WINTER – 19 EXAMINATIONS

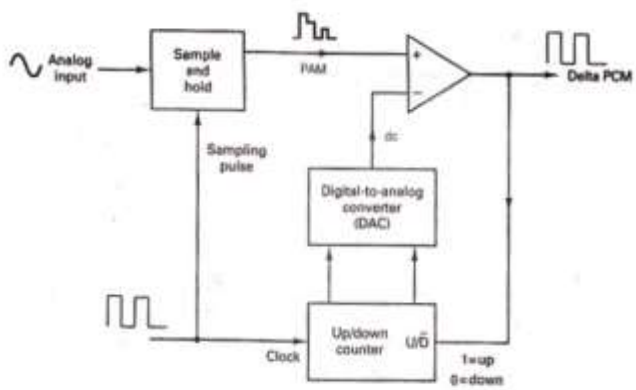
Subject Name: Digital Communication Systems **Model Answer** Subject Code: 22428

Important Instructions to examiners:

1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given more Importance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in the figure. The figures drawn by candidate and model answer may vary. The examiner may give credit for any equivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constant values may vary and there may be some difference in the candidate's answers and model answer.
6) In case of some questions credit may be given by judgement on part of examiner of relevant answer based on candidate's understanding.
7) For programming language papers, credit may be given to any other program based on equivalent concept.

Q. No.	Sub Q. N.	Answer	Marking Scheme
Q.1		**Attempt any FIVE of the following:**	**10-Total Marks**
	a)	**Define (i)Bit rate (ii)Baud rate**	**2M**
	Ans:	**(i)Bit rate :-** Bit rate is simply the number of bits transmitted during one second and is expressed in bits per second (bps). Mathematically bit rate is given by:- $Rb = 1 / T_b$ where Tb is time interval of one bit **(ii)Baud rate: -** Baud is the unit of symbol rate. Baud rate is the number of symbols transmitted during one second and is expressed in symbols per second or baud. Mathematically, baud rate is the reciprocal of the time of one output signaling element and a Signaling element (symbol) may represent several information bits. Baud rate is expressed as, $Rs = 1 / T_s$ Where, baud rate = symbol rate (symbols per second) and T_s = time interval of one symbol.	**1M each**
	b)	**State the Hartley's law with mathematical expression.**	**2M**
	Ans:	**Hartley's Law / Nyquist Theorem:-** **Statement:** Hartley's Theorem/Law states that the channel capacity of the transmission channel of bandwidth 'B' which carries a signal having 'M' levels in the total absence of noise is given by: $\mathbf{C = 2\,B\,log\,2\,M}$ where, C – channel capacity (bits/sec) B – channel bandwidth M – number of coding levels (2 or more) In the absence of noise, Hartley's Law shows that greater the number of levels in the coding system, the greater the information rate that can be sent through the channel.	

	c)	**State sampling theorem. Define Nyquist rate.**	**2M**
	Ans:	**SAMPLING THEOREM:** Sampling theorem states that a band-limited signal of finite energy having the highest frequency component f_m Hz can be represented and recovered completely from a set of samples taken at a rate of f_s samples per second provided that $\boldsymbol{f_s \geq 2f_m}$. Here f_s *is* the *sampling frequency*. This theorem is also known as the *Sampling Theorem for Baseband or Low-pass Signals.* **Nyquist rate:-** Sampling frequency should be equal to or greater than twice the maximum signal frequency ($f_s \geq 2f_m$)	**1M each**
	d)	**Classify the modulation techniques.**	**2M**
	Ans:	**Classification of the modulation techniques:-** 1. Amplitude Shift Keying (ASK) 2. Frequency Shift Keying (FSK) **3.** Phase Shift Keying (PSK)	**2M**
	e)	**State two advantages of WDM technique.**	**2M**
	Ans:	**ADVANTAGES OF WDM:** 1. WDM has enhanced capacity. 2. WDM can be used for full duplex transmission with a single fiber. 3. It is inherently easier to reconfigure (addition or removal of channels). 4. Fiber optic cable networks use optical components which are simpler and more reliable and often less costly than their electronic counterparts	**Any 2 1M each**
	f)	**List the various multiple access techniques.**	**2M**
	Ans:	1. Frequency Division Multiple Access (FDMA) 2. Time Division Multiple Access (TDMA) 3. Code Division Multiple Access (CDMA) 4. Space Division Multiple Access (SDMA)	**½ M each**
	g)	**Define the concept of spread spectrum.**	**2M**
	Ans:	**Concept of spread spectrum :-** Spread-spectrum techniques are methods by which a signal (e.g. an electrical, electromagnetic, or acoustic signal) generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth. **OR** Spread spectrum systems are intended to provide such secure and reliable communication. In this system the spectrum of the transmitted signals spreaded over a very wide bandwidth. This achieved in these systems by modulating for a second time, an already modulated signal in such a way as to spread the power of the transmitted spread spectrum signal over a very large bandwidth.	**2M**
Q.2		**Attempt any THREE of the following:**	**12-Total Marks**
	a)	**State the advantages and disadvantages of digital communication system.**	**4M**

Ans:	**Advantages of Digital Communication : (any 2)** 1. High noise interference tolerance due to digital nature of the signal. 2. With channel coding, error detection and correction at receiver is possible. 3. It provides us added security to our information signal i.e. Data encryption is possible for greater security. 4. Cheaper due to advances in digital VLSI technology. 5. Digital information can be saved and retrieved when necessary. 6. Large data storage is possible. **Disadvantages of Digital Communication : (any 2)** 1. Large System Bandwidth: - Digital transmission requires a large system bandwidth to communicate the same information in a digital format as compared to analog format. 2. High power consumption (Due to various stages of conversion). 3. Needs synchronization 4.Sampling Error.	**1M each**
b)	**Draw the block diagram of DM transmitter.Explain each block in detail.**	**4M**
Ans:	**Block diagram of DM transmitter:-**  **Explanation:-** **Sample and Hold:-** The input analog is sampled and converted to PAM signal, which is compared with the output of the DAC. The output of the DAC is a voltage equal to the regenerated magnitude of the previous sample, which was stored in the up-down counter as a binary number. **Up-down counter:-** The up-down counter is incremented or decremented depending on whether the previous sample is larger or smaller than the current sample. The up-down counter is clocked at a rate equal to the sample rate. Therefore the up-down counter is updated after each comparison. Initially the up-down counter is zeroed and DAC output is 0v. The first sample is taken and converted to a PAM signal, and compared with zero volts. The output of the comparator is a logic 1 condition (+v), indcating that the current sample is larger in amplitude than the previous sample. On the next clock pulse, the up- down counter is incremented to a count of 1.The DAC now outputs a voltage equal to the mgnitude of the minimum step size (resolution). The	**2M** **2M**

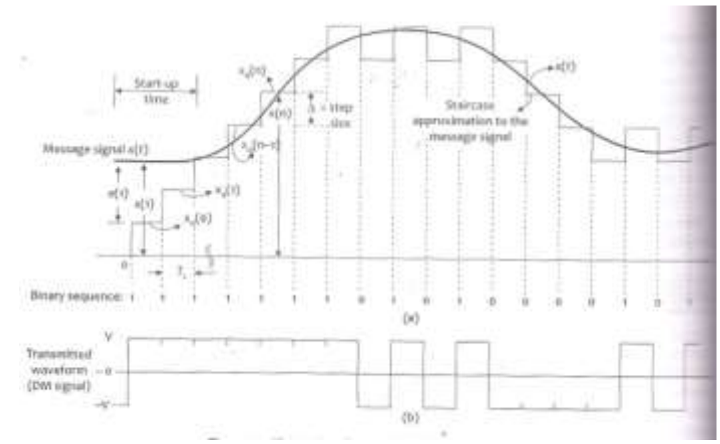
steps change at a rate equal to the clock frequency (sample rate).

Consequently, with the input signal shown, the up-down counter follows the input analog signal up until the output of the DAC exceeds the analog sample; then the up-down counter will begin counting down until the output of the DAC drops below th sample amplitude.

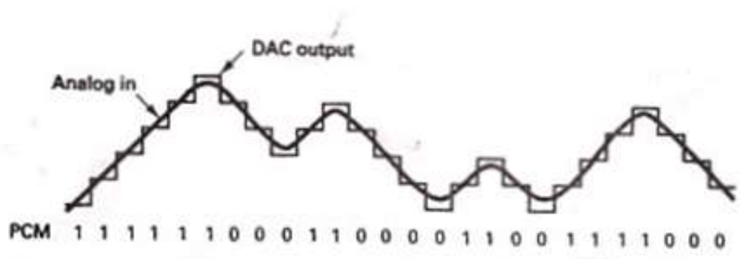
Digital to Analog Converter (DAC):-

In the idealized situation the DAC output follows the input signal. Each time the up-down counter is incremented, a logic 1 is transmitted, and each time the up-down counter is decremented, alogic 0 is transmitted.

<u>(Waveform is optional):-</u>



OR



c)	**Explain flat top sampling with circuit diagram.Draw flat top sampled signal.**	**4M**

Ans:

Flat top sampling:

- In flat top sampling, the top of the samples remains constant and equal to the instantaneous value of the modulating signal at the start of the sampling.
- Thus the amplitude of the pulse after sampling is kept constant and the top of the sampled pulse do not follow the contour of the modulating signal unlike Natural sampling.
- The duration of each sample is τ and the sampling rate is : $F_s = 1/T_s$. $T_s = 1/F_s$
- Sample and hold circuit is used for the generation of the sampled signal to attain flat top sampling, which is shown in the Figure below.

Diagram 1M

Explanation 2M

Waveform 1M

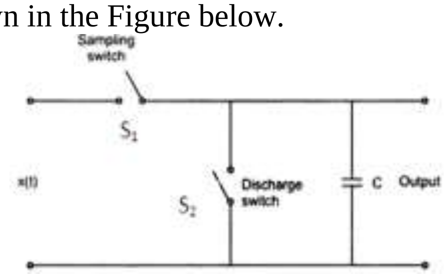
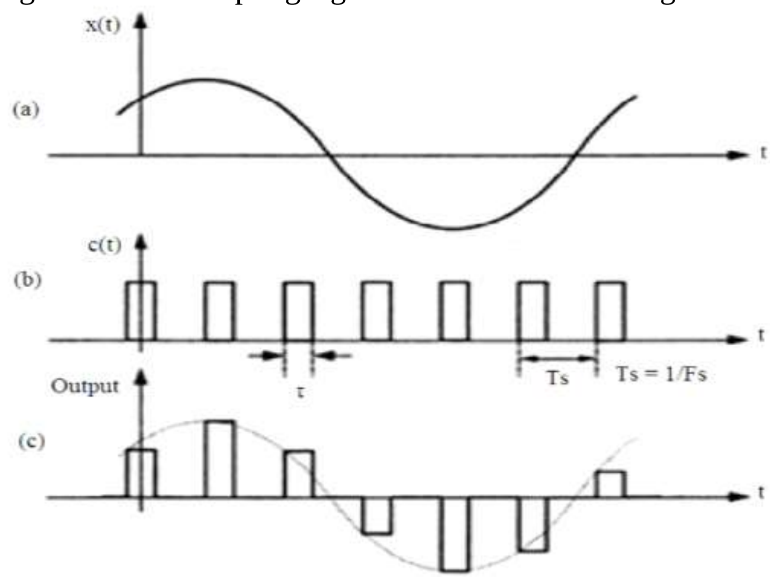
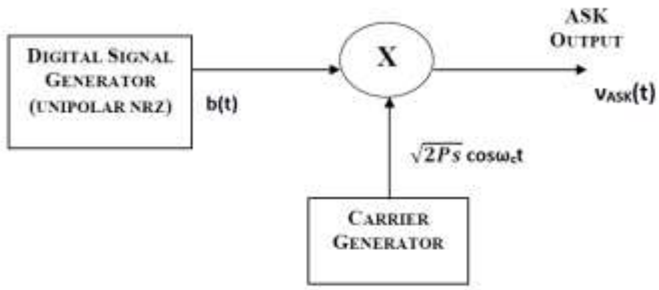
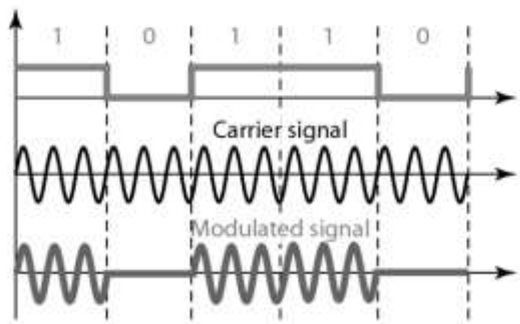
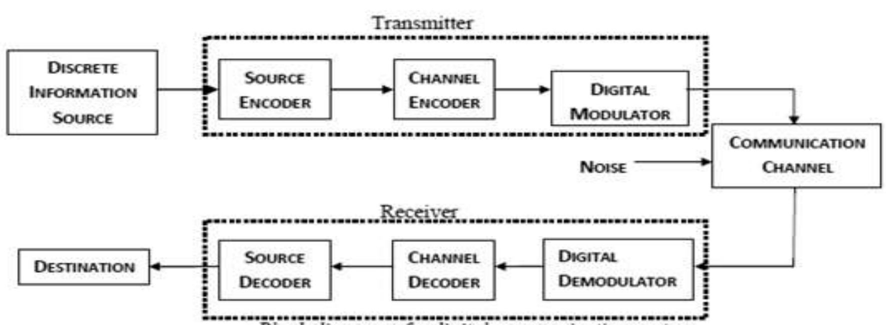
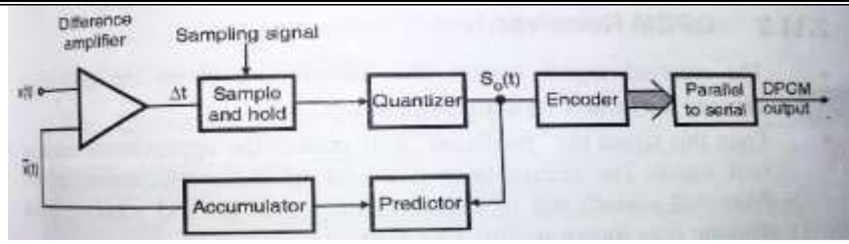


Figure shows the Sample and hold circuit to generate flat top samples

		• The switch S_1 closes at each sampling instant to sample the modulating signal. • The capacitor C holds the sampled voltage for period τ at the end of which switch S_2 is closed in order to discharge the capacitor. • Thus the signal generated as a result of sample and hold process is the flat top sampled signal. The spectrum of the generated flat top sampling signal along with the modulating signal and the sampling signal is shown below in Figure 2 below.  Figure.2 (a) Modulating signal (b) sampling signal and (c) Flat top sampling spectrum • The starting edge of the pulse corresponds to the instantaneous value of the modulating signal x (t). • Flat top sampling can be mathematically considered as convolution of the sampled signal and the pulse signal. • Flat top sampling is mostly used in digital transmission	
	d)	**Describe amplitude shift keying (ASK) modulation with suitable circuit diagram.**	**4M**
	Ans:	**Explanation:-** **ASK MODULATOR:** • The process where a binary information signal directly modulates the amplitude of an analog carrier. The digital signal is used to switch the carrier between amplitude levels is called Amplitude Shift Keying (ASK). • The ASK technique of binary modulation is illustrated in Figure where modulating signal consists of unipolar pulses. Because in this case the carrier is switched ON and OFF, this method is also known as *ON-OFF keying*. • For the entire time the binary input is high, the output is a constant amplitude, constant frequency signal and for the entire time the binary input is low, the carrier is off. • ASK is given by: $v_{ASK}(t) = b(t)\sqrt{2P_s}\cos\omega_c t$ **Block diagram of ASK Transmitter / ASK modulator:-**	**2M** **1M**

		 **Waveforms:-**  	1M
Q.3		**Attempt any THREE of the following:**	**12- Total Marks**
	a)	**Draw the block diagram of digital communication system.Explain the function of source encoder and channel encoder.**	**4M**
	Ans:	 *Block diagram of a digital communication system* **[Note: Any other similar diagram to be considered]** **Source Encoder :** • The input to the source encoder (also referred to as the source coder) is a string of symbols occurring at a rate symbols/sec. • The source encoder converts the symbol sequence into a binary sequence of 0's and 1's by assigning code words to the symbols in the input sequence by using either assigning fixed-length binary code word to each symbol or assigns variable length code words to these blocks. Second function it performs is data compression. **Channel Encoder :** • The channel coder provides some amount of error controlled capability to the data to be transmitted. • It adds some extra bits to the output of the source coder. While these extra bits make it possible for the receiver to detect and/or correct some of the errors in the information bearing bits.	**2M Diagram** **1M for source encoder** **1M for channel encoder**
	b)	**Explain DPCM with block diagram.**	**4M**

Ans:



(Block Diagram of DPCM)

2M diagram

- The above figure shows the block diagram of DPCM transmitter x(t) is the analog input signal and x^(t) is its approximated signal. What is important to know is whether x^(t) is larger or smaller than x(t) and by how much.
- At each sampling instant the difference amplifier compares x(t) and x^(t) and the sample and hold circuit will hold the result of this subtraction.
- The difference signals at the output of sample and hold circuit is quantized by the quantizer. The quantizer output S0(t) is the transmitted as it is or it is encoded into a stream of bits as explained in conventional PCM system.
- The quantizer output is also used to produce the approximated signal x^(t)by passing the quantizer output through a predictor and accumulator.

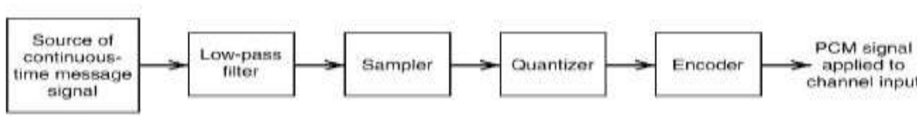
2M explanation

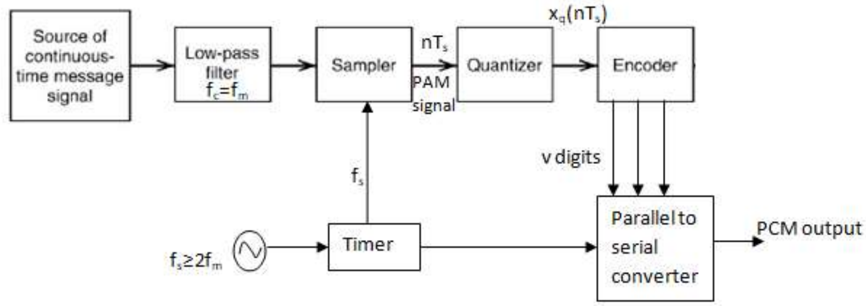
c) **Distinguish between TDMA and CDMA (any four points)** **4M**

Ans:

Sr. No.	Parameter	TDMA	CDMA
1.	Definition	Entire bandwidth is shared among different subscribers at Fixed predetermined or dynamically assigned time intervals/slots.	Entire bandwidth is shared among different users by assigning unique codes.
2.	Bandwidth Available	Time sharing of satellite transponder takes place	Sharing of bandwidth and time both takes place
3.	Synchronization	Synchronization is essential	Synchronization is not necessary
4.	Interference	Due to incorrect synchronization there can be interference between the adjacent time slots.	Both type of interference will be present
5.	Guard bands	Guard times between adjacent timeslots are necessary.	Guard bands and Guard times both are necessary
6.	Active terminals	Terminals are active in their specified slot on same frequency	All terminals active on same frequency
7.	Signal separation	Synchronization in time	Code separation
8.	Near Far Problem	No	Yes
9.	Handoff	Hard handoff	Soft handoff
10.	Application	Advanced mobile phone, system(AMPS), Cordless telephone	IS95 Wide band, CDMA 2000,2.5G and 3G

1 mark for Each point (Any 4 points)

	d)	**Compare FDM & TDM systems (any four points).**	**4M**
	Ans:	<table><tr><th>Sr. No.</th><th>FDM</th><th>TDM</th></tr><tr><td>1</td><td>Divides the channel into the two or more frequency ranges that do not overlap.</td><td>Divides and allocates certain Time periods to each channel.</td></tr><tr><td>2</td><td>Code word is not required</td><td>No coding</td></tr><tr><td>3</td><td>Needs guard bands</td><td>Needs guard time</td></tr><tr><td>4</td><td>Problem of crosstalk</td><td>No problem of crosstalk</td></tr></table>	**1M for each point**
Q.4		**Attempt any THREE of the following :**	**12-Total Marks**
	a)	**State the Shannon Hartley's theorem for channel capacity. Explain the effect of S/N ratio and bandwidth on channel capacity.**	**4M**
	Ans:	In information theory, the **Shannon–Hartley theorem** tells the maximum rate at which information can be transmitted over a communications channel of a specified bandwidth in the presence of noise. According to Shannon, the bandwidth of the channel and signal energy and noise energy are related by the formula $C = W \log_2(1 + S/N)$ where C is channel capacity in bits per second (bps) W is bandwidth of the channel in Hz S/N is the signal-to-noise power ratio (SNR). SNR generally is measured in dB using the formula $(S/N)\ dB = 10 \log(\text{Signal power / Noise power})$ **Effect of S/N on Channel Capacity C:** ● If the communication channel is noiseless then N = 0. Therefore, $S/N \rightarrow \infty$ and so C also will tend to ∞. Thus the noiseless channel will have an infinite capacity. **Effect of Bandwidth B on Channel Capacity C:** ● Consider that some white Gaussian noise is present. Hence (S/N) is not infinite as $N \neq 0$. Now as the bandwidth approaches infinity, the channel capacity C does not become infinite because, $N = \eta B$ will also increase with the bandwidth B. This will reduce the value of S/N with increase in B,assuming the signal power S to be constant.	**1M for statement** **1M for formula** **2M effect of s/n & Bandwidth B on Channel Capacity**
	b)	**Describe PCM transmitter with block diagram.**	**4M**
	Ans:	 **OR**	**2M for block diagram**



Block diagram of PCM transmitter

2M for description

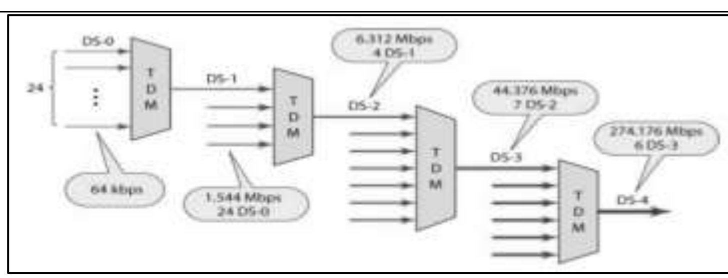
- The analog signal x (t) is passes through band limiting / low pass filter, which has a cut-off frequency fc=W Hz. This will ensure x (t) will not have any frequency component higher than "W". In other words, suppresses high frequency components and passes only low frequency signal to avoid 'aliasing error'.

- The band limited analog signal is then applied to sample and circuit where this circuit acts as modulator and both modulating input signal and sampling signal with adequately high sampling rate are inputs to this circuit. Output sample and hold block is a flat topped PAM signal.
- These samples are subjected to operation "quantization" in the "quantizer". The quantizer is used to reduce effect of noise. Quantization is a process of approximation of the value of respective sample in to a finite number that will reduce data bits. The combined effect of sample and quantization produces is 'Quantized PAM' at the quantizer output.
- The Quantized PAM output is analog in nature. So to transmit it through digital communication system the quantized PAM pulses are applied to an encoder which is basically A to D convertor. Each quantized level is converted into N bit digital word by A to D converter.
- The communication system is normally connected to each other using a single cable i.e. serial communication. But the output of ADC is parallel which cannot be transmitted through serial communicating links. So this block will convert the parallel data into serial stream of data bits.
- A pulse generator produces train of rectangular pulses of duration "t" seconds. This signals acts as sampling signals for the sample and hold block. The same signal acts as "clock" signals for parallel to converter .the frequency "f" is adjusted to satisfy the criteria.

c) **Describe North American (T-carrier) digital multiplexing hierarchy with neat diagram.** **4M**

Ans:

2M for diagram



Explanation:-

T1 Carrier System

T1 carrier systems were designed to combine PCM and TDM Techniques for the

Transmission of 24 64Kbps channels with each channel Capable of Carrying Digitally. Encoded voice band telephone signals or data. The transmission bit rate (line speed) for a T1 carrier is 1.544 Mbps.
All 24 DS-0 channels combined has a data rate of 1.544Mbps, this digital signal level is Called DS-1. Therefore T1 lines are referred as DS-1 lines.

2M for explanation

Service	Line	Rate (Mbps)	Voice Channels
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

DS and T Line rates

T2 Carrier System

T2 carriers time division multiplex 96 64-Kbps voice or data channels into a single 6.312 Mbps data signal for transmission over twisted pair copper wire up to 500 miles over a special metallic cable.

T3 Carrier system

T3 carriers Time division multiplex 672 64-kbps voice or data channels for transmission over a single coaxial cable. The transmission rate is 44.736 Mbps.

T4 Carrier System

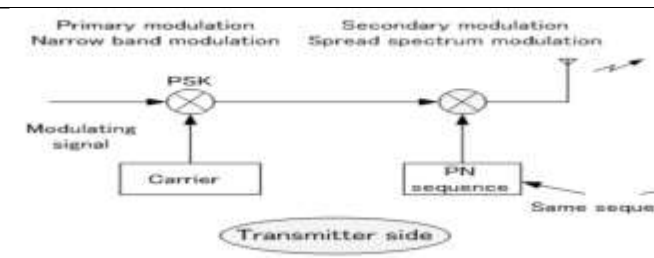
T4 carriers time division multiplex 4032 64-kbps voice or data channels for transmitting over a single T4 coaxial cable upto 500 mile. The transmission rate is very high i.e. 274.16Kbps.

T5 Carrier System

T5 carriers time division multiplex 8064 64Kbps voice or data channels and transmit them at 560.16Mbps over a single coaxial cable.

d) **Explain direct sequence spread spectrum (DSSS) transmitter with block diagram.** **4M**

Ans:



2M for diagram

In direct sequence, the serial binary data is mixed with a higher frequency pseudorandom binary code at a faster rate and the result is used to phase-modulate a carrier.
The information signal undergoes primary modulation by PSK, FSK or other narrow band modulation and secondary modulation with spread spectrum modulation. Spread spectra are obtained by multiplying the primary modulated signal and the square wave, called the PN sequence. Contrariwise, as with commercial radio, there are cases where spread modulation is applied to the data first, and narrow band modulation such as PSK or FSK is applied afterwards. The figure below is an example **Hamming code for the data 1010 with odd parity.** le of spread spectrum modulation and demodulation using PSK for primary modulation.

2M for explanation

e) **Construct the Hamming code for the data 1010 with odd parity.** **4M**

Ans: Let us find the Hamming code for binary code, $d_4d_3d_2d_1$ = 1010. Consider even parity bits.
The number of bits in the given binary code is n=4.
We can find the required number of parity bits by using the following mathematical relation.

$$2k \geq n+k+12k \geq n+k+1$$

1M for calculating no. of

Substitute, n=4 in the above mathematical relation.

$$\Rightarrow 2k \geq 4+k+1 \Rightarrow 2k \geq 4+k+1$$
$$\Rightarrow 2k \geq 5+k \Rightarrow 2k \geq 5+k$$

The minimum value of k that satisfied the above relation is 3. Hence, we require 3 parity bits p_1, p_2, and p_3. Therefore, the number of bits in Hamming code will be 7, since there are 4 bits in binary code and 3 parity bits. We have to place the parity bits and bits of binary code in the Hamming code as shown below.

- For a 4-bit code there are 3 parity bits **p1, p2 and p3** at location **1, 2 and 4** resp.
- So, the code will be: **"p1 p2 n1 p3 n2 n3 n4"** where, **n1, n2, n3, n4** are bits of the code and **p1,p2 and p3** are parity bits to be calculated
- Therefore, the code for even parity is calculated as below:

statement	Bits			
Odd Parity for bits 1,3,5,7	P1	n1	n2	n4
		1	0	0
	0	1	0	0
Odd parity for bits 2,3,6,7	P2	n1	n3	n4
		1	1	0
	1	1	1	0
Odd parity for bits 4,5,6,7	P3	n2	n3	n4
		0	1	0
	0	0	1	0

Therefore ODD parity hamming code will be 0110010

parity bits.

3 marks for calculating hamming code

Q.5 **Attempt any TWO of the following:** **12-Total Marks**

(a) **A discrete memory less source has an alphabet of seven symbols with probabilities foe its output given in the following table:** **6M**

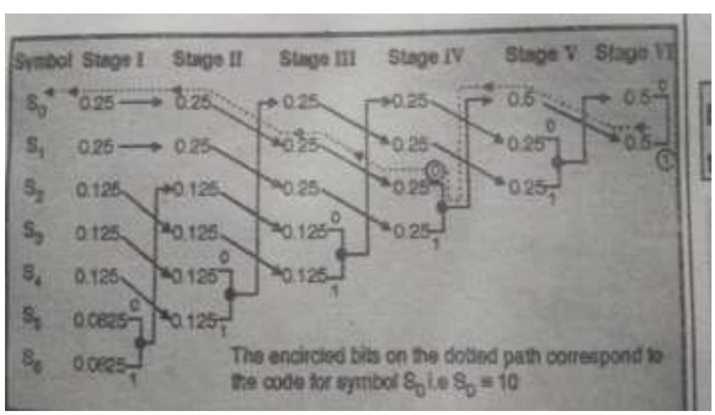
Symbol	**S_0**	**S_1**	**S_2**	**S_3**	**S_4**	**S_5**	**S_6**
Probability	**0.25**	**0.25**	**0.125**	**0.125**	**0.125**	**0.0625**	**0.0625**

Compute:

(i) Huffman code for the above source.

(ii) The coding efficiency of the designed Huffman code.

Ans: **The Huffman code for the source is:** **2M**



Symbol	Probability	Codeword	Codeword length
S_0	0.25	10	2 bit
S_1	0.25	11	2 bit
S_2	0.125	001	3 bit
S_3	0.125	010	3 bit
S_4	0.125	011	3 bit
S_5	0.0625	0000	4 bit
S_6	0.0625	0001	4 bit

2M

To compute the efficiency :

I. The average code length = L

$$= \sum_{k=0}^{6} P_k \times \text{(length of symbol in bits)}$$

From Table P. 2.7.3(b)

$$L = (0.25 \times 2) + (0.25 \times 2) + (0.125 \times 3) \times 3 + (0.0625 \times 4) \times 2$$

$$\therefore \quad L = 2.625 \text{ bits/symbol}$$

The average information per message

$$= H = \sum_{i=0}^{6} p(x_i) \log_2 [1/p(x_i)]$$

$$\therefore \quad H = [0.25 \log_2 (4)] \times 2 + [0.125 \log_2 (8)] \times 3 + [0.0625 \log_2 (16)] \times 2$$

$$= [0.25 \times 2 \times 2] + [0.125 \times 3 \times 3] + [0.0625 \times 4 \times 2]$$

$$\therefore \quad H = 2.625 \text{ bits/message.}$$

Code efficiency $\eta = \frac{H}{L} \times 100 = \frac{2.625}{2.625} \times 100$

$$\therefore \quad \eta = 100\%$$

2M

(b)	**Compare binary ASK, FSK &PSK modulation techniques (any six points).**	**6M**

Ans:

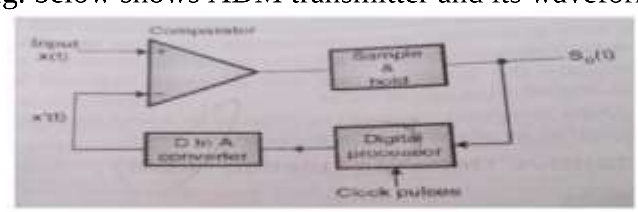
Sr. No	Parameter	Binary ASK	Binary FSK	Binary PSK
1.	Variable Characteristic	Amplitude	Frequency	Phase
2.	Maximum bandwidth(Hz)	$2f_b$	**5** $f_b/3$	$2f_b$
3.	Noise immunity	low	high	high
4.	Error probability	high	low	low
5.	Performance in presence of noise	poor	Better than ASK	Better than FSK
6.	Complexity	Simple	Moderately complex	Very complex
7.	Bit rate	Suitable upto 100 bits/sec	Suitable upto about 1200 bits/sec	Suitable for high bit rates
8.	Detection method	Envelope	Envelope	Coherent

1M each for any 6 valid points

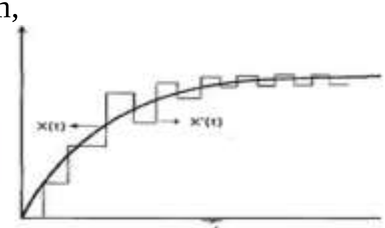
(c) **"Adaptive Delta modulation reduces slope overload distortion and granular noise present in delta modulation". Justify the above statement regarding ADM.** **6M**

Ans: In delta modulation, the step size is constant so slope overload distortion and granular noise both cannot be controlled. These drawbacks can be controlled by using adaptive delta modulation wherein the step size is variable.

Fig. below shows ADM transmitter and its waveform,



ADM transmitter



ADM waveform

- As shown, X (t) is the analog input signal & x' (t) is the quantized version of x(t). Both these signal are applied to comparator.
- Comparator output goes high if x(t) >x'(t) & it goes low if x(t)<x'(t). Thus the comparator output is either 1 or 0.
- Sample & hold circuit will hold this level for entire clock cycle.
- If the signal is rising in the same direction then step size will be made 2 times ,3 times etc.If the signal is falling step size is increased in that direction .
- The variable step size will make the modulator to track fast rising and falling signal.thus slope overload distortion can be minimized
- To reduce granular distortion step size should be made small .
- ADM reduces the step size for a slow varying signal reducing the effect of granular noise

1M for drawbacks in DM

1M for ADM Block diagram

2M for ADM waveform

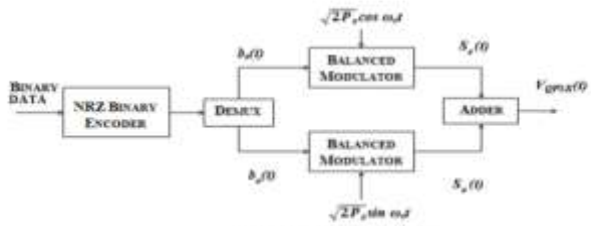
2M explanation

Q.6 **Attempt any TWO of the following:** **12 Marks**

(a) Explain QPSK transmitter with block diagram its constellation diagram. **6M**

Ans:

2M for block diagram



QPSK Transmitter (non - offset)

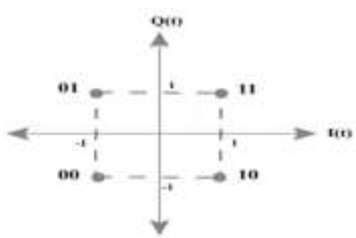
2M explanation

Operation:

- The input data sequence is first converted into a bipolar NRZ signal b(t). The value of b(t) = + 1 for logic 1 input and b(t) = - 1 when the binary input is equal to 0.
- The Demultiplexer(DEMUX) will divide b(t) into twoseparate bit streams bo(t) and be(t). The bit stream be(t) consists of only the even numbered bits 2, 4, 6, 8, ... whereas bo(t) bit stream consists of only the odd numbered bits i.e., 1, 3, 5, ... as shown in Figure 3.18.
- Each bit in the even and odd stream will be held for a period of 2Tb. This duration is called as symbol duration Ts. Thus, every symbol contains two bits.
- The bit stream be(t) is superimposed on a carrier cosωct and the bit stream bo(t) is superimposed on a carrier sinωct by using two balanced modulators (or multipliers) to generate se(t) and so(t). These two signals are basically BPSK signals.
- These signals are then added to generate the QPSK output signal given by, vQPSK(t) = bo(t) sinωct + be(t)cosωct

	2 bits	2 bits	2 bits	2 bits
Bit sequence	00	01	10	11
Phase shift	Zero	90°	180°	270°

2M constellation diagram



Constellation diagram of QPSK

(b) Distinguish between m-ary PSK & m-ary FSK techniques.(Any six points) **6M**

Ans:

1M each for any 6 valid points

Parameter	**M-ary PSK**	**M-ary FSK**
Number of bits per symbol	N [$M = 2^N$]	N [$M = 2^N$]
Symbol duration	$T_S=NT_b$	$T_S=NT_b$
Variable parameter	Phase	Frequency
Demodulation	Coherent	Non-Coherent

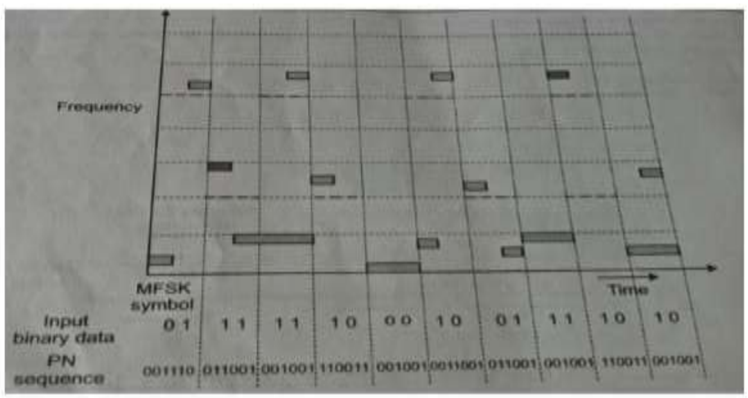
Method		
Bandwidth	2fb/N	2^{N+1}fb/N
Probability of Error	More than that in M-ary FSK	Less than that in M-ary PSK
Transmitted signal	$\sqrt{2f_s}\cos(\omega_c t + \phi_m)$ $\therefore \phi_m = (2m+1)\pi/4$	$\sqrt{2P_s}\cos\omega_c t$

(c)	**Explain fast frequency hopping techniques with suitable waveforms. State its advantage and disadvantages.**	**6M**

Ans:

- In Fast Frequency hopping multiple frequencies or hops are used to transmit one symbol.
- The hop rate is higher than symbol rate but chip rate is equal to hop rate
- For each symbol several hops takes place. So several frequencies changes for one symbol such that symbol rate Rs< Hop rate Rh
- A jammer cannot detect this signal because one symbol is transmitted using more than one carrier frequency.

2M explanation



2M diagram

Advantages :

1. The processing gain is higher
2. More secured transmission as only transmitter and receiver are aware of PN Codes
3. Shorter time for acquisition
4. Robust technology

Disadvantages :

1. Bandwidth requirement is more [GHz]
2. Lower Coverage range due to high SNR requirement at receiver
3. Complex and expensive digital frequency synthesizers are required
4. Data rates are low

1M each for any one adavantage and disadvantage