

1. Introduction of digital communication

Ques: Block diagram of Digital communication system

Ans :-

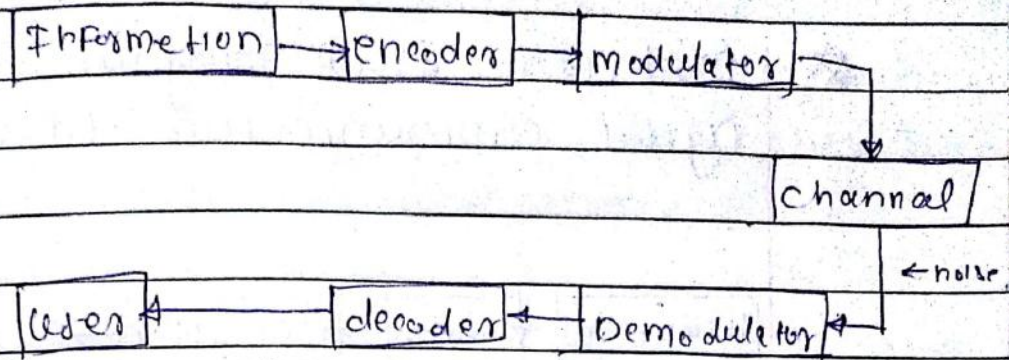


Fig. Block diagram of Dc.

① The above Block diagram of digitel communi. system. It is consist of xⁿ section and the Resliver section.

② In the transmitter section First we can give the signal from the source to the encoder.

③ The signal which will be encoded by the encoder and signal is given to the modulator to modulate the signal.

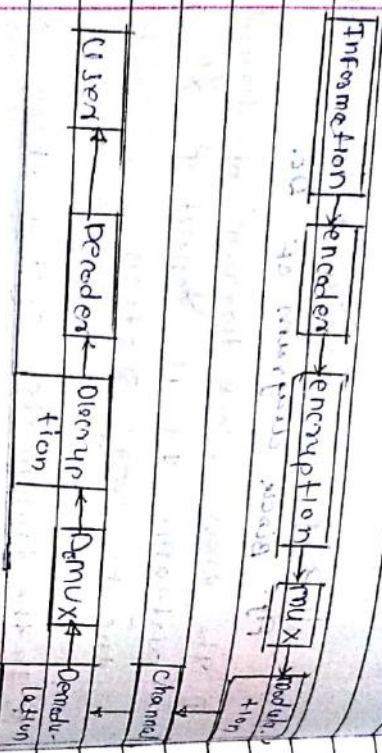
④ The modulated signal will be transmitted with the help of channel.

Resliver section :-

In the Resliver section the signal will be demodulated with the help of demodulation process.

② The the signal will be given to the decoder to decode the signal to the # get the signal user successfully

★ Modified Block diagram of Digital communication system:-



① The above modified diagram the additional block diagram can be added in digital communication

Encryption :-

Encryption prevent unknown user from understanding the message and from injecting false message into the system.

MUX :- It is use for combining signal from different source so that they share the portion of the communication system.

Demux :- Demux is used for separating the signal so that they reach there respective destination

Description :- It does the reverse operation of the encoder.

★ Synchronization:-

It will be used to both the time and frequency need to synchronize there frequency reference with carrier in both frequency and phase.

★ Advantages of DE :-

- ① Interference is less
- ② Regenerative Repeater can be used at fixed distance along the link
- ③ The Hardware encipherment is more flexible than analog Hardware.

* Disadvantages :-

- (i) Long Bandwidth will be Required
- (ii) Synchronization is Required

* Review of Random Variable :-

- (1) A sample space consist of all the possible outcome of an experiment.

Eg:- The instantaneous value of a noise voltage on the no. of error in a digital message.

- (2) These term is used to simplify a rule by which a real no. is assigned to each possible outcome. let the possible outcome to ~~there of~~ an experiment is represented by $\{s\}$ then the Random variable is represented by $X(s)$

There are two types of Random variable.

- (1) Discrete Random Variable
- (2) Continuous Random Variable.

(1) Discrete Random Variable

(1) It is associated with an experiment or relationship $X(s)$ that a sign the real no. x . From every sample point $s \in F$.

- (2) If the sample point s contains a countable no. of sample. sample point then $X(s)$ is a discrete random variable.

Eg:- Rolling of die $\{1, 2, 3, 4, 5, 6\}$
(2) Tossing of coin $\{H, T\}$

(2) Continuous Random Variable

- (1) These term denote that those variable has an uncountable no. of possible value.
- (2) Such that there system generate a infinite no. of output within a finite period of time.

Eg:- Noise voltage generated in electronic amplification.

ADPCE

* PDF and CDF and their properties

* Probability Distribution function :-

1.17 (1) Commutative Distribution Function
(CDF) :-

CDF is a random variable is defined as the probability that the random variable X takes value less than or equals to x . $X \leq x$.

i.e. CDF :

$$CDF : F_X(x) = P(X \leq x)$$

Properties :-

- (1) The CDF is always bounded between 0 and 1
- (2) This property says that

$$F_X(\infty) = 1$$

this included probability of the all possible outcome.

- (3) There property state that random variable can not have any value less or equal to $(-\infty)$

- (4) $F_X(x_1) \leq F_X(x_2)$ if $x_1 < x_2$, these property indicate that monotone non decreasing function of x .

(2) PDF :- Probability Density function :-

- (1) PDF is more convenient way to describe the continuous random variable. It is defined as the derivative of CDF.

$$PDF : f_X(x) = \frac{d}{dx} F_X(x)$$

* Properties :-

- (1) The CDF can be derived from the PDF by integration of PDF

$$F_X(x) = \int_{-\infty}^x f_X(x) dx$$

i.e. $CDF = \int_{-\infty}^x (PDF) dx$

- (2) PDF is a non negative function for all the values of x .

$$f_X(x) \geq 0 \text{ for all value of } x$$

- (3) The PDF is increasing monotonically. So the derivative of CDF is always positive.

(4) Area under PDF curve is unity

$$\int_{-\infty}^{\infty} f_X(x) dx = 1$$

* Probability of error

- ① Probability of error due to Noise
- ② Probability of error due to ISI

① POE Due to Noise :-

Since the match filter is the optimum detector of a known pulse in additive noise. Then the probability of error can be calculated due to Noise.

The received signal in additive noise is

$$x(t) = \begin{cases} +A + w(t) & \text{Symbol 1 is sent} \\ -A + w(t) & \text{Symbol 0 is sent} \end{cases}$$

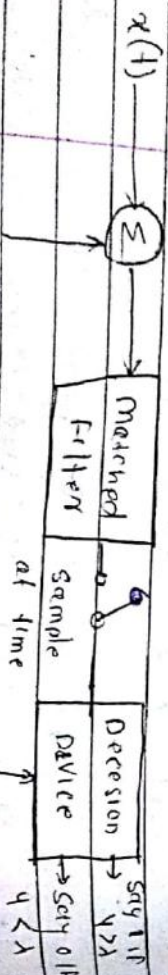


Fig. POE Due to Noise

① Let us assume the receiver starts with starting and receiving signal.

② The receiver receives sample at the output of match filter at a time $t = T_b$. It will decide whether the transmitted symbol is 0 or 1.

③ The decision device uses compared the sample value and threshold value.

④ If the sample signal is above A , the receiver assumed that the transmitted symbol is 1.

⑤ If the sample signal is below A then the receiver assume that the transmitted symbol is 0.

⑥ If sample signal is equal to the threshold value A the receiver gets that the outcome does not alter the average probability of error.

② Intersymbol Interference (ISI)

① ISI happens when the channel is a frequency dependant amplitude spectrum.

② The signal is band limited channel.



In the telecommunication ISI for the distortion of signal in which one symbol interferes with subsequent symbol and spreading of the pulse beyond pulse then it will call the spreading of interference with spreading pulse

Numerical:

$$f(x) = \begin{cases} 0 & \text{for } x < 0 \\ 4x^2 & \text{for } 0 \leq x < 10 \\ 100 & \text{for } x > 10 \end{cases}$$

① calculate the value k .

② plot corresponding PDF

Ans we have

$$f(x) = 100k \quad (x > 10)$$

$$f(x) = 1$$

$$100k = 1$$

$$k = \frac{1}{100}$$

For

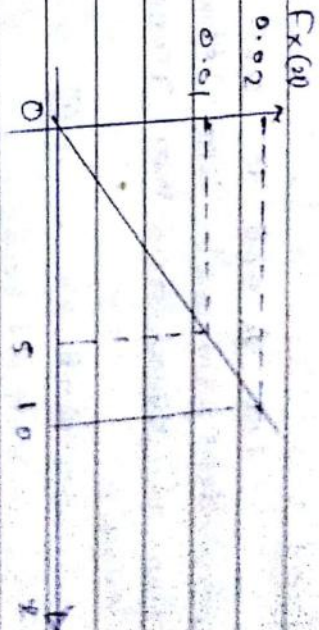
$$f(x) = \frac{d}{dx} (F(x))$$

$$= \frac{d}{dx} [F(x)] \text{ for } 0 < x < 10$$

$$= \frac{d}{dx} [4x^2]$$

$$= 2 \times 4x$$

$$f(x) = 8x$$



* Central limit theory

Statement:-

To obtain the PDF of several independent variable central limit theorem is used.

the PDF

Under certain condition sum of a large no of independent variable tends to approach gaussian PDF.

Proof:-

Let A and B are two independent variable and their addition is equal another random variable

$$C = A + B \quad \dots \quad (1)$$

The probability that $C \leq c$.

C is denoted by $P(C \leq c)$ and it is given by.

$$F_C(c) = P(C \leq c)$$

$$= P(A + B \leq c)$$

A/c to the definition of CDF we have.

$$P(C \leq c) = F_C(c)$$

$$\text{Hence } F_C(c) = P(A + B \leq c)$$

Therefore event $C \leq c$ joint event

$$F_C(c) = P(C \leq c) = P[A \leq \infty, B \leq (c-A)]$$

The joint CDF can be express in terms of joint PDF as under

$$P(a_1 < A < a_2, b_1 < B < b_2) =$$

$$\int_{a_1}^{a_2} \int_{b_1}^{b_2} F_{AB}(a, b) da db$$

$$F_C(c) = \int_{-\infty}^{\infty} \int_{-\infty}^{c-A} F_{AB}(a, c-A) da db$$

there is the expression for CDF of random variable.

of c

To obtain the PDF we have to differentiate of equation of CDF. we have.

$$f_C(c) = \frac{d}{dc} F_C(c)$$

$$= \frac{d}{dc} \int_{-\infty}^{\infty} \int_{-\infty}^{c-A} F_{AB}(a, c-A) da db$$

$$= \int_{-\infty}^{\infty} F_{AB}(a, c-A) da \cdot \frac{d}{dc} \int_{-\infty}^{c-A} da db$$

$$f_c(c) = \int_{-\infty}^{\infty} f_{AB}(a, c-a) da$$

A and B are independent random variable then the joint PDF is $f_{AB}(a,b)$

$$= f_{AB}(a,b) = f_A(a) * f_B(b)$$

$$= f_A(a) * f_B(c-a)$$

$$f_c(c) = \int_{-\infty}^{\infty} f_A(a) * f_B(c-a) da$$

These is the equn PDF

* MATCH FILTER :-

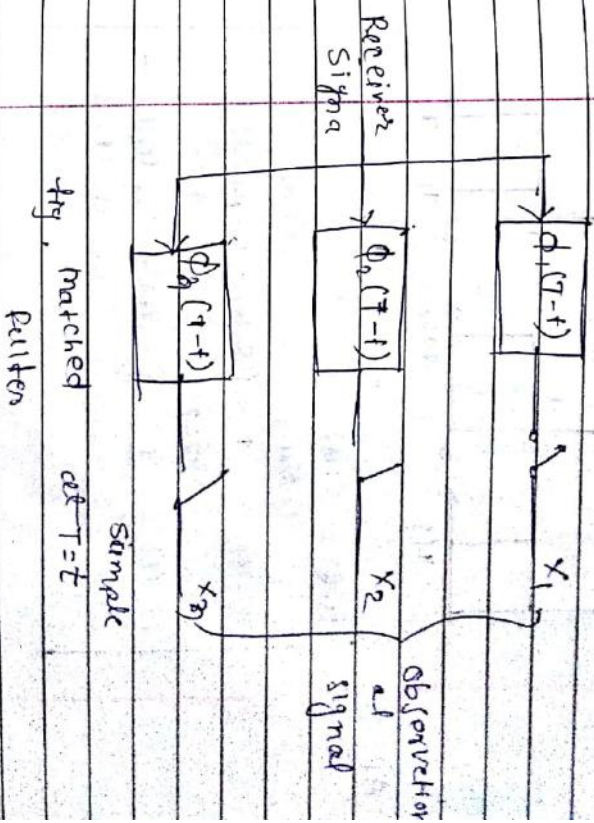
- ① In binary communication we can transmit the signal in form of '1' and '0'. respectively
 1 Represent present of pulse and
 0 Represent absent of pulse.

- ② It will decide when the pulse is present or absent with a reduce error probability is

Match Filter.

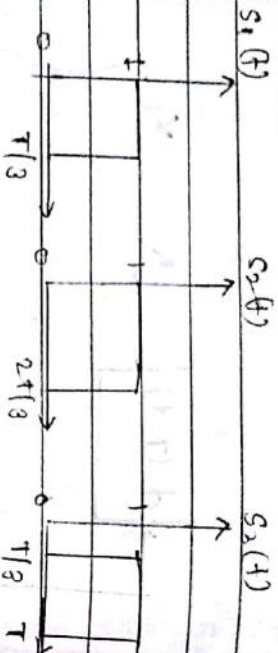
- ② The match filter is increasing signal components and decreasing the noise components at a given time

Q. So that signal to noise ratio will be increases at the output.



or Gram Smith orthogonalization:

Ques. Consider the given signal $s_1(t), s_2(t)$ as shown in Fig. Find orthonormal basis signals for the set of signal using Gram Smith orthogonalization.



Ans:-

$$u_1(t) = s_1(t)$$

S_{11}

$$u_2(t) = \frac{1}{S_{22}} (s_2(t) - S_{21}u_1(t))$$

S_{22}

$$u_3(t) = \frac{1}{S_{33}} (s_3(t) - S_{31}u_1(t) - S_{32}u_2(t))$$

$$S_{11}(t) = 1 \text{ given}$$

$$S_{11} = \left[\int_0^T s_1^2(t) dt \right]^{1/2}$$

$$S_{11} = \int_0^{T/3} (1)^2 dt \int^{1/2}$$

$$= \left[t \right]_0^{T/3} \int^{1/2}$$

$$= \left[\frac{T}{3} \right]^{1/2}$$

$$S_{11} = \sqrt{\frac{T}{3}}$$

$$\therefore u_1(t) = \frac{s_1(t)}{S_{11}}$$

$$= 1$$

$$\int_0^T \frac{1}{3}$$

$$u_1(t) = \sqrt{\frac{3}{T}}$$

$$\therefore S_{22}(t) = 1 \text{ given}$$

$$S_{22} = \left[\int_0^T s_2^2(t) dt \right]^{1/2}$$

$$S_{22} = \int_0^{2T/3} (1) \left[\frac{3}{T} \right]^{1/2} dt$$

$$= \left[\frac{3}{T} \right]^{1/2} \times \int_0^{2T/3} dt$$

$$= \left[\frac{3}{T} \right]^{1/2} \times \left(\frac{2T}{3} \right)$$

$$= \frac{2T}{3} \times \left[\frac{3}{T} \right]$$

$$S_{22} = \int_0^{2T/3} \int_0^{2T/3} [S_2(t) - S_{21}(u_1(t))]^2 dt$$

$$= \int_0^{2T/3} \int_0^{2T/3} \left[1 - \frac{2T}{3} \times \left[\frac{3}{T} \times \left[\frac{3}{T} \right] \right] \right]^2 dt$$

$$= \int_0^{2T/3} \left[1 - \frac{2T}{3} \times \frac{3}{T} \right]^2 dt$$

$$= \int_0^{2T/3} (-1)^2 dt \int_0^{2T/3}$$

$$S_{22} = \left[\frac{2T}{3} \right]$$

$$u_2(t) = \left[\frac{3}{2T} \right] \quad 1 - \frac{2T}{3} \times \left[\frac{3}{T} \times \left[\frac{3}{T} \right] \right]$$

$$= \left[\frac{3}{2T} \right] \quad 1 - 2$$

$$u_2(t) = - \left[\frac{3}{2T} \right]$$

$$u_3(t) = \frac{1}{3} \{ S_3(t) - S_{31}(u_1(t)) - S_{32}(u_2(t)) \}$$

$$S_{31} = \int_0^{2T/3} S_3(t) u_1(t) dt$$

$$= \int_0^{2T/3} \int_0^{2T/3} 1 \times \left[\frac{3}{T} \right] dt$$

$$= \left[\frac{3}{T} \right] \int_0^{2T/3} \int_0^{2T/3} (1) dt$$

$$= \left[\frac{3}{T} \right] \int_0^{2T/3} \left[t - \frac{t}{3} \right] dt$$

$$= \left[\frac{3}{T} \right] \left[\frac{2T}{3} \right]$$

$$= \left[\frac{3}{T} \times \frac{2T}{3} \right]$$

$$S_{32} = \int_0^{2T/3} S_3(t) u_2(t) dt$$

$$= \int_0^{2T/3} (1) \times \left[\frac{3}{2T} \right] dt$$

$$S_{32} = \left(- \left[\frac{3}{2T} \right] \times \left[t - \frac{t}{3} \right] \right) = - \left[\frac{3}{2T} \times \frac{2T}{3} \right]$$

$$S_{33} = \int_{\pi/3}^T \left(S_3(t) - S_{31} u_1(t) - S_{32} u_2(t) \right)$$

$$= \int_{\pi/3}^T \left[1 - \frac{3}{T} \times \frac{2t}{3} \times \frac{3}{t} + \frac{3}{2T} \times \frac{2t}{3} \right]$$

$$= \int_{\pi/3}^T \left[1 - \frac{2}{T} \times \frac{2t}{3} + \frac{3}{2T} \times \frac{2t}{3} \right] dt$$

$$= \left[\int_{\pi/3}^T (1 - 2 + 1)^2 dt \right]^{1/2}$$

$$= \left[\int_{\pi/3}^T [-2]^2 dt \right]^{1/2}$$

$$= \left[4 \left[T - \frac{t}{3} \right] \right]^{1/2}$$

$$= \sqrt{\frac{8T}{3}}$$

$$u_2(t) = \frac{-2}{\sqrt{\frac{8T}{3}}}$$

$$u_2(t) = -2 \times \frac{3}{8T} = -\frac{3}{2T}$$

$$G_4(t) = \frac{-8/3}{\frac{8}{3} \sqrt{\frac{2\pi}{3}}}$$

$$G_4(t) = \begin{bmatrix} 3 \\ 2\pi \end{bmatrix}$$

Que *

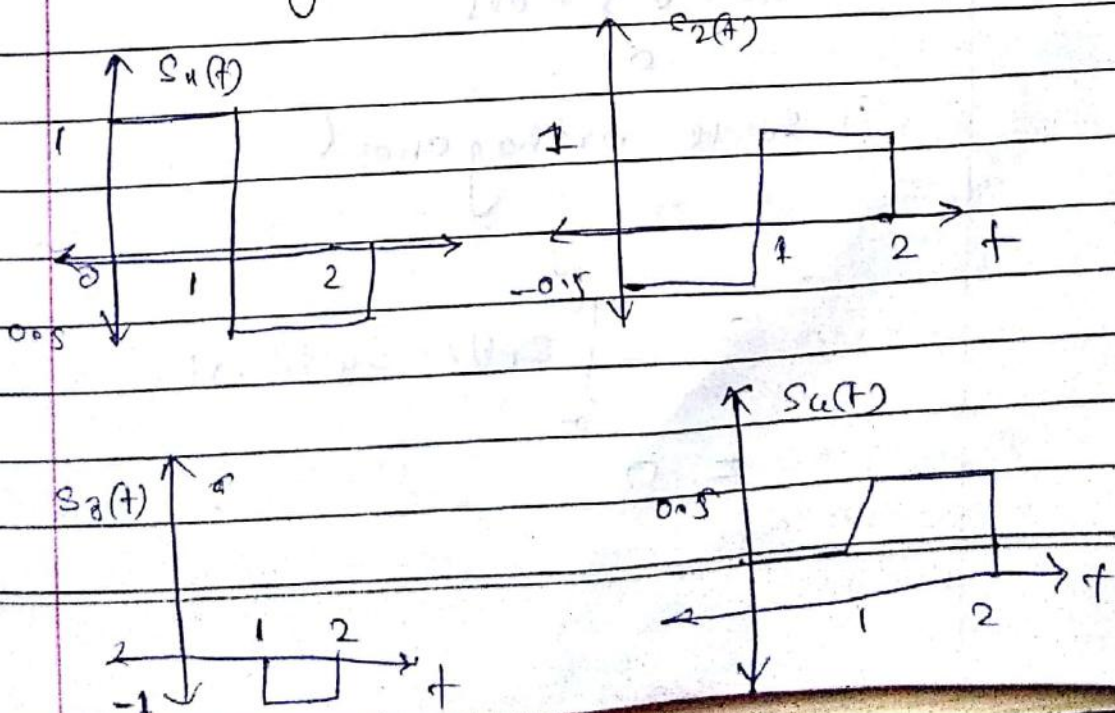
Signal space concept :

numerical..

5.10.2020

Que

A signal space consist of three or four signal $s_1(t)$, $s_2(t)$, $s_3(t)$, $s_4(t)$ as shown in Fig. determine suitable setup, basic factor and the dimensionality of the signal. Represent the in geometrical in vector space.



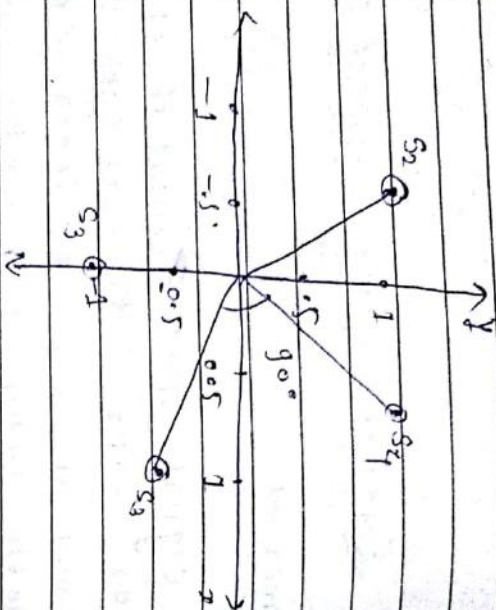
Ans :-

$$s_1(t) = (1, 0.5)$$

$$s_2(t) = (0.5, 1)$$

$$s_3(t) = (0, -1)$$

$$s_4(t) = (0.5, 2)$$



$$s_1 - s_4$$

$$= 0.5 - 0.5$$

$$= 0$$

$s_1 s_4$ is orthogonal.

$$= \int_{-\infty}^{\infty} s_1(t) \cdot s_4(t) dt$$

$$= 0$$

que :- Correlation Receiver :-

Ans :-

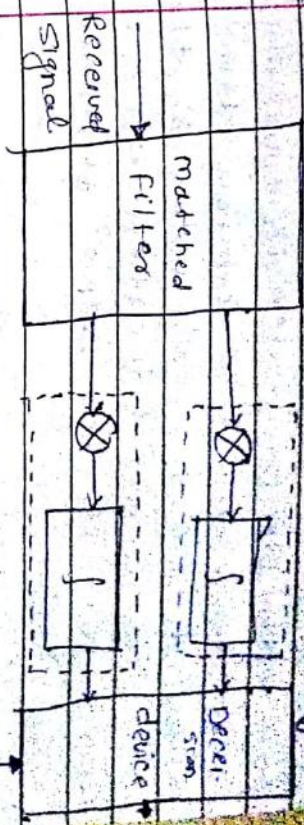


Fig :- Correlation Receiver.

Threshold value (X)

In the above diagram of the Correlation receiver is consist of match filter division device and the correlator.

First the signal will be x_n through the match filter. The match filter is used to match the signal which will be required for the system and given to the correlator. The correlator is combination of multiplier and integrator.

The signal will be multiply and integrate given to the decision device.

and the another input of decision device. threshold value that is λ

the decision device is working as a comparator. It is comparing the ^{signal} to the threshold value (λ). then the output of the decision device will be set with the help of threshold value.

Q. Limit : Source and coding method.

Source: cooling method :-

① Source coding method for discrete

② however in many practical situations there are constraints that favour coding in perfect residualant.

ex. The constraint imposed in a comm. channel at may place upper limits on permissible code

③ The information source may have a continuous amplitude but such a continuous signal. The information may quantize the amplitude and generate the source code.

④ The branch that deals with the rate of distortion.

② when the average code word is less than code entry then the rate of distortion. it will find in application that is 2 nature-

① Information $\propto$ that higher rate of channel capacitance

② source coding where permitted coding. x -Baud can not exactly represent the the information which are sent the the information which are data compression.

* Huffman coding

① Important formula

② Entropy of source :-

$$① H(X) = \sum_{i=1}^m P(x_i) \log_2 P(x_i) \text{ bit/symbol}$$

③ Average code word length :-

$$L = \sum_{i=1}^m P(x_i) n_i$$

④ Code efficiency η :-

$$\eta = \frac{H(X)}{L}$$

⑤ Code redundancy

$$r = 1 - \eta$$

Ex 17

①

Consider a DMS with alphabets s_1, s_2, s_3 with probability 0.45, 0.35, 0.20 respectively.

② Find the average code word length of Huffman code word

③ Let the source be extended of order 2 applying the Huffman algorithm. the average code word length of new code and its entropy

④ Compare the average code word length calculate in part B with entropy of original source.

Ans :- Step 1 :-

Letter	Probability
s_1	0.45
s_2	0.35
s_3	0.20

Step 2 Letter prob code length

s_1	0.45	1
s_2	0.35	0
s_3	0.20	0 1

Step 3 :-

Entropy of source :-

$$H(X) = - \sum_{i=1}^n P(x_i) \log_2 \left(\frac{1}{P(x_i)} \right)$$

$$= 0.45 \log_2 \left(\frac{1}{0.45} \right) + 0.35 \log_2 \left(\frac{1}{0.35} \right) + 0.20 \log_2 \left(\frac{1}{0.20} \right)$$

$$H(X) = 1.51 \text{ byte/symbol}$$

Step ⑤:

Average code word length

$$L = \sum_{i=1}^M P(x_i) n_i$$

• n_i = code word length

$$= 0.45 \times 1 + 0.35 \times 2 + 0.20 \times 2$$

$$L = 1.55 \text{ bit}$$

Step ⑥

Coding efficiency :-

$$\eta = \frac{1.51}{1.55} = \frac{H(X)}{L}$$

$$= 0.97 \times 100\%$$

$$= 97.41$$

Step ⑥

Redundancy

$$\gamma = 1 - \eta$$

$$= 1 - 97.41$$

$$\gamma = 0.0259$$

Ex

Hempel 2id loading

Que

consider a binary sequence
101011010010011010100001100
1110101100011011.

Ans.

Numeric position	Sub-sequence	Variable represent.	Binary encoded code
1	1	-	00001
2	0	-	00000
3	10	12	00010
4	11	11	00011
5	01	21	00101
6	00	22	00100
7	100	32	00110
8	111	41	01001
9	010	52	01010
10	1000	72	01110
11	0111	51	01011
12	001	61	01101
13	110	42	01000
14	0101	31	00111
15	10001	701	01001
16	1011	141	11101

② Find the Lempel-Ziv Code for the Binary sequence.

Ans :- 010010000001100010000001000001
010001000001

Recover the original sequence back from the extract source code.

Ans	Numeric Position	sub sequence	Variable represent	Binary Encoder block
	1	0	-	00000
	2	1	-	100001
	3	00	11	00010
	4	10	21	00100
	5	000	31	00110
	6	001	32	00101
	7	100	41	01000
	8	0010	61	01100
	9	0000	51	01010
	10	00100	81	10000
	11	00001	92	10001
	12	01	12	00011
	13	0001	52	01011
	14	00001	912	01001

Binary Encoded	Number position	Original Sequence
00000	1	0
00001	2	1
00010	3	00
00100	4	10
00110	5	000
00101	6	001
01000	7	100
01100	8	0010
01010	9	1000
10000	10	00100
10011	11	00001
00011	12	01
01011	13	0001
10011	14	000001
	15	
	16	
	17	
	18	
	19	
	20	

Halpman coding

Lempelzid coding

Prima-
for

Proba-
model
of
source

① Halpman coding is required the probability model of the code

② It does not require the knowledge of probability model of source code and parsing source data string into segment.

Code-
word
length

③ It having a variable code length

④ It having fixed code length

Compa-
tion

⑤ It achieve a coding in 43% (less than LZ coding)

⑥ It achieve a compaction is high as compared to Halpman coding

Imple-
mentation

⑦ It is implement is difficult

⑧ It is difficult to implement

Efficien-
cy

It will be required to calculate the efficiency in halpman coding. But it is less efficient as compared to LZ coding.

It will no required to calculate the efficiency but LZ efficient is more as compared to Halpman coding

★ Rate distortion theory in optimum quantization :-

we have the various source coding them for DMS AIC to which the average code word length must be At least as long as source entropy for perfect coding.
But in actual procedure the coding will be imperfect due to some constraint that create a distortion in a system.

Rate distortion function

Consider a discrete memoryless source (dms) defined by memory source.

i.e. $x_i = 1$ to M .

which consist of state of statistical index. Symbol associated with probability P_i .

Let R the average code length in bit/word
Source coding from state that

The alphabet provide a perfect representation of source.

Let $P\left(\frac{y_i}{x_i}\right) = \text{joint probability}$

$P(x_i, y_i) =$ the joint probability of occurrence of source symbol x and y .

$$P(x_i, y_i) = P(y_i | x_i) \cdot P(x_i)$$

$P(x_i) =$ individual probability

★ Quantization :-

① It is process which is Round off the Range or value is called "Quantization".

It is process of mapping a continuous a discrete scaling or vector produce by a source into a set of digital symbol that can be x^n or stored using a finite no. of bit

Ques Explain the Huffman's coding and Lempel coding.

Ans. Huffman's coding.

① Find out the no. of msg. or letter.

② arrange the letters in order of their probability in descending order.

③ Find out the code, which is a fast tree value of code probability can be added and arranged that value in descending value.

④ Find out the code and the length of code.

⑤ length of code is equal to no. of binary value present in code.

⑥ then find entropy of the code which is find out by using formula $H = -\sum p_i \log_2 \left(\frac{1}{p_i} \right)$.

⑦ then find out the length of code which is denoted by PL .

⑧ which is find out by:

$L = \text{Probability} \times \text{length of code}$
probability code

⑨ then, find out efficiency of the code which is find by $\eta = \frac{H}{L}$

$$\eta = \frac{H}{L}$$

efficiency is always less than or equal to 1

⑩ the redundancy is find out by using formula

$$R = 1 - \eta$$

ADPCM: Adaptive differential Pulse Code Modulation

- 1) In pcm system the speech coding rate is required 6400 per sample. So that a high bandwidth is needed for transmitting the signal.
- 2) In the radio channel with low capacity required a speech coding at low speech rate for maintaining the Fidelity and quality reproduction.
- 3) The speech coding at low bit rate it required a waveform coder for optimise the and removing the unwanted data of speech signal.
- 4) In ADPCM (Adaptive) it required 32 kb/sample for transmitting the signal we can used the combination of adaptive quantization and adaptive prediction.
- 5) In Adaptive quantization it required quantizer to round of the value of the system and it will be varied by step size will will denoted by Δ



There are two type of adaptive quantization with forward estimation

- 1) Adaptive quantization with forward estimation
 - 2) Adaptive quantization with backward estimation
- 1) In adaptive quantization with backward estimation is derived with backward estimation.

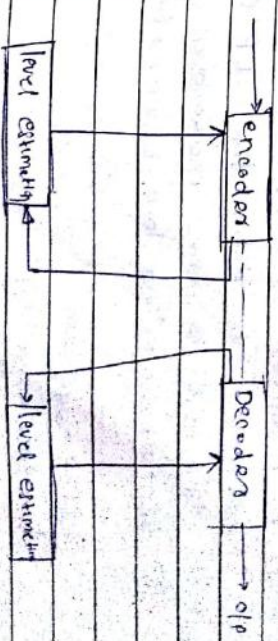
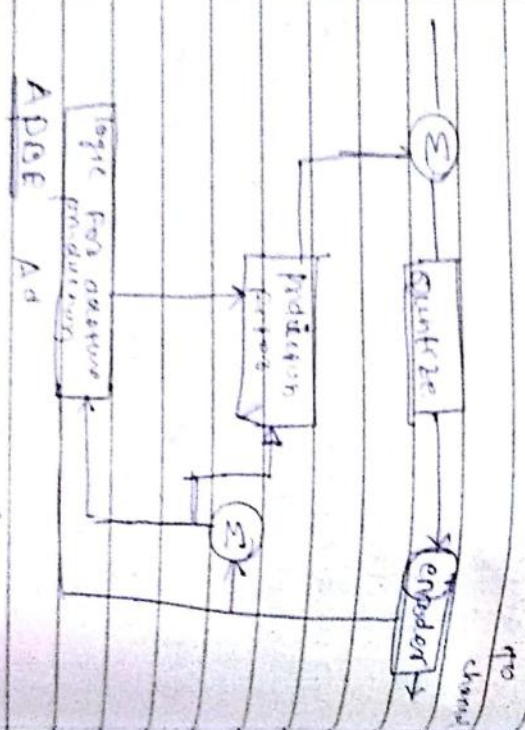


Fig: Adaptive Quantization

③

The above prediction having disadvantages which can be removed by Adaptive Prediction Residual Estimation.

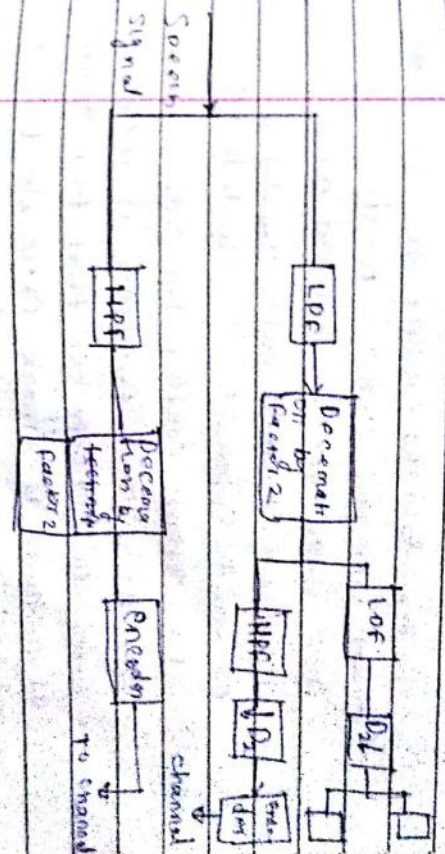
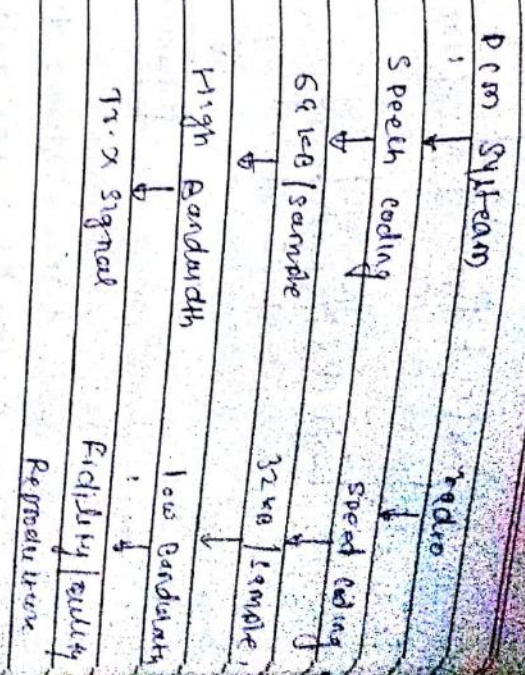
- ① side information
- ② quantizing
- ③ delay

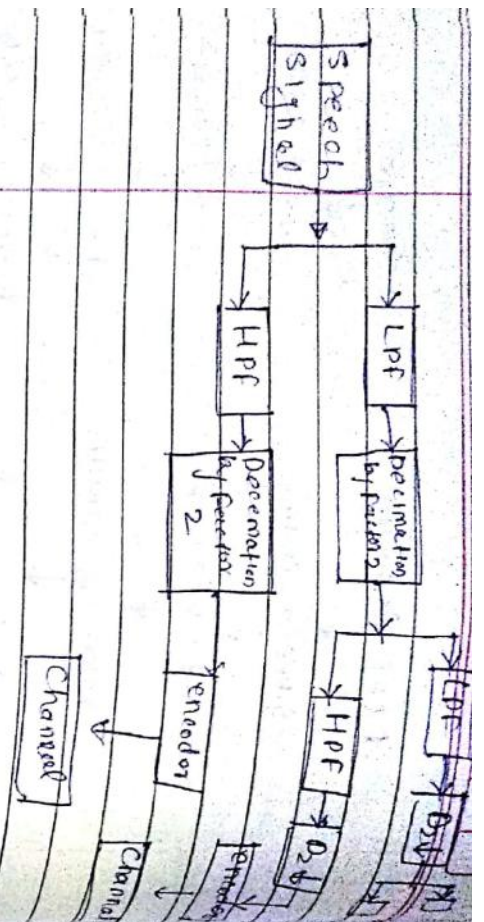


- ① It required 22 KHz sample
- ② restoring is better than DPCM
- ③ It remove the redundancy
- ④ It required a low channel BW

→ PCM-system
 → speech coding
 → 64 KHz / sample
 → High BW
 → transmission signal

* Subband coding:





① The above diagram shows that sub-band coding low pass filter, high pass filter, decimation and encoder are used for the operation

② In the subband coding the speech signal (image) signal is divided into small no. of subband and the time waveform in each subband is encoded separately subband

③ In speech coding for eg. At lower frequency is most spectral energy so that the quantization noise is more noticeable for the hearing bit

④ more bits are used and fewer bits are used higher band signal

⑤ The filter is designed for achieving the good performance in sub band coding so that we are using the low pass filter and HPF.

⑥ The LPTF will pass the low frequency and the HPF will pass the higher frequency

⑦ The output of the filter is given to the decimation to decrease the signal by factor 2. then output of decimation is given to the encoder in which the signal will be encoded and given to the channel

⑧ But in the LP operation the signal will not be encoded and it will be repeated respectively

⑨ In the sub-band coding the lower frequency repeatedly sub divided by a factor of 2.

In the decimation it will be decreased by 2 so that the sampling of the system is reduced

⑩ For eg. the speech signal having frequency of 3200 Hz in the first stage the lower frequency 0 to 1600 Hz

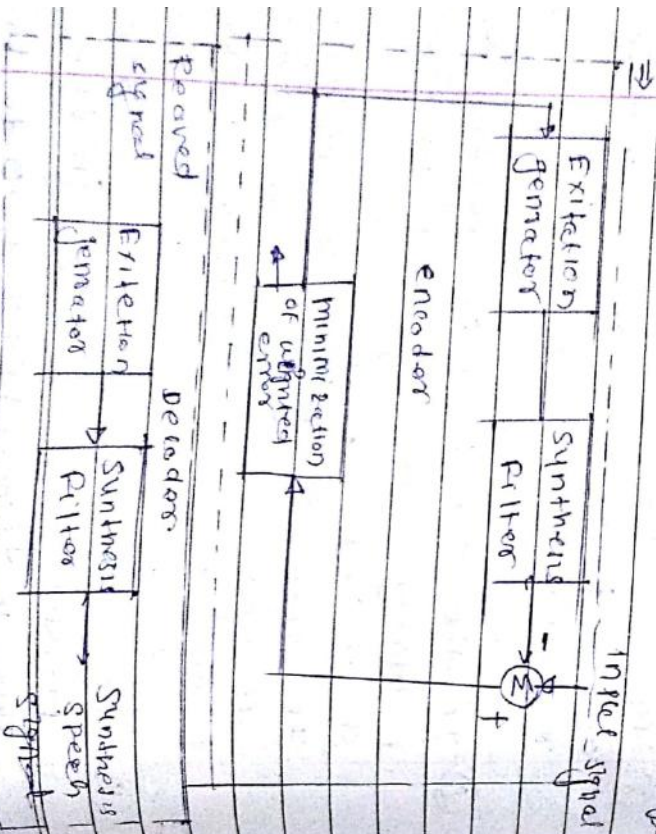
and High frequency 1600 to 3200 Hz

Then the lower frequency can be split in 2nd stage. Lower frequency is 0 to 800 Hz and Higher frequency 800 to 1600 Hz

Then in 3rd stage lower frequency is 0 to 400 Hz and Higher stage 400 to 800 Hz

* LP and CELP :-

(Linear Prediction Coding)
(code excited linear prediction coding)



①

The above diagram shows the LP encoder. the encoder consists of three blocks. Excitation generator, Synthesis filter, Minimization of weighted error.

②

In the Synthesis filter the main function of the Synthesis filter is produce the Synthesized speech. Attenuation of original version.

③

The excitation generator generate the No. of Pulse having 5 to 50 msec. every.

④

The minimization of weighted error original signal and synthesized speech.

⑤

First the input speech signal is excited with the help of excitation generator. The output of excitation generator is the synthesized speech.

⑥

The free parameter of synthesis filter are computed using the actual speech sample as input to the summer or adder.

⑦

Another input of the summer is coming from original signal as new signal.

⑧ The output of the summer if the signal is errorless the we can add to the signal to the system.

⑨ If the error is present in the signal the we can go minimization of weighted error block

⑩ The decoder is consist of excitation generator and synthesis filter

⑪ the decoder decodes the Residual signal to produce a synthesis version of the original speech signal which is achieve by by raising the excitation generator of synthesis filter whose parameter is set equal to present in the encoder.

⑫ CELP :

① In CELP uses the code predetermine of excitation code book as the ~~code~~ source of excitation for the synthesis of filter.

② In the CELP there are four block minimization of weighted error, excitation, synthesis filter, gain factor.

③ first we can give the input signal directly to the summer and another input coming from synthesis filter. Then both the signal applied to minimized the weighted error. The choice of factor G that is gain to optimize the minimum average power of a weighted error between original and synthesis speech.

④ The ~~average~~ ^{address} of the code book is selected from excitation code book and the corresponding gain together ~~and~~ ^{used} for transmitting the signal.

⑤ An identical copy of the code book is made available in the decoder which enables the decoder to at current result at the. thereby producing the synthesis of original speech signal.

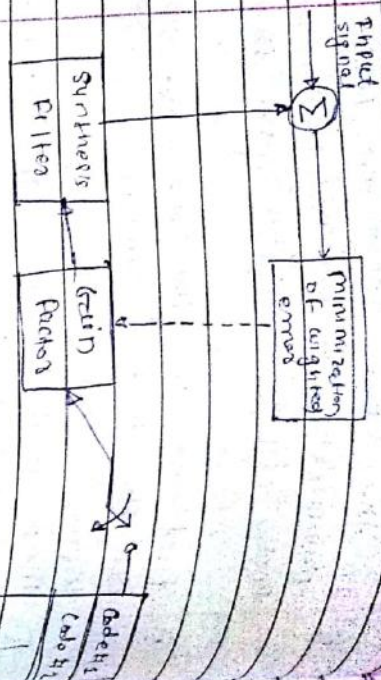


Fig. CELP

Excitation
(code book
of STEN)

3. Digital Modulation Techniques

* Amplitude Shift Key (ASK)

(ON-OFF Key)

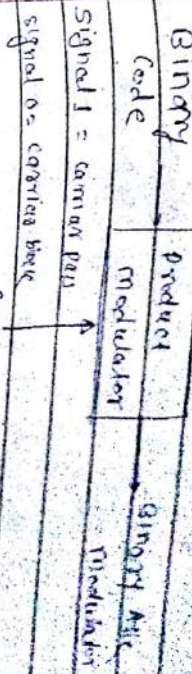


Fig. Generalization of ASK

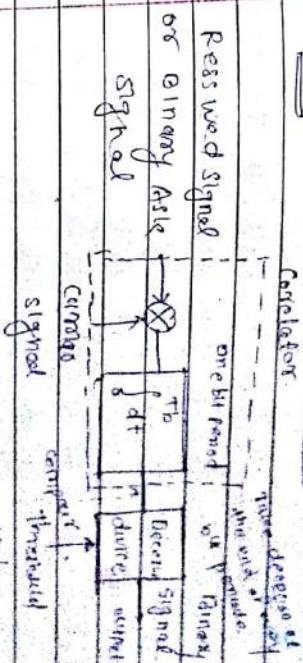


Fig. Detection of ASK

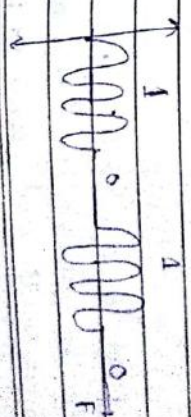


Fig. waveform

① ASK or on-off key is simplest digital modulation technique. In those techniques there is only one unit energy carrier and it is switch on-off depending upon input binary sequence.

② the ASK waveform can be represented as

$$s(t) = \int_0^T \cos 2\pi f_c t \quad \text{for '1'}$$
$$s(t) = 0 \quad \text{for '0'}$$

④ the symbol '1' pulse is Tx and the symbol '0' No pulse is Tx.

⑤ Generation of ASK :-

① diagram of ASK

② the input binary sequence is applied to the product modulator. Another

③ the product input of the product modulator is carrying from carrier signal

④ the product modulator amplitude when it passes the carrier when the input is '1' and when input is

0, the carrier is blank.

⑥ At the output of the product modulator binary ASK signal.

5. Detection of ASK :-

① diagram of Detection of ASK

② the ASK signal or receive signal is applied to the correlator. The correlator is combination of multiplication and integrator

③ the locally generated coherent carrier is applied to the multiplier. The output of the multiplier is integrated over one bit period.

④ The decision device takes the decision at the end of every bit period. It compares the output of integrator with the threshold. Decision is taken in favour of 1 when the threshold is high otherwise zero.

Frequency shift key 8-

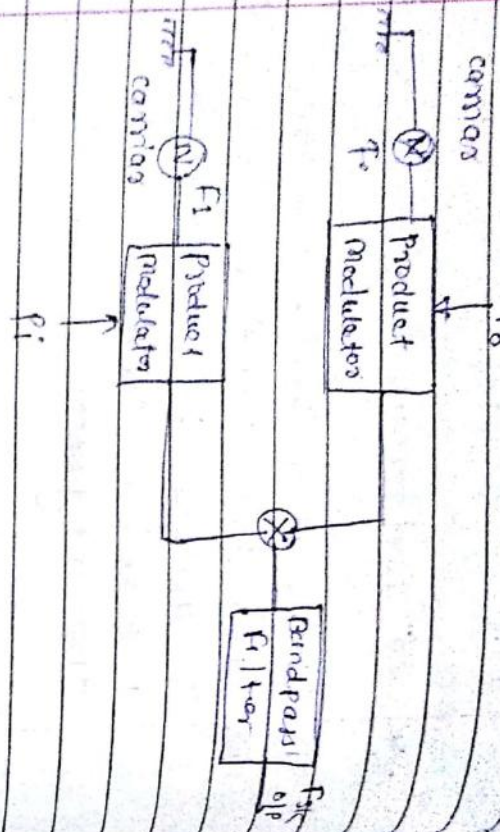


Fig generation of fsk

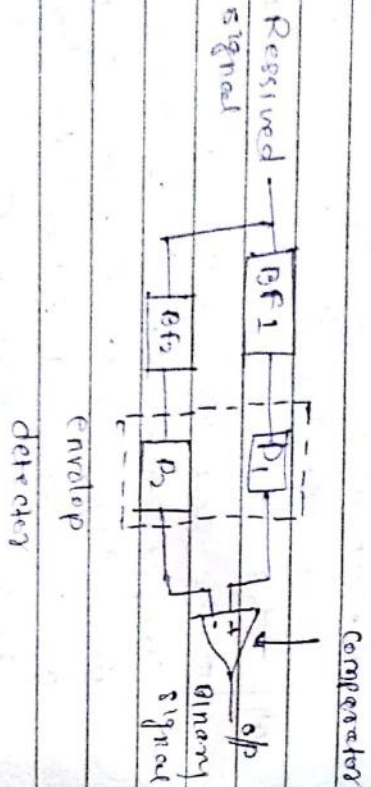
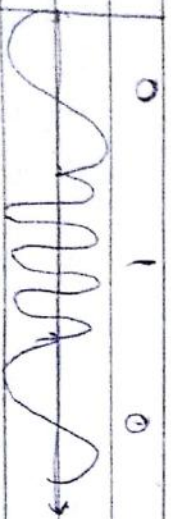


Fig:- detection of fsk



① The above diagram shows that FSK is consist of two oscillator product modulator. BPF.

② It is consist of two oscillator which produce the sine wave at frequency f_c and f_i respectively.

③ The output of oscillator are apply to the input of two product modulator & the circuit of product modulator are the signal P_i & f_c .

Binary 0 :-	-1	1	1	0
Binary 1 :-	+1	1	0	1

⑤ when Binary 0 is to be transmitted P_i is equal 1 and P_i is equal to 0. Therefore the output of 1st modulator is present and the output of 2nd modulator is absent at frequency f_i .

⑦ when Binary 1 is transmitted P_i is equal 0 and P_i is equal 1. Therefore the output of 1st modulator is absent at frequency f_c and second modulator is present at frequency f_i .

- ⑥ By using adder both the o/p of modulator is added and result of these is 1.
- ⑦ And the o/p of Adder passes through BPF which is part of system and o/p of the BPF is we get a FSK signal.

Detection of FSK

① diagram

- ① the FSK receiver is consist of two BPF with frequency f_0 and f_1 two envelop detector which is simple diode detector which rectifier and filter the signal

- ② If the binary 1 receive first mean the o/p of BPF-1 and D_1 is also 1 and o/p of BPF-2 and D_2 is zero. and the output to the comparator. It will compare and produce the logic and the o/p of comparator. ~~also~~ (VLSI version)

Advantages :-

- ① The when binary 0 is transmitted signal is present. And in ASK signal is not.

② Disadvantages

- ① The transmission part is complex as compared to ASK
- ② maintenance is more
- ③ cost is more
- ④ Bandwidth required is more

Quadrature

QPSK (Polarization phase shift keying) :-

Trellis mitter :-

Binary 1 = 1V
Binary 0 = -1V

even signal
odd signal

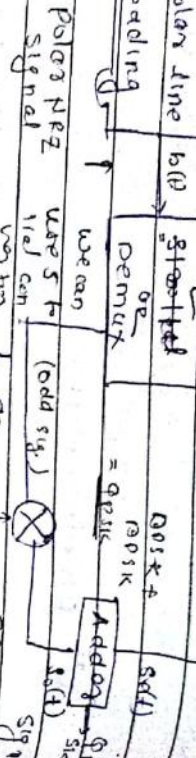


Fig: Tx of QPSK

Example, 1100101101

odd signal = {1, 0, 1, 1, 1, 1}

even signal = {1, 0, 0, 1, 1, 0}

even

be(t)

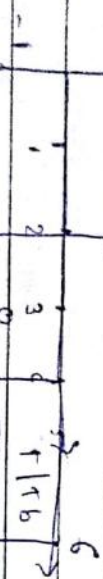


Fig. even bit

odd

bo(t)

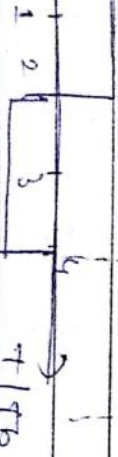


Fig. odd bit

1 the above block diagram is the QPSK transmission system

2 It consists of a polar line coding, a MUX or signal to received carrier, a demultiplexer, and a decoder.

3 The binary code signal is applied to polar line coding which is converted to the polarize signal. that signal at the output of polar line coding is 1V and -1V.

4 If the binary 1 is represented by 1V and binary 0 is represented by -1V.

5 $b_e(t)$ is applied to the signal to the carrier vector of demux the signal is the output of demux to get a two-separate bit stream. that is even bit or odd bit.

6 the even signal is denoted by $b_e(t)$ and odd signal is applied to $b_o(t)$.

7 this signal is applied to the multiplexer product modulator and another input of the multiplexer product modulator is a common carrier signal. the output is a modulated signal with a sine wave as a carrier signal.

* Advantages *

- ① 1st Bandwidth as compared to BPSK
So the v/r rate is high.
- ② Good noise immunity
- ③ Disadvantages are
- ④ 1st Bandwidth efficiency

* Application *

It is used in various wireless
cellular standard like GSM

S/P sequence

Differential
encoding

Modulated wave

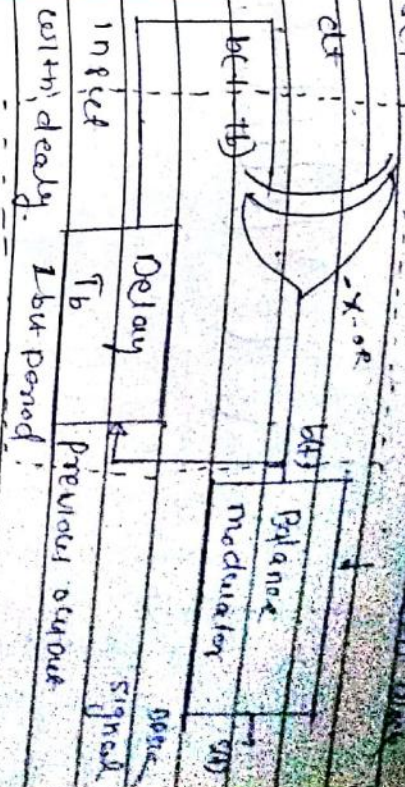


Fig. Tx of DPSK

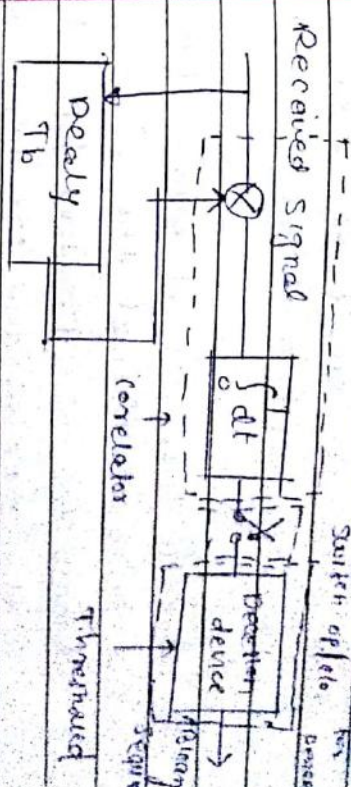
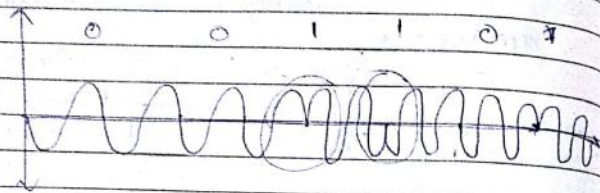


Fig Rx of DPSK

Example.



① waveform of DPSK



Tx

- ① The above Fig shows that the generation of DPSK is consist of Differential encoding, and modulator block
- ② the I/P sequence of binary bits is modified such that the next bit depends upon a previous bit therefore in the receiver the previous received bit are used to detect the present bit
- ③ The I/P sequence is denoted as $d(t)$ and the o/p sequence is denoted by $b(t)$ and $b(t-T_b)$, is the previous output delay by 1 bit period
- ④ depending upon the value $b(t)$ and $b(t-T_b)$ the X-OR gate generate the output sequence $b(t)$

- ⑤ And then apply to the B.M and the output of B.M is the DPSK signal which is represented $s(t)$

Rx of DPSK

- ① In the Rx section of DPSK it is consist of correlator, switch and comparator device.
- ② first the Rx signal is applied the correlator is applied and another signal applied to delay T_b
- ③ The o/p of multiplier is given to the integrator and if the signal is present at the o/p of correlator the switch will be closed and its output will be given to the input of decision device and another input is given to the threshold
- ④ It is act as comparator with it compare the signal and we get a binary sequence.

Advantages

- ① Synchronization is not required Tx and Rx

- ② the carrier signal is not required at the Rx
- ③ B.W of DSK is reduced

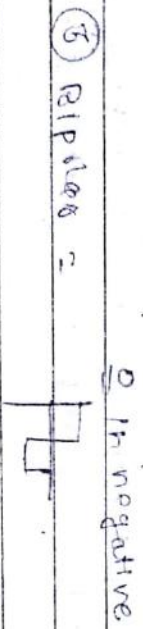
Disadvantages:

- ① Error rate of DSK is high than BPSK

* Numbered on Line coding :-

Unipolar, polar, Bipolar, Manchester
 RZ - return to zero
 NRZ - Non return to zero

- ① Unipolar - Pulse present at 1 for 0H
- ② polar - pulse present in 1 and 0 is negative



- ④ split Manchester

Binary 1 - (voltage + half)
 Binary 0 - (-v or positive)

1000101
 ① unipolar ② polar, ③ bipolar ④ split Manchester

Ans.	1	0	0	0	1	0	1
unipolar							

polar

--	--	--	--	--	--	--	--

bipolar

--	--	--	--	--	--	--	--

Manchester

--	--	--	--	--	--	--	--

* Memoryless modulation method.

① QAM :-

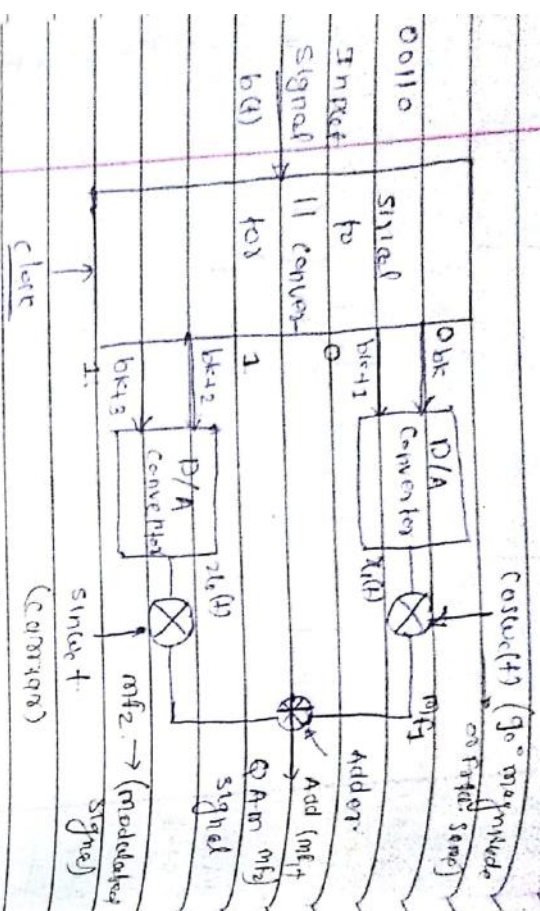
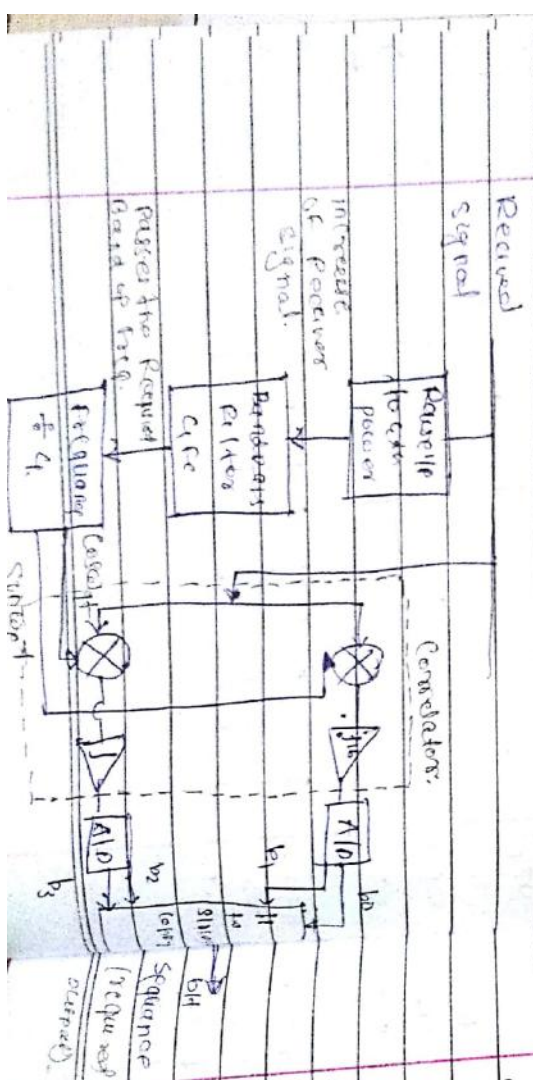


Fig: Tx of QAM or QASK



QAM Tx :-

① The fig shows that the transmitter of quadrature Amplitude Modulation. It is consist of signal to be modulated, P/A converter, multiplier and adder.

② First 1/r signal which is denoted by 1 is fed back to the signal to be modulated. The signal is converted into 1/r signal $b(t)$, $b(t+1)$, $b(t+2)$, $b(t+3)$...

③ Then these signal is applied to the digital to analog convert. The op of d/a convert $x_1(t)$ and $x_2(t)$.

④ These signal is applied to the multiplier and another is coming from carrier signal and are in the form of sine and cosine.

⑤ Again the carrier signal having 90° distance so that the magnitude and frequency of the signal is same as of multiplier. First modulated signal and second modulated signal in the form of m_1 and m_2 both the modulated signal are applied to the adder and produce the qam signal which is used for Tx.

* QAM Rx

① the above fig shows that the Rx of QAM. It is consist of Phase locked loop, Band Pass Filter, Frequency divider, Correlator, ~~the~~ A/D converter and 1101 to signal converter.

② The receive signal
If the signal is not proper so that the receive signal is passed Phase locked loop power.

③ This block is increases the H.F. Received signal. then it will be applied to the Band Pass Filter with frequency 4fc.

④ the frequency which passed to the limited with 4fc. the output of filter is given to the frequency divider.

⑤ the frequency divider is divided the frequency then it will be cancel and the carrier frequency in the form of sin and cosine.

⑥ Both the carrier frequency provided to the correlator
⑦ then the output of correlator is applied to the A/D converter. which converts the Analog signal into digital in the form of bit denoted as b_1, b_2, b_3 .

⑧ then these bit are applied to the 1101 to signal converter. These bit are produce the binary and produce the bit sequence that is bit's required output.

* Advantages :-

- ① B.W is less as compared to PSK
- ② more efficient
- ③ simple net.

* Disadvantages :-

- ① B.W is very bit rate increases
- ② more bit information can be required per symbol

* Application :-

- ① telephone cable
 - ② optical fiber cable
- system to produce long bit error

Linear modulation with memory

Modulation order $\rightarrow$ M-ary DSK

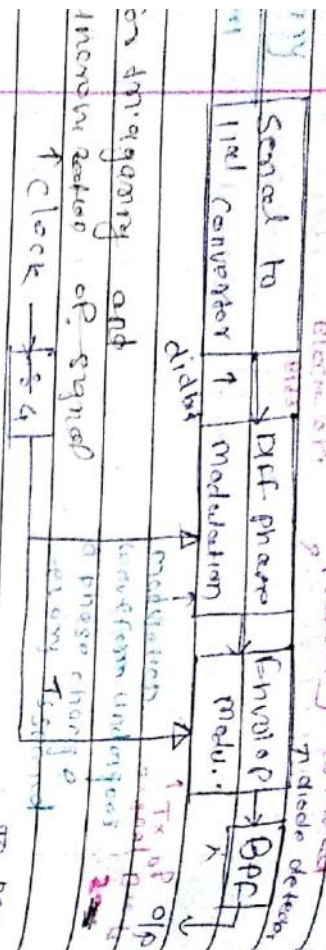


Fig. Tx of m-ary DSK

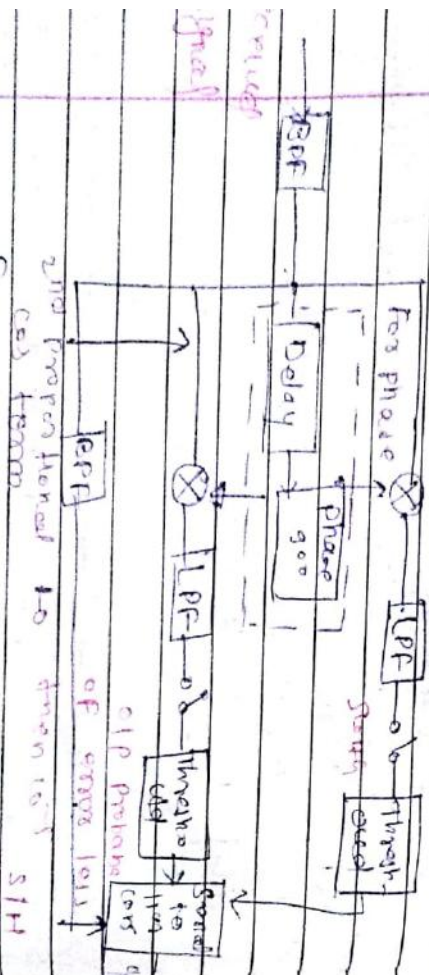


Fig. Rx of m-ary DSK

Fig. Rx of m-ary DPSK

- 1) The above Fig shows that Tx of m-ary DSK
- 2) First the binary data is converted into signal to parallel form then the binary data is group into clock of two bit or in the form of parallel
- 3) At differential phase modulator the data rate and carrier frequency fixed
- 4) The BW of the Tx signal is in the order of two.
- 5) The clock is used for frequency and synchronization of signal
- 6) The non-rectangular signal energy the useful information used to the Receiver without introducing error.
- 7) Because of the differential phase shift key the modulated waveform under goes a phase changes every second then the required band of frequency will be gate from BPF to produce the output.

Rx.

10 - The above fig shows that the Rx of m-ary DSK.

11 In the m-ary DSK Receiver Produces the two output.

The first o/p is proportional to sine wave and 2nd o/p is $\propto$ to cos wave.

12 The 1/p bit can be decoded uniquely from the sine of these two o/p.

13 The signal is filtered by g_c with the help of delay then the signal is passed through LPF.

14 It will pass the low frequency and signal will compared to the stored value and provided to the 1/p to serial converter.

15 Thus the signal is converted in the serial form.

16 Finally the PLL to signal converter interface the diodes to 2L 9

17 signal binary of the o/p probability of error is less than 10⁻⁵ and $\frac{3}{4}$ ratio of the o/p channel is above 18 dB.

Advantages :-

- 1 cost is low
- 2 maintenance is low
- 3 No. of possible outcome is more
- 4 Simple construction
- 5 Reduced loss

Disadvantages :-

- 1 Baud rate synchronization is required
- 2 The LPF is required and good phase shift required as compared to DSK.

Application :-

- 1 optical cable.
- 2 television
- 3 cordless communication.