MODEL Question PAPER

Paper Code:- ETIT 208

Sub:- Communication Systems

NOTE :- Q.no.1 is Compulsory. Attempt any one question from each unit.

Q1. (a) Explain Different Sampling techniques.

Soln: There are three sampling techniques:

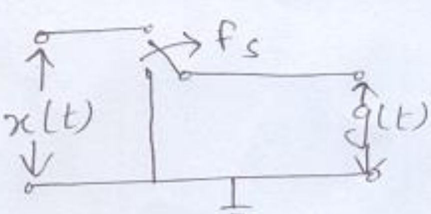
(a) Instantaneous or Ideal Sampling.

(b) Natural Sampling.

(c) Flat top sampling.

a) Ideal Sampling:

In this type, the sample function is a train of impulses.



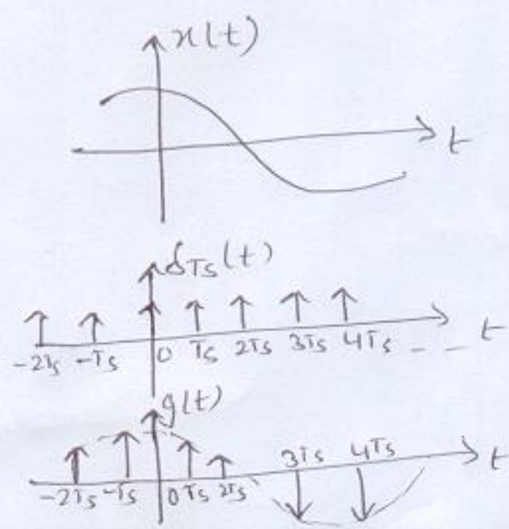
$$\delta_{Ts}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$g(t) = x(t)\,\delta_{Ts}(t)$$

$$= x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(t - nT_s)$$

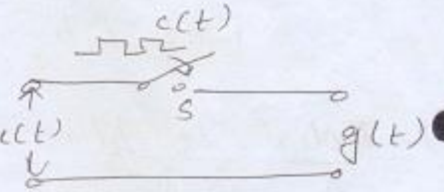
$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$



(2)

Natural Sampling:- Instantaneous sampling results in the samples whose width τ approaches zero. Due to this, the power content in the instantaneously sampled pulse is negligible. This method is not suitable for x-mission.

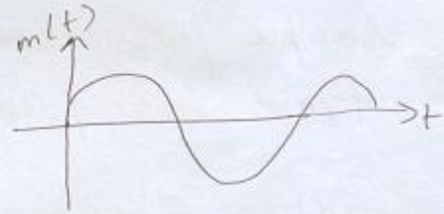
In natural sampling, the pulse has a finite width 'τ'.



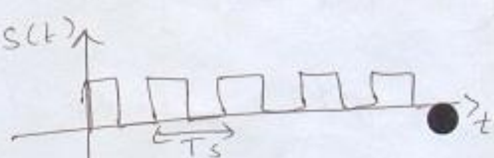
When $c(t)$ goes high, switch (S) is closed.

$\therefore g(t) = x(t)$ when $c(t)$ is A

$g(t) = 0$ " " $= 0$

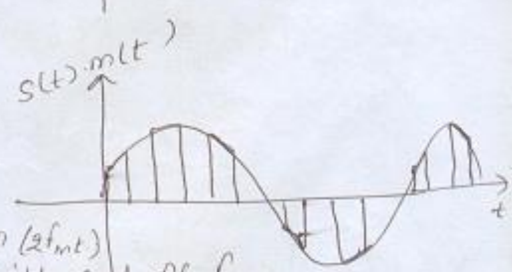


$$s(t) = \frac{\tau}{T_s} + \frac{2\tau}{T_s}\left[C_1 \cos 2\pi \frac{t}{T_s} + C_2 \cos 2\times 2\pi \frac{t}{T_s} - - -\right]$$



where $C_n = \dfrac{\sin n\pi\tau/T_s}{n\pi\tau/T_s}$

$T_s = \dfrac{1}{2f_m}$



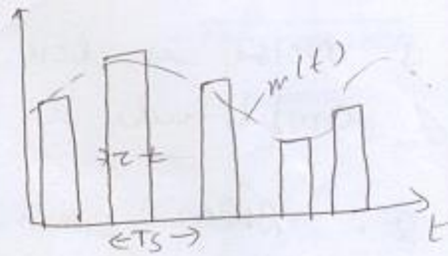
$$s(t)\,m(t) = \frac{\tau}{T_s}m(t) + \frac{2\tau}{T_s}\left[m(t)\,C_1 \cos 2\pi(2f_m)t + m(t)C_2 \cos 2\pi(2f_m t)\right]$$

In instantaneous sampling, LPF with cut off f_m delivers an o/p i.e.

$$s_o(t) = \frac{\tau}{T_s}m(t)$$

Flat top sampling In this the pulse has constant amplitude at some point within the pulse interval.

Original signal can not be recovered exactly by passing the samples through LPF.



The distortion is there but not so large, & is known as aperture effect.

Q4(b) A binary channel with bit rate $R_b = 36000$ bits per second (b/s) is available for PCM voice transmission. Find appropriate values of the sampling rate f_s, the quantizing level L, & the binary digits n, assuming $f_m = 3.2$ KHz.

Sol. we know, $f_s \geq 2f_m = 6400$ ∴ $nf_s \leq R_b = 36000$

$$n \leq \frac{R_b}{f_s} \leq \frac{36000}{6400} = 5.6$$

So $n = 5$, $L = 2^5 = 32$

$$\& \ f_s = \frac{36000}{5} = 7200 \text{ Hz} = 7.2 \text{ KHz}.$$

Q1(C) Give advantages & disadvantages of PWM & PPM.

Soln : PWM :
advantages

1. Noise is less as compared to PAM because amplitude is held const.

2. PWM comm. does not require synchronization between transmitter & receiver.

Disadvantages -

1. Pulses are varying in width. Therefore their power contents are variable.

2. Large bandwidth is reqd. for PWM comm.

PPM .

advantages :

1. Like PWM in PPM, Amplitude is held const. Thus ~~less~~ noise interference.

2. Because of const. pulse width & amplitude transmission power for each pulse is same.

Disadvantage :-

1) Synchronization between X-mitter & receiver is required.

2) Large Bandwidth is reqd.

Q11) How many AM broadcast station can be accomodated in 100kHz BW if the highest freq. modulating a carrier is 5kHz.

Sol. $f_{max} = 5kHz$

$BW \text{ of station} = 2f_m = 10kHz.$

$$\text{No. of stations} = \frac{100 \times 10^3}{10 \times 10^3} = 10 \text{ stations}$$

Q12) A broadcast radio transmitter radiates 10kW when modulation percentage is 60. How much of this is carrier power!

Soln $$P_c = \frac{P_t}{1+\frac{m^2}{2}} = \frac{10}{1+\frac{.6^2}{2}} = \frac{10}{1.18}$$

$$= 8.47 kW.$$

Q1 Give the comparison between AM and FM

Soln.

AM	FM
1. AM has two Sidebands	1) FM has infinite namber of sidebands.
2. AM Signals are less immune to noise.	2) FM are more immune to noise. To reduce noise we increase its freq. deviation.
3. It does not provide guard bands.	3) It provides guardband between FM Stations
4. It operates in MF and HF range	4) It operates in VHF and UHF range.
5. 10kHz of channel is required for AM broadcast.	5. Much wider channel i.e. 200 KHz is required.
6. Its equipments are easier and much Cheaper.	6. Its equipments are more complex and costly.

Q1(g) Explain the Information & properties of Information theory?

Soln. Information may be defined as the probability of occurrence or non-occurence of an event.

Let any comm. system have messages $m_1, m_2 --- m_n$ with probability of occurence $p_1, p_2 -- p_n$.

So amount of Information

$$I_{s(k)} = \log_2 \frac{1}{P_K}$$

The unit of information is bits.

Properties:-

1. $I(s_K) = 0$ for $P_K = 1$ where $S = S_1 + S_2 -- S_K$ sample space

i.e. no information for absolutely certain of the outcome of an event.

2. $I(s_K) \geq 0$ for $0 \leq P_K \leq 1$

i.e. occurence of an event either provides some or no information.

3. $I(s_K) > I(s_i)$ for $p_K < p_i$

i.e. less probable an event, the more information we gain.

4. $I(s_K s_i) = I(s_K) + I(s_i)$

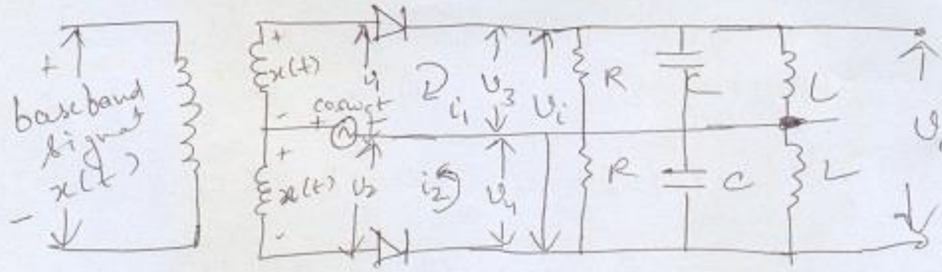
Unit - I

Q 2 (a) Explain the method of generation of DSB-SC ?

Soln. There are two methods of generation of DSB-SC.

1) Balanced Modulator
2) Ring Modulator

Balanced Modulator :- DSB-SC contains only two sidebands, thus two non-linear devices are connected in balanced mode so as to suppress the carrier of each other, then only sidebands are left i.e. DSB-SC signal is generated.



Modulating signal x(t) is applied to the two diodes through a centre-tapped transformer with the carrier signal $\cos\omega_c t$.

The non-linear V-I relationship is expressed as-

$$i = av + bv^2$$

The two voltages v_1 & v_2 across the two diodes are

$$v_1 = \cos\omega_c t + x(t)$$
$$v_2 = \cos\omega_c t - x(t)$$

For diode D_1,

$$i_1 = av_1 + bv_1^2$$

Similarly for D_2,

$$i_2 = av_2 + bv_2^2$$

Putting the value of v_1 in i_1

$$i_1 = a[\cos\omega_c t + x(t)] + b[\cos\omega_c t + x(t)]^2$$

$$= a\cos\omega_c t + ax(t) + b[\cos^2\omega_c t + x^2(t) + 2x(t)\cos\omega_c t]$$

$$= a\cos\omega_c t + ax(t) + b\cos^2\omega_c t + bx^2(t) + 2bx(t)\cos\omega_c t$$

Similarly, $i_2 = a[\cos\omega_c t - x(t)] + b[\cos\omega_c t - x(t)]^2$

$$= a\cos\omega_c t - ax(t) + b\cos^2\omega_c t + bx^2(t) - 2bx(t)\cos\omega_c t$$

The net voltage v_i at the I/P of BPF.

$$v_i = v_3 - v_4$$

$$= i_1 R - i_2 R = R[i_1 - i_2]$$

Substituting the value of i_1 & i_2

$$v_i = R[2ax(t) + 4bx(t)\cos\omega_c t]$$

$$= 2R[ax(t) + 2bx(t)\cos\omega_c t$$

The BPF is centered at $\pm\omega_c$, it will pass a NB frequencies centered at $\pm\omega_c$ with a small Bandwidth of $2\omega_m$ to preserve the sidebands.

O/P of BPF is given as -

$$v_o = 4bRx(t)\cos\omega_c t$$

$$= Kx(t)\cos\omega_c t$$

which is the expression of DSB-SC

Q2

Q6) An Audio Signal given as $15 \sin 2\pi (1500\,t)$ Amp. modulates a carrier given as $60 \sin 2\pi (100{,}000\,t)$

Determine the following

a) Sketch the audio signal.

b) Sketch the carrier signal.

c) Construct the modulated wave.

d) Determine the modulation index & % modulation.

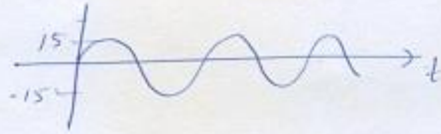
e) What are the frequencies of audio signal & carrier.

f) What frequencies would present in a spectrum analysis of the modulated wave?

Sol $x(t)$ = Audio signal $= 15 \sin 2\pi (1500\,t)$

Carrier $= 60 \sin 2\pi (100{,}000\,t)$

a)

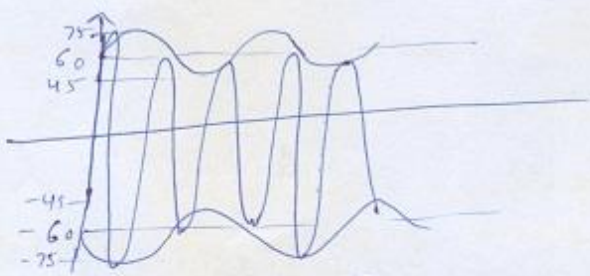


b)

d) $m_a = \frac{15}{60} = .25$

$= 25\%$

c)



e) $v_m = V_m \sin 2\pi f_m t$

$f_m = 1500$

$f_c = 100{,}000\ Hz$

f) $f_c = 100{,}000\ Hz$

$f_c + f_m = 100{,}000 + 1500 = 101{,}500\ Hz$

$f_c - f_m = 98{,}500\ Hz$

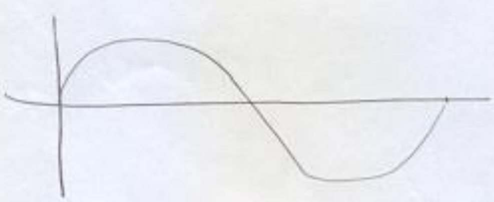
} freq. of Modulated wave

Q.3 Explain Generation & demodulation of PAM.?

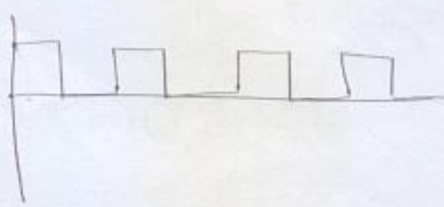
Soln. Amplitude of a regularly spaced rectangular pulses vary acc. to the instantaneous value of the modulating or message signal. Pulse in PAM may be flat top type or natural type.

Flat top PAM is widely used because during the x-mission the noise interferes with the top of the x-mitted pulse & this noise can be easily removed if the PAM has flat top.

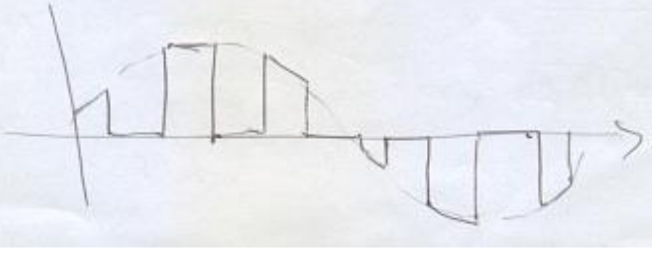
In case of natural samples, when these pulses are recd. at the receiver, it is always contaminated by noise. It becomes difficult to determine the shape of the top of the pulse & thus amplitude detection is not exact.



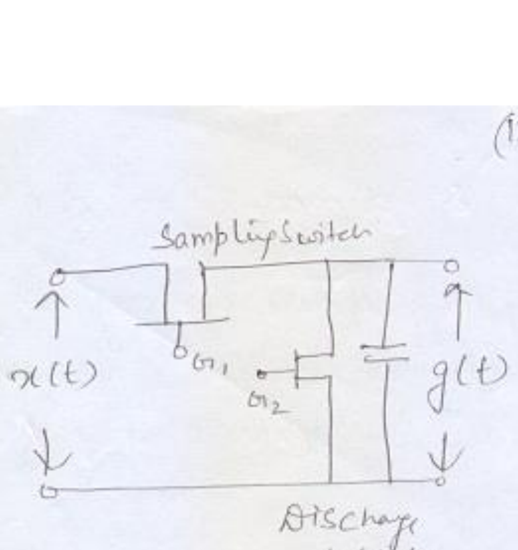
message signal



carrier pulse



PAM



Mathematical Analysis :-

Train of impulses may be represented as-

$$\delta_{T_S}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

$$s(t) = x(t) \cdot \delta_{T_S}(t)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s).$$

Now $g(t) = s(t) \circledast h(t)$

$$g(t) = \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau.$$

$$= \int_{-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s)\, h(t-\tau)\, d\tau.$$

Acc. to shifting property

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t-nT_s)$$

$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)\, H(f)$$

Demodulation of PAM :-

Recd. Signal $\rightarrow$ Holding ckt $\rightarrow$ LPF $\rightarrow$ demodulated Signal.

From the PAM, demod. is done using a holding ckt. & then passing thr. LPF.

Q4: The Joint Prob. func. of two random variables X & Y is given by –

$$f(x,y) = \begin{cases} c(x^2+2y) & x = 0,1,2,\ y = 1,2,3,4 \\ 0 & \text{otherwise} \end{cases}$$

Find (a) The value of c (b) $P(X=2, Y=3)$
c) $P(X \le 1, Y > 2)$ (d) Marginal Prob. func. of X & Y.

Soln

x \ y	1	2	3	4	
0	2c	4c	6c	8c	20c
1	3c	5c	7c	9c	24c
2	6c	8c	10c	12c	36c
	11c	17c	23c	29c	80c

80c must be 1

so. $80c = 1$, $c = \frac{1}{80}$

b) $P(X=2, Y=3) = 10c \Rightarrow 10 \times \frac{1}{80} = \frac{1}{8}$

c) $P(X \le 1, Y > 2) = (6c + 8c + 7c + 9c) = 30c \Rightarrow 30 \times \frac{1}{80} = \frac{3}{8}$

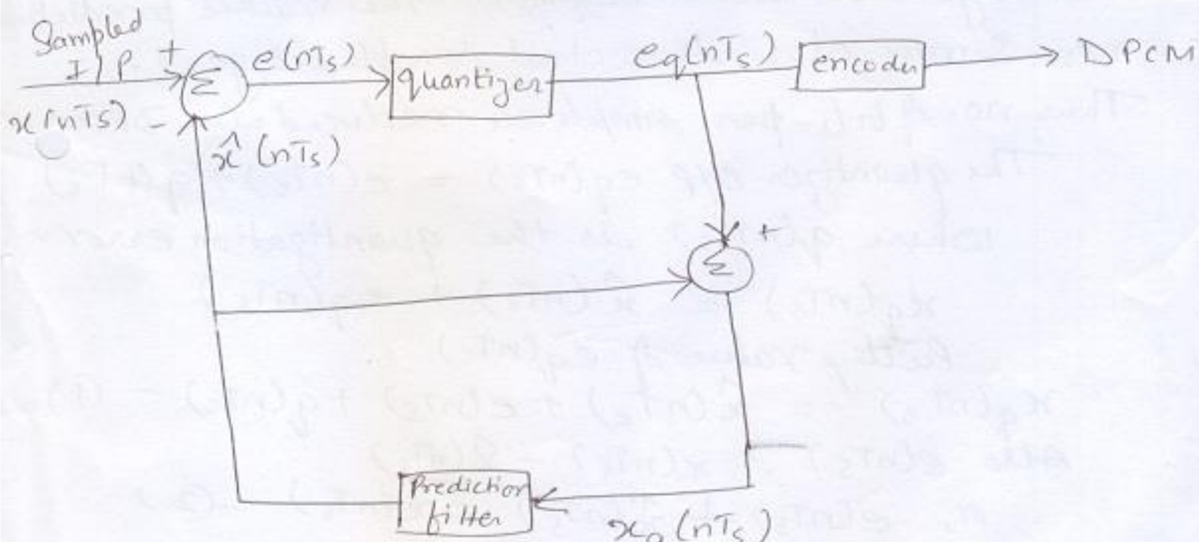
d) $$f_1(x) = P(X = x) = \begin{cases} 20c = \frac{1}{4} & x = 0 \\ 24c = \frac{3}{10} & x = 1 \\ 36c = \frac{9}{20} & x = 2 \end{cases}$$

$$f_2(y) = P(Y = y) = \begin{cases} 11c = \frac{11}{80} & y = 1 \\ 17c = \frac{17}{80} & y = 2 \\ 23c = \frac{23}{80} & y = 3 \\ 29c = \frac{29}{80} & y = 4 \end{cases}$$

(14)

Q.5 Explain the Generation of DPCM?

Soln. The samples of signal are highly correlated with each other. Due to this any signal does not change so fast. This means its value from the present sample to the next sample does not differ by large amount. When these samples are encoded by a PCM system, the resulting encoded signal contains some redundant information. If this redundancy is reduced then the overall bit rate will decrease & no. of bits reqd. to transmit one sample will also be reduced. This scheme is known as DPCM.



The DPCM works on the principle of Prediction ∴ The value of the present sample is predicted from the past sample. The prediction may not be exact but it is very close to the actual sample. The sampled signal is denoted by $x(nT_s)$ and predicted signal is denoted by $\hat{x}(nT_s)$. The Comparator finds out the difference b/w the actual sample value $x(nT_s)$ & predicted sample value $\hat{x}(nT_s)$. This is known as prediction error denoted by $e(nT_s)$.

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

This error is difference b/w two signals. The predicted value is produced by prediction filter. This signal is called $(e_q(nT_s))$. This makes prediction more & more close to actual sampled signal. Thus no. of bits per sample are reduced in DPCM.

The quantizer O/P $e_q(nT_s) = e(nT_s) + q(nT_s)$

where $q(nT_s)$ is the quantization error.

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

Putting value of $e_q(nT_s)$

$$\therefore x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s) \quad -(1)$$

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

$$\text{or, } e(nT_s) + \hat{x}(nT_s) = x(nT_s) \quad -(2)$$

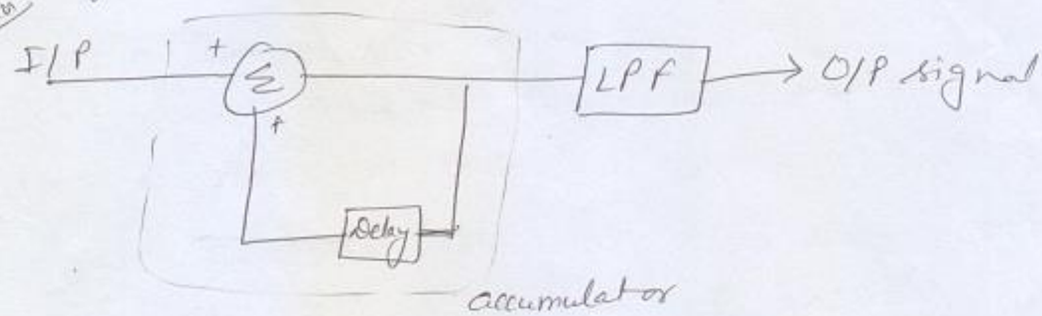
Substituting 2 in eqn(1)

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

∴ quantized version of the signal is the sum of original sample value & quantization error.

Q6 Explain the Receiver, advantages & disadvantages of Delta Modulation?

Soln:



At the Receiver, accumulator & LPF are used. This accumulator generates the Stair-case approximated signal and is delayed by one sampling period to get previous O/P. It is then added to the I/P signal. If I/P is binary 1, then it adds +Δ step to the previous O/P. If I/P is binary 0, the one step Δ is subtracted from delayed signal. LPF Smoothens the staircase signal to reconstruct the original message signal.

Advantages:

1. It transmits only one bit per sample so signalling rate & transmission channel bandwidth is small.

Disadvantages

1. Slope overload Distortion :-

This distortion arises due to large dynamic range of the I/P signal. i.e. I/P signal is so high that the staircase cannot approximate it. Hence there is large error b/w staircase & original signal called slope overload distortion.

To reduce this distortion, step size must be increased when slope of signal is high.

2. Granular Noise :-

It occurs when the step size is too large as compared to small variations in the input signal. To overcome this problem, step size should be made smaller.

Q7. Explain <u>MATCHED FILTER</u> & Properties of Matched filter

<u>Soln</u>: It is a linear, time invariant filter designed to provide the maximum S/N ratio at its o/p for a given x-mitted symbol waveform.

Let the I/P x(t) to the filter consists of signal s(t) corrupted by additive noise n(t)

i.e. $x(t) = s(t) + n(t) \quad 0 \le t \le T$

where T is arbitrary observation interval.

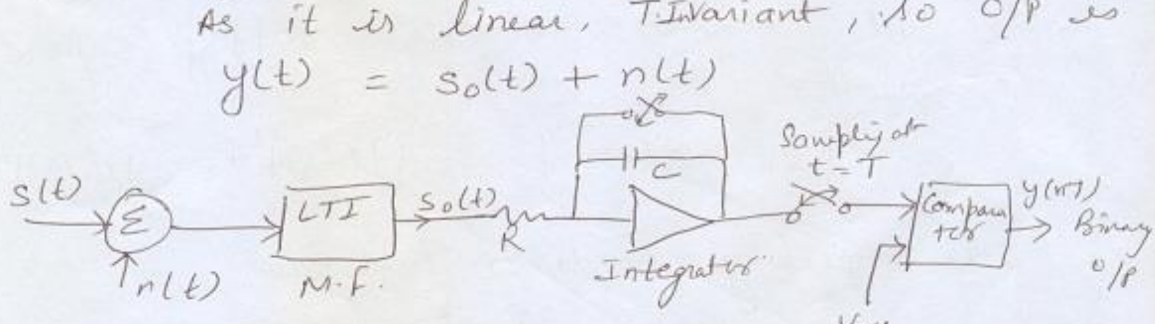
- The noise n(t) is additive white gaussian noise whose mean is zero & noise spectral density is $\frac{N_0}{2}$. The func. of the receiver is to detect the signal s(t) in an optimum manner given the recd. signal x(t).

The matched filter can be implemented as an integrator & dump correlation receiver.

- <u>Matched filter as an optimum Receiver</u> :-

As it is linear, T.Invariant, so O/P is

$y(t) = s_0(t) + n(t)$



where $s_0(t)$ & n(t) are produced by signal & noise component of I/P s(t).

It is reqd. to maximize the signal to noise power ratio i.e. $\eta = \frac{|s_0(t)|^2}{E[n^2(t)]}$ → inst. power at o/p ↳ avg. o/p noise power

$S(f) \rightarrow$ F.T. of $S(t)$
$H(f) \rightarrow$ " " $h(t)$ i.e. x-fer func. of filter.

$S_o(t) \rightarrow$ Inverse F.T. of $S_o(f)$ (o/p)

i.e. $S_o(t) = \int_{-\infty}^{\infty} H(f) S(f) e^{j2\pi ft} df$

Sampled at $t = T$

$$|S_o(T)|^2 = \left| \int_{-\infty}^{\infty} H(f) S(f) e^{j2\pi fT} df \right|^2$$

The power spectral density of o/p noise $n(t)$ is given by –

$$S_N(f) = \frac{N_0}{2} |H(f)|^2$$

The avg. power of o/p noise –

$$E[n^2(t)] = \int_{-\infty}^{\infty} S_N(f) df = \frac{N_0}{2} \int_{-\infty}^{\infty} |H(f)|^2 df$$

The resulting signal to noise ratio is –

$$\eta = \frac{|S_o(t)|^2}{E[n^2(t)]} = \frac{\left| \int_{-\infty}^{\infty} H(f) S(f) e^{j2\pi fT} df \right|^2}{\frac{N_0}{2} \int_{-\infty}^{\infty} |H(f)|^2 df}$$

To get the max. ratio, we apply Schwarz's inequality i.e.

$$\left| \int_{-\infty}^{\infty} \phi_1(x) \phi_2(x) dx \right|^2 \leq \int_{-\infty}^{\infty} |\phi_1(x)|^2 dx \int_{-\infty}^{\infty} |\phi_2(x)|^2 dx$$

The equality condition holds when

$$\phi_1(x) = K \phi_2^*(x)$$

$$\therefore \left| \int_{-\infty}^{\infty} H(f) S(f) e^{j2\pi fT} df \right|^2 \leq \int_{-\infty}^{\infty} |H(f)|^2 df \int_{-\infty}^{\infty} |S(f)|^2 df$$

or, $\eta = \frac{\cancel{\int_{-\infty}^{\infty} |H(f)|^2 df} \int_{-\infty}^{\infty} |S(f)|^2 df}{\frac{N_0}{2} \cancel{\int_{-\infty}^{\infty} |H(f)|^2 df}} = \frac{2}{N_0} \int_{-\infty}^{\infty} |S(f)|^2 df$

∴ η will be max.

$$\eta_{max} = \frac{2}{N_0}\int_{-\infty}^{\infty}|S(f)|^2 df$$

T.F. of filter is also at the optimum value.

$$\therefore \quad H_{opt}(f) = K S^*(f)\, e^{-j2\pi fT}$$

$K \Rightarrow$ scaling factor

$e^{-j2\pi fT} \rightarrow$ const. delay.

In the time domain.

$$h_{opt}(t) = K\int_{-\infty}^{\infty} S^*(f)\, e^{-j2\pi fT}\, e^{j2\pi ft}\, df$$

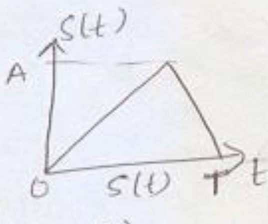
$$= K\int_{-\infty}^{\infty} S^*(f)\, e^{-j2\pi f(T-t)}\, df$$

By conjugate property —

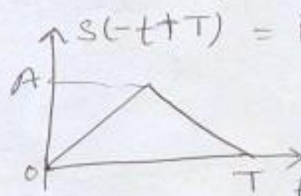
$$S^*(f) = \int_{-\infty}^{\infty} s(t)\, e^{j2\pi ft}\, dt = S(-f)$$

$$\therefore \quad h_{opt}(t) = \int_{-\infty}^{\infty} S(-f)\, e^{-j2\pi ft\,(T-t)}\, dt = s(T-t)$$

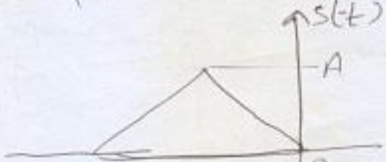
∴ Impulse response of matched filter is the time reversed & delayed version of I/P signal.



(a)



c Matched filter response



(b)

All shows that
$h(t) = 0$ for $t < 0$
so it is causal system

~~What are~~ The Properties of Matched Filter ?

(21)

Properties of Matched Filter :-

we know $h(t) = s(T-t)$

In freq. domain

$$H(f) = S^*(f) e^{-j2\pi fT}$$

Based on these two relations, the properties of Matched filter are –

I. The spectrum of O/P signal of M.F. with I/P signal is proportional to Energy spectral density of I/P signal except for a time delay factor

$$S_o(f) = S(f) H(f) = S(f) \cdot S^*(f) e^{-j2\pi fT}$$

$$= \underbrace{|S(f)|^2}_{\text{Energy spectral density of I/P}} e^{-j2\pi fT}$$

2. The O/P signal of M.F. is proportional to shifted version of auto correlation func. of I/P to which the filter is matched.

we know ACF & ESD forms a fourier pair

$$s_o(t) = \underbrace{R_s(t-T)}_{\text{ACF of I/P } s(t)}$$

3. The O/P signal to noise ratio $(SNR)_o$ of M.F. depends on the ratio of signal Energy to power spectral density of white noise at the I/P.

$$|SNR_o|_{max} = \frac{\int_{-\infty}^{\infty} |S(f)|^2 df}{N_o/2}$$

from Rayleigh Energy theorem

$$E = \int_{-\infty}^{\infty} |S(f)|^2 df$$

$$\therefore |(SNR)_o|_{max} = \frac{E}{N_0/2} = \frac{2E}{N_0}.$$

4. we know

$$S_o(f) = \int_{-\infty}^{\infty} H(f) S(f) e^{j2\pi ft} df$$

for a M.F.

$$H(f) = S(-f) e^{-j2\pi ft}$$

So, $S_o(f) = \int_{-\infty}^{\infty} S(-f) S(f) e^{-j2\pi ft} e^{j2\pi ft} df.$

$$= \int_{-\infty}^{\infty} |S(f)|^2 df) = E \text{ (energy of the signal)}$$

5. Signal to Noise power at the o/p is given as -

$$(SNR)_o = n = \frac{|S_o(t)|^2}{E[n^2(t)]}$$

$$E[n^2(t)] = \frac{|S_o(t)|^2}{n} = \frac{E^2}{E/N_0/2} = \frac{EN_0}{2}$$

Q8. A DMS has an alphabet of five symbols with probabilities for its output. "Compute Huffman's code. (ii) Determine average length of the code word & Entropy.

Soln:

Prob.	Symbols	Stage I	Stage II	Stage III	Stage IV	Code word
.4	S_0	.4	.4	.4	.6	00
.2	S_1	.2	.2	.4	.4	10
.2	S_2	.2	.2	.2		11
.1	S_3	.1	.2			010
.1	S_4	.1				011

$$\bar{L} = \sum_{K=0}^{K-1} p_K l_K = .4\times .2 + .2\times 2 + .2\times 2 + .1\times 3 + .1\times 3 = 2.2$$

(average Length)

$$H(\mathcal{S}) = \sum_{K=0}^{K-1} p_K\left(\log_2 \frac{1}{p_K}\right) = .4\log_2\frac{1}{.4} + .2\log_2\frac{1}{.2} + .2\log_2\frac{1}{.2} + .1\log_2\frac{1}{.1} + .1\log_2\frac{1}{.1}$$

(Entropy)

$$= 2.12193 \text{ bits/symbol}$$

Q. 9(a) What is kraft inequality?

Soln. Let X be a DMS with alphabet x_i $i = 1, 2, 3 - m$. Assume that the length of the assigned binary code word corresponding to x_i is n_i.

A necessary & sufficient condition for the existence of an instantaneous binary code is -

$$K = \sum_{i=1}^{m} 2^{-n_i} \leq 1$$

Which is known as kraft inequality.

Q. 9(b) Encode the Sequence 000101110010100101 by Lemple Ziv Algorithm. 0 & 1 already exist in dictionary.

Soln.

Numerical Position	Dictionary Location	Sub Sequence	Num. Representation	Binary encoded block
1	0001	0	1	0000
2	0010	1	2	0001
3	0011	00	11	0010
4	0100	01	12	0011
5	0101	011	42	1001
6	0110	10	21	0100
7	0111	010	41	1000
8	1000	100	61	1100
9	1001	101	62	110