1 Explain the purpose of communication system.

- To transfer the information from source to desti-nation
- there systems are widely used in varity of applications
  Ex: telecommunication etc.
- To co-ordinate the activities
- To build the relationship b/n source and destination
- it can be used to control machines and devices remotely
- This allows us to stay connected with others
- To provide early warning of disasters
- To promote education.
- To support scientific ~~research~~ research.

2. Describe the need for modulation in communication system.

- To increase the range of transmission.
  low frequency signal, such as audio signals cannot travel very far so modulation allows to shift signal to higher frequency.
- To reduce the height of Antennae
- Avoiding mixing of signals by using carrier signal
- Multiplexing.
  passing of mor than one-signal in one-channel
  it allows to achieve adjustment in Bandwidth
- Noise will be rdeced.
- To enable wireless communication
  Wireless communication is only because of modulation
  Modulation allows us to transmit signals through air.

5. Identify a suitable analog modulation techique to transmit an audio signal for long distance requiring minimum bandwidth, and explain with appropriate expressions and block diagram.

- The most suitable technique to transmit an audio signal for long distance is SSB AM.
- Single SB AM is a type of AM in which only one ~~type~~ side band of modulator signal is transmitted.
- This reduces the BW of transmitted signal ~~usage~~ by half.
- SSB is more efficient interms of power usage.

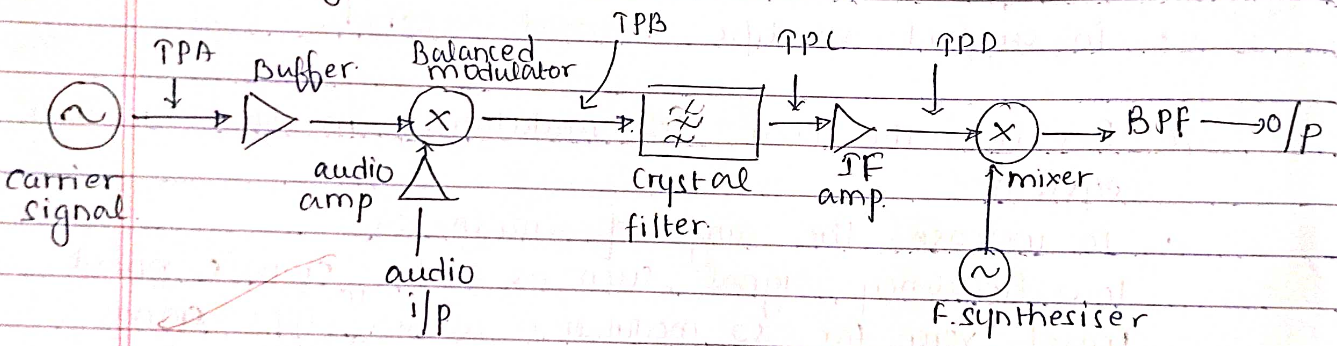
Expressions

BW of SSB AM is given by

$BW = 2f_m$

F_m - max frequency

Block diagram.



Advantages of SSB AM.

- high B.W Efficiency / Reducing the bandwidth.
- high power efficiency.
- long-distance transmission.

disadvantage

complex to implement

Requires coherent reciver.

6.

Given

$A_c = 100\,V$

$R_c = 50\,\Omega$

$\mu = 0.8$

i) it is a narrow Band signal

$$P_c = \frac{A_c^2}{2R_c}$$

$$P_c = 100\,W$$

ii

$$P_{sb} = P_{cs} + P_{us}$$

$$P_{cs} = P_{us} = \frac{A_c^2 \mu^2}{8R_c}$$

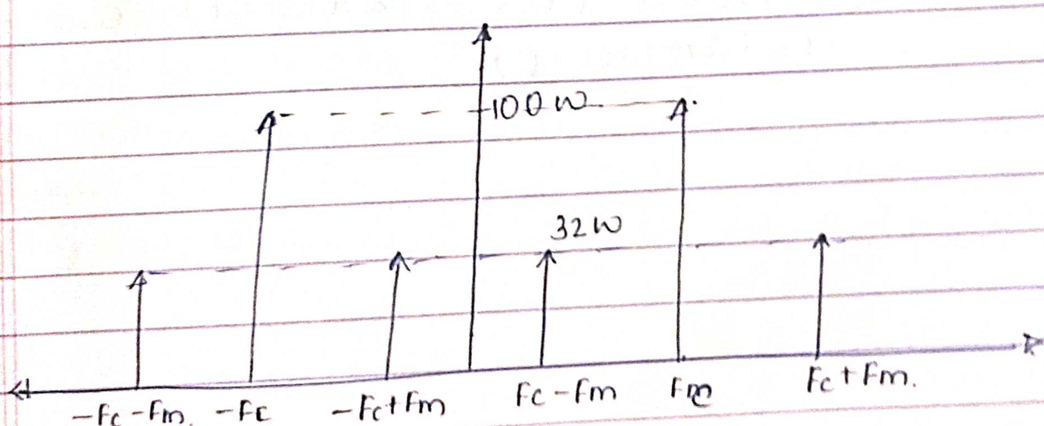
$$= 16\,W$$

$$P_{sb} = 16 + 16 = 32\,W$$

iii

$$P_T = 100 + 32 = 132\,W$$

iv. Power spectrum



v AM varients

for DSBFC signals

$P_T = 132\,W = P_c + P_{ls} + P_{us}$

for DSBSC

$P_T = 32\,W$

SSB

$P_T = 16\,W$

for VSB

$P_T = P_{ls} + P_{uB}$

$(16 + P_{uB})\,W$

By observing power of all AM varients we conclude that SSB AM is uses less power $P_{uB} = 16\,W$.

L2 PO1.3.1
L2 PO1
L2 P

11) A crrier signal $v_c(t) = 5\sin(2\pi \times 10^6)t]$ is amplitude modulated by a modulating sinusoidal signal and $v_m(t) = \sin[(4\pi \times 10^3)t]$. Determine the amp of F of the cs and the modulating signal. Also write expression for the resulting AM signal.

$V_c(t) = 5\sin((2\pi \times 106)t]$

$V_m(t) = \sin(4\pi \times 103)t.$

$\cancel{f_c = 106\,Hz}$ $= 10^6\,Hz$

$\cancel{f_m = 206\,Hz}$ $= 206\,Hz$

$\cancel{\mu = \frac{A_m}{A_c} = 0.12}$ $\mu = \frac{A_m}{A_c} = 0.2$

$\cancel{A_c = 5V}$ $A_c = 5V$

$\cancel{A_m = 1V.}$ $A_m = 1V.$

$s(t) = 5\sin(2\pi(106)t) + \frac{5\times 0.2}{2}\cos(2\pi(100)t)$

$\cancel{+\frac{5\times 0.2}{2}}\cos(2\pi(312)t)$

$= 5\sin(2\pi(106)t) + 0.5\cos(2\pi(100t) - 0.5\cos(2\pi(312)t)$

12.

Given

$P_T = 10\,KW$

$\mu = 0.8.$

$P_T = P_c\left(1 + \frac{\mu^2}{2}\right)$

$P_c = P_{LS} = P_{US} = \frac{A_c^2 \mu^2}{8R_c}.$

$10 = P_c\left(1 + \frac{(0.8)^2}{2}\right)$

$P_c = 7.57K$

So from

$P_T = P_C\left(1 + \frac{m^2}{2}\right)$

$P_t = \frac{10 \times 10^3}{1 + \frac{0.5^2}{2}}$

$P_C = 7.5 KW$

$P_1 = P_{LS} = P_{US} = \frac{\mu^2 P_C}{4}$

$= \frac{0.5^2}{4} \times 7.5K\,W$

$= 1.2 KW$

$\eta = \frac{\mu^2}{2+\mu^2} \times 100$

$\eta = 26.26\%$

$P_{LSB} = P_{USB} = \frac{P_T - P_C}{2} = 1.21K$

Transmission eff $= \frac{P_{SB}}{P_T} \times \cancel{100}$

$= \frac{2.43}{10} = 24.24\%$

20

Given

$s(t) = 2\cos(2\pi 110 t) \cdot \cos(2\pi 10 t)$

$N = 250$ samples

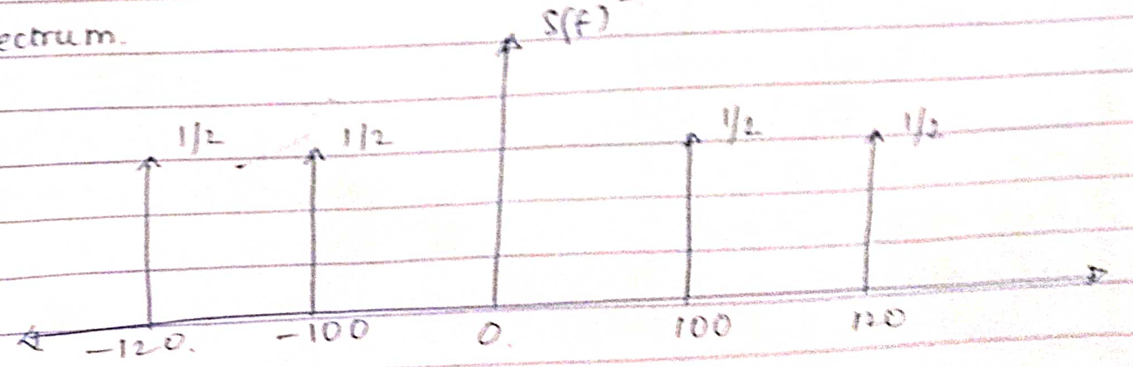
Now,

$s(t) = 2 \times \frac{1}{2}\left[\cos(2\pi \times 120 t) + \cos(2\pi \times 100 t)\right]$

$= \cos(2\pi \times 120 t) + \cos(2\pi \times 100 t)$

$S(f) = \frac{\delta(f-120) + \delta(f+120) + \delta(f-100) + \delta(f+\;)}{2}$

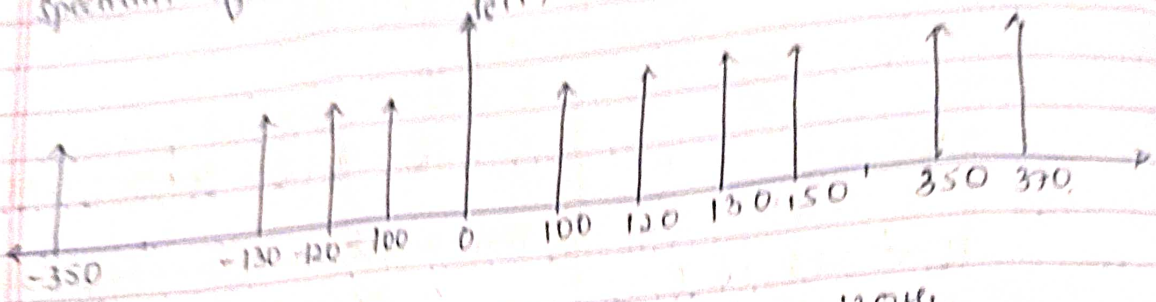
Spectrum



max frequency = 140Hz = f_m

$f_s = 2f_m = 280Hz$

Spectrum of sampled signal



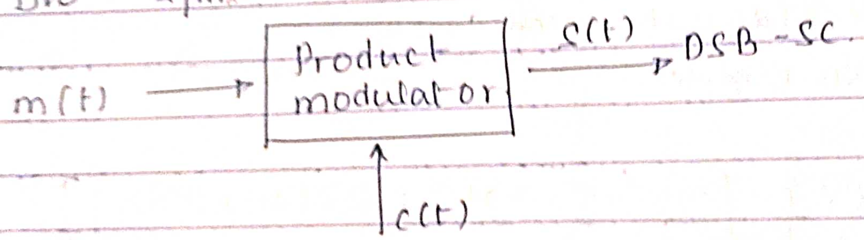
cutoff frequency of reconstruction filter = 120Hz

8. Define Double sideband SC wave and illustrate its generation using a balanced modulator.

DSBSC → 2 side bands are avilable.

carrier wave is suppressed

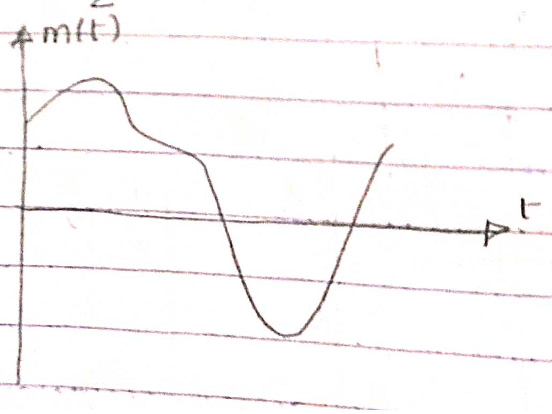
BW − $2f_m$

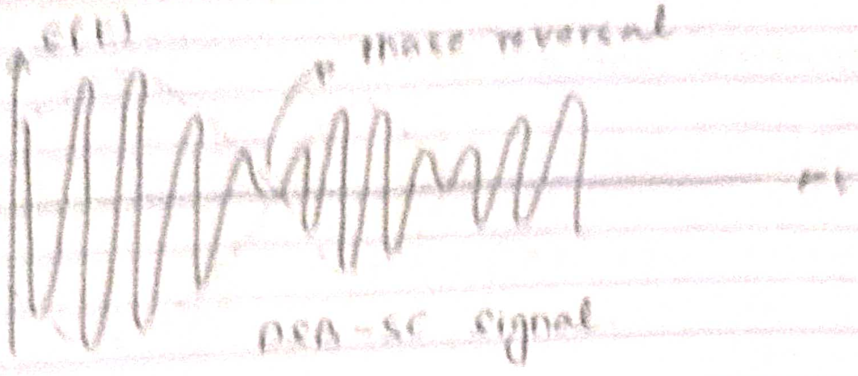


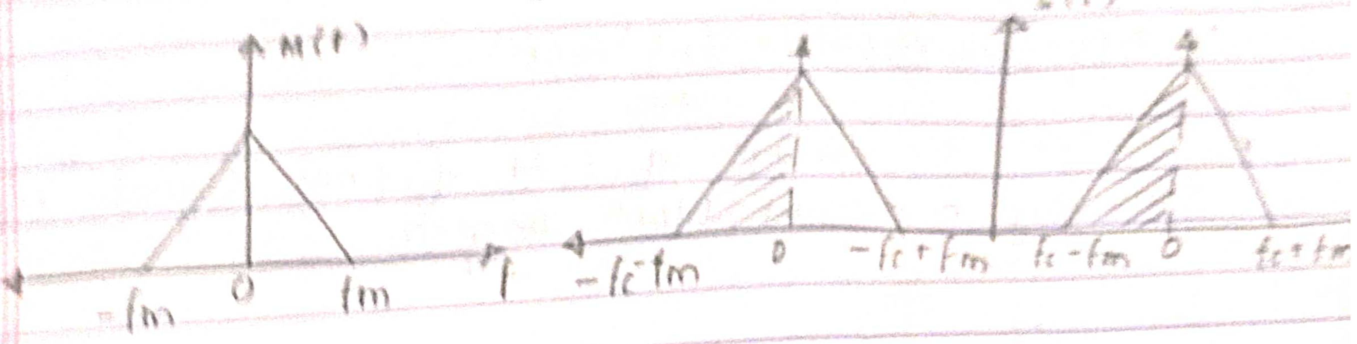
$$s(t) = m(t)\, c(t)$$

$$m(t) \cdot A_c \cos(2\pi f_c t)$$

$$s(f) = \frac{1}{2} A_c [M(f - f_c) + M[f + f_c)]$$



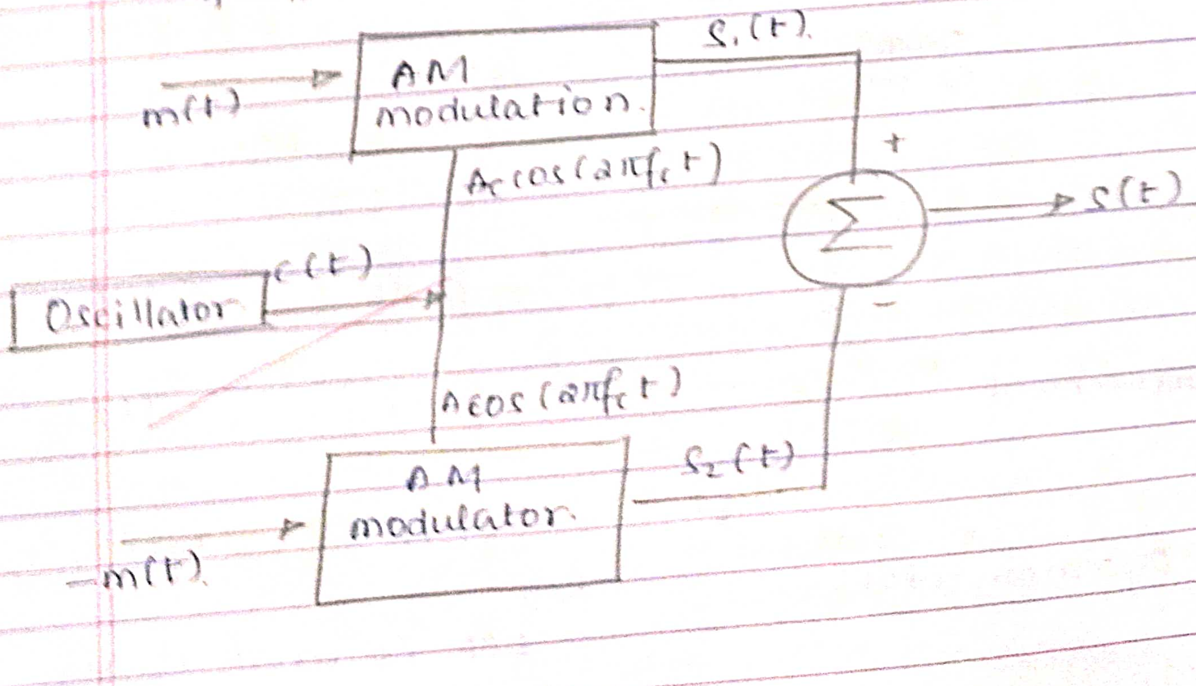




Generation of DSB-SC using Balanced modulator

- Use of 2 non-linear devices
- non-linear devices — diode, transistor of JFET
- 1 non-linear can generate carrier wave + a side band
- But using 2 non-linear devices carrier wave is suppresed.
- Balanced modulated circuit is used to supress the carrier from AM signal

Principle used in Balanced Modulation.



O/p of 2 modulators can be

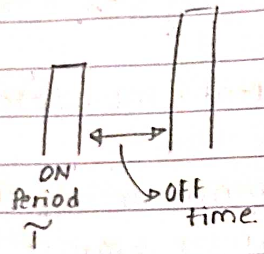
$s_1(t) = A_c[1 + k_a m(t)]\cos(2\pi f_c t)$

$s_2(t) = A_c(1 - k_a m(t)]\cos(2\pi f_c t)$

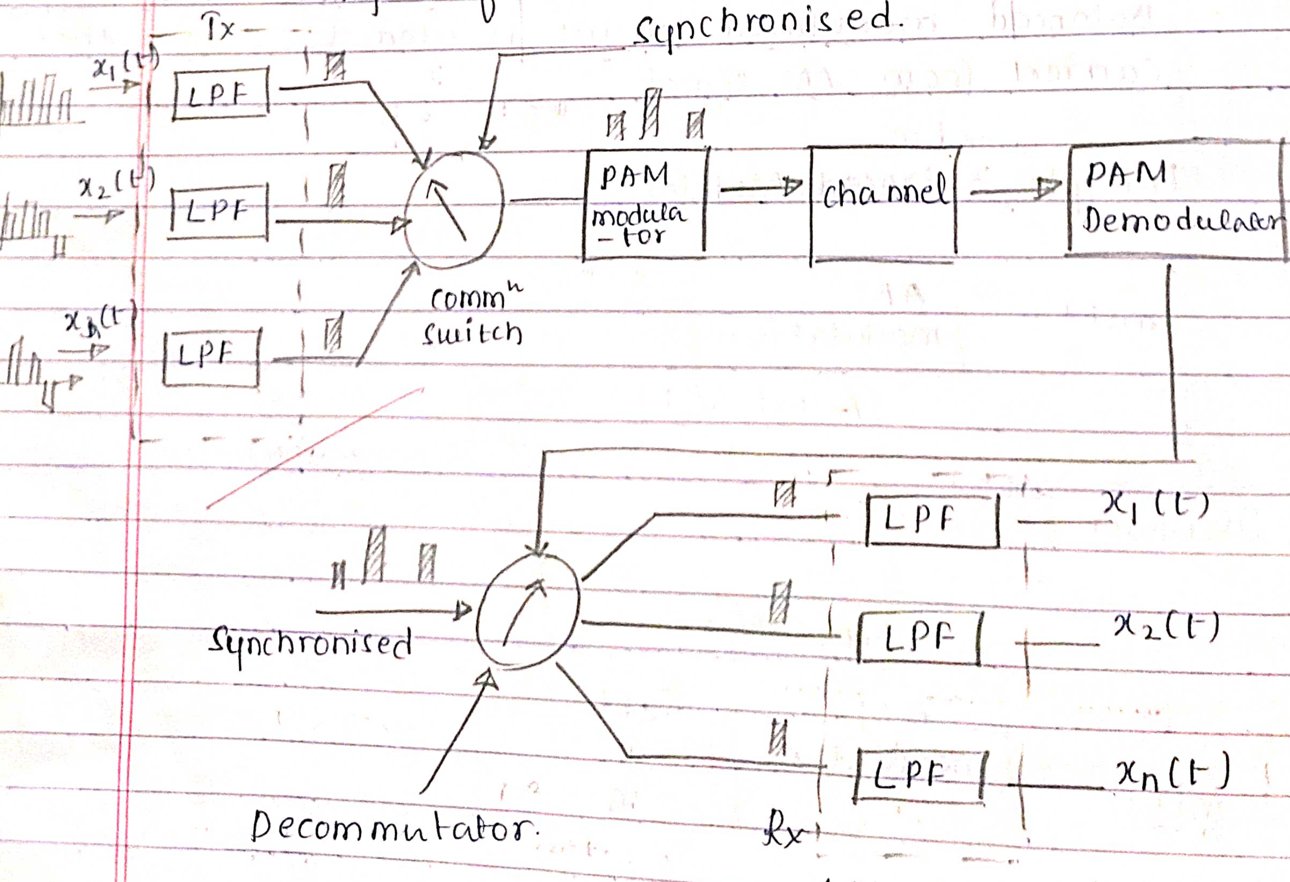
$s(t) = s_1(t) - s_2(t)$

$s(t) = 2 k_a A_c \cos(2\pi f_c t)\, m(t)$.

(17) Explain the concept of TDM of PAM signals by drawing a relevant block diagram.



During 'off' time, the channel can be used to transmit other samples of PAM.

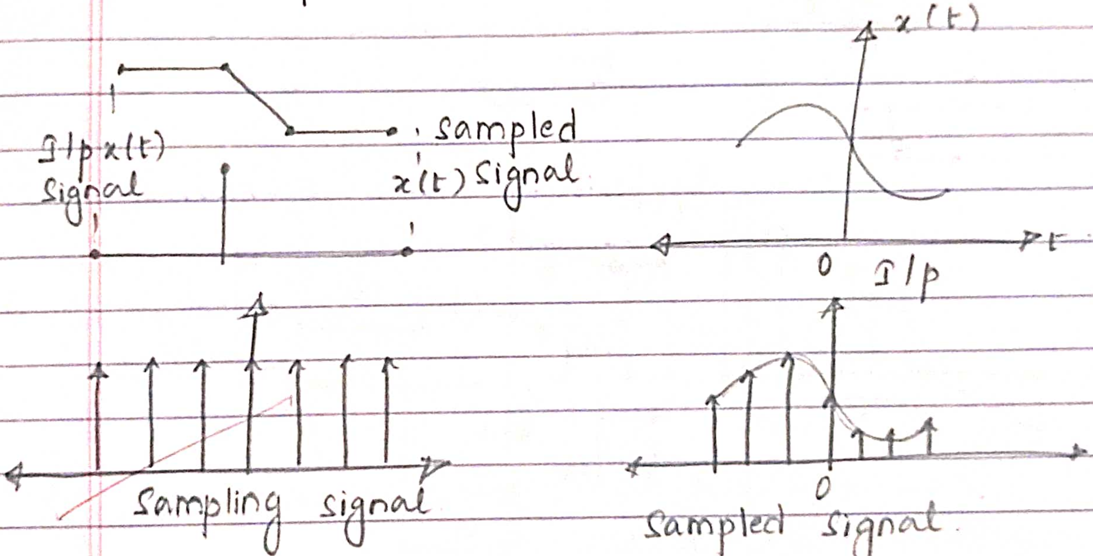


LPF is used to get

16. Compare ideal and practical sampling

Ideal Sampling.

- also known as instantaneous or impulse train sampling.
- In this method, the sampling signal is a periodic impulse train
- The area of each impulse in the sampled signal is equal to the instantaneous value of the i/p signal $x(t)$
- Duration of impulse is zero ($T_s = 0$)
- Practically it is not possible

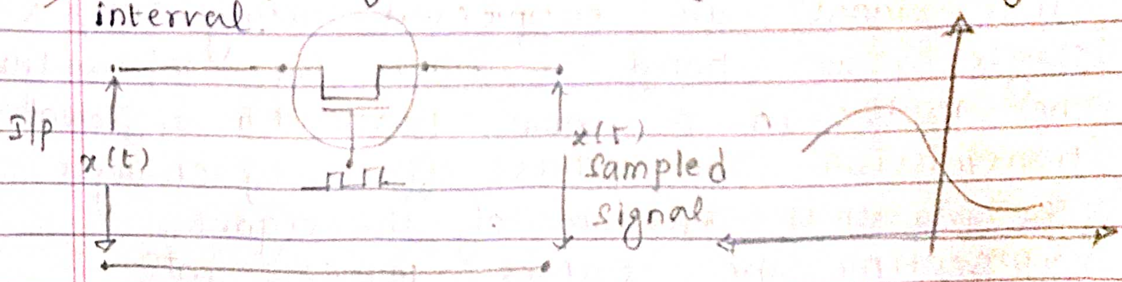


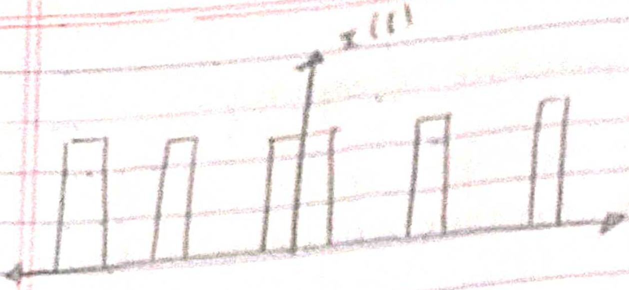
Practical sampling.

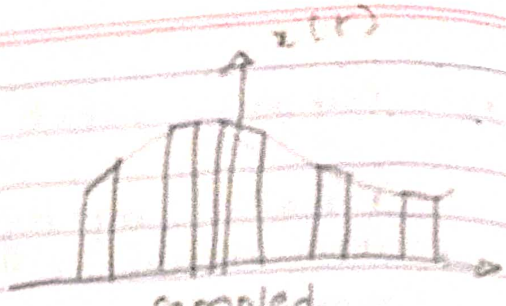
There are 2 types of practical sampling.

Natural Sampling → practical sampling.

- The sampling signal is a pulse train and pulse has a Finite width of τ.
- the top of each pulse in the sampled signal retains the shape of the input signal $x(t)$ during pulse interval.

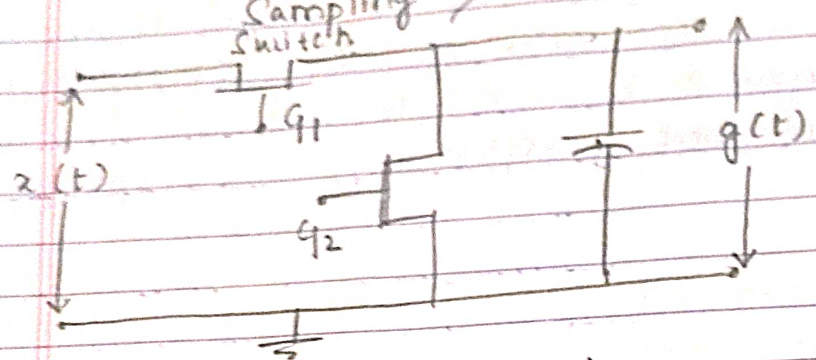


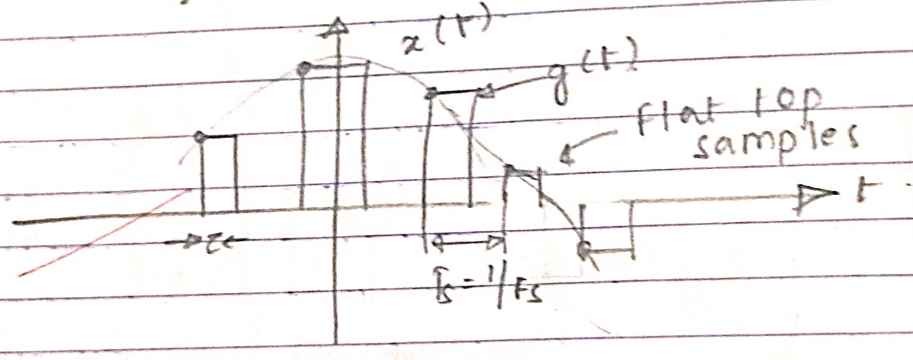




Sampled signal

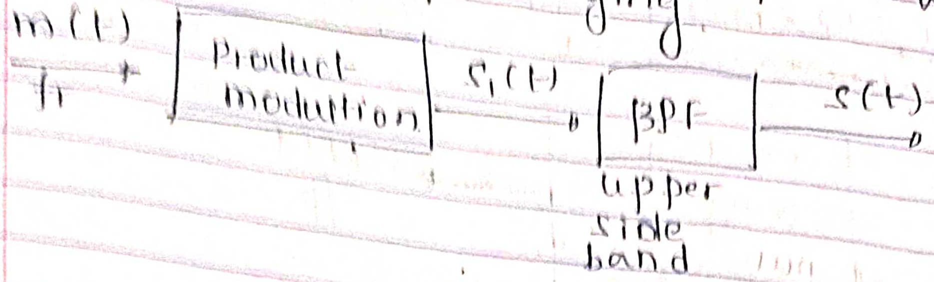
• Flat top sampling

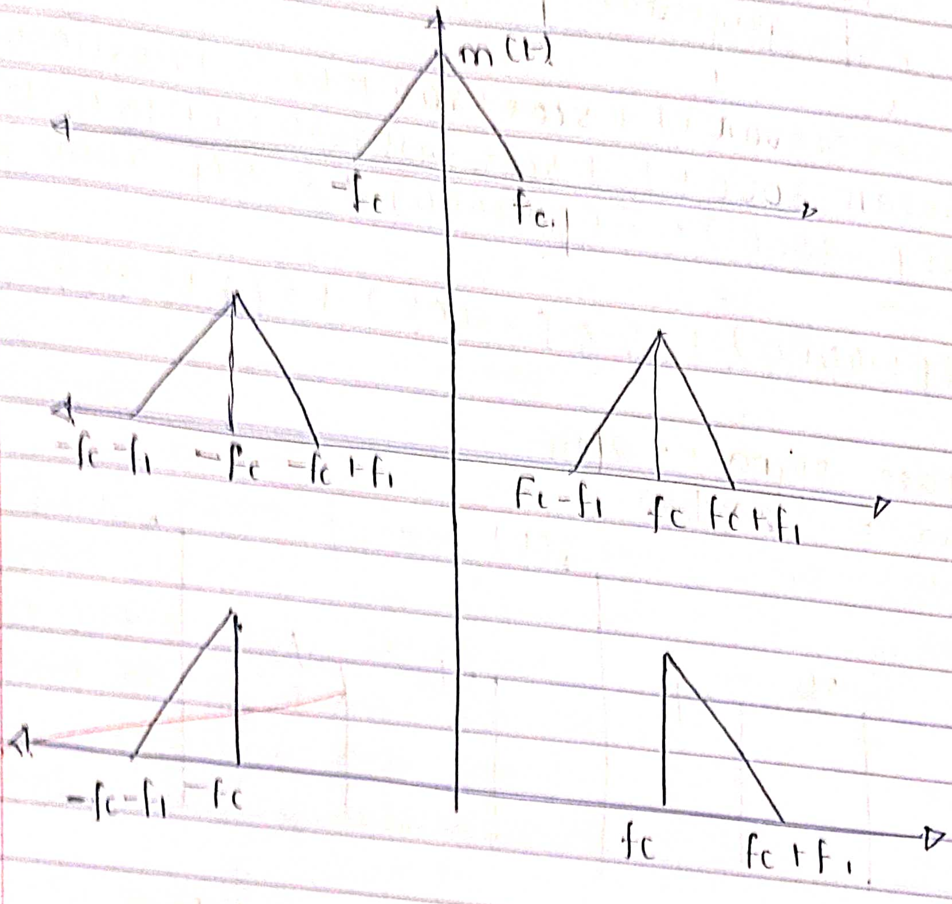




10.

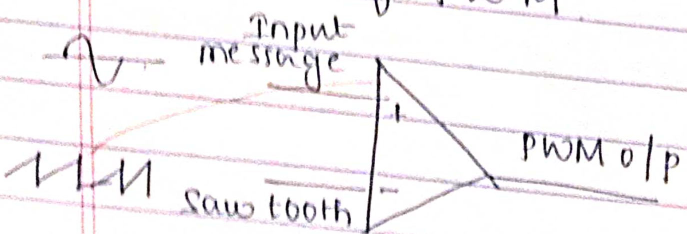
It refers to frequency changing because it involves shifting the entire information containg signal to differentiate frequency range. In SSB modulation one of the side bands and the carrier signal are suppressed leaving only a single side band to carry the information The results in a more bandwidth-efficient transmission. The process effectively changes the frequency spectrum of the signal. Converting the signals power into a narrow frequency range compared to traditional AM. This shift in frequency is why SSB described as freq changing.





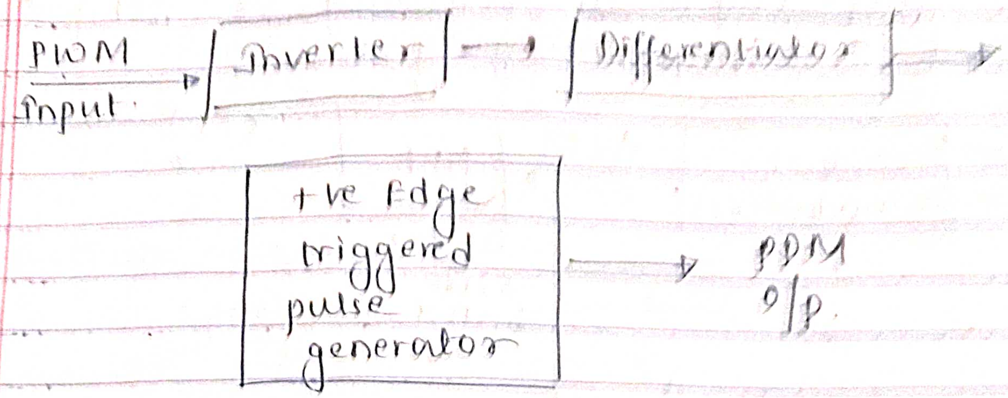
15. Write a short note on PPM PAM PWM.

Generation of PWM



Generation of PWM by a comparator.

> Generation of PPM



2) $s(t) = 5\cos(6000\pi t) + 5\cos(800\pi t) + 10\cos(10000\pi t)$

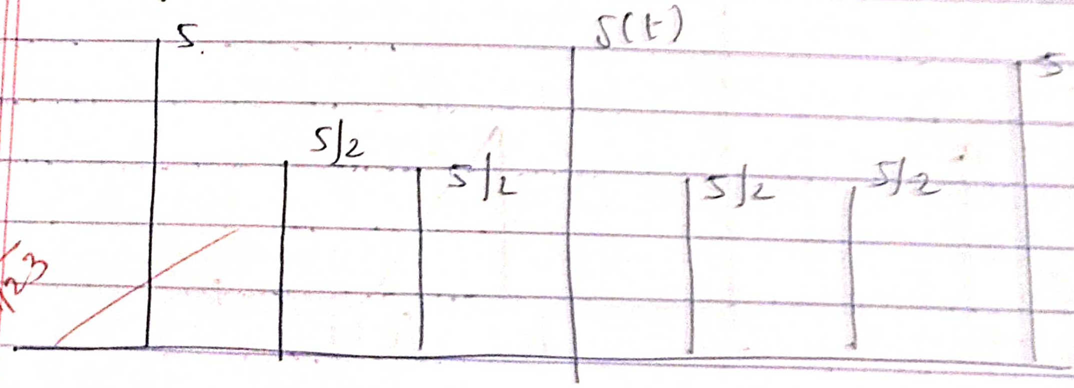
$= 5\cos(2\pi 3000 t) + 5\cos(2\pi 4000 t) + 10\cos(2\pi 5000 t)$

$= \frac{5}{2}\delta(f-3000) + \delta(f+3000) + \frac{5}{2}\delta(f-4000)$

$+ \delta(f+4000) + \frac{10}{2}(f-5000) + \delta(f+5000)$

a) Low pass signal : $2f_m$

band pass $= 2 \times BW$



Saraj 16/11/23

Low pass signal : $f_s = 2f_m = 10000\,Hz$

Band pass $= f_s = 2 \times BW = 4000\,Hz$

Master tutorial - 2

7) Consider a PCM system with a signal bandwidth of 4 KHz. The sampling rate is 10KHz, and each sample is represented using 16 bits. Calculate the following Bit rate - Nyquist frequency - signal-to-quantization

Given $BW = 4 KHz$ $BW = 4KHz$

$fs = 10 KHz$ $fs = 10KHz$

$N = 16$ $N = 16$

Bit rate $\Rightarrow fs \times N$ 2^{16}

$10 \times 10^3 \times 16$

$= 160 \times 10^3$.

Nyquist frequency $= 2fm$

$= 8KHz$.

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

$= 1.8 + 6 \times 16$

$= 1.8 + 96$

$= 97.8 \, dB$

8. In a PCM system ~~wi~~ the input signal ranges from -5V to +5V. The quantization is performed using a 10 bit ADC. Determine the following.

- Number of quntization levels.
- Quantization step size.
- Maximum quantizatio error.

$I/p \Rightarrow [-5, 5)$ $\frac{5+5}{L}$

* Number of quantization $= 2^{bits}$

$= 2^{10}$

$= 1024$ levels.

Quantization step size $\Delta = \frac{10}{1024}$ $\frac{5+5}{L}$

$= 0.00976$ // 9.76×10^{-3}

maximum quantization error $= \frac{\Delta}{2} = \frac{9.76 \times 10^{-3}}{2}$

$= 4.88 \times 10^{-3}$

9. A PCM system is used to encode an analog signal with a frequency range of 0 to 4 kHz. If the quantization error should be less than 1% of the maximum i/p signal amp. calc the ~~maximum~~ minimum number of bits required for quantization. Additionally, determine the bit rate if the sampling rate is 16 kHz.

Given,

$f_s = 16\ kHz$

error $= 0.01$

$\Delta = 2 \times$ error

$= 0.02\ V$

min. no of bits $= \dfrac{(4-0)\ K}{\cancel{0-0.2}\ 0.02}$

bits $= 2.00\ K$ //

$2^N = 200K$

$2^N = 2,00,000$

$N \log 2 = \log(2,00,000)$

$N = 17.60 \approx 18$

levels $= 2^N$

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

$= \log_2[200K]$

$= 17.60 \approx 18$

if $f_s = 16\ kHz$

$T_s = \dfrac{1}{16 \times 10^3}$

Bit Rate $= f_s \times N$

$= 16K \times 18$

$= 288K\ bps$

10. A speech signal has a total duration of 10s. It is sampled at the rate of 8 kHz and then encoded. The signal-to(qun) noise ratio is required to be 40 dB. Calculate the minimum storge capacity needed to accommudate this

digitized speech signal.

Give, $T_0 = 10s$

$f_s = 8KHz$

$(SNR)_{pcm} = 40 dB$

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

$40 = 1.8 + 6N$

$\boxed{N = 6.366 \approx 6 \text{ bits}}$

no. of samples $= 8K * 10 \quad = f_s * t$

$= 80K$ samples

no. of bits in 10s sample $= 80K \times 6$ 1 bytes → 8 bits

$= 480K$ bits n → 480 bits.

$= \underline{\underline{60K \text{ bytes}}}$

13. A Delta modulation system input applied 10 KHz, 1Vp-p. The signal is sample ten times more than Nyquist rate, what is the minimum step size ~~req~~ required to prevent slop overload?

WKT,

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

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

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

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

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

So, $A_m = \frac{1}{2} = 0.5V$

$f_m = 10 KHz$

$f_s = 10 * 2f_m = 10 \times 2 * 10K$

$= \underline{\underline{200 KHz}}$

$2\pi f_m A_m < \Delta f_s$

$2\pi \times 10K \times 0.5 < \Delta \times 200K$

$\frac{2\pi \times 0.5}{20} < \Delta$

$\frac{\pi}{20} < \Delta$

$\Delta \geq 0.157$

Note: condition for slope overload $2\pi f_m A_m < \Delta f_s$ (1)

14. A 4 KHz sinusoidal message signal having amplitude 4 V is fed to a delta modulator operating at a sampling rate of 32 KHz. The mimimum step size required to avoid slope overload noise in the DM is ___ V.

$A_m = 4V$

$f_s = 32KHz$

$f_m = 4KHz$

$2\pi f_m A_m < \Delta f_s$

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

$\Delta \leq 3.142V$

15. A sinusoidal signal of 2KHz frequency is applied to a delta modulator. The sampling rate and step - size Δ of the delta modulator are 20,000 samples per second and 0.1V respectively. To prevent slope overload, the max amp of the sinusoidal signal (in Volts) is

$f_m = 2KHz$

$f_s = 20K$ samples / second.

$\Delta = 0.1V$

$2\pi f_m A_m < \Delta f_s$

$AA \times WK \times Am \times \overset{AM}{WK} = 0.1$

$Am \times \frac{0.5}{\pi}$

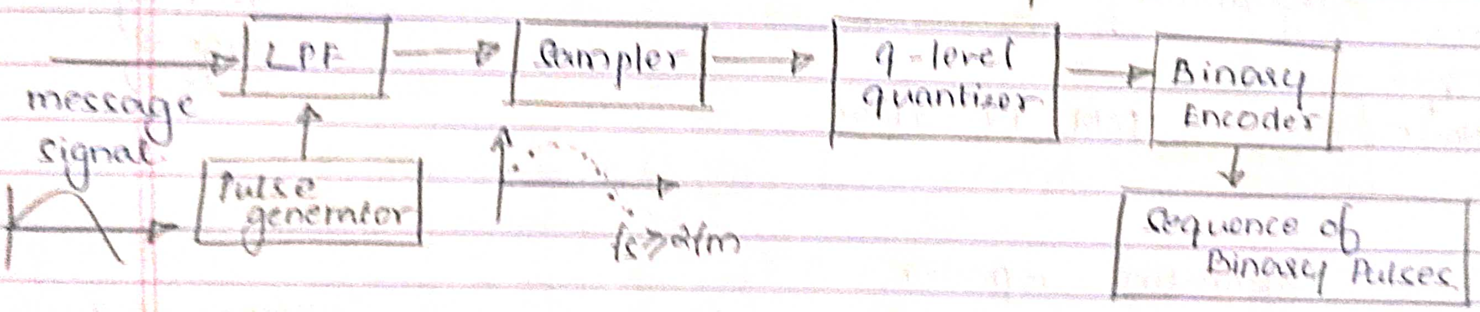
$Am = \frac{0.5}{\pi}$

$Am = 0.159V$

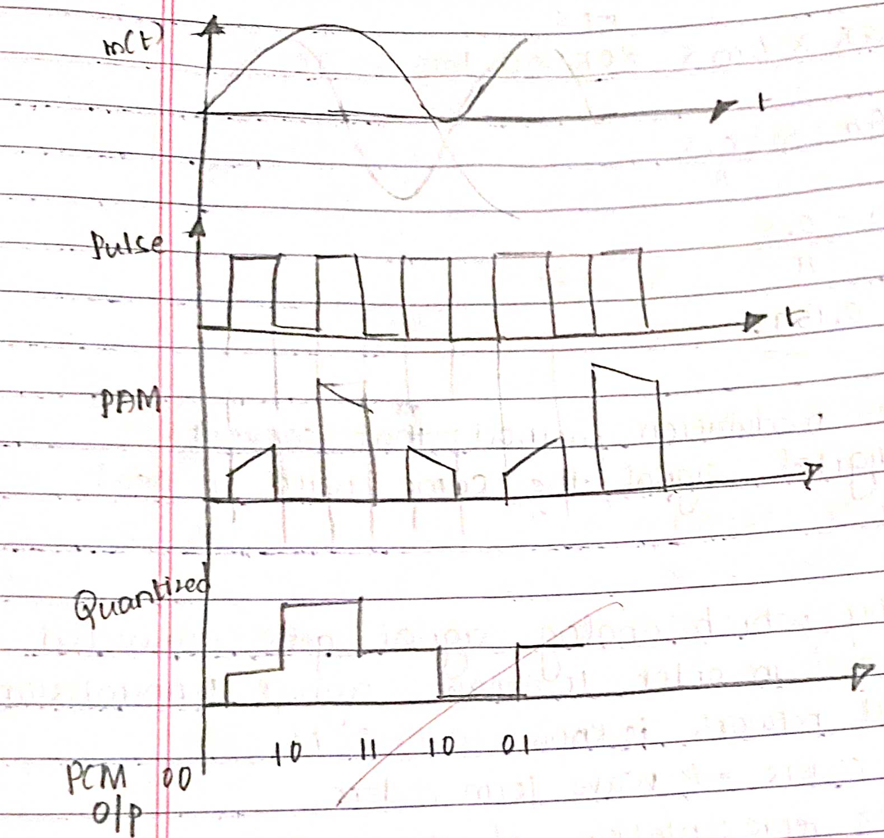
(1) Identify the modulation method that converts analog to digital signal the same with a neat block daigram

- A techinque by which analog signal gets converted into digital form in order to have signal transmission through a digital network is known as PCM
  PCM → signal coders → wave form coders.
- PCM allows the representation of the continuous time signal as a sequence of binary coded pulse

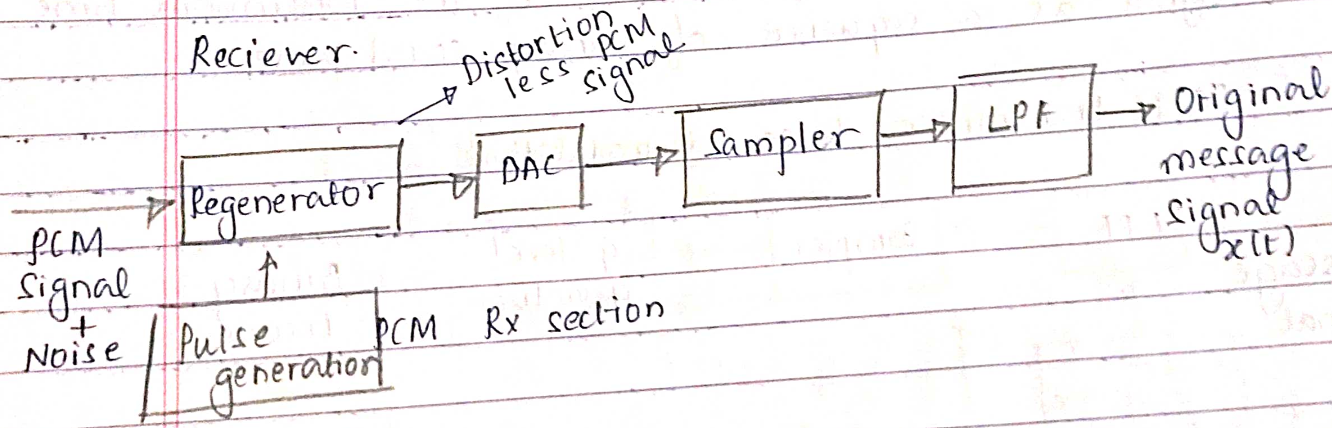
PCM(tranemitter) Pulse code modulation



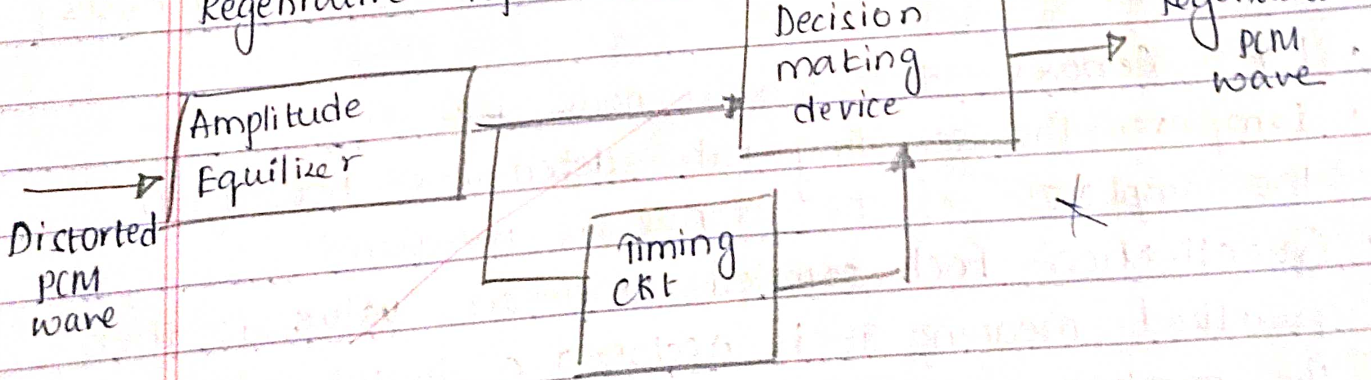
- LPF - Removes the sample noise i.e
- Sampler - The sampling rate determines how often the amplitude of the signal is measured
- Quantization - Each sampled amplitude value is then quantized, meaning it is assigned a digital code. The number of bits used for ~~representation~~ quant^n determines the resolution of digital representation
- Encoding - The quantized values are encoded into a digital bit stream.



Reciever.



Regenrative repeater



Advantage

- Immune to channel (Better noise immunity)
- Encoders allow secured data transmission

Disadvantage

- PCM increases the transmission Bandwidth ~~system~~
- System is complex

Application

Digital telephone

Digital audio appⁿ

Space commⁿ

5. Signal to noise Ratio Derivation.

Consider i/p M of continuous amplitude which is symetrically occupies the range [$-M_{max}$, M_{max}] of uniform ~~mid~~ mid rise quantizer, then the step size is given by

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

$$= \frac{2M_{max}}{L} \quad \text{--- (1)} \qquad L = \text{no. of levels}$$

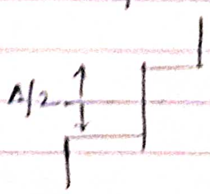
Let R denote the no. of bits / sample used to construct binary code

$$L = 2^R$$

Substitute in Eq (1)

$$\Delta = \frac{2M_{max}}{2^R} \quad \text{--- (2)}$$

Quantization error (Q) for uniform midrise quantizer will have samples b/n $-\frac{\Delta}{2}$ to $\frac{\Delta}{2}$



If step size is sufficiently small, then quantization error is uniformly distributed random func^n

∴ Probability density func^n of Quantization noise is

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

$$\sigma_Q^2 = E[Q] = \int_{-\Delta/2}^{\Delta/2} q^2 f_Q(q)\, dq$$

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

$$= \frac{1}{\Delta} \frac{\left[\left(\frac{\Delta}{2}\right)^3 - \left(\frac{-\Delta}{2}\right)^3\right]}{3}$$

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8}\right] = \frac{1}{3\Delta}\frac{\Delta^3}{4}$$

$$\sigma_Q^2 = \frac{1\,\Delta^2}{12} \quad \text{--- (1)}$$

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

$$\sigma_Q^2 = \frac{4M_{max}^2}{2^{2R} \times 12}$$

$$= \frac{M_{max}^2}{2^{2R} \times 3}$$

Let P be the power of original message signal then o/p signal to Noise ratio of uniform quantizer is

$$(SNR)_0 = \frac{P}{\sigma_Q^2} = \frac{P \times 2^{2R} \times 3}{M_{max}^2}$$

$\Rightarrow$ SNR ↑ as R ↑

also, $P = \frac{A_m^2}{2} = \frac{M_{max}^2}{2}$

$$(SNR)_0 = \frac{M_{max}^2 \times 2^{2R} \times 3}{M_{max}^2 \times 2}$$

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

Taking log

$$10 \log_{10} (SNR)_0 = 10 \log_{10} \left(\frac{3}{2}\right) 2^{2R}$$

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

$$= 1.7609 + 10(2R) \log_{10} 2$$

$$= 1.7609 + 20R \, (301.029$$

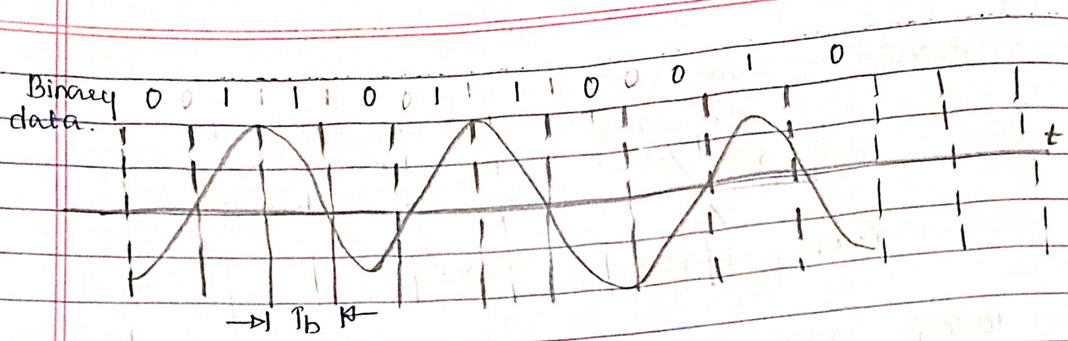
$$10 \log_{10} (SNR)_0 = 1.7609 + 6.02R.$$

26/11 $\Rightarrow$ $SNR = [1.760 + 6.02R] \, dB.$

$$\approx \underline{[1.8 + 6R] \, dB}$$

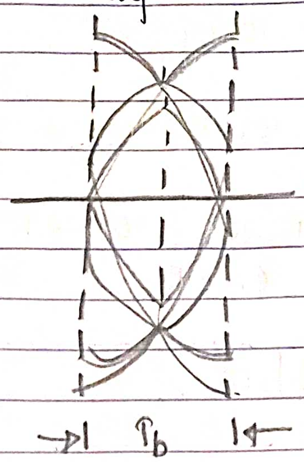
18.

- In a bandlimited channel, the pulse spreading at the o/p of the system will dispersed our an interval which is longer than that of the transmitted pulse
- Eye diagram is very effective tool for digital signal analysis during real time experiment
- Eye pattern is a practical technique for determining the severity of the degradation introduced by ISI and channel noise into the line coded digital pulses in base band transmission



(a)

Binary data sequence and its waveform.

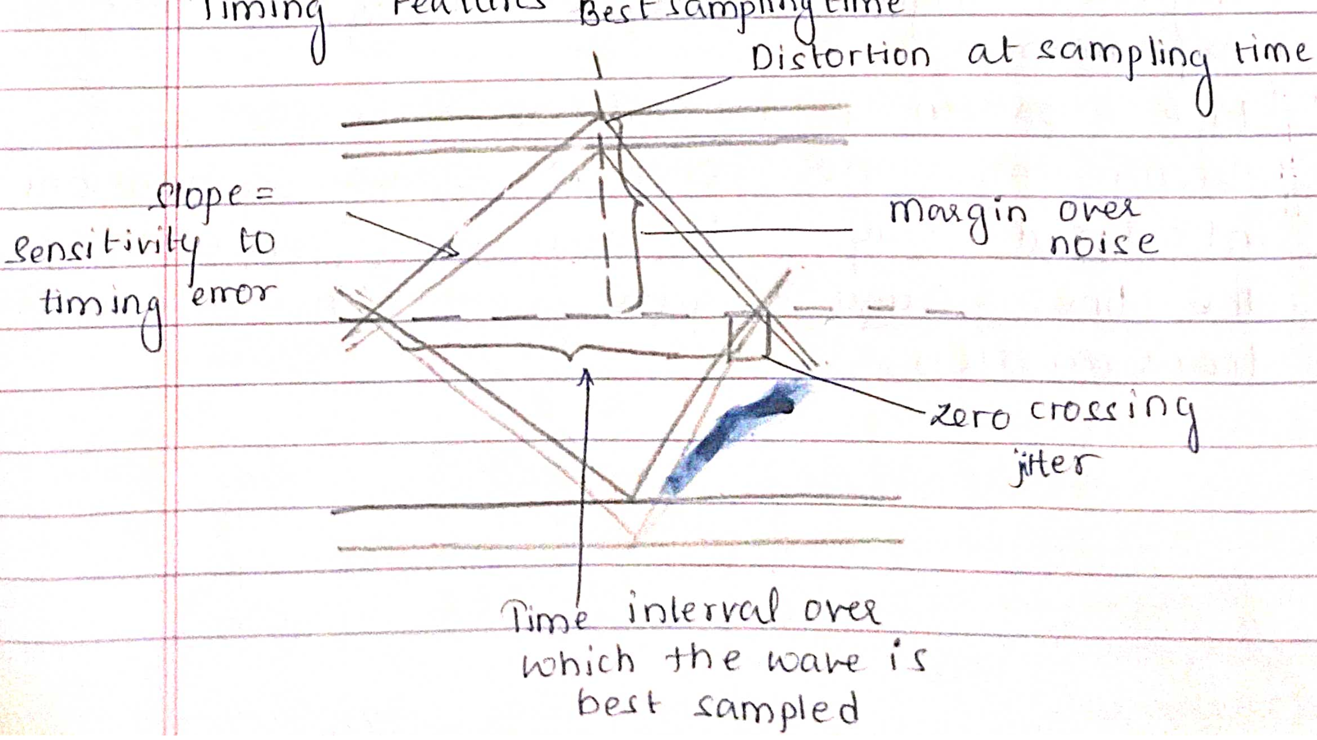


(b) Corresponding eye pattern.

the eye pattern offers 2 compelling virtues

- The simplicity of eye-pattern generation.
- The provision of a great deal of insightful information about the characteristics of the data transmission system

Timing Features



In fig 8:13,
The horizontal axis → represing time, spans the symbol interval from $-T_b/2$ to $T_b/2$.
T_b → bit duration.

1. Optimum sampling time. — The width of the eye opening defines the time interval over which the distorted binary waveform appearing at the o/p of the receive filter in the PAM system can be uniformly sampled without decision errors. Clearly, the optimum sampling time is the time at which the eye opening is at its widest.

2. Zero - crossing jitter - the timing signal is extracted from the ZC of the waveform that appears at the receive - filter output. There will always be irregularities in the zero - crossings, which, in turn, give rise to jitter and, therefore, non - optimum sampling times.

3. Timming sensitivity : Another timing - related feature is the sensitivity of the PAM system to timing errors. This sensitivity is determined by the rate at which the eye pattern is closed as the sampling time is varied.

6. Channel noise and error probability
Probability of error. — the frequency with which a certain probabilistic testing procedure will lead to a type I erro or a type II error.
average probability of symbol error is

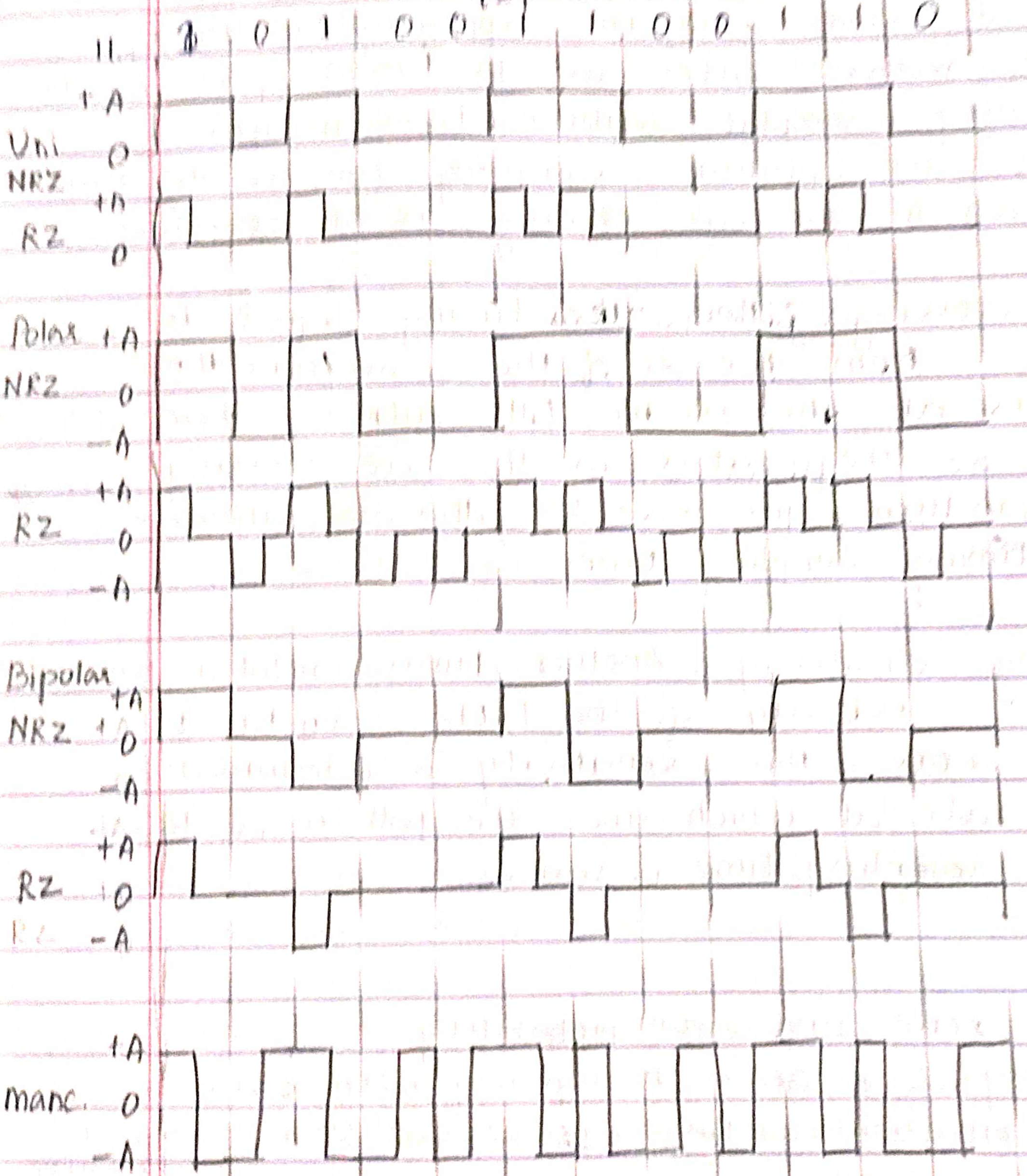
$$P_e = \sum_{i=1}^{M} \pi_i \, \mathbb{P}(x \text{ does not lie in } Z_i | m_i \text{ sent})$$

$$= \frac{1}{M} \sum_{i=1}^{M} \mathbb{P}(x \text{ does not lie in } Z_i | m_i \text{ sent}), \quad \pi_i = 1/M$$

$$= 1 - \frac{1}{M} \sum_{i=1}^{M} P(x \text{ lies in } Z_i | m_i \text{ sent})$$

since x is the sample value of random vector X,

$$P_e = 1 - \frac{1}{M}\sum_{i=1}^{M}\int_{Z_i} f_X(x|m_i)\,dx$$

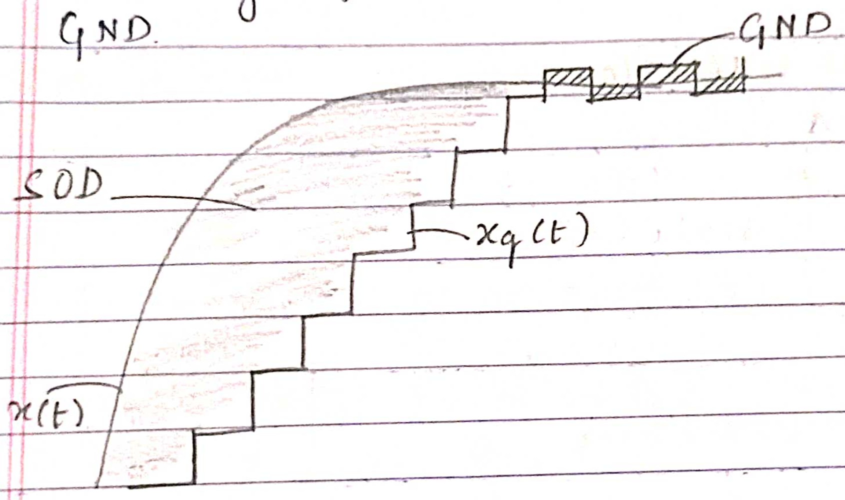


(3). Two types of Quantization noise.

1) Slope overload distortion - when the slope of $x(t)$ is much higher than $x_q(t)$ over a long preiod of duration the $x_q(t)$ can not follow the $x(t)$ which forms SOD.

i,e $x(t) - x_q(t)$

2 Granular nois distortion. when i/p signal $x(t)$ is relatively constant for in amplitude the $x_q(t)$ is bouncing up and down whill which will cause GND.

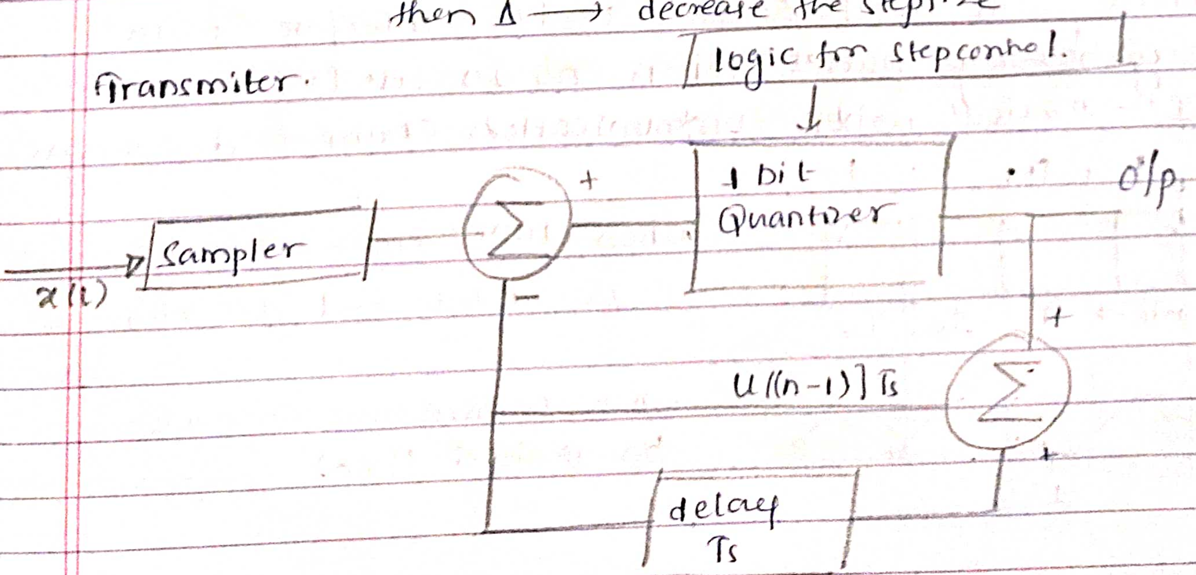


To over come this adaptive DM is required.

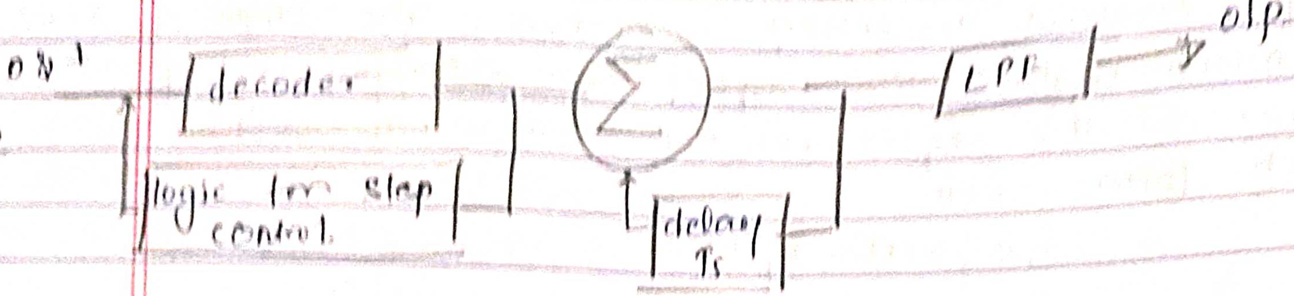
If $x(t) \longrightarrow$ high slope
the $\Delta \longrightarrow$ increase the step size

If $x(t) \longrightarrow$ less slope
then $\Delta \longrightarrow$ decrease the stepsize

Transmiter.



Receiver :



Step size $\Delta(n) = \Delta(n-1)\, e(n) + \Delta(0)\, e(n-1)$

$\Delta(0) = \Delta$

$e(0) = 1$

$\Delta(1) = \Delta(0)\, e(1) + \Delta(0)\, e(0)$

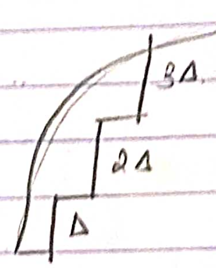
$= \Delta 1 + \Delta$

$= 2\Delta$

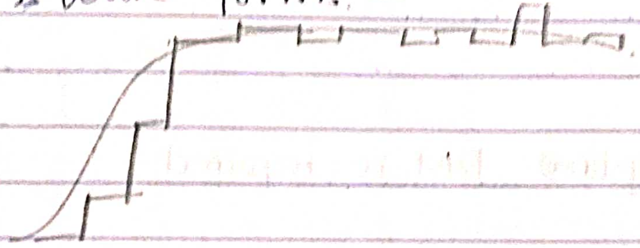
$\Delta(2) = \Delta(1)\, e(2) + \Delta(0)\, e(2)$

$= 2\Delta + \Delta$

$= 3\Delta$

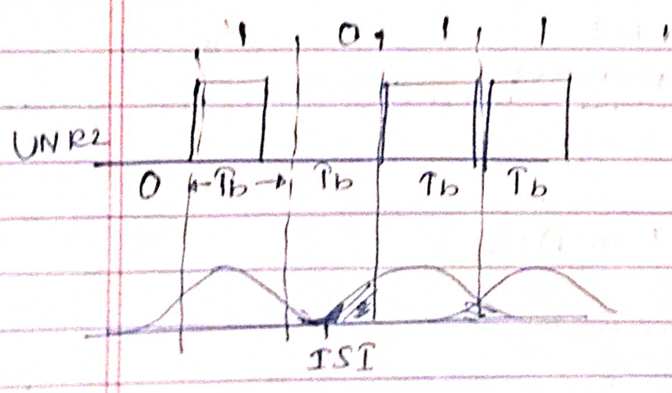


Δ wave forms :-



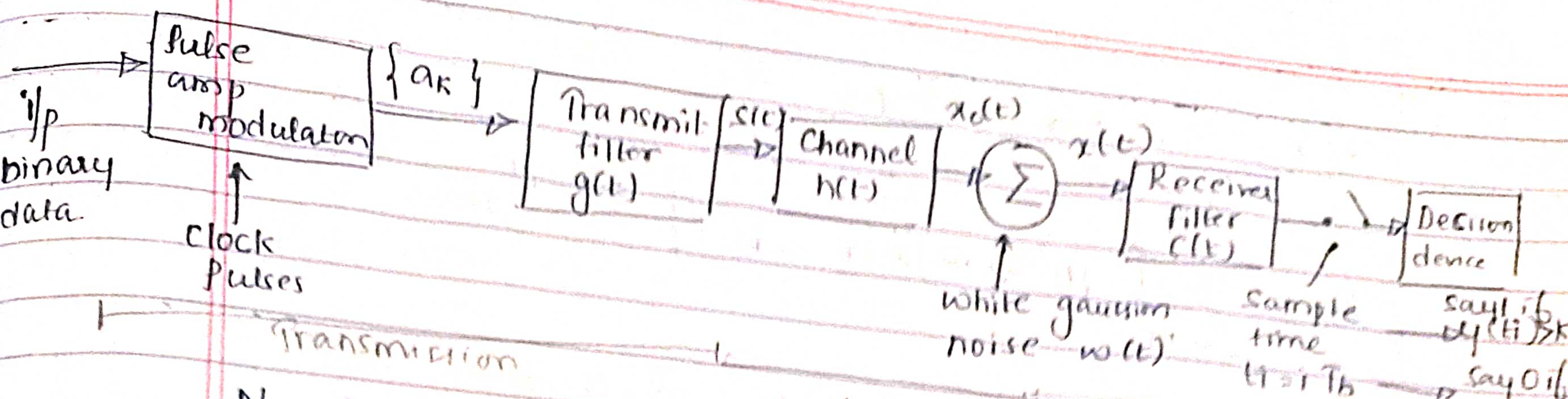
Inter symbol Interference

- Spreading of a pulse beyond its allotted time interval T_b will cause it to interfere with neighbouring pulses. This is known as ISI.
- ISI $\rightarrow$ arise when communication channel despersive



when transmission channel is linear and distorted

when transmission channel has limited B.W.



~~Nyqui~~ Nyquist

at ISI = 0, we get $y(t_i) = a_i$

for extracting, sample the o/p $y(t)$ at $t = iT_b$

$$y(t) = \sum_K a_k \, p(t - kT_b) \quad \text{--- } a$$

if we do sample for above eq.

$$y(t) = \sum_K a_K \, P(iT_b - KT_b)$$

if we put $i = k$ in $P(iT_b - kT_b)$
then it will become $P(0) = 1$ when $i = K$

$$\therefore P(iT_b - KT_b) = \begin{cases} 1 \\ 0 & i \neq 0 \end{cases} \quad \text{--- } (1)$$

Eq^n 1 is called Nyquist pulse, reffered as creation for distortionless binary baseband data transmission.

* FD $\{p(nT_b)\}$ — R_b - bit rate $= 1/T_b$ $\frac{\text{bits}}{\text{sec}}$

where $n = 0, \pm 1, \pm 2 \ldots$

$$P_\delta(f) = F[p(nT_b)]$$

$$P_\delta(f) = R_b \sum_{n=-\infty}^{\infty} P(f - nR_b) \quad \text{--- } (2)$$

∴ PSF left hand side FT
Given as

$$P_\delta(f) = \int_{-\infty}^{\infty} \sum_{m=-\infty}^{\infty} [P(mT_s)\,\delta(t-mT_b)] \cdot e^{-j2\pi f t} \cdot dt$$

Let $m = i - K$ when $i = k$ $m = 0$
$i \neq k$ $m \neq 0$

$$P_\delta(f) = \int_{-\infty}^{\infty} P(0)\,\delta(t)\,e^{-j2\pi f t} \cdot dt \quad \text{— (4)}$$

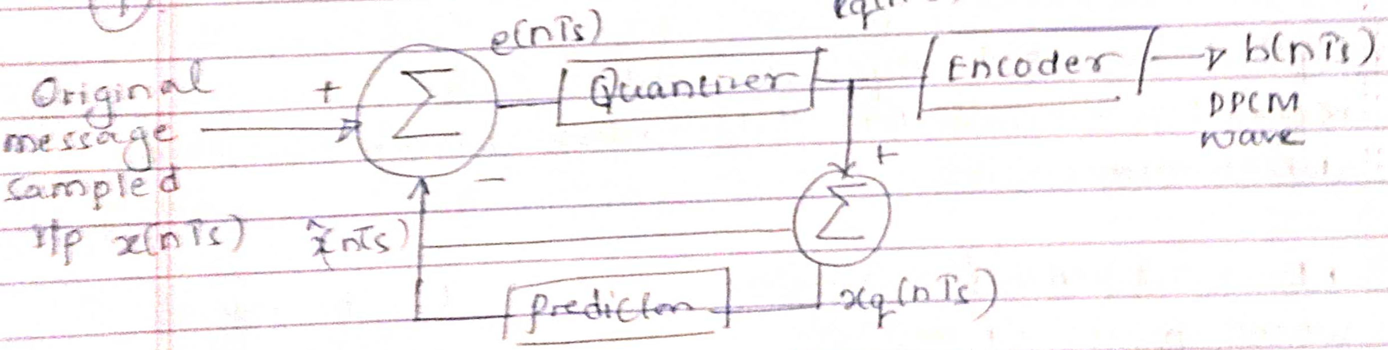
The frequency domain condition for zero ISI form eq 1 we get

$$\sum_{n=-\infty}^{\infty} P(f - nR_b) = T_b \quad \text{— (5)}$$

$$T_b = 1/R_b$$

The frequency $P(f)$ eliminates intersymbol interference for samples taken at intervals T_b provided that it satisfies Eqn (5).
where $P(f)$ is the overall s/m.

(4) DPCM transmitter



$x(t)$ → message signal $x(nT_s)$ → sampled signal
$\hat{x}(nT_s)$ predicted o/p
$e(nT_s)$ output of adder → Predictor error
$e_q(nT_s)$ o/p Quantizer

$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \longrightarrow (1)$

$e_q(nT_s) = e(nT_s) + q(nT_s) \quad (2) \quad q_e(nT_s) \to Q \sim$ error

$x_q(nT_s) = e_q(nT_s) + \hat{x}(nT_s) \quad (3)$

$x_q(nT_s) = \cancel{e_q(nT_s) - \hat{x}(nT_s) + \hat{x}(nT_s)}$

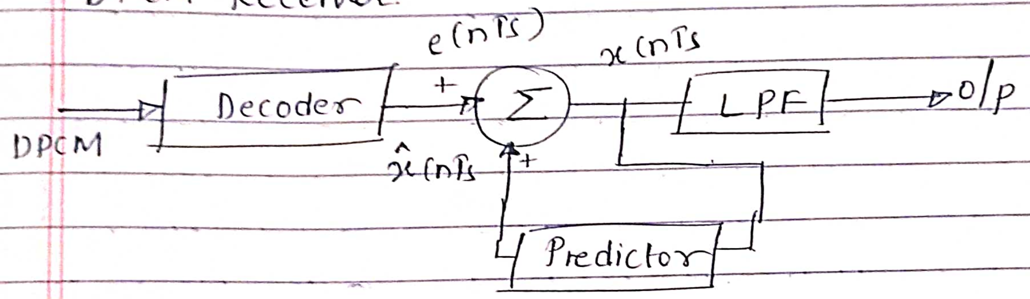
$= \cancel{x(nT_s)}$

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

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

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

DPCM Receiver



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

8. Channel noise :

Channel noise is any unwanted signal which added to the transmitted signal during the journey through the communication channel

Thermal noise
Interference
distortion noise

error
Probability

Error probability error aim to calc the probility incorrect information sent to destination ~~a~~ due to channel noise

Bit erro rate
Packet erro rate
Symbol " "

$$P_e = \sum_{i=1}^{M} \pi_i \, \mathbb{P}\{x \text{ does not lie in } Z_i | m_i\}$$

$$P_e = \frac{1}{M} \sum_{i=1}^{M} \cancel{\pi_i} \, \mathbb{P}\{x \text{ does not lie in } Z_i | m_i\} \quad \pi_i = 1/M$$

$$= 1 - \frac{1}{M} \sum_{i=1}^{M} \mathbb{P}\{x \text{ lie in } '' \}$$

x is the sample of any random vector X.

(10) SSB $\boxed{}$ AM

(or) heterody

- here,

$m(t)$