

SRM Institute of Science and Technology College of Engineering and Technology

DEPARTMENT OF ECE

SRM Nagar, Kattankulathur – 603203, Chengalpattu District, Tamilnadu

Academic Year: 2024-2025 (Even)

Test: FT- III Course Code / Title:21ECC302T/ Analog and Digital Communication Year & Sem:III&VI

Date: 03.04.2025 **Duration:12.30 – 2.15PM** Max. Marks:50

Batch 2

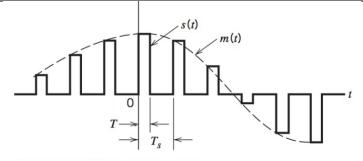
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Course Articulation Matrix:

	21ECC302T/ Analog and Digital Communication		PROGRAM OUTCOMES (PO)			PROGRAM SPECIFIC OUTCOMES										
S.NO	COURSE OUTCOMES	1	2	3	4	5	6	7	8	9	10	11	12	1	2	3
1	Explain the Various Analog Modulation Techniques	3	-	-	-	-	-	-	-		1	-	2	2	-	-
2	Analyze the Noise performance of Radio transmitters and Receivers	3	3	ı	-	-	-	-	-	-	i	-	2	-	3	-
3	Demonstrate the demodulation and detection of received digital data	3	2	ı	-			-	1	1	ı	-	-	-	ı	3
4	Apply the suitable passband techniques for real time applications	3	ı	ı	-	3	-	-	1	1	i	-	-	-	ı	2
5	Exposed to the concepts of information theory and channel capacity	3	-	3	-	-	-	-	-	1	-	-	-	3	-	-

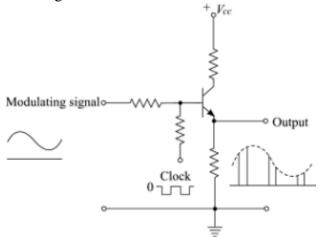
Q. No	Part A (11x1=11 Marks) Answer ALL the question	Marks	BL	CO
1	If a PCM system uses 8 bits per sample and has a sampling rate of 8 kHz, what is the bit rate? a) 64 kbps b) 32 kbps c) 16 kbps d) 128 kbps	1	2	3
2	A PCM system has a maximum input voltage of 5V and a minimum voltage of -5V. It uses 8-bit quantization. What is the step size of the quantizer? a) 0.02 V b) 0.04 V c) 0.08 V d) 0.1 V	1	2	3
3	The main purpose of a matched filter in a communication system is to: a) Reduce noise in the channel. b) Maximize the signal-to-noise ratio (SNR). c) Minimize the bit error rate. d) Amplify weak signals.	1	1	3
4	Inter Symbol Interference (ISI) occurs due to: a) Channel noise b) Overlapping of successive symbols c) Insufficient quantization levels d) Low sampling rates	1	1	3
5	A major drawback of Delta Modulation (DM) is: a) Quantization noise b) Slope overload distortion c) High complexity d) Low bandwidth efficiency	1	1	3
6	If a BPSK signal has a carrier frequency of 100 MHz and a data rate of 5 Mbps, what is the spectral bandwidth of the BPSK signal? a) 2.5 MHz b) 5 MHz c) 10 MHz d) 20 MHz	1	2	4
7	The probability of error in Quadrature Phase Shift Keying (QPSK) is given by: a) $P_e = \frac{1}{2} erfc \left(\sqrt{\frac{E_b}{N_0}} \right)$ b) $P_e = \frac{1}{2} erfc \left(\sqrt{\frac{2E_b}{N_0}} \right)$	1	2	4

	c) $P_e = erfc(\sqrt{\frac{2E_b}{N_0}})$ d) $P_e = \frac{1}{2} erfc(\sqrt{\frac{E_b}{2N_0}})$			
8	Which of the following has the highest bandwidth efficiency? a) BPSK b) QPSK c) 8-PSK d) 16-PSK	1	1	4
9	is a passband modulation technique a) Pulse Amplitude Modulation (PAM) b) Frequency Shift Keying (FSK) c) Pulse Code Modulation (PCM) d) Delta Modulation (DM)	1	1	4
10	A 16-QAM system has how many distinct symbols? a) 4 b) 8 c) 16 d) 32	1	2	4
11	Which modulation scheme has the lowest bit error rate (BER) for a given signal-to-noise ratio (SNR)? a) BPSK b) QPSK c) 16-PSK d) BFSK	1	1	4
	Part B (3x8=24 Marks) Answer ALL the question			
12. a.	Explain Pulse Amplitude Modulation (PAM) and describe how a PAM signal is generated and demodulated, including the necessary waveforms. Answer:	8	3	3
	 In pulse-amplitude modulation (PAM), the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate shape. Pulse-amplitude modulation as defined here is somewhat similar to natural sampling, where the message signal is multiplied by a periodic train of rectangular pulses. However, in natural sampling the top of each modulated rectangular pulse varies with the message signal, whereas in PAM it is maintained flat The waveform of a PAM signal is illustrated in Figure 3.1. The dashed curve in this figure depicts the waveform of a message signal m(t), and the sequence of amplitude modulated rectangular pulses shown as solid lines represents the corresponding PAM signal s(t) For transmission of digital data is discrete pulse amplitude modulation(PAM). In discrete PAM, the amplitude of the pulse varies in discrete manner according to the input binary data. The discrete PAM can have only two amplitude levels corresponding to binary '1' and '0'. Successive binary bits can be combined into symbols. There can be multiple amplitude levels corresponding to these symbols. They generate discrete PAM signals. These signals can be transmitted (without any modulation) over the channel in baseband transmission. In PAM, amplitude of pulses is varied in accordance with instantaneous value of modulating signal. 			

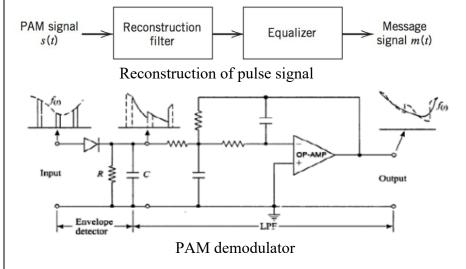


PAM Generation (Explanation- 1 Mark; diagram 2 Marks)

- The circuit is simple emitter follower., In the absence of the clock signal, the output follows input. The modulating signal is applied as the input signal. Another input to the base of the transistor is the clock signal. The frequency of the clock signal is made equal to the desired carrier pulse train frequency.
- The amplitude of the clock signal is chosen the high level is at ground level(0v) and low level at some negative voltage sufficient to bring the transistor in cutoff region.
- When clock is high, circuit operates as emitter follower and the output follows in the input modulating signal. When clock signal is low, transistor is cutoff and output is zero. Thus the output is the desired PAM signal.

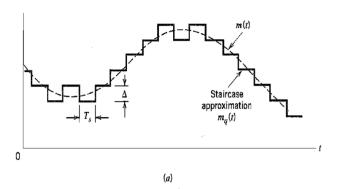


PAM Demodulation (Explanation- 1 Mark; diagram 2 Marks)



• A PAM (Pulse Amplitude Modulation) demodulator recovers the original message signal from a modulated waveform.

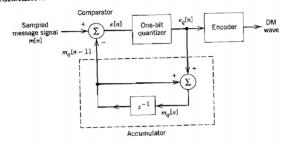
	It consists of an envelope detector and a second-order op-amp low pass			
	filter.			
	• The envelope detector, made using a diode, capacitor, and resistor, tracks the amplitude variations of the PAM pulses and converts the amplitude information into a continuous signal.			
	• The second-order low pass filter, designed using an operational amplifier, provides sharper roll-off and better high-frequency noise attenuation.			
	• It effectively removes the sampling frequency and residual carrier components.			
	• Compared to a first-order filter, the second-order filter is more effective in noise reduction.			
	This ensures accurate retrieval of the original message signal.			
	• PAM demodulators are widely used in analog communication systems, data transmission, and audio signal processing.			
	• Combining an envelope detector and a second-order filter enhances demodulation performance, making it reliable for various practical applications.			
12. b.	(OR) With neat diagram, explain the working of Delta modulation.	8	2	3
	Answer:			
	Delta modulation Concept (2 Marks)			
	The delta modulation is a special case of DPCM.			
	DM is 1-bit version of DPCM.			
	In DM, an incoming message signal is oversampled to increase the correlation between adjacent samples of the signal.			
	It provides the staircase approximation to the oversampled version of the message signal.			
	The difference between the input and the approximation is quantized into only two levels $\pm \Delta$, corresponding to positive and negative differences.			
	If the approximation falls below the signal at any sampling epoch, it is increased by Δ			
	If the approximation lies above the signal ,it is decreased by Δ			
	If there is no rapid change from sample to sample, then the stair-case approximation remains within $\pm \Delta$ of the input signal.			
	The staircase approximation $\underline{mq}(t)$ follows variations in the input signal $\underline{m}(t)$.			



(Diagram 1 Mark)

(Transmitter Diagram 1 Mark: Expression with Explanation 2 Marks)

Transmitter.



• Let m(t) – input message signal $m_q(t)$ – its staircase approximation m[n] –sequence of samples m[n] T_s - sample of m[t] taken at time t=n T_s , where T_s is sampling period. we have,

$$m[n] = m(nT_s),$$
 $n = 0, \pm 1, \pm 2, ...$
 $e[n] = m[n] - m_q[n - 1]$
 $e_q = \Delta \operatorname{sgn}(e[n])$
 $m_q[n] = m_q[n - 1] + e_q[n]$

where

e[n] – error signal representing the difference between the present sample m[n] of input signal and the latest approximation m_q [n-1]

 $e_q[n]$ – quantized version of e(n)

sgn(.) -signum function

The quantizer output $m_q[n]$ is coded to produce the DM signal.

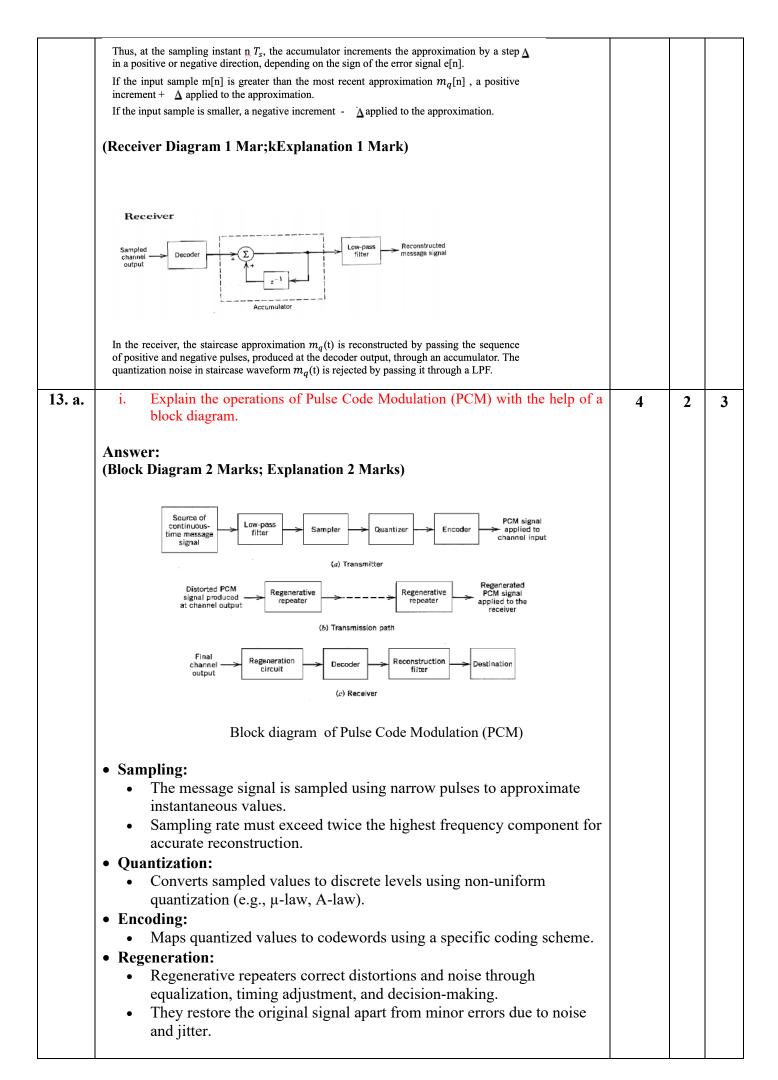
The rate of information transmission is equal to the sampling rate $f_s = 1/T_s$.

The delta modulated wave is generated by applying the sampled version of the incoming message signal to a modulator that involves a comparator , quantizer and accumulator.

z⁻¹ is unit delay.

The comparator computes the difference between its two inputs . The quantizer consists of hard limiter with an input-output relation that is scaled version of the signum function . Then the quantizer output is applied to the accumulator, producing the result,

$$m_q[n] = \Delta \sum_{i=1}^n \operatorname{sgn}(e[i])$$
$$= \sum_{i=1}^n e_q[i]$$



•	Decoding:
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- Converts codewords back to quantized values, forming a Pulse Amplitude Modulated (PAM) signal.
- Filtering:
 - A low-pass filter with a cutoff at the message bandwidth reconstructs the original signal.
 - ii. A Television signal with a bandwidth of 4.2 MHz is transmitted using binary PCM. The number of quantization levels is 512. Calculate, (a) Code word length (1 Mark), (b) Final bit rate (2 Marks) and Transmission bandwidth (1 Mark)

3 3

Answer:

$$W = 4.2 \text{ MHz}$$

Quantization levels q = 512

i) Number of bits and quantization levels are related in binary PCM as,

$$q = 2^v$$

.e. $512 = 2^{v}$

 $\log 512 = v \log 2$

$$v = \frac{\log 512}{\log 2} = 9 \text{ bits}$$

code word length is 9 bits.

(1 Mark)

The final bit rate will equal to signaling rate. The signaling rate is given as,

$$r = v f_s$$

Sampling frequency $f_s \ge 2W$ by sampling theorem.

$$f_s \ge 2 \times 42 \text{ MHz}$$
 since $W = 4.2 \text{ MHz}$

$$f_s \ge 8.4 \text{ MHz}$$

 p_{utting} this value of ' f_s ' in equation for signaling rate,

$$r = 9 \times 8.4 \times 10^6 = 75.6 \times 10^6$$
 bits/sec

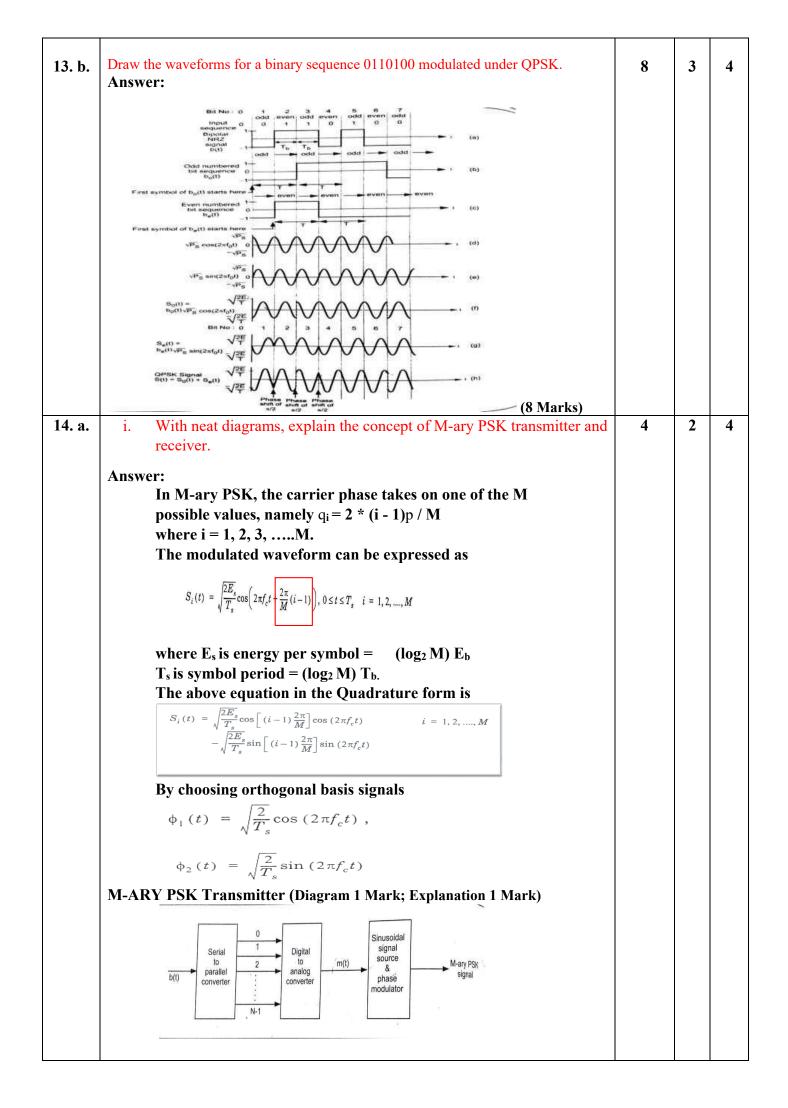
(2 Marks)

The transmission bandwidth is obtained as,

$$B_T \ge \frac{1}{2} r \ge \frac{1}{2} \times 75.6 \times 10^6$$
 bits/sec

$$B_T \ge 37.8 \text{ MHz} (1 \text{ Mark})$$

(OR)



Input Binary Data (b(t)):

The input is a stream of binary data (0s and 1s).

Serial to Parallel Converter:

This block converts the incoming serial bit stream into N parallel bit streams.

Digital to Analog Converter (DAC):

The parallel data is converted into an analog signal that represents a unique value or symbol. Each unique symbol corresponds to a specific phase in the constellation diagram.

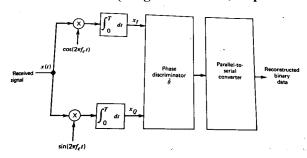
Sinusoidal Signal Source and Phase Modulator:

A carrier wave (sinusoidal signal) is generated. The phase modulator shifts the phase of the carrier wave based on the analog signal. This results in an **Mary PSK modulated signal**.

Output M-ary PSK Signal:

The modulated signal is transmitted over the communication channel. Each symbol carries log₂M bits of information, providing efficient data transmission.

M-ARY PSK Receiver (Diagram 1 Mark; Explanation 1 Mark)



Uses a reference carrier signal that is phase-aligned with the transmitted signal.

I/Q Demodulation: Separates the signal into in-phase and quadrature components for accurate detection.

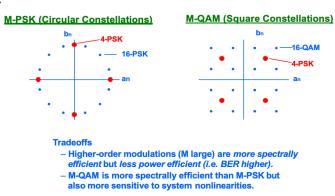
Phase Discrimination: Determines the most likely transmitted symbol based on phase shifts.

Efficient for M-PSK: Works for BPSK (M=2), QPSK (M=4), 8-PSK (M=8), and higher-order PSK.

This receiver efficiently demodulates M-PSK signals by **extracting phase information** from the received signal.

ii. Explain the difference between M-PSK and M-QAM in terms of constellation diagrams.

Answer:



Constellation Diagram for M-PSK (1 ½ Marks)and M-QAM (1 ½ Marks)

Any two differences 1 Mark (1/2 Mark for each)

4 3

3

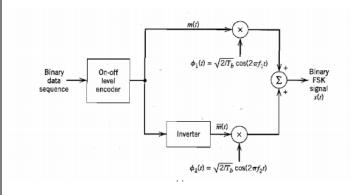
Describe the generation and detection of binary FSK signal with necessary diagram and equation.

Answer:

Generation of FSK (2 Marks – Diagram; 2 Marks - Explanation with Expressions)

8 2

4



In a binary FSK system symbol '1' and '0' are transmitted as

$$S_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_1 t \text{ for symbol 1}$$

$$S_2(t) = \sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_2 t \text{ for symbol 0}$$

Frequency

$$f_i = \frac{n_c + i}{T_b}$$
 for some fixed integer nc and i=1, 2

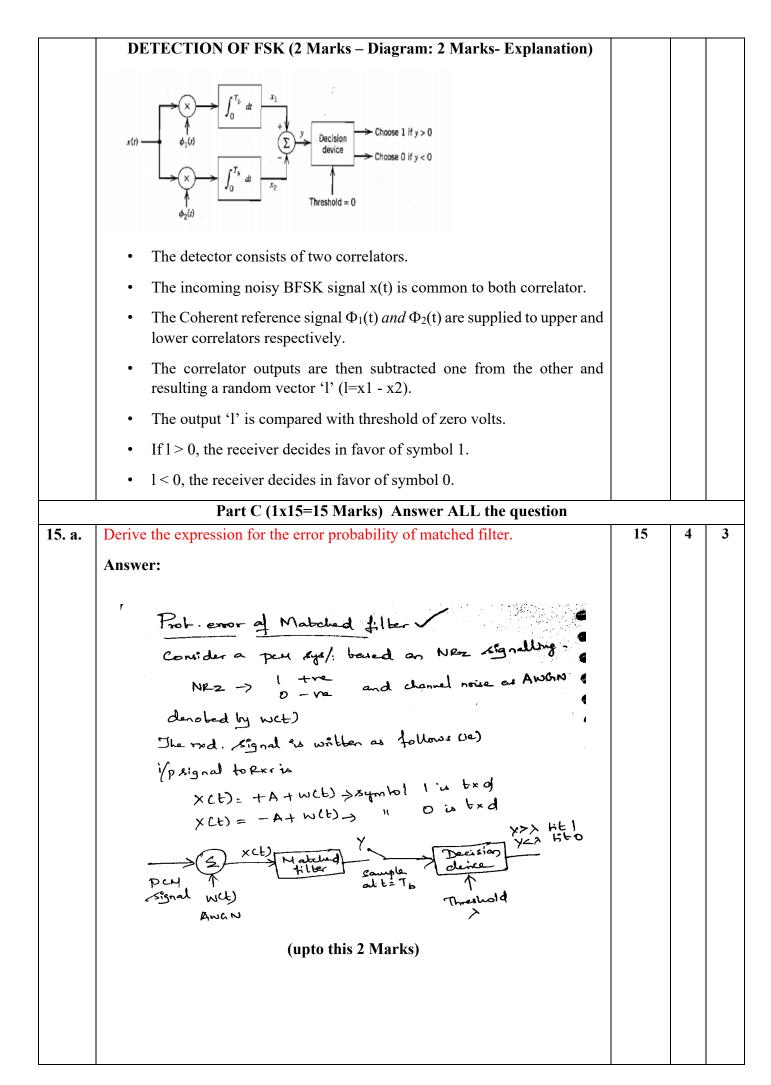
The basic functions are given by

$$\begin{split} \phi_1(t) = & \sqrt{\frac{2}{T_b}} \, Cos2\pi f_1 t & for \quad 0 \leq t \leq T_b \\ \phi_2(t) = & \sqrt{\frac{2}{T_b}} \, Cos2\pi f_2 t & Zero \ Otherwise \end{split}$$

Therefore FSK is characterized by two dimensional signal space with two message points i.e. N=2 and m=2.

The two message points are defined by the signal vector $S_1 = \begin{bmatrix} \sqrt{E_b} \\ 0 \end{bmatrix} \quad \text{and} \quad S_2 = \begin{bmatrix} 0 \\ \sqrt{E_b} \end{bmatrix}$

- The incoming binary data sequence is applied to on-off level encoder.
- The output of encoder is \sqrt{Eb} volts for symbol 1 and 0 volts for symbol '0'.
- When we have symbol 1 the upper channel is switched on with oscillator frequency f1, for symbol '0', because of inverter the lower channel is switched on with oscillator frequency f2.
- These two frequencies are combined using an adder circuit and then transmitted.



```
2 possible Biterors are
   i) chosing tit I when o was txd.
   i) "
 care i) symbol o was tood
   then tod. signal & xct) = - A+ wct)
 rectangular '(p'-A' to matched filter can be
 replaced by integrator and dump circuit
  The integrator computes the area under the
   ... Y= 1 xct) dt
  rectangular pulse
         Y= Th To [-A+wet)]dt
      Y= Tb [ ] - Adt + ] well dt
        = 1 [[ NE] + ] will de]
       = In-ATb + In I bwill dt
     y = -A + Tb Jwch) de -> 0
 · · Y+A = I Jouce dt > 2
  random variable y is gaussian distributed with a
  - A . The variance of y on written as
       5-2 E [ y- 772
          = E[Y-(-A)]2= E[Y+A]2
    Wing 2
    ozy = E[ + Jouchot]
      = [ ] b ] b With wear dt du]
      = 1 To To E[wet) wew) ] dt du
can be withered to Jo Pru(t,u) dt du > (3)

The of Ru(t,u) dt du > (3)

white nouse
       PWCF, W= No f(F-W) -> (4)
```

(upto Average probability of symbol error 8 Marks)

For optimum threshold:

Using Leibnit's rule

$$\frac{1}{du} \int_{0}^{du} f(z,u) du = f(b(u),u) \frac{db(u)}{du} - f(a(u),u) \frac{da(u)}{du}$$

$$+ \int_{0}^{du} \frac{1}{du} \frac{1}{du} dz$$

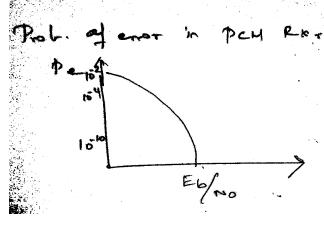
660 wider of $f(z,u) = \frac{1}{2} \exp(-z^{2})$

$$\frac{d}{du} \operatorname{er} \sharp c(u) = -\frac{1}{\sqrt{n}} \exp(-u^2) \longrightarrow 9$$

differentiating eq @ using @ and equality to zero wa get the optimum value.

> optimum tereshed d

From eq (8)
$$P_{e} = P_{0} \frac{1}{2} \operatorname{erfc} \left[\frac{A+\lambda}{\sqrt{N_{0}/T_{b}}} \right] + P_{1} \frac{1}{2} \operatorname{erfc} \left[\frac{A-\lambda}{\sqrt{N_{0}/T_{b}}} \right]$$



(1 Mark)

(OR)

15

4

4

15. b.

(a) In digital CW communication system, the bit rate of NRZ data stream is 1 Mbps and carrier frequency is 100 MHz. Find the symbol rate of transmission and bandwidth requirement of the channel in the following cases of different techniques used. (1) BPSK system (2) QPSK system (3)16ary PSK system. (6 Marks)

Answer:

- (1) BPSK system (2 Marks-> Symbol rate 1 Mark BW 1 Mark)
- (2) QPSK system (2 Marks-> Symbol rate 1 Mark BW 1 Mark)
- (3)16ary PSK system (2 Marks-> Symbol rate 1 Mark BW 1 Mark)

Sol: Given data

The bit rate is,

 $f_b = 1 \text{ Mbps} = 1 \times 10^6$

Hence

$$T_b = \frac{1}{f_b} = \frac{1}{1 \times 10^6} = 1 \times 10^{-6} \text{ or } 1 \text{ µs.}$$

Carrier frequency $f_0 = 100 \text{ MHz}$

(i) BP\$K system

Bandwidth of BPSK system is given as,

$$BW = 2 f_b$$
$$= 2 \times 1 \times 10^6 = 2 \text{ MHz}$$

In BPSK, one bit is considered as one symbol.

$$T_s = T_b = 1 \times 10^{-6}$$

$$\therefore \text{ Symbol rate } = \frac{1}{T_s} = \frac{1}{1 \times 10^{-6}} = 1 \times 10^{-6} \text{ symbols/sec.}$$

Note that this rate is same as bits/sec.

(ii) QPSK system

Bandwidth of QPSK is given as,

BW =
$$f_b = 1 \times 10^6$$
 i.e. 1 MHz

$$T_s = 2 T_b$$

=
$$2 \times 1 \times 10^{-6}$$
 or $2 \, \mu sec$

Hence, symbol rate =
$$\frac{1}{T_s}$$

$$=\frac{1}{2\times10^{-6}}=500\times10^{3} \text{ symbols/sec}$$

(iii) 16-ary PSK system

Equation 3.6.13 gives the bandwidth of M-ary PSK as,

$$BW = \frac{2f_b}{N}$$

Here 'N' is the number of bits used to make one symbol. Here there are total 16 sy

$$N = \log_2 M$$
$$= \log_2 16 = \frac{\log_{10} 16}{\log_{10} 2} = 4$$

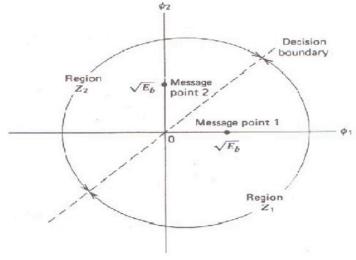
Thus 4 bits make 16 symbols. Hence bandwidth of the 16-ary PSK becomes,

$$BW = \frac{2 \times 1 \times 10^6}{4} = 500 \text{ kHz}$$

$$T_s = NT_b$$

= $4 \times 1 \times 10^{-6} = 4 \times 10^{-6}$
rate = $\frac{1}{T_s}$
= $\frac{1}{4 \times 10^{-6}} = 250 \times 10^3$ symbols/sec

(b) Draw the constellation diagram of FSK, PSK and QPSK. (6 Marks)



(2 Marks)

Fig. Signal Space diagram of Coherent binary FSK system.

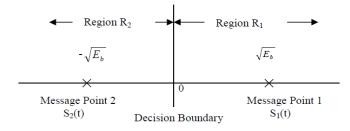
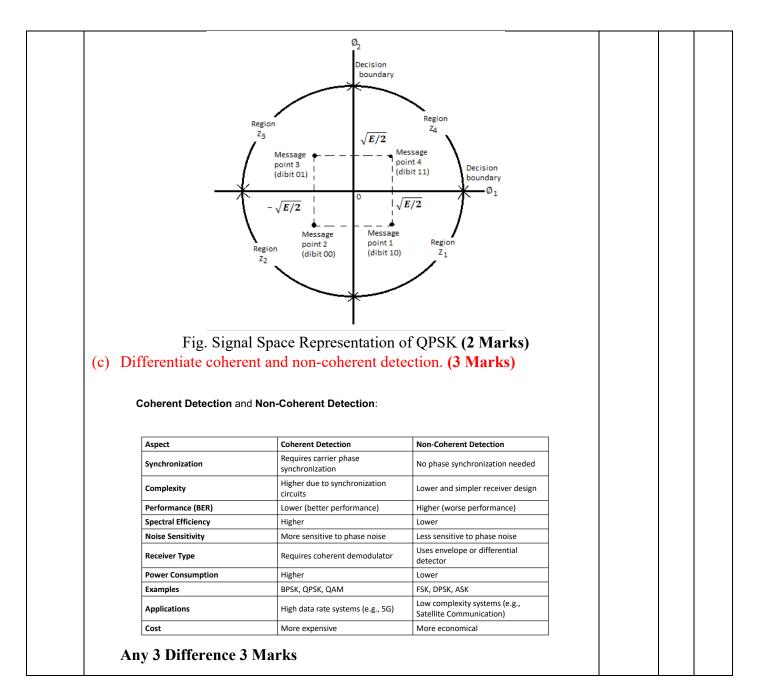
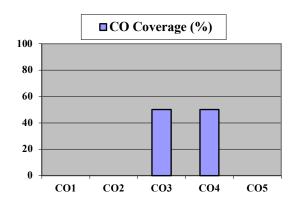


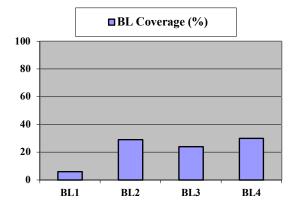
Fig. Signal Space Representation of BPSK

(2 Marks)



Course Outcome (CO) and Bloom's level (BL) Coverage in Questions





Evaluation Sheet

Name of the Student:

Register No.:

		Part- A (11x 1= 11	Marks)	
Q. No	CO	Maximum Marks	Marks Obtained	Total
1	3	1		
2	3	1		
3	3	1		
4	3	1		
5	3	1		
6	4	1		
7	4	1		
8	4	1		
9	4	1		
10	4	1		
11	4	1		
		$Part - B (3 \times 8 = 24)$	Marks)	
12 a	3	8		
12 b	3	8		
13 a	3	8		
13 b	4	8		
14 a	4	8		
14 b	4	8		
		Part - C (1 x 15 = 15	Marks)	
15 a	3	15		
15 b	4	15		

Consolidated Marks:

СО	Maximum Marks	Marks Obtained
3	44	
4	45	
Total	89	

Signature of Course Teacher

Signature of the Course Coordinator

Signature of the Academic Advisor