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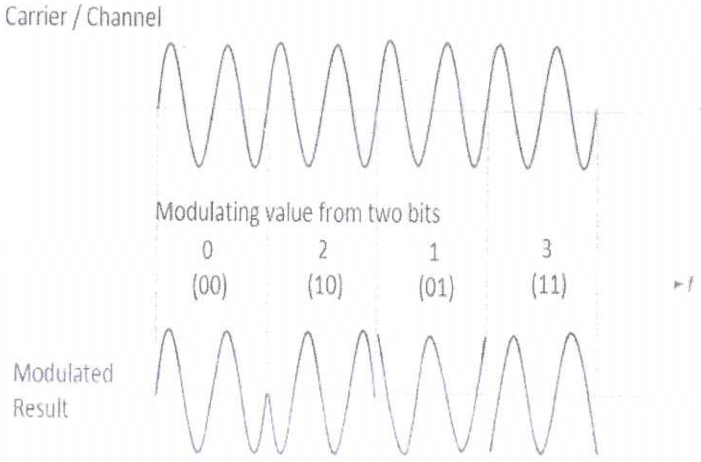
April: 2024

SCORING INDICATORS

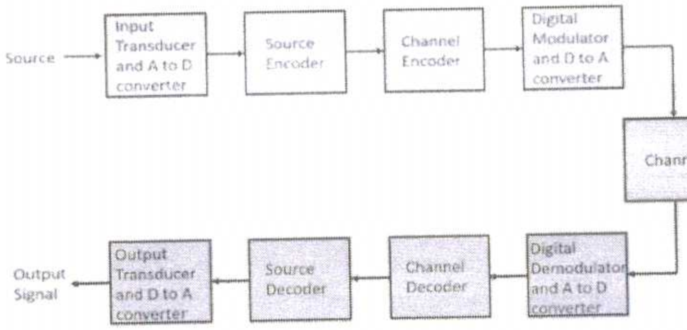
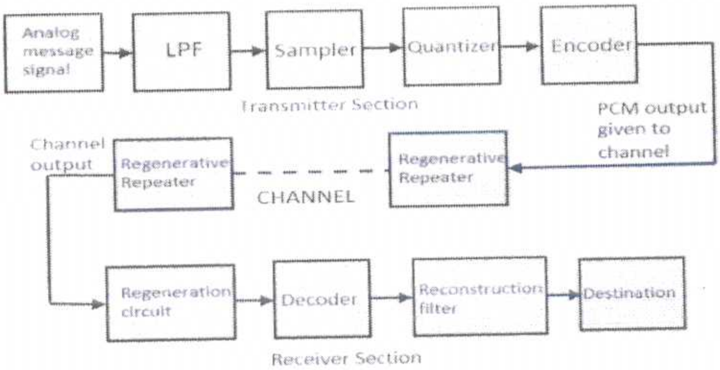
COURSE NAME : DIGITAL COMMUNICATION
COURSE CODE : 5043D QID:

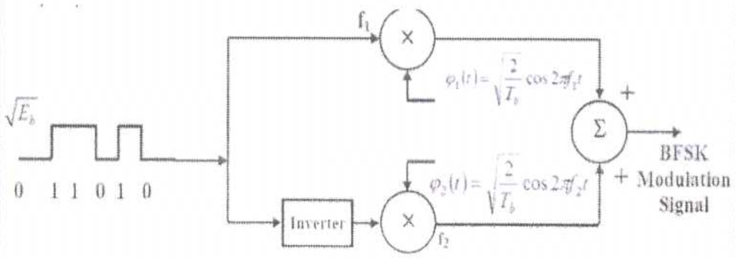
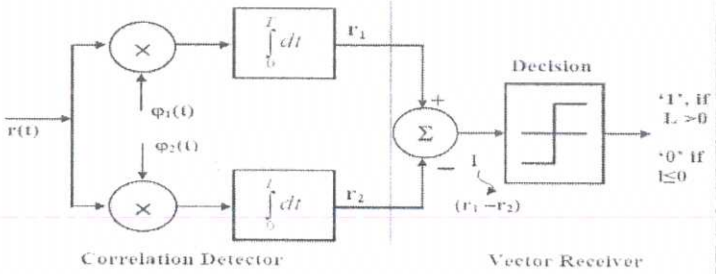
Q No	Scoring Indicators	Split score	Sub Total	Total score
	PART A			**9**
I. 1	Pulse Code Modulation.	1	1	9
I. 2	Quantisation.	1	1	
I. 3	1.Slope overload distortion. 2. Granular noise.	0.5+0.5	1	
I. 4	Amplitude Shift Keying (ASK).	1	1	
I. 5	Time Division Multiplexing (TDM) and Frequency Division Multiplexing (FDM).	0.5+0.5	1	
I. 6	Entropy is the measure of the average information content per source symbol denoted by H(x).	1	1	
I. 7	Hamming code / Convolution code. (any one code can be written)	1	1	
I. 8	FHSS (Frequency Hop Spread Spectrum)	1	1	
I. 9	TDMA and CDMA	0.5+0.5	1	
	PART B			**24**
II. 1	The sampling theorem states that a continuous time signal may be completely represented in its samples and recovered back if the sampling frequency $f_s \geq 2f_m$. Here f_s is the sampling frequency and f_m is the maximum frequency present in the signal. $f_s \geq 2f_m$ *Definition (2 marks) + equation (1mark)*	Definition (2 marks) + equation (1mark)	3	
II. 2	Companding is a technique of achieving non-uniform quantization. Companding is done to improve the signal to noise ratio of weak signals. Companding model consists of a compressor, a uniform quantizer and an expander.  Companding Model	Figure (2 marks + explanation (1 mark)	3	

	Companding is formed by merging the compression and expanding. Initially at the transmitting end the signal is compressed and further at the receiving end the compressed signal is expanded in order to have the original signal. *Figure (2 marks + explanation (1 mark)*			
II. 3	1. The effect of distortion, noise and interference is much less in digital signals. 2. Digital circuits are more reliable. 3. Digital circuits are easy to design and cheaper than analog circuits. 4. Cross-talk is very rare in digital communication. 5. Encryption and compression are employed in digital circuits to maintain the secrecy of the information. 6. Probability of error occurrence is reduced by employing error detecting and error correcting codes. 7. Spread spectrum technique is used to avoid signal jamming. 8. The capacity of the channel is effectively used by digital signals. 9. Digital signals can be saved and retrieved more conveniently than analog signals. *(Any 3 can be written)*	1 x 3	3	
II. 4	DPCM is a technique of analog to digital signal conversion. If the redundancy in PCM is reduced, then the overall bitrate will decrease and the number of bits required to transmit one sample will also reduce. This type of digital pulse modulation technique is called differential pulse code modulation. The DPCM works on the principle of prediction. The value of the present sample is predicted from the previous samples. The prediction may not be exact, but it is very close to the actual sample value. This technique samples the analog signal and then quantizes the difference between the sampled value and its predicted value, then encodes the signal to form a digital value.		3	

II. 5			3	
II. 6	Coding is done for the purpose of designing efficient and reliable data transmission methods. This involves 1. Removal of redundancy 2. Detection of errors. 3. Correction of errors in the transmitted data.	1 x 3	3	
II. 7	A coded sequence of 1s and 0s with certain auto-correlation properties is called as Pseudo-Noise sequence. It is used in spread spectrum techniques. It is a maximum-length sequence, which is a type of cyclic code. Both the transmitter and receiver are informed of the PN sequence, which is a deterministic, periodic signal. The signal appears to have the statistical properties of sampled white noise since the signal is deterministic. Thus to an unauthorized listener, it appears as a random signal.		3	
II. 8	Code Division Multiple Access (CDMA) is a sort of multiplexing technique that facilitates various signals to occupy a single transmission channel. It optimizes the use of available bandwidth. The technology is commonly used in ultra-high-frequency (UHF) cellular telephone systems, bands ranging between the 800-MHz and 1.9-GHz. In CDMA, a user has access to the whole bandwidth for the entire duration. The basic principle is that different CDMA codes are used to distinguish among the different users. Techniques generally used are direct sequence spread		3	

	spectrum modulation (DS-CDMA), frequency hopping or mixed CDMA detection (JDCDMA). Here, a signal is generated which extends over a wide bandwidth. A code called **spreading code** is used to perform this action. Using a group of codes, which are orthogonal to each other, it is possible to select a signal with a given code in the presence of many other signals with different orthogonal codes.			
II.9	Orthogonal Frequency Division Multiplexing (OFDM) is a method of digital data modulation, whereby a single stream of data is divided into several separate sub-streams for transmission via multiple channels. OFDM uses the principle of frequency division multiplexing (FDM), where the available bandwidth is divided into a set of sub-streams having separate frequency bands. In OFDM, the sub-streams in which the main signal is divided are orthogonal to each other.		3	
II.10	1. Direct-sequence spread spectrum (DSSS) is a technique that transmits a data signal over a range of frequencies, spreading it uniformly across the allocated spectrum. 2. DSSS is used to ensure that a particular frequency band and its corresponding range of frequencies is kept free from interference. 3. This technique can be related to escaping the problem of co-channel interference (like two different wireless networks transmitting on the same frequency band) and cross-talk interference. 4. Direct-sequence spread spectrum employs error correction coding at the transmitter and the receiver. *(Any 3 features)*	1 x 3	3	

	PART C			42
III	 Basic Elements of a Digital Communication System *Figure (4 marks) + Explanation (3 marks)*	Fig (4 marks) + Expln. (3 marks)	7	7
IV	PCM block diagram consists of a transmitter section, transmission path (channel) and a receiver section.  • The transmitter section consists of low pass filter, sampler, quantizer and encoder. • The LPF prior to sampling prevents aliasing of the message signal. • In sampling section the analog signal is sampled according to sampling theorem resulting in a discrete time signal. The quantizing and encoding operation are performed in the same section which is known as analog to digital converter and the signal is transmitted. • The transmitting signal is travelled through the transmission path. The signal may get distorted. The distorted signal is regenerated by a regenerative repeater. This section increases the signal strength. • At receiving section, there will be a regenerative circuit,	Fig (4 marks) + Expln. (3 marks)	7	

	decoder and a filter. The PCM signal is regenerated and decoded (i.e., actual it is DAC) that maybe filtered by the filter and the analog signal maybe reproduced. *Figure (4 marks) + Explanation (3 marks)*			
V	FSK Modulator The FSK modulator block diagram comprises of two oscillators with a clock and the input binary sequence.  Input random binary sequence is represented by '1' and '0' where '0' represents no voltage at the input of the multipliers. A '0' input to the inverter results in a '1' at its output. Inverter, along with the two multipliers and the summing unit behave as a 'switch 'which selects output of one of the two oscillators. FSK Demodulator  r(t) is the noisy received signal. Demodulator consists of two correlators. With common input r(t). Locally generated coherent reference signal $\Phi1(t)$ and $\Phi2(t)$. Correlator output are r1 and r2. Correlated output is subtracted and the difference 'L' is compared with a threshold of '0' volts.L= r1-r2 If L > 0, '1' is transmitted If L < 0 , '0' is transmitted *Modulator (4 marks) + Demodulator (3 marks)*	Modulator (4 marks) + Demodulator (3 marks)	7	

<table>
<tr>
<td>VI</td>
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<table>
<tr><th>Key</th><th>TDM</th><th>FDM</th></tr>
<tr><td>Definition</td><td>TDM stands for Time Division Multiplexing</td><td>FDM stands for Frequency Division Multiplexing</td></tr>
<tr><td>Signal</td><td>TDM works well with both anlaog as well as digital signals.</td><td>FDM works only with analog signal.</td></tr>
<tr><td>Conflict</td><td>TDM has low conflict.</td><td>FDM has high conflict.</td></tr>
<tr><td>Wiring</td><td>Wiring or chip of TDM is simpler.</td><td>Wiring or chip of FDM is complex.</td></tr>
<tr><td>Efficiency</td><td>TDM is efficient.</td><td>FDM is quite inefficient.</td></tr>
<tr><td>Sharing</td><td>Time is shared in TDM.</td><td>Frequency is shared in FDM.</td></tr>
<tr><td>Required input</td><td>Synchronization pulse is mandatory in TDM.</td><td>Synchronization pulse is not mandatory.</td></tr>
</table>

(Any 7 differences)
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<td>1 x 7</td>
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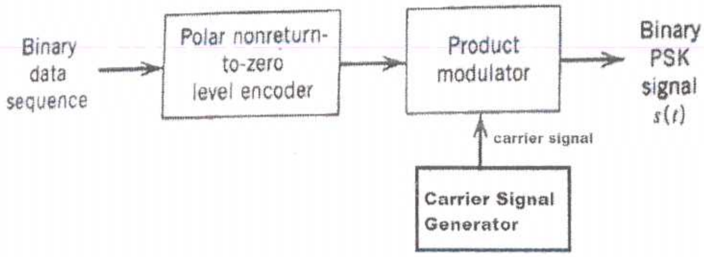
<td>VII</td>
<td>
<u>BPSK Modulator</u>



It consists of

- Polar non return to zero encoder.

- Balance modulator (Product Modulator) which has the carrier sine wave as one input and the binary sequence as the other input.
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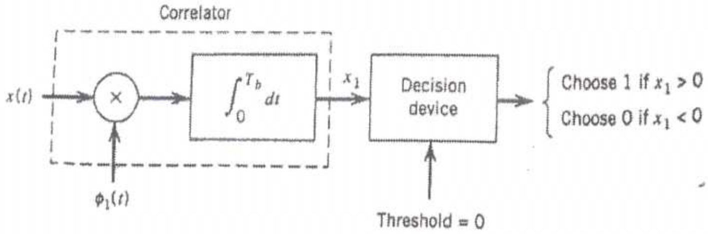
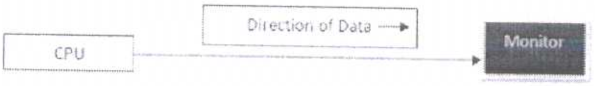
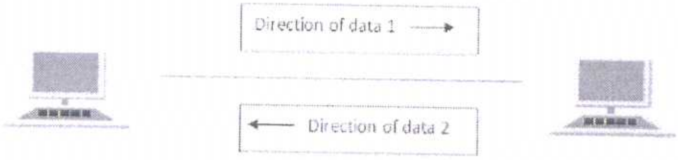
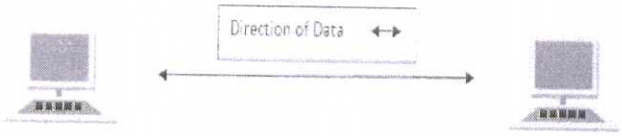
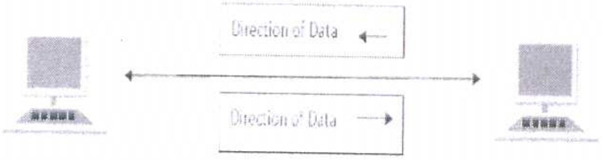
	The i/p of the bipolar NRZ encoder is a binary signal (0 or 1) which is converted into a NRZ signal by the encoder. The bipolar signal is applied as modulating signal to the balanced modulator. The balanced modulator converts that signal into BPSK by modulating with the carrier signal. If the digital signal is binary '0' bipolar NRZ equals -1 and BPSK output signal is $-\sqrt{2P}\cos(2\pi f_c t)$. If the digital signal is binary '1' bipolar NRZ equals +1 and BPSK output signal is $\sqrt{2P}\cos(2\pi f_c t)$. BPSK Demodulator  BPSK demodulator consists of - Correlator & Decision device Noisy PSK x(t) is applied to a correlator to which reference carrier signal $\Phi_1(t)$ is also given. Reference signal frequency is equal to the carrier frequency in transmitter. When two sinusoids of equal frequency but different phase is multiplied, the result is a sinusoid of double the frequency plus an offset that depends on the difference between the two phases. If the phase of the received signal is equal to the phase of the reference signal, the offset is cos0°, which equals 1. If the phase of the received signal is 180° different from the phase of the reference signal, the offset is cos180°which is –1. This offset can be used to interpret each symbol as a '0' or '1'. Correlator output x_1(offset)is compared with a threshold value of '0'. If $x_1 >$ '0', receiver decides in favor of symbol 1. If $x_1 <$ '0', receiver decides in favor of symbol 0. *Modulator (4 marks) + Demodulator (3 marks)*	Modulator (4 marks) + Demodulator (3 marks)	7	

VIII	1. <u>Simplex Mode</u> - In this type of transmission mode data can be sent only through one direction i.e. communication is unidirectional. We cannot send a message back to the sender. Examples of simplex mode is loudspeaker, television broadcasting, television and remote, keyboard and monitor etc.  2. <u>Half Duplex Mode</u> - Half-duplex data transmission means that data can be transmitted in both directions on a signal carrier, but not at the same time. Example of half duplex is a walkie- talkie in which message is sent one at a time and messages are sent in both the directions.  3. <u>Full duplex mode</u> - In full duplex system data is send in both directions as it is bidirectional. Data can be sent in both directions simultaneously. Data can be send as well as received simultaneously. Example of Full Duplex is a Telephone Network in which there is communication between two persons by a telephone line, through which both can talk and listen at the same time.  In full duplex system there can be two lines one for sending the data and the other for receiving data.  	Simplex (2 marks) + duplex (2 marks) + full duplex (3 marks)	7	

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<td>IX</td>
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$$H = \begin{bmatrix} 1 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 1 & 0 \\ 1 & 0 & 1 & 1 & 0 & 0 & 1 \end{bmatrix} \quad \text{Also, n=7, k=4}$$

$H = [\, P^T : I_{n-k} \,]$

$$P^T = \begin{bmatrix} 1 & 1 & 1 & 0 \\ 1 & 1 & 0 & 1 \\ 1 & 0 & 1 & 1 \end{bmatrix}$$

$$\therefore\ P = \begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 0 \\ 1 & 0 & 1 \\ 0 & 1 & 1 \end{bmatrix}$$

Generator matrix is given by $G = [\, I_k : P \,]$

$$\therefore\ G = \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 & 1 \\ 0 & 1 & 0 & 0 & 1 & 1 & 0 \\ 0 & 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 0 & 1 & 0 & 1 & 1 \end{bmatrix}$$

Find P (3marks) + Find G (4 marks)

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<td>Find P (3marks) + Find G (4 marks)</td>
<td>7</td>
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<td>X</td>
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<table>
<tr><th>X</th><th>P(x)</th><th>S₁</th><th>S₂</th><th>S₃</th><th>S₄</th><th>Code</th><th>Code Length</th></tr>
<tr><td>x1</td><td>0.3</td><td>0</td><td>0</td><td></td><td></td><td>00</td><td>2</td></tr>
<tr><td>x2</td><td>0.25</td><td>0</td><td>1</td><td></td><td></td><td>01</td><td>2</td></tr>
<tr><td>x3</td><td>0.2</td><td>1</td><td>0</td><td></td><td></td><td>10</td><td>2</td></tr>
<tr><td>x4</td><td>0.12</td><td>1</td><td>1</td><td>0</td><td></td><td>110</td><td>3</td></tr>
<tr><td>x5</td><td>0.08</td><td>1</td><td>1</td><td>1</td><td>0</td><td>1110</td><td>4</td></tr>
<tr><td>x6</td><td>0.05</td><td>1</td><td>1</td><td>1</td><td>1</td><td>1111</td><td>4</td></tr>
</table>

$$H(x) = \sum_{i=1}^{m} P(x_i)\log_2(1/P(x_i))$$

$$= 0.3\log_2 1/0.3 + 0.25\log_2 1/0.25 + 0.2\log_2 1/0.2 + 0.12\log_2 1/0.12 + 0.08\log_2 1/0.08 + 0.05\log_2 1/0.05$$

$\therefore\ H = 2.36$ bits/symbol

Find Shannon Fano code (4 marks) + Find H(x) (3 marks)

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<td>Find Shannon Fano code (4 marks) + Find H(x) (3 marks)</td>
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<td>XI</td>
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In convolution code a block of n bits generated by encoder in a time slot depends not only on the k message bits but also on the preceding L blocks of the message bit where L is greater than one.

Encoding by convolutional codes

For generating a convolutional code, the information is passed sequentially through a linear finite-state shift register. The

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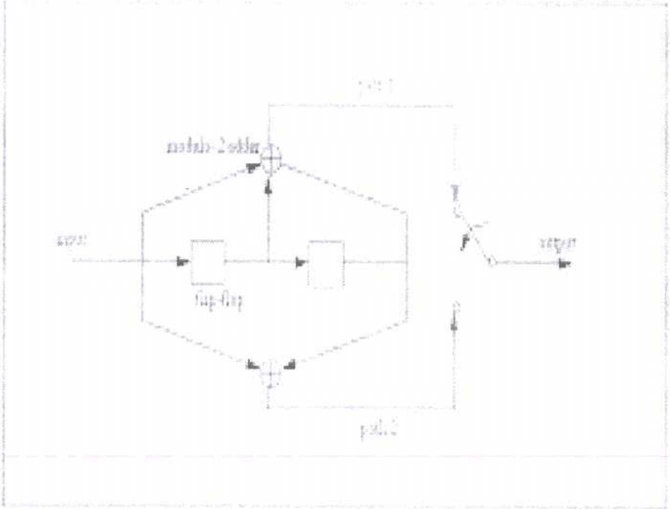
	shift register comprises of (-bit) stages and Boolean function generators. A convolutional code can be represented as (n,k, K) where - *k* is the number of bits shifted into the encoder at one time. Generally, *k = 1*. - n is the number of encoder output bits corresponding to k information bits. - The code-rate, $R_c = k/n$. - The encoder memory, a shift register of size k, is the constraint length. - *n* is a function of the present input bits and the contents of *K*. - The state of the encoder is given by the value of $(K - 1)$ bits.  *(Explanation 4 marks + figure 3 marks)*	Expln. (4 marks) + figure (3 marks)		
XII	Hamming codes are defined as (n,k) linear block codes ○ Number of check bits $m \geq 3$ ○ Block length $n = 2^m - 1$ ○ Number of message bits $k = n-m$ ○ Minimum distance $d_{min} = 3$ ○ Code rate $r = k/n = (n-m)/n$ $r = 1-(m/n)$			

	◦ Detect 2 errors ($d_{min} \geq t+1$) ◦ Correct one error ($d_{min} \geq 2t+1$) Generator Matrix ▶ Matrix used to generate the code words from the given data bits ▶ In an (n,k) linear block code C , a code vector c is given by c =dG $\mathbf{c} = \mathbf{d}G = [d_1\ d_2\ \dots\ d_k]\begin{bmatrix} 1 & 0 & \dots & 0 & p_{11} & p_{21} & \cdots & p_{m1} \\ 0 & 1 & \dots & 0 & p_{12} & p_{22} & \cdots & p_{m2} \\ \dots & \dots & \dots & \dots & \dots & \dots & \dots & \dots \\ 0 & 0 & \dots & 1 & p_{1k} & p_{2k} & \cdots & p_{mk} \end{bmatrix}$ $G = [I_k P]$ ▶ Generator matrix have 'k' rows and 'n' columns i.e. G is a 'k x n' matrix ▶ Generator matrix is used in the encoder side. Parity Check Matrix ▶ Used in the receiver side to check whether the received word is correct or not ▶ Denoted by H; $H = [\ P^T : I_m]$ ▶ H is an 'm x n' matrix ▶ C is the codeword $cH^T = dGH^T = 0$ Syndrome decoding ▶ Let 'r' be the received word of length 'n' when codeword 'c' of length 'n' was sent over a noisy channel ▶ Then $r = c \oplus e$; where 'e' is the error pattern. ▶ To check whether the received vector is erroneous or not • Find the syndrome vector $S = rH^T$ • If the syndrome vector is '0', no error occurred • If the syndrome vector is not '0', error occurred	Hamming code generation (4marks) + syndrome decoding (3marks)	7	

<table>
<tr><td></td><td colspan="3">▸ To correct the detected error, find the bit position where error has occurred. i.e., find the row position of syndrome vector in H^T matrix.
▸ Row number will give the error bit position in the received word.
▸ Change the bit'0' to '1' accordingly for error correction.
Hamming code generation (4marks) + syndrome decoding (3marks)</td><td></td><td></td><td></td></tr>
<tr><td rowspan="9">XIII</td><td>Sl. No.</td><td>FDMA</td><td>TDMA</td><td rowspan="9">1 x 7 = 7</td><td rowspan="9">7</td><td rowspan="9"></td></tr>
<tr><td>1.</td><td>FDMA stands for Frequency Division Multiple Access.</td><td>TDMA stands for Time Division Multiple Access.</td></tr>
<tr><td>2.</td><td>Overall bandwidth is shared among number of stations.</td><td>Time sharing of satellite transponder takes place.</td></tr>
<tr><td>3.</td><td>Guard bands between adjacent channels is necessary.</td><td>Guard time between adjacent slots is necessary.</td></tr>
<tr><td>4.</td><td>Synchronization is not necessary.</td><td>Synchronization is necessary.</td></tr>
<tr><td>5.</td><td>Power efficiency is less.</td><td>Power efficiency is high.</td></tr>
<tr><td>6.</td><td>It requires stability of high carrier efficiency.</td><td>It does not require stability of high carrier efficiency.</td></tr>
<tr><td>7.</td><td>It is basically used in GSM and PDC.</td><td>It is basically used in advanced mobile phone systems.</td></tr>
<tr><td colspan="3">(Any 7 differences can be written)</td></tr>
<tr><td>XIV</td><td colspan="3">In the frequency-hopping spread spectrum (FHSS), each component is transmitted at a different carrier frequency. In FHSS systems, the transmitted power is concentrated on one or a few carriers at a time. The carrier frequencies are chosen</td><td></td><td></td><td></td></tr>
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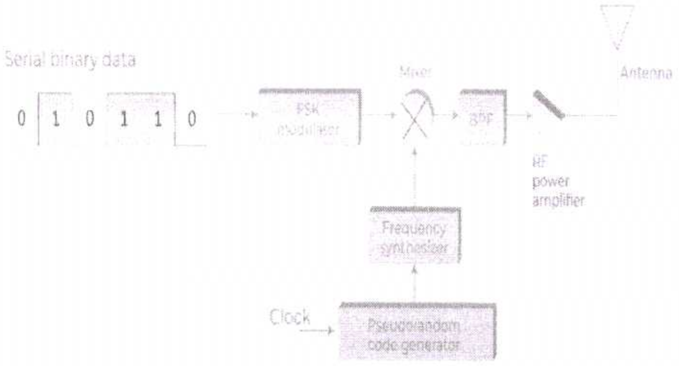
	in accordance with a pseudo-random sequence or hopping sequence that changes periodically, so as to prevent long-term predictability of the carrier frequencies used. The receiver correlates received signals against the sequence of the received signals to determine which doesn't interfere from noise and interference.  Advantages of FHSS - The processing gain PG is higher than that of DSSS system. - The synchronization is not greatly dependent on the distance. - The serial search system with FHSS needs shorter time for acquisition. *Figure (3 marks) + explanation (4 marks)*	Figure (3 marks) + expln. (4 marks)	7	