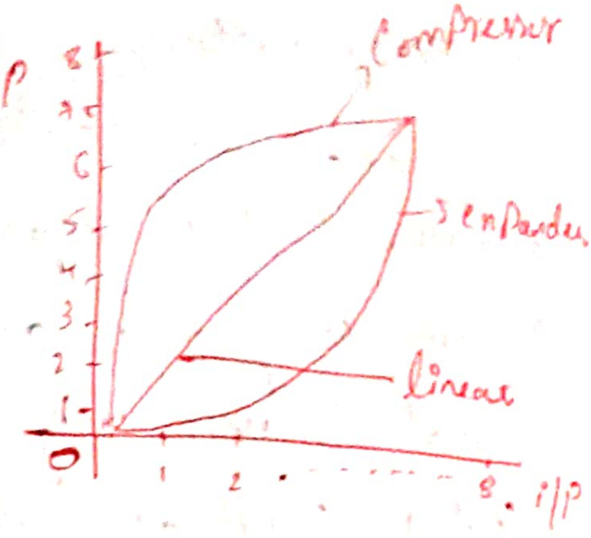
levels = if bits = 4

levels = $2^4 = 16$

Expander:-

Reverse the Compressor operation





Compander — μ-law
— A law

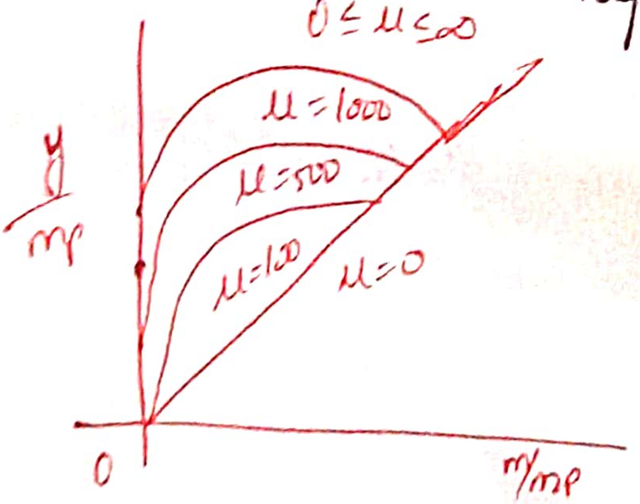
μ- law:-

$$\frac{y}{m_p} = \frac{1}{\ln(1+\mu)} \ln\left(1 + \frac{\mu |m|}{m_p}\right) \qquad 0 < \frac{|m|}{m_p} < 1$$

y = Output

μ = unitless value (the point where the curve signal is changing)

m = instantaneous magnitude of msg signal / input signal



Same for A-law

A - law:-

$$\frac{y}{m_p} = \frac{A}{1+\ln A}\left(\frac{|m|}{m_p}\right) \qquad 0 < \frac{m}{m_p} < \frac{1}{A}$$

$$= \frac{1}{1+\ln A}\left(1+\ln\frac{A|m|}{m_p}\right) \qquad \frac{1}{A} < \frac{m}{m_p} < 1$$

modulator:- / modulation:-

two types

-> Analog modulation

-> Digital modulation

=> Digital modulation

-> Base Band

-> Band Pass

Base Band:-

=> where we are transmitting the signal through wires

=> Where we didn't change the frequency of signal

Band Pass:-

=> Where we can change the frequency of signal

=> Basic Band modulation are of 2 types

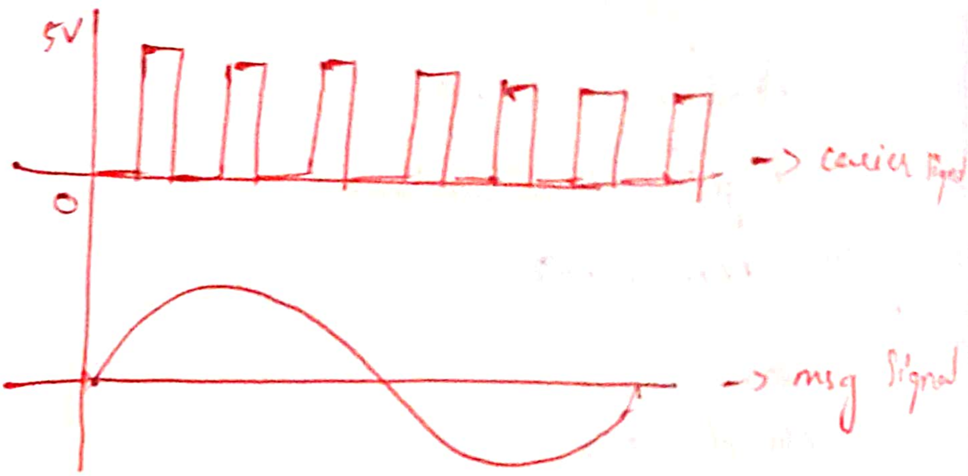
-> Pulse analog modulation

-> Pulse Digital modulation

(1) Pulse analog modulation

- Pulse Amplitude modulation
- Pulse width modulation
- Pulse Position modulation

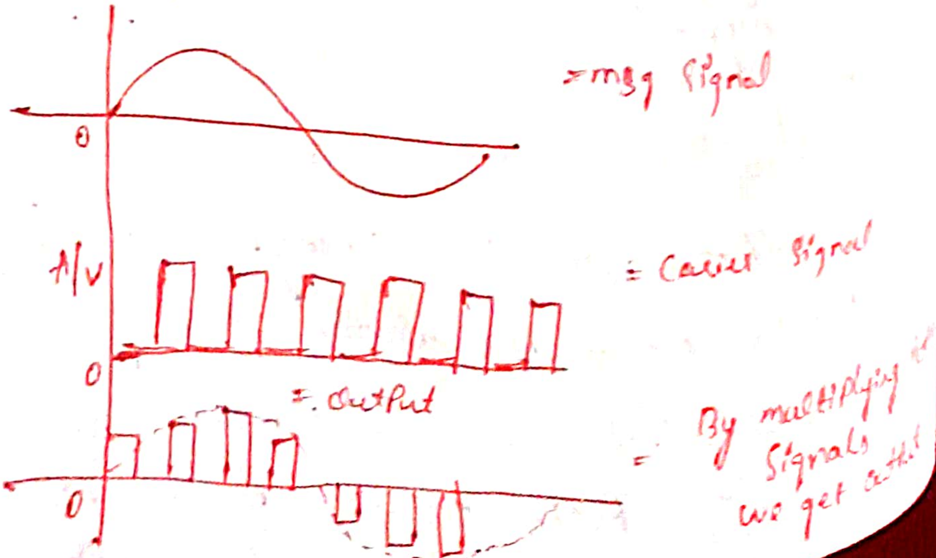
* Square wave Signal is used as Carrier Signal in Pulse analog modulation.



⇒ Pulse amplitude modulation:-

The amplitude of a carrier signal which vary according to your message / modulating signal is called PAM.

Pulse Amplitude modulation

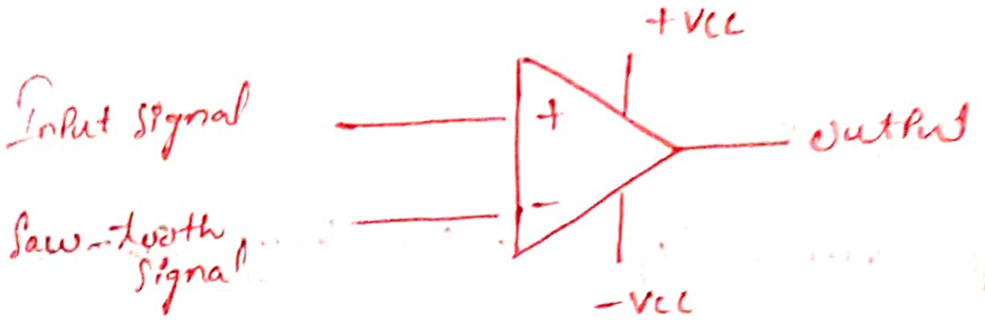


Pulse width Modulation:-

-> The width of the carrier signal will vary according to your message signal

-> magnitude of carrier signal rises if magnitude of message signal rises. and vice versa

-> Consider an op amp which will act as comparator. and the inputs will be input signal and sawtooth signal.

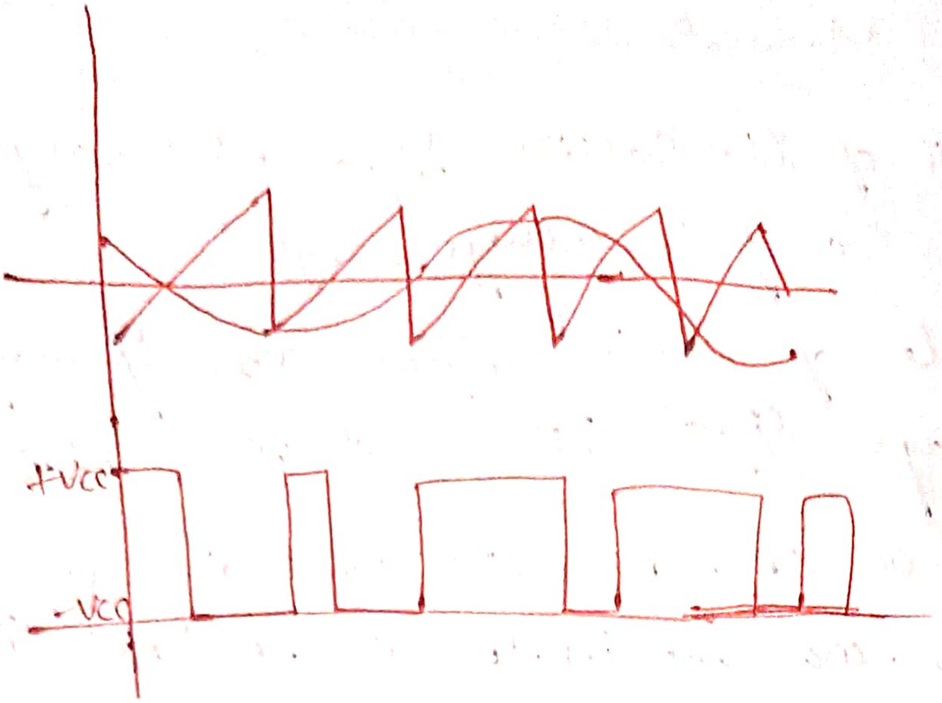


-> The comparator will compare the two signals and gives the output. Based on some conditions the comparator generates the output.

-> If the magnitude of input signal > magnitude of saw tooth signal.

the signal will get to +vcc

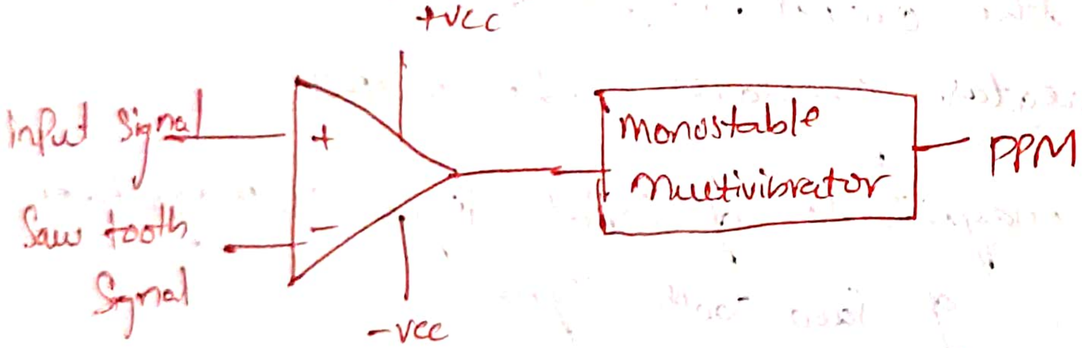
If the magnitude of input signal < magnitude of saw tooth signal. the output of a signal get to the −vcc.



Pulse width modulation

Pulse Position Modulation

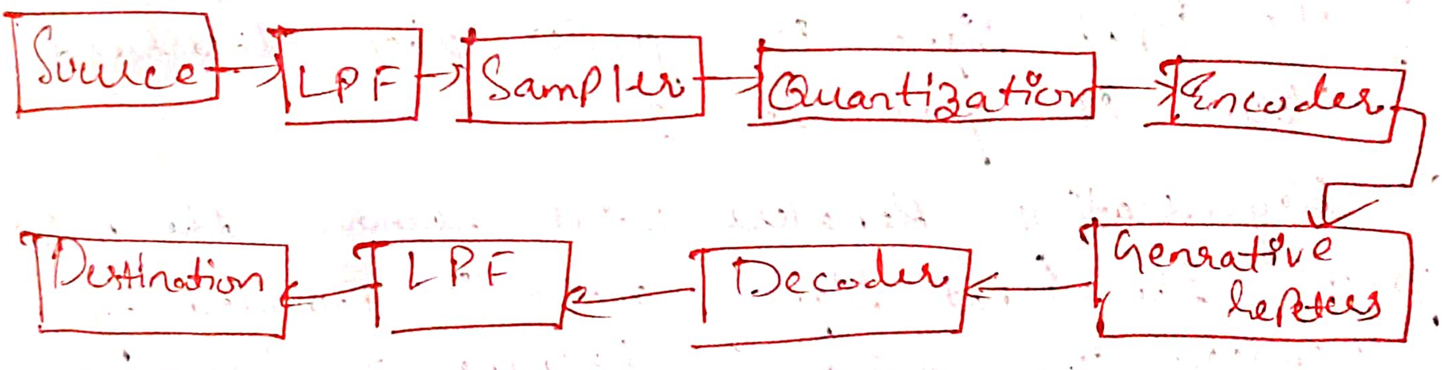
The Position of Carrier Signal will Vary accordingly to your message Signal



When the output of PWM is Passed through monostable multivibrator the Point where the Output Signal trigger it will remain for Same duration and after the duration it get to '0'. The duration is constant

Pulse Digital Modulation:-

(1) Pulse code Modulation (PCM):-



Source:-

an entity which gives ~~data~~ input

→ LPF is used to remove the noise in the signal

→ Sampler will samplizer the signal

Nyquist Rate $> 2f_m$

* The input signal will map the signal to its respective quantization level.

* Encoder will give the code for each quantization level

Regenerating Repeaters:-

=> This block have threshold value

=> Threshold value be the centre of the signal

$\hookrightarrow \frac{V_{max} - V_{min}}{2}$

=> The block starts comparing the input signal with threshold value.

=> If the input signal is less than. It goes to 0.
If the input signal is greater than threshold value it goes to 1.

=> Regenerating Repeters will remove the noise from the input signal which is provided by encoder. The noise is because of some distortions happen in the process.

Decoder:-

=> Decoder will map the input signal to their quantization level

LPF:-

-> and passes through LPF then we get the destination.

Line Coding:-

=> way of Representation of 0's and 1's

Three types of Line coading.

-> unipolar

-> polar

-> Bipolar

(i) unipolar:- — RZ
— NRZ

-> It is also known as ON-OFF

-> It is ranges from 0-V (V= any value)

(ii) Polar:- — RZ
— NRZ

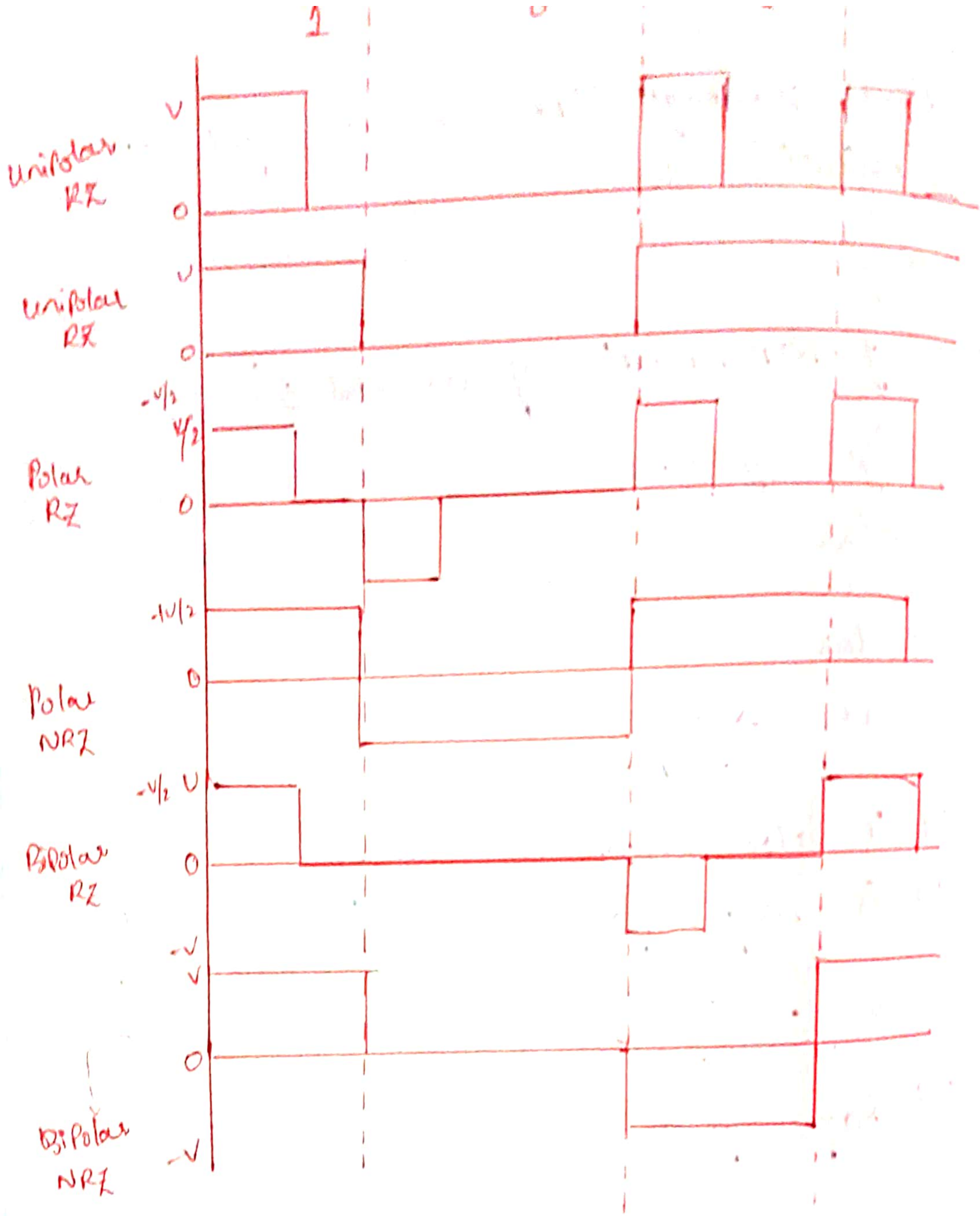
-> It ranges from $\frac{V}{2}$ - $\frac{-V}{2}$

(iii) Bipolar:- — RZ
— NRZ

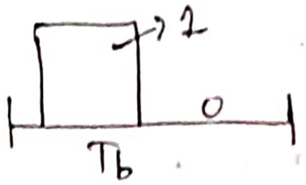
-> It ranges from -V to V.

RZ = Returning to zero

NRZ = Non-returning to zero



Since RZ => T_b will be divided into two Parts

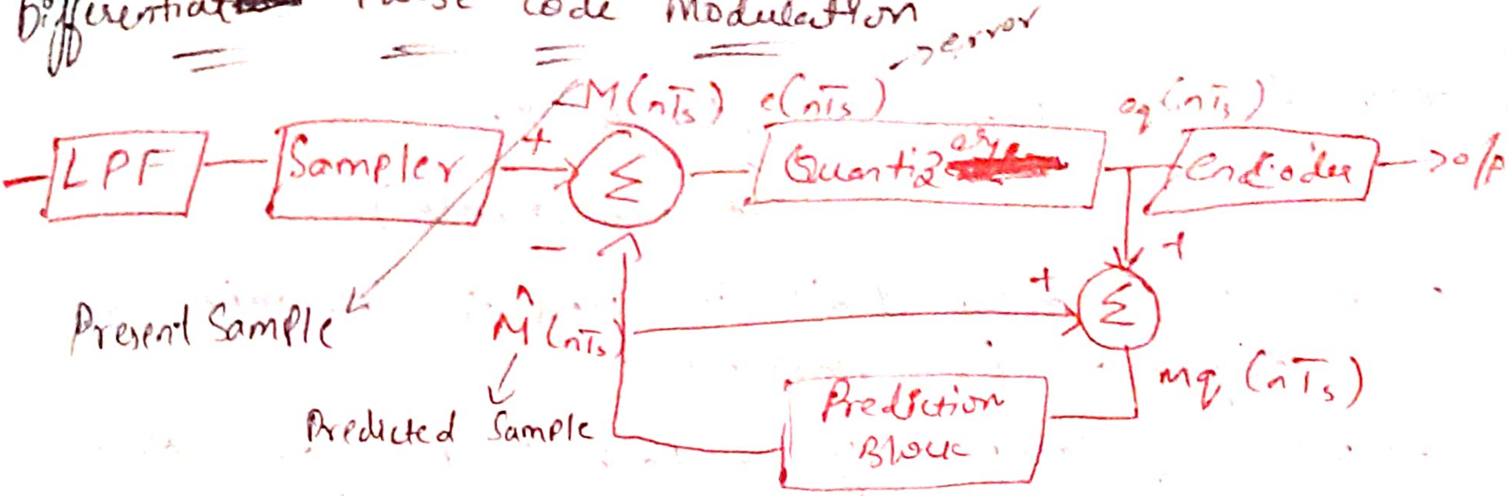


T_B = time duration Pu

NRZ => Tb will not divided into two Parts

DPCM:-

Differential Pulse code Modulation



In order to overcome the disadvantage of PCM. we use DPCM

$$Q.E_{max} = \frac{1}{2} \cdot \frac{Range}{2^n}$$

to decrease Q.E we have two ways

-> Reduce the Range
-> Increase no. of bits "n".

∴ QE is Small

$$M_q(nT_s) = \hat{M}(nT_s) + e_q(nT_s)$$
$$= \hat{M}(nT_s) + e(nT_s) \pm q(nT_s)$$
$$= M(nT_s) \pm q(nT_s)$$

-> here we are giving the Present Sample to Predict the future Sample
-> By this we are Reducing the Range of the Signal
-> By Reducing the Range we can Reduce the Quantization error.

->

A.W.

Delta Modulation

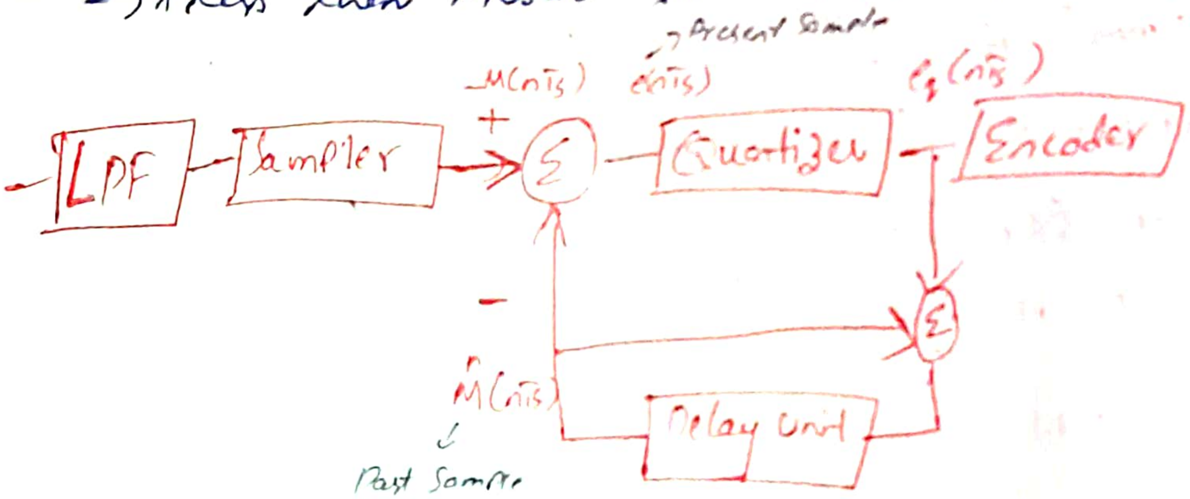
A.W.
we To further Reduce the Bandwidth
(Bits Per Sample)

→ we use only 1 bit in DM to transmit the Signal. (uses least bits)

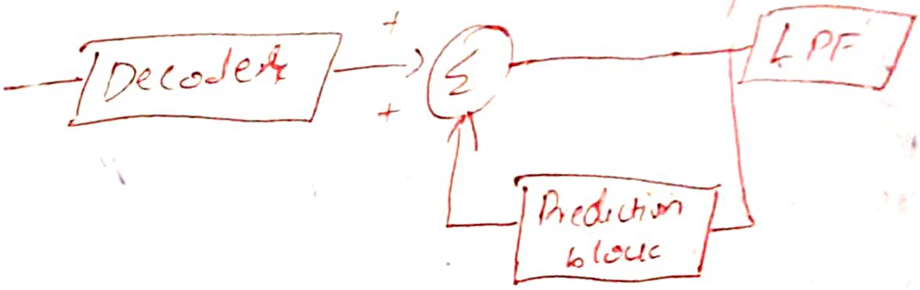
→ It will compare your Past Sample and Present Sample.

Present to Prev compare → If Past Sample greater than Present Sample it will transmit 1

→ If Past Sample less than Present Sample it will transmit 0



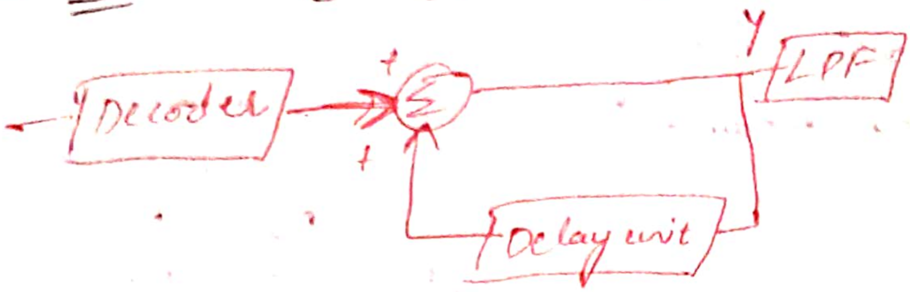
Demodulation & DPCM



$$y(nT_s) = e_q(nT_s) + \hat{m}(nT_s)$$
$$= e(nT_s) \pm q(nT_s) + \hat{m}(nT_s)$$
$$= m(nT_s) \pm q(nT_s)$$

2528
2590

Demodulation of Delta modulation



Under Delta modulation there are 2 types of Errors.

→ Granular error

$$\frac{dm(t)}{dt} < \frac{\Delta}{T_{S0}}$$

reconstructed signal

$$\boxed{\Delta_{opt} < \Delta}$$

msg signal

⇒ 0V

→ Overload error

$$\frac{d}{dt} m(t) > \frac{\Delta}{T_S}$$

$$\frac{\Delta_{opt}}{\cancel{T_S}} > \frac{\Delta}{\cancel{T_S}}$$

$$\boxed{\Delta_{opt} > \Delta}$$

→ When the slope of msg signal is less than reconstructed signal then it is called Granular error.

→ When 1 bit O/P is received at decoder 0 voltage is added when 1 is received and subtracted when 0 is received.

→ The value of $\pm\Delta$ must be choosen carefully and can be calculated by using to reconstruct original signal.

→ The slope of both the signals must be same

$$d/dt\ m(t) = \frac{\Delta_{opt}}{T_S}$$

→ When $\frac{d}{dt} m(t) > \Delta/T_S$, Overloaded occurs slope of reconstructed signal

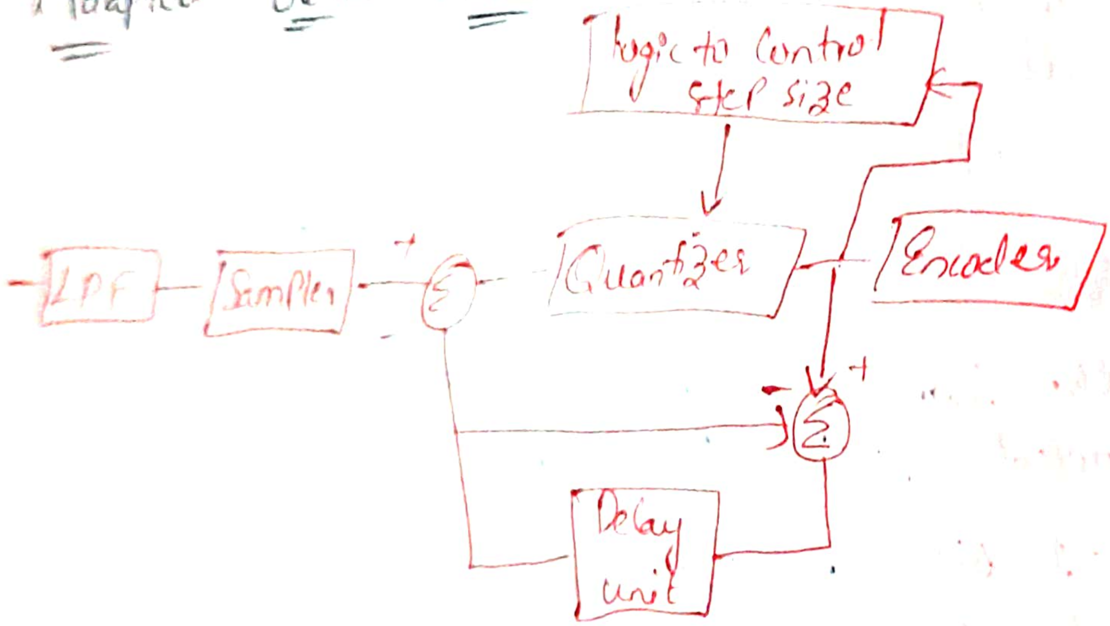
$$\frac{\Delta_{opt}}{\cancel{T_S}} > \frac{D}{\cancel{T_S}}$$

$$\Delta_{opt} > \Delta$$

To overcome the disadvantage of the Delta modulation we use Adapted Delta modulation

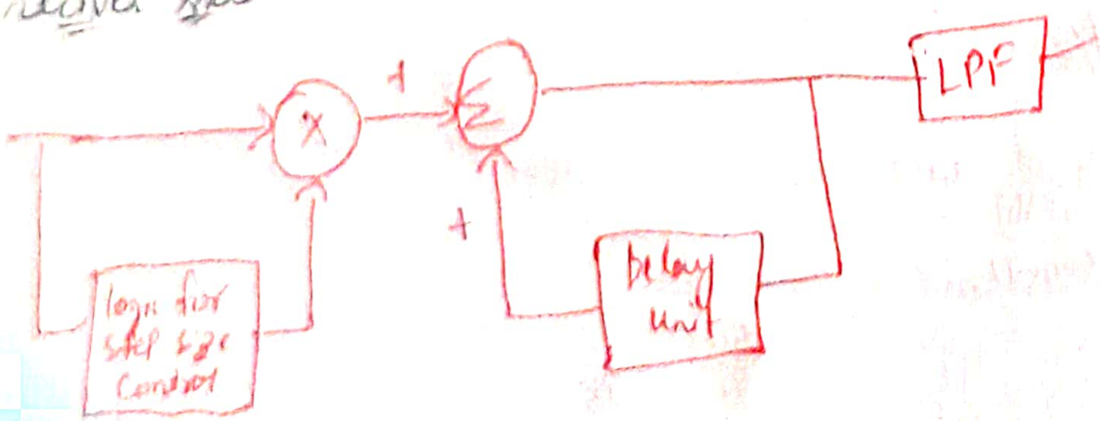
Δ = 1
Must be added when 1
Subtracted when 0

Adapted Delta Modulation



- Based on the slope of the msg signal the step size varies time to time
- Based on error we can control the step size
- Disadvantage :- logic for step size control is too complex

Receiver side



Q) A Continuous signal of $8\sin 8\pi \times 10^{3t}$ is Passed through Delta modulation whose Pulse rate is 4000 Pulses per second. find optimal step size of the receiver.

$$\frac{\Delta opt}{Ts} = \frac{d}{dt} 8\sin 8\pi \times 10^3 t$$

the given signal is $8\sin(8\pi \times 10^3)t$

$fs = 4000$ Pulse/sec

$$\frac{\Delta opt}{Ts} = \frac{d}{dt}(m(t))$$

$$\frac{\Delta opt}{Ts} = \frac{d}{dt} 8\sin(8\pi \times 10^3)t$$

$$Ts = \frac{1}{fs} = \frac{1}{4000}$$

$$\Delta opt \cdot 4000 = (8\pi \times 10^3)\ 8\cos(8\pi \times 10^3)t$$

$$= 8 \times 8\pi \times 10^3$$

$$\Delta opt = \frac{\cancel{8}^2 \times 8\pi \times 10^{\cancel{3}}}{\cancel{4000}}$$

$$= 2 \times 8\pi$$

$$= 2 \times 8(3.14)$$

$$= \underline{\underline{50.24}}$$