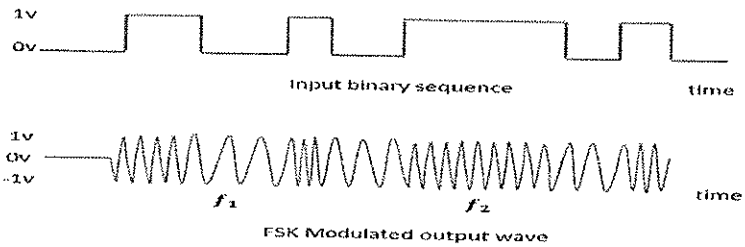
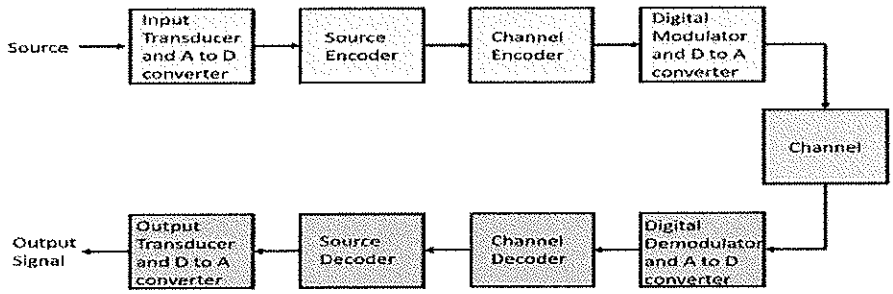
SCHEME OF VALUATION

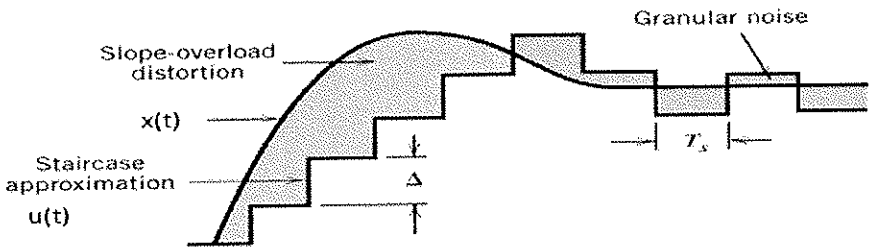
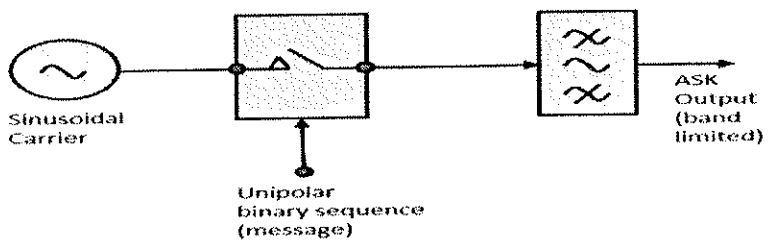
(Scoring Indicators)

Revision : 2021	Course Code :5043D	
Course Title : Digital Communication		
Qn No:	Scoring Indicator	Split up score
PART A		
1	A band limited signal can be reconstructed exactly if it is sampled at a rate twice the maximum frequency component in it. fs>=2fm	1
2	It is the representation of the sampled values of the amplitude by a finite set of levels. which means converting a continuous-amplitude sample into a discrete-time signal.	1
3	a. Impulse sampling b. Natural sampling c. Flat top sampling (Any two)	1
4	Amplitude Shift Keying (ASK)	0.5x2
5	Walkie - Talkie	1
6	Entropy can be defined as a measure of the average information content per source symbol.	1
7	Hamming code , Convolution code (Any two)	0.5x2
8	The theorem states that the channel capacity is equal to the bandwidth times the logarithm of the signal-to-noise ratio (SNR) plus one . $C = B \log_2(1 + S/N)$	1
9	Spread spectrum is a technique used for transmitting radio or telecommunications signals in a protective envelope so that more secure transmission is possible with extended bandwidth.	0.5*2
PART - B		
1.	The conventional methods uses analog signals for long distance communications, which suffer from many losses such as distortion, interference, and other losses including security violations. The digitized signals allow the communication to be more clear and accurate without losses. Digital circuits are easy to design and cheaper than analog circuits.	3

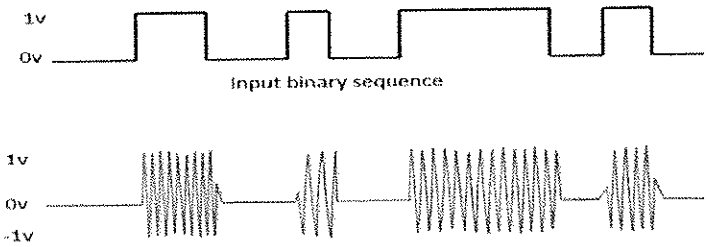
	Digital circuits are more reliable.	
2.	There are two types of Quantization d. Uniform Quantization e. Non-uniform Quantization **Uniform Quantization**: The type of quantization in which the quantization levels are uniformly spaced Two types a. **Mid-Rise** type: The origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number. b. **Mid-tread** type: The origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number. **Non-uniform Quantization**: It is the type of quantization in which the quantization levels are unequal and the relation between them is logarithmic. In this the step size is reduced with the reduction in signal levels to improve the signal to quantization noise ratio for weak signals. Hence the step size is varied according to the signal levels to keep the signal-to-noise ratio adequately high.	1.5*2
3.	It is Differential PCM DPCM requires less Bandwidth than PCM for transmitting the same signal. DPCM is used to decrease quantization noise. DPCM is a signal encoder that uses the baseline of pulse-code modulation (PCM) but adds some functionalities based on the prediction of the samples of the signal.	
4.	DM / ADM The step size cannot be varied / The step size can be varied according to the variation of the signal. Slope overload distortion and granular noise can be present / Slope overload distortion and granular noise is not present Inefficient utilization of bandwidth compared to ADM / Utilizes the bandwidth a lot more efficiently than DM	3
5.	 FSK is the digital modulation technique in which the frequency of the carrier signal varies according to the input digital signal changes.	1.5 + 1.5

	The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input.	
6.	$[H] = \begin{bmatrix} 1 & 0 & 1 & 1 & 1 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 1 & 0 \\ 1 & 1 & 1 & 0 & 0 & 0 & 1 \end{bmatrix}$ $[H] = [P^T : I]$ $[P^T] = \begin{bmatrix} 1 & 0 & 1 & 1 \\ 1 & 1 & 0 & 1 \\ 1 & 1 & 1 & 0 \end{bmatrix}$ Generator matrix $[G] = [I : P]$ Parity matrix $[P] = \begin{bmatrix} 1 & 1 & 1 \\ 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \end{bmatrix}$ Hence Generator matrix $[G] = \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 & 1 \\ 0 & 1 & 0 & 0 & 0 & 1 & 1 \\ 0 & 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 0 & 1 & 1 & 1 & 0 \end{bmatrix}$	3
7.	• The increasing demand for wireless communications has problems due to limited spectrum efficiency and multipath propagation. • The use of spread spectrum communication has simplified these problems. • In the spread spectrum, signals from different sources are combined to fit into larger bandwidth. It offers • Higher channel capacity. • Ability to resist multipath propagation. • The spread spectrum offers multiple access capabilities. **(Any three points)**	3
8.	Time Division Multiple Access (TDMA) is a digital cellular telephone communication technology. It facilitates many users to share the same frequency without interference. It requires an accurate synchronization between the transmitter and the receiver. TDMA is used in digital mobile radio systems.	3

9.	A coded sequence of 1s and 0s with certain auto-correlation properties, called as Pseudo-Noise coding sequence. It is used in spread spectrum techniques. It is a maximum-length sequence, which is a type of cyclic code.	3
10.	Orthogonal Frequency Division Multiplexing (OFDM) is a method of digital data modulation used in telecommunications. It is a transmission technique that divides the whole bandwidth into many subchannels that are closely spaced with overlapping spectra and mutually orthogonal to each other. OFDM is a specialised FDM having the constraint that the sub-streams in which the main signal is divided, are orthogonal to each other.	3
	PART C	
1.	 Basic Elements of a Digital Communication System Source: The source can be an analog signal. Example: A Sound signal Input Transducer: This is a transducer which converts the physical input to an electrical signal (Example: microphone). And it is converted into digital for further processes. The output is a binary sequence. Source Encoder: The source encoder compresses the data into minimum number of bits. This helps in effective utilization of the bandwidth. Channel Encoder: Due to the noise in the channel, the signal may get altered. The channel encoder does error correction by adding some redundant bits to the transmitted data. Digital Modulator: The signal to be transmitted is modulated here by a carrier and is also converted to analog form, in order to make it travel through the channel or medium. Channel: The channel or a medium, allows the analog signal to transmit from the transmitter end to the receiver end. Digital Demodulator: This is the first step at the receiver end. The received signal is demodulated as well as converted again from analog to digital. The signal gets reconstructed here.	7

	Channel Decoder: Due to the noise in the channel, the signal may get altered. The channel decoder does error correction by adding some redundant bits to the received data. This addition of bits helps in the complete recovery of the original signal. Source Decoder: The resultant signal is once again digitized by sampling and quantizing so that the pure digital output is obtained without the loss of information. Output Transducer: This is the last block which converts the electrical signal into physical output (Example: loud speaker). Output Signal: This is the output which is produced after the whole process. Example – The sound signal received.	
2.	⇒ Slope overload distortion ⇒ Granular or idle noise  ▪ <u>Slope Overload Distortion</u> This distortion arises because of large dynamic range of the input signal. From fig. , the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it. Hence, there is a large error between the staircase approximated signal and the original input signal x(t). This error or noise is known as slope overload distortion . ▪ Granular or Idle Noise Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal. Fig. shows that when the input signal is almost flat , the staircase signal u(t) keeps on oscillating. The error between the input and approximated signal is called granular noise.	7
3.	 ▪ The ASK modulator block diagram consists of the carrier signal generator, the binary sequence from the message signal and the band-limited filter.	7

- The carrier generator, sends a continuous high-frequency carrier.
- The binary sequence from the message signal makes the unipolar input to be either High or Low.
- The high input closes the switch, allowing a carrier wave. Hence, the output will be the carrier signal.
- The low input opens the switch , allowing no voltage to appear. Hence, no output.
- The band-limiting filter, shapes the pulse depending upon the pulse-shaping filter.



4.

TDM	FDM
Works with both analog and digital signals.	Primarily used in analog systems.
Shares the timescale for various signals.	Requires guard bands and has lower spectral efficiency compared to TDM.
Divides the available time on the channel into discrete time slots, with each slot allocated to a different signal or data stream.	Allocates bandwidth to different sources.
Data from each input source is interleaved into the allocated time slots, creating a composite signal for transmission.	Enables simultaneous transmission of multiple signals without interference.

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5.

There are 3 types of transmission modes which are given below: Simplex mode, Half duplex mode, and Full-duplex mode.

1. **Simplex**
   - ⇒ It is a unidirectional communication.
   - ⇒ In this mode Sender can send the data but the sender can't receive the data.
   - ⇒ It uses one channel for the transmission of data.
   - ⇒ Less performance than half duplex and full duplex.
   - ⇒ Examples: Keyboard and monitor.

3.5 * 2

2. Full Duplex

⇒ It is two-way directional communication

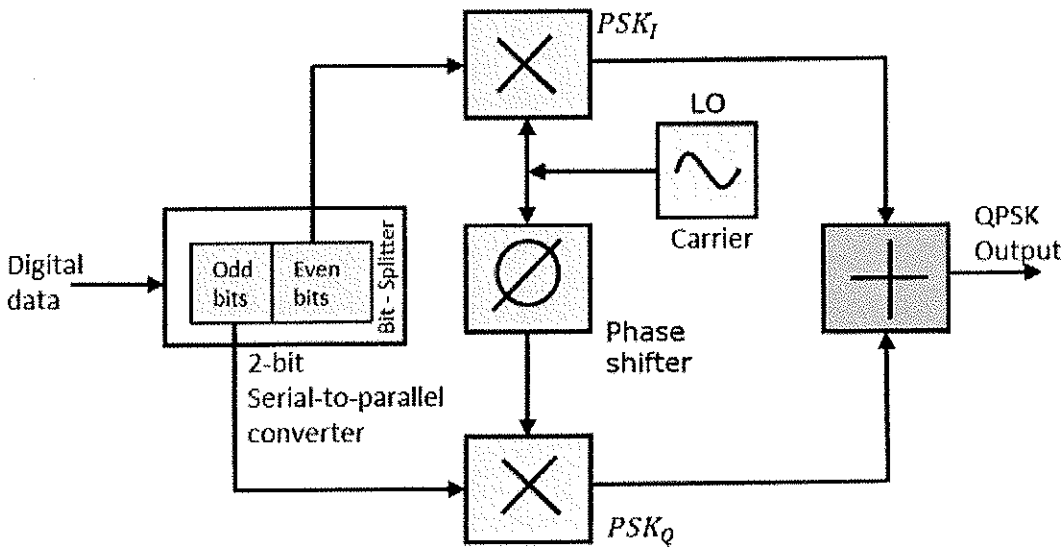
⇒ In this mode, Sender can send the data and also can receive the data simultaneously.

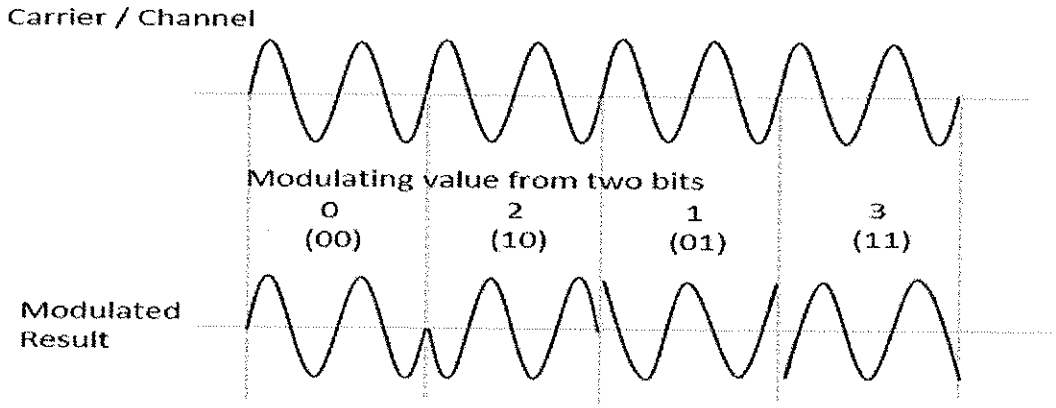
⇒ It uses two channels for the transmission of data.

⇒ Better performance than simplex and half duplex mode.

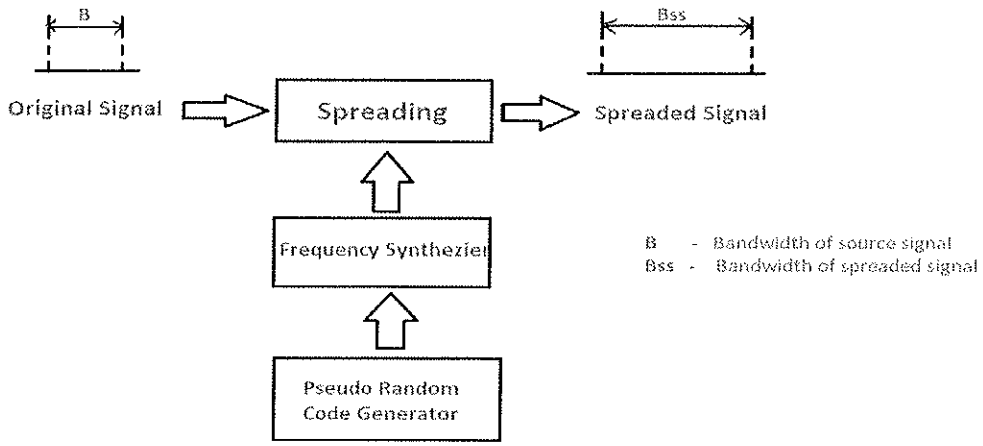
⇒ Example: Telephone.

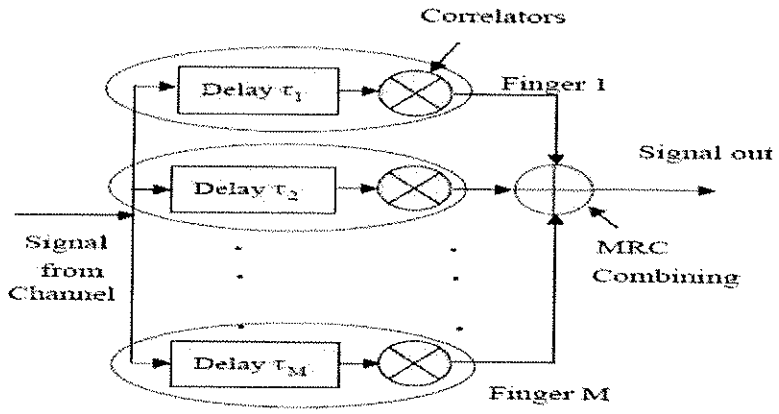
6. It is a form of Phase shift keying in which two bits are modulated at once. Selecting one of four possible carrier phase shifts (0,90,180 or 270 degrees). QPSK allows the signal to carry twice as much information as ordinary PSK using the same bandwidth. 7



	At the modulator's input, the message signal's even bits (i.e., 2nd bit, 4th bit, 6th bit, etc.) and odd bits (i.e., 1st bit, 3rd bit, 5th bit, etc.) are separated by the bits splitter and are multiplied with the same carrier to generate odd BPSK (called as **PSK_I**) and even BPSK (called as **PSK_Q**). The **PSK_Q** signal is anyhow phase shifted by 90° before being modulated. The QPSK waveform for two-bits input is as follows, which shows the modulated result for different instances of binary inputs. 	
7.	1. Data compression: reducing the size of data to save storage space or bandwidth. Cryptography: protecting the confidentiality, integrity, and authenticity of data from unauthorized access or modification. Error detection and correction: detecting and correcting errors that occur during data transmission or storage due to noise or interference. Data transmission: sending data over a communication channel, such as a wire, a radio wave, or an optical fibre. 2. Linear block codes are a type of error-correcting code in which the actual information bits are linearly combined with the parity check bits so as to generate a linear codeword that is transmitted through the channel.	7
8.	The parity bit method is a simple error detection method that involves adding an extra bit to a data transmission. The parity bit is an extra bit included in a binary message to make the total number of 1's either odd or even. There are two parity systems - even and odd parity checks. <u>Even parity check</u> The total number of 1's in the given data bit should be even. So if the total number of 1's in the data bit is odd then a single 1 will be appended to make the total number of 1's even else 0 will be appended. Eg: If the binary message **1011**	7

	Since there are three 1s in the message, we add a 1 as the parity bit to make the total number of 1s even. So the message we send is **10111**. <u>Odd parity system</u> The total number of 1's in the given binary string (or data bits) are even then 1 is appended to make the total count of 1's as odd else 0 is appended. Eg: If the binary message **1011** Since there are three 1s in the message, we add a 0 as the parity bit to keep the total number of 1s odd. So the message we send is **10110**.	
9.	SNR = 20dB = 10^2 BW = 30MHz = $30x10^6 = 3*10^7$ Channel Capacity C = B $\log_2(1+ SNR)$ C = B $\log_2(1+10^2)$ = $3*10^7 \log_2 (101)$ = $3*10^7$ (log101/log2) log101/log2 ≅ 10 ∴ C = $3*10^7*10$ = $3*10^8 = 300*10^6$ **C = 300Mbps**	
10.	Received Hamming Code D7 D6 D5 P4 D3 P2 P1 1 0 1 0 1 1 1 **<u>Detecting errors</u>** Step 1: Analysing bits 1,3,5,7 P1 D3 D5 D7 1 1 1 1 ➟ Even Parity . So error exists. So put **P1=1** Step 2: Analysing bits 2,3,6,7 P2 D3 D6 D7 1 1 0 1 ➟ Odd Parity . So no error. So put **P2=0** Step 3: Analysing bits 4,5,6,7 P4 D5 D6 D7 0 1 0 1 ➟ Even Parity . So error exists. So put **P4=1** P1 & P4 ≠ 0 So received code is wrong.	7

	Locating the bit in error Error word P4 P2 P1 1 0 1 ➡ 5 So **5th** bit is in error. **Corrected word** For correcting the received word invert the 5th bit Corrected word - **1000111**	
11.	⇒ This is frequency hopping technique, where the users are made to change the frequencies of usage, from one to another in a specified time interval, hence called as frequency hopping. ⇒ In Frequency Hopping Spread Spectrum (FHSS), different carrier frequencies are modulated by the source signal ⇒ For example, a frequency was allotted to sender 1 for a particular period of time. Now, after a while, sender 1 hops to the other frequency and sender 2 uses the first frequency, which was previously used by sender 1. This is called as frequency reuse. ⇒ The frequencies of the data are hopped from one to another in order to provide a secure transmission. ⇒ The amount of time spent on each frequency hop is called as Dwell time. ⇒ The general block diagram of FHSS is shown in the below figure. ⇒ A pseudorandom code generator generates Pseudo-random Noise of some pattern for each hopping period. ⇒ The frequency corresponding to the pattern is used for the hopping period and is passed to the frequency synthesizer. ⇒ The synthesizer generates a carrier signal of that frequency.  	7

12.	A Rake Receiver is a radio receiver which is designed for the purpose to counter the effects of multipath fading. Due to reflections from multiple obstacles in the environment, the radio channel can consist of multiple copies of the transmitted signal having different amplitude, phases or delays. A rake receiver can resolve this issue and combine them. For this purpose, several sub-receivers are used which are known as "fingers". Working of Rake Receiver: The received signal from multiple paths arrives at the antenna with different delays and phases. The signal is first passed through a bank of matched filters, each of which corresponds to a specific path. The output of each matched filter is sampled at the symbol rate and the resulting samples are combined. The combining process is done using a technique called maximum ratio combining (MRC) which gives more weight to the signals that have higher signal-to-noise ratio (SNR). The combined signal is then demodulated to obtain the transmitted symbols. The rake receiver is able to recover the transmitted signal even in the presence of severe multipath fading and interference.  	7