MODEL ANSWER

SUMMER– 17 EXAMINATION

Subject Title:Digital Communication Subject Code: 17535

Important Instructions to examiners:

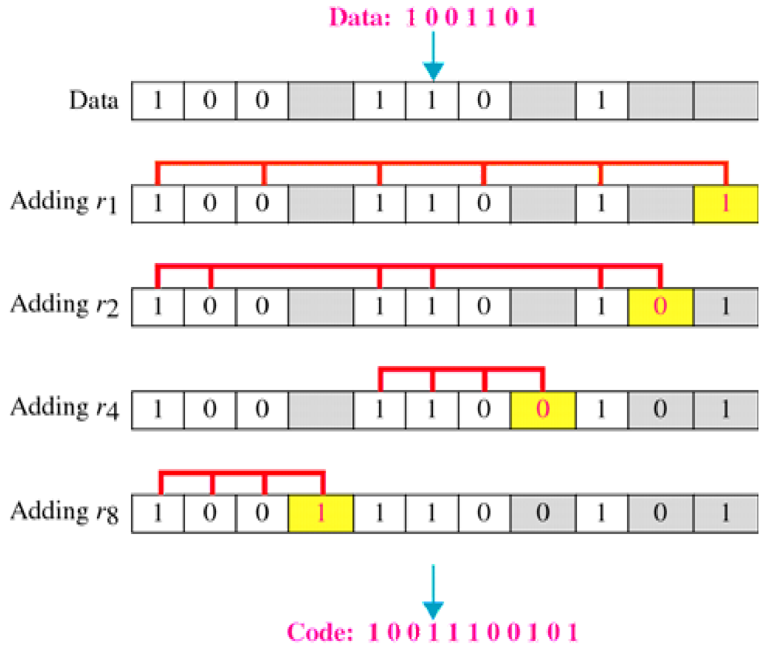
1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling 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 anyequivalentfigure 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 THREE of the following:**	**12-Total Marks**
	(a)	**Compare analog pulse modulation with digital pulse modulation. (any four points)**	**4M**
	Ans:	(see table below)	**1 M each (any 4 points)**

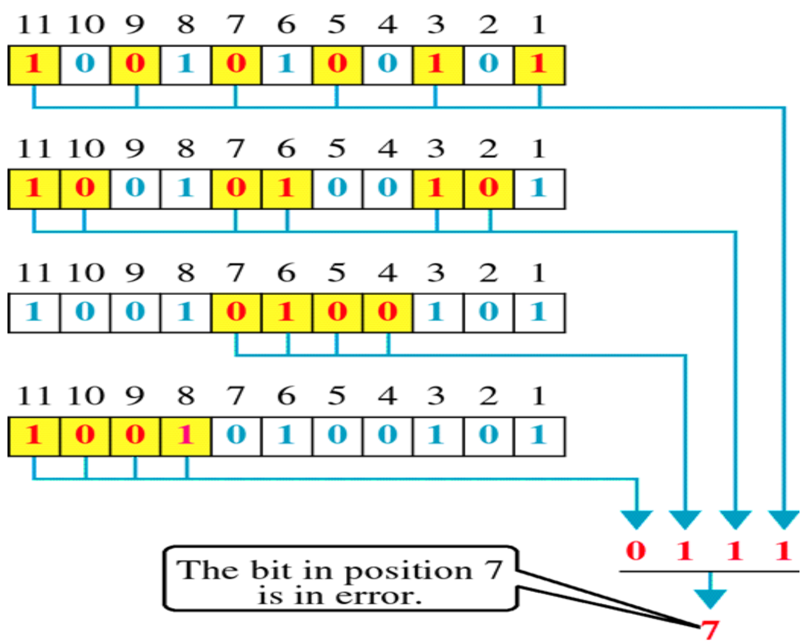
SR.NO	PARAMETER	ANALOG PULSE MODULATION	DIGITAL PULSE MODULATION
1	Nature of transmitted signal	Pulse with varying parameters(amplitude, width or position of the pulse)	Digital signal i.e in the form of one's and zero's
2	Noise immunity	Poor	Excellent
3	Bandwidth requirement	Lower then digital	Higher due to higher bit rate
4	Multiplexing used	FDM/TDM	TDM
5	Types	PAM,PPM,PWM	DM,ADM,PCM,DPCM

	(b)	**Explain with example how hamming code is used for single bit error correction.**	**4M**
	Ans:	*Note:- Any relevant example can be considered*	**4M**

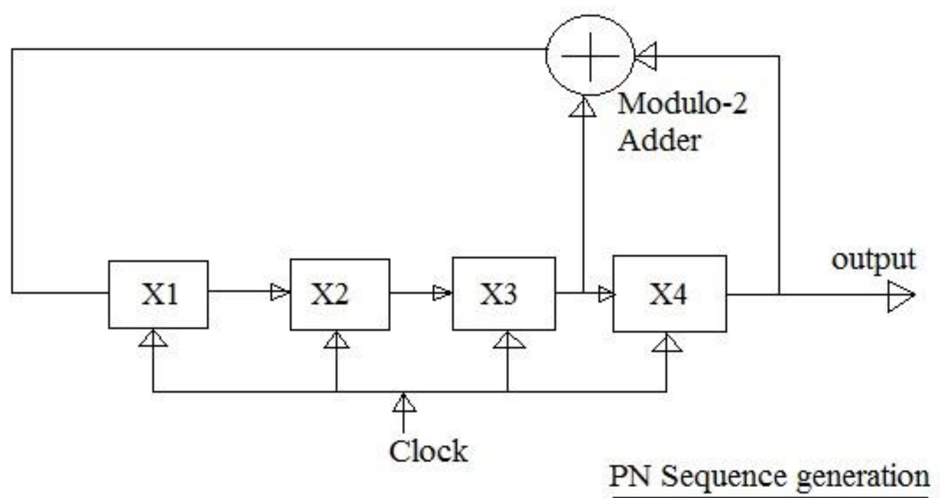
At Transmitter end

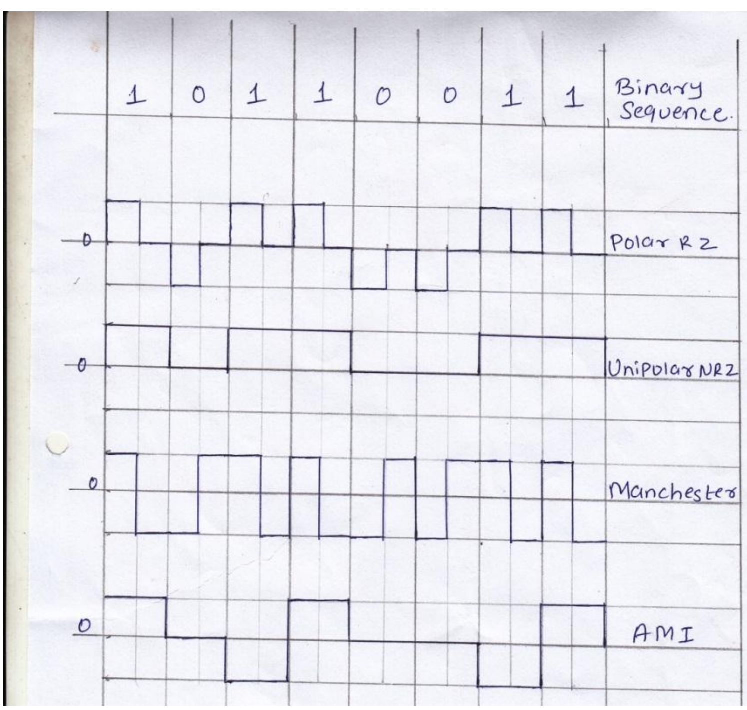


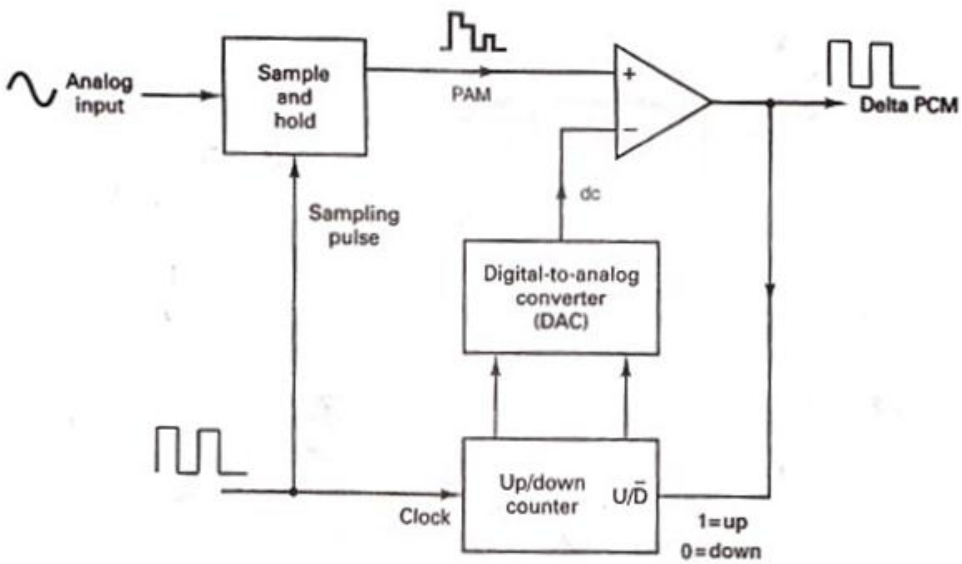
At Receiver end



(c)	**State the principle of orthogonality. Explain the concept of single carrier & multi-carrier system.**	**4M**
Ans:	**<u>Principle of orthogonality-</u>** • Orthogonality between two signals means that the two co-existing signals are independent of each other in a specified time interval and do not interact with each other. • Orthogonality is a property that allows multiple information signals to be transmitted	**2M**

		perfectly over a common channel and detected without interference. Loss of orthogonality results in blurring between these information signals and degradations in communication. **<u>Single carrier system.</u>** • In order to use the available radio spectrum efficiently, in single carrier system, the modulated sub carrier should be placed as close to each other as possible without causing interference. Guard bands are required to be inserted between adjacent spectrum to avoid interference but these increases bandwidth & reduce spectrum efficiency **<u>Multi-carrier system</u>** • The basic idea of OFDM is to divide the available spectrum into several sub-channels (or subcarriers). By making all the sub-channels narrow band. • The OFDM provides a technique allowing the bandwidth of modulated carriers to overlap without interference.	**1M** **1M**
	(d)	**Draw block diagram of PN sequence generator using 4 D-Flip flop.**	**4M**
	Ans :	 **Where X1,X2,X3,X4 are D flip flops.**	**4M**
	B)	**Attempt any ONE of the following :**	**6M**
	a)	**Explain the effects of noise on the channel. Also state the need of channel modelling.**	**6M**
	Ans :	**<u>Effects of noise on the channel</u>** • The signal is corrupted by unwanted, an unpredictable electrical signal is known as noise. • Greater the amount of noise, the lower the channel capacity. The presence of noise reduces the amount of information that can be transmitted in a given bandwidth. • While some of the degrading effects of the channel can be removed or compensated	**3M**

		for, the effects of noise cannot be completely removed. • One of the ways to minimize the noise is to increase the signal power. **<u>Need of channel modeling</u>** • In the analysis and design of communication system, it will be necessary to model the channel as system and incorporate in to that model as many details of electrical behavior of the channel as possible, so as to make it represent the actual situation as accurately as possible. • It is more convenient and appropriate to classify the channels as linear & non linear channels, time invariant and time varying channels and bandwidth limited and power limited channels as these characteristic can easily be incorporated in to the system used for modeling the channel. .	**3M**
	b)	**Define line coding. Draw the waveforms for a binary sequence 10110011 for following signal codes :** **(i) Polar R Z** **(ii) Unipolar NRZ** **(iii) Manchester** **(iv) Alternate Mark Invasion (AMI)**	**6M**
	Ans :	**<u>Definition :</u>** It is a coding technique that converts the stream of binary digits into a format or code which is suitable for transmission over a cable or any other medium. **<u>Waveforms :</u>** 	**(Define 2 M, line code 1 M each)**

Q 2		Attempt any TWO of the following :	16M
	(a)	**Draw and explain the block diagram of Delta Modulation. Also explain slope overload and granular noise in linear delta modulation.**	8M
	Ans :		2M
		Explanation: • Delta modulation is a type of digital modulation in which one bit per sample is transmitted. This reduces the bandwidth requirement to a great extent. • DM is a special case of DPCM in which only the polarity of the difference signal is encoded by one bit. If the difference between the analog input and the feedback signal is positive, it is encoded as binary 1 and transmitted and if the difference is negative, binary 0 is transmitted. • 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. • 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 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. In the idealized situation the DAC output follows the input signal.	2M

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.

SLOPE-OVERLOAD DISTORTION:

2M

- If the slope of the analog signal *x(t)* is much higher (steep) than that of the approximated signal $x_q(t)$ over a long duration then $x_q(t)$ will not follow *x(t)* at all as shown in Figure
- The difference between *x(t)* and $x_q(t)$ is called the ***slope-overload distortion or the slope-overload error***. Thus, slope-overload error occurs when the slope of *x(t)* is much higher than $x_q(t)$.

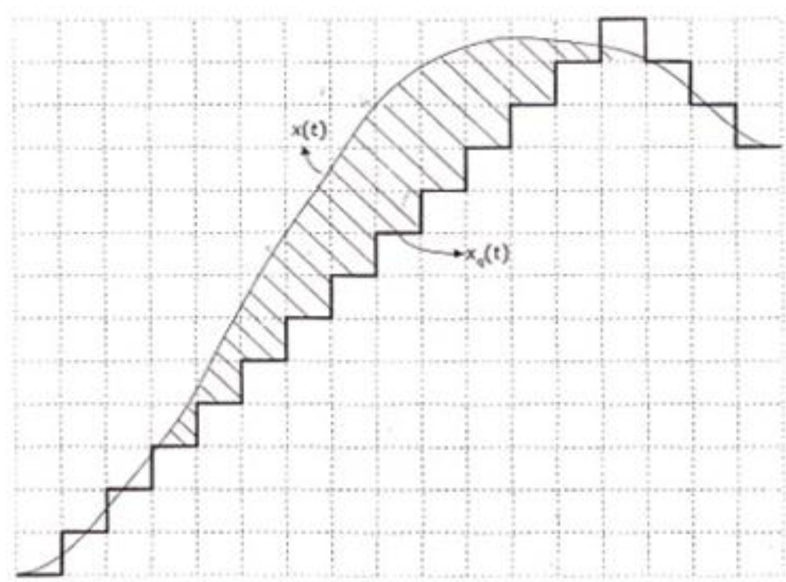


Fig. Illustrating the phenomenon of slope overload in linear delta modulation

GRANULAR NOISE:

2M

- When the input signal *x(t)* is relatively constant in amplitude, the approximated signal $x_q(t)$ will *hunt* above and below *x(t)* as shown in Figure. This leads to a noise called ***granular noise***.
- It increases with increase in step size δ. To reduce granular noise, the step size should be as small as possible. However, this will increase slope-overload distortion.

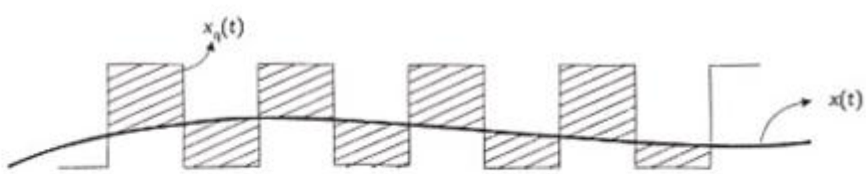


Fig. Granular noise

(b) **State bandwidth required for BASK, BFSK, BPSK and QPSK. Also draw waveforms for binary data 10110010 in ASK, FSK, PSK and QPSK modulation.** **8M**

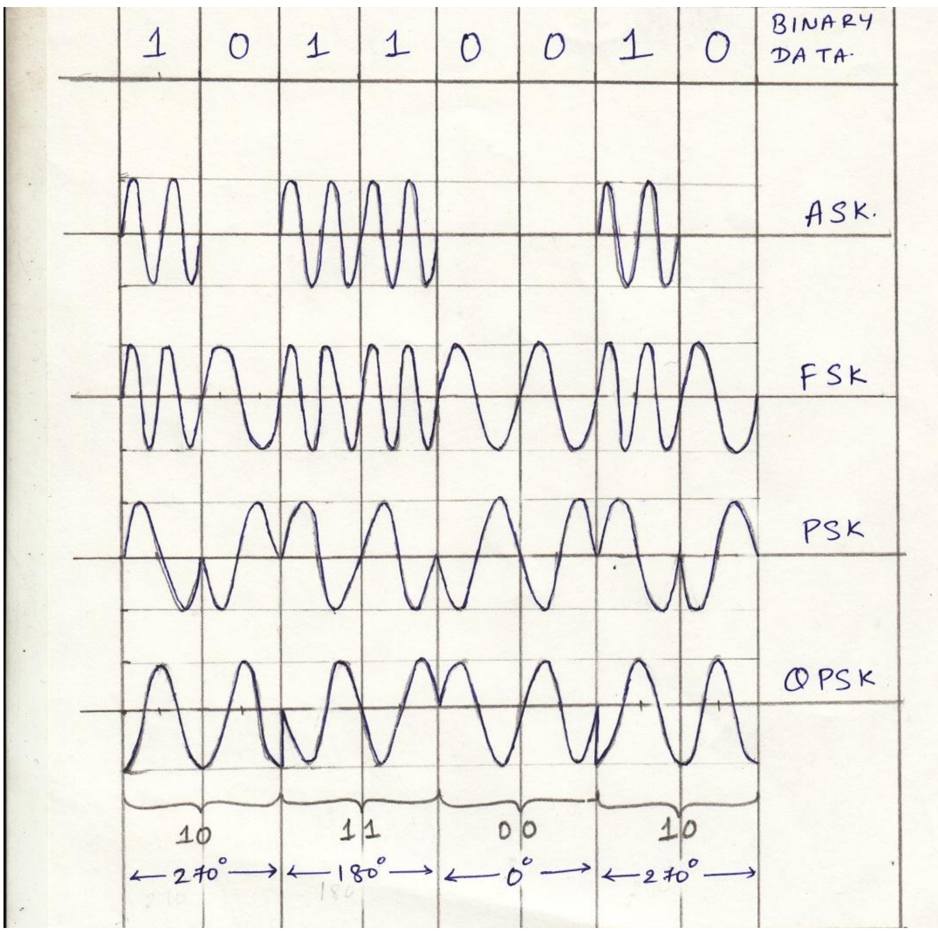
Ans: **Bandwidth :**

(Bandwidth-1 M each, Waveform - 1 M each)

- **BASK = 2fb**
- **BFSK = 4fb**
- **BPSK = 2fb**
- **QPSK = fb**

Where fb is Bit Frequency/Bit rate

<u>Waveforms:</u>



(c) **Explain the CCITT digital multiplexing hierarchy with block diagram.** **8M**

Ans: **<u>Diagram :</u>** **4M**

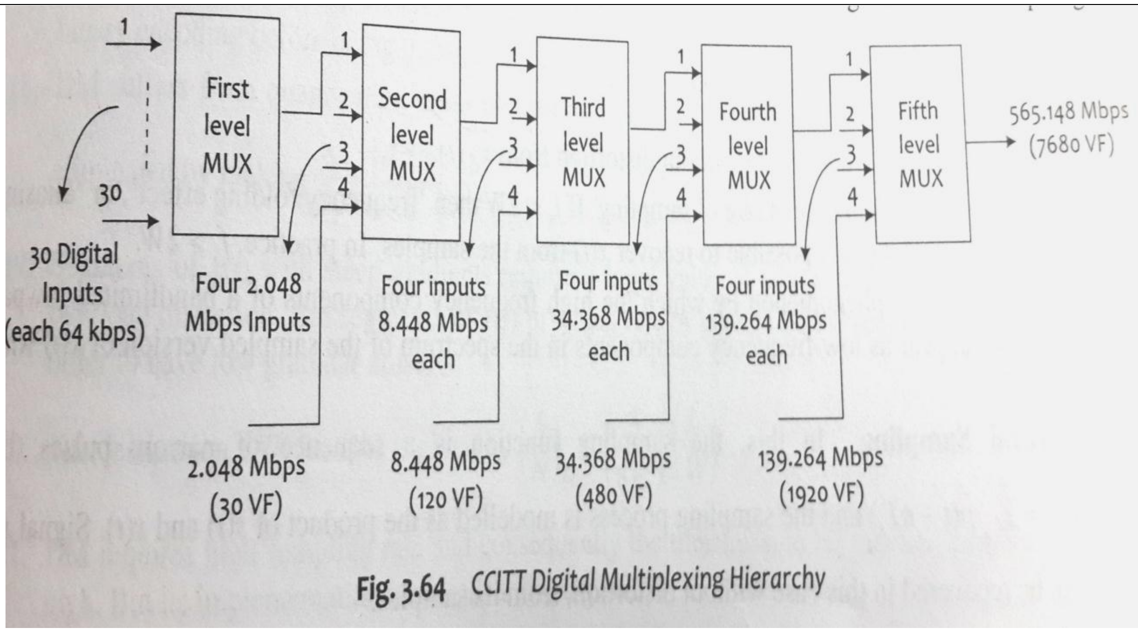
			

Fig. 3.64 CCITT Digital Multiplexing Hierarchy

		<u>Explanation :</u> • In this hierarchy the first level of multiplexing involves 30 numbers of 64 kbps PCM-ed voice channels. • This gives a 2.048 Mbps digital signal. Four such signals are multiplexed in the second –level multiplexing to obtain an 8.448 Mbps digital signal. • The third also involves only four inputs to give a 34.368 Mbps multiplexed signal. Four such signals are multiplexed in the fourth –level multiplexer to obtain a 139.246 Mbps digital signal. • Again four such signals are multiplexed in the 5^{th} level to get a 565.148 Mbps signal.	**4M**
Q. 3		**Attempt any FOUR of the following :**	**16M**
	(a)	**State sampling theorem. Calculate Nyquist rate for voice signal of range 300H_Z to 3400 H_Z.**	**4M**
	Ans :	**<u>Sampling theorem:</u>** Sampling theorem states that a band-limited signal of finite energy having the highest frequency component *fm* Hz can be represented and recovered completely from a set of samples taken at a rate of *fs* samples per second provided that ***fs ≥ 2fm***. Where, fs = sampling frequency fm = maximum frequency of continuous original signal fs = 2*W W = bandwidth = 3400-300 = 3100 Hz Nyquist rate = 2*3100 = 6200 Hz.	**2M** **2M**
	b)	**Explain quantization and quantization error.**	**4M**
	Ans :	**<u>Quantization:-</u>** The quantization process is the process of approximation of the sampled signal. It assigns a particular level to which the sampled value is near to.	**1M**

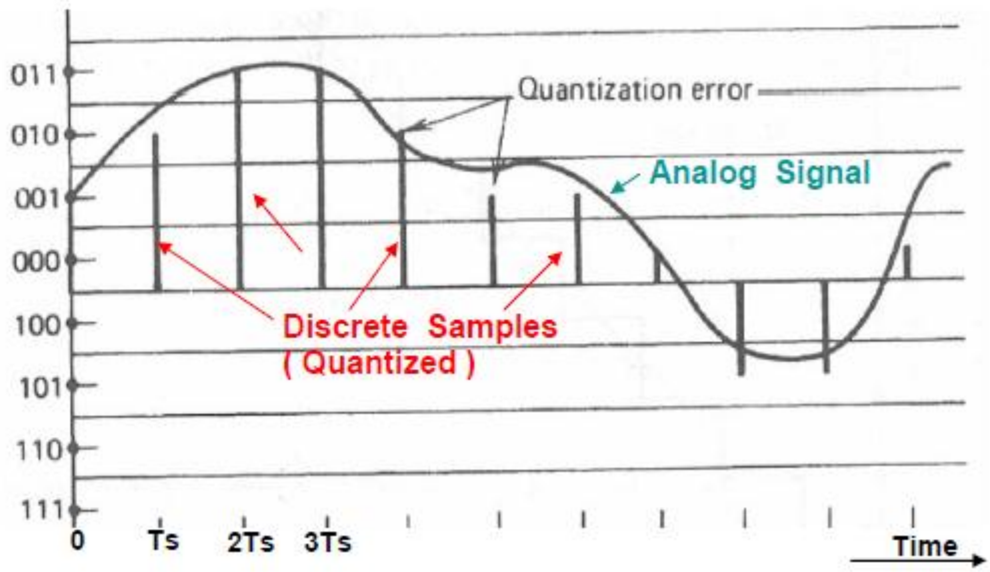


Fig:3.4 Typical Quantization process.

2M

Quantization error:-

The signal with discretized amplitude value is termed as quantized signal.

The difference between the sampled signal and its quantized version which is measured and is represented in terms of quantization noise or quantization error.

1M

c) Compare QPSK and QASK (any 4 points) **4M**

Ans:

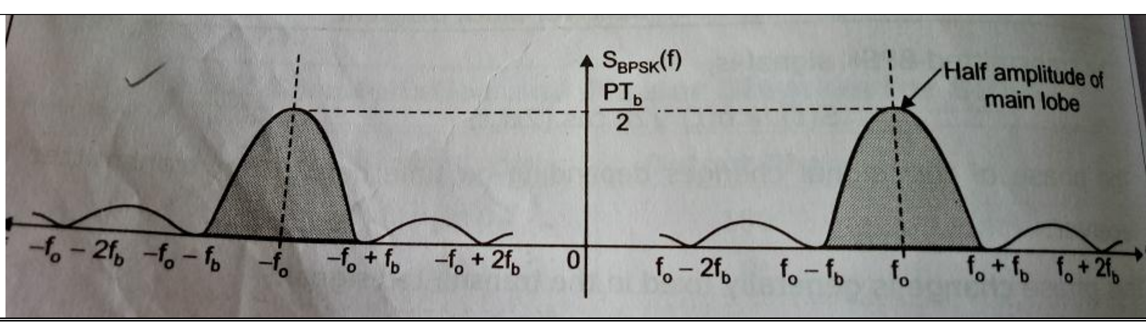
Sr. No.	Parameter	QASK /QAM	QPSK
1	Parameters carrying information signal	Both Amplitude and Phase	Phase only
2	Performance of the system	Better than QPSK	Less than QASK
3	Noise immunity	Better than QPSK	Comparatively less than QASK
4	System complexity	More complex than QPSK	Less complex than QASK
5	Probability of error	Less than QPSK	More than QASK
6	Location of signal points on signal space diagram	Equally spaced and placed symmetrically about origin.	On the circumference of a circle

(Any 4 points.)

(1M For Each Point)

d) Draw and explain the power spectral density of BPSK. **4M**

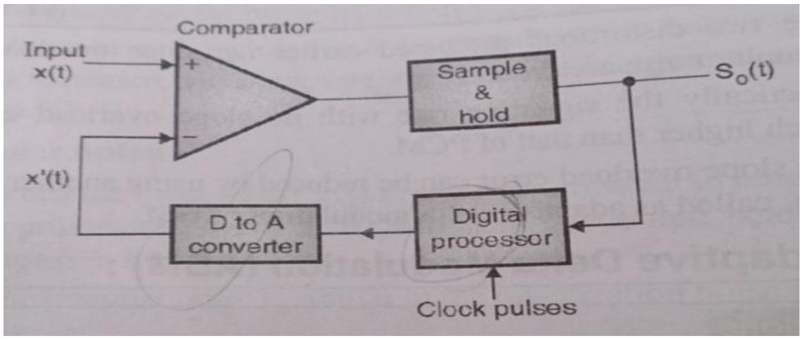
Ans:



2M

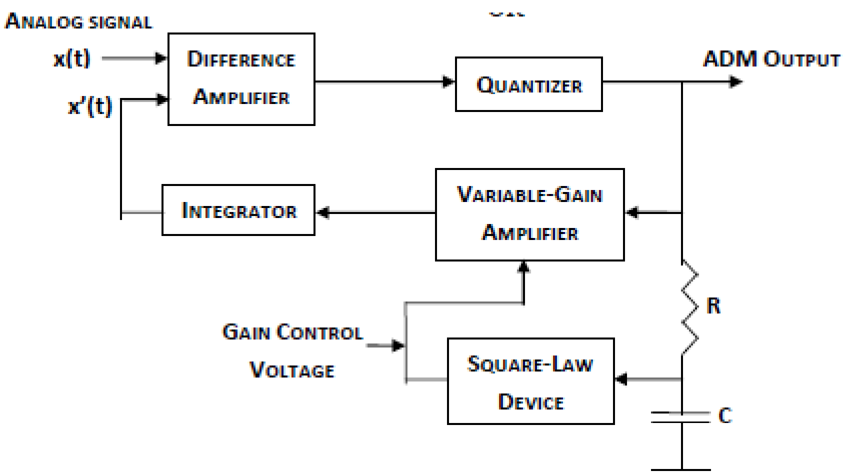
		The spectrum of BPSK signal is centered around the carrier frequency f0. For BPSK the max frequency in the baseband signal will be fb shown in fig. Bandwidth of BPSK signal is, BW = 2 fb.	**2M**
	e)	**List any four advantages of TDMA over FDMA.**	**4M**
	Ans :	**Advantages:- (any four)** 1.In TDMA since only one station is present at any given time the generation of intermodulation products will not take place. 2. The entire channel band width can be allowed to a single channel at given instant of time. This is particularly advantageous for the digital channel which demands large bandwidth. 3. The frequency selective fading does not affect the TDMA to extent it affect of FDMA. 4. As only one channel is being transmitted at a time it is not necessary to separate out various channels at the receiver. 5. TDMA by default can work well with the digital therefore it can be easily used for data transmission.	**(1M Each)**
Q. 4	**(A)**	**Attempt any THREE of the following :**	**12M**
	(a)	**State two advantages and disadvantages of digital communication system.**	**4M**
	Ans :	**Advantages of digital communication system:-** 1) Immunity to transmission noise and interference. 2) Regeneration of the coded signal along the transmission path is possible (Repeater can be used). 3) Digital signals are better suited than analog signals for procession and combining using technique called multiplexing. 4) Communication can be kept "private" and "secured" through the use of encryption. 5) Digital transmission systems are more resistant to analog systems to additive noise because they use signal regeneration rather than signal amplification. 6) Digital signals are simpler to measure and evaluate. 7) It is possible to store the signal and process it further. 8) In digital systems transmission errors can be corrected and detected more accurately. 9) Using data encryption only permuted receivers can be allowed to detect the transmission data. 10) Wide dynamic range. 11) Techniques such as data compression and image enhancement can be used. 12) Because of the advances of IC technologies and high speed computers, digital communication systems are simpler and cheaper. **Disadvantages of digital communication system:-** 1)Digital signal does not provide continuous representation of original signal. 2)It requires synchronization in case of synchronous modulation. 3)Bandwidth requirement is high.	**2M** **2M**
	(b)	**Draw the block diagram of Adaptive Delta modulation transmitter and illustrate its working with waveforms.**	**4M**
	Ans	**The ADM transmitter is shown in figure.**	

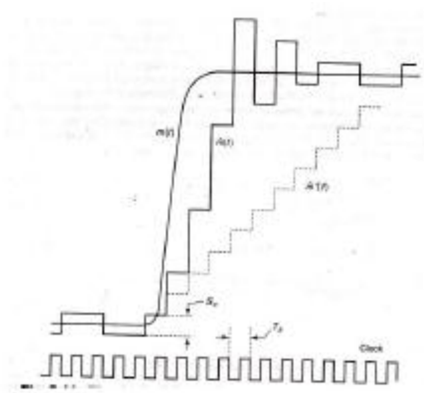
:



2M

OR



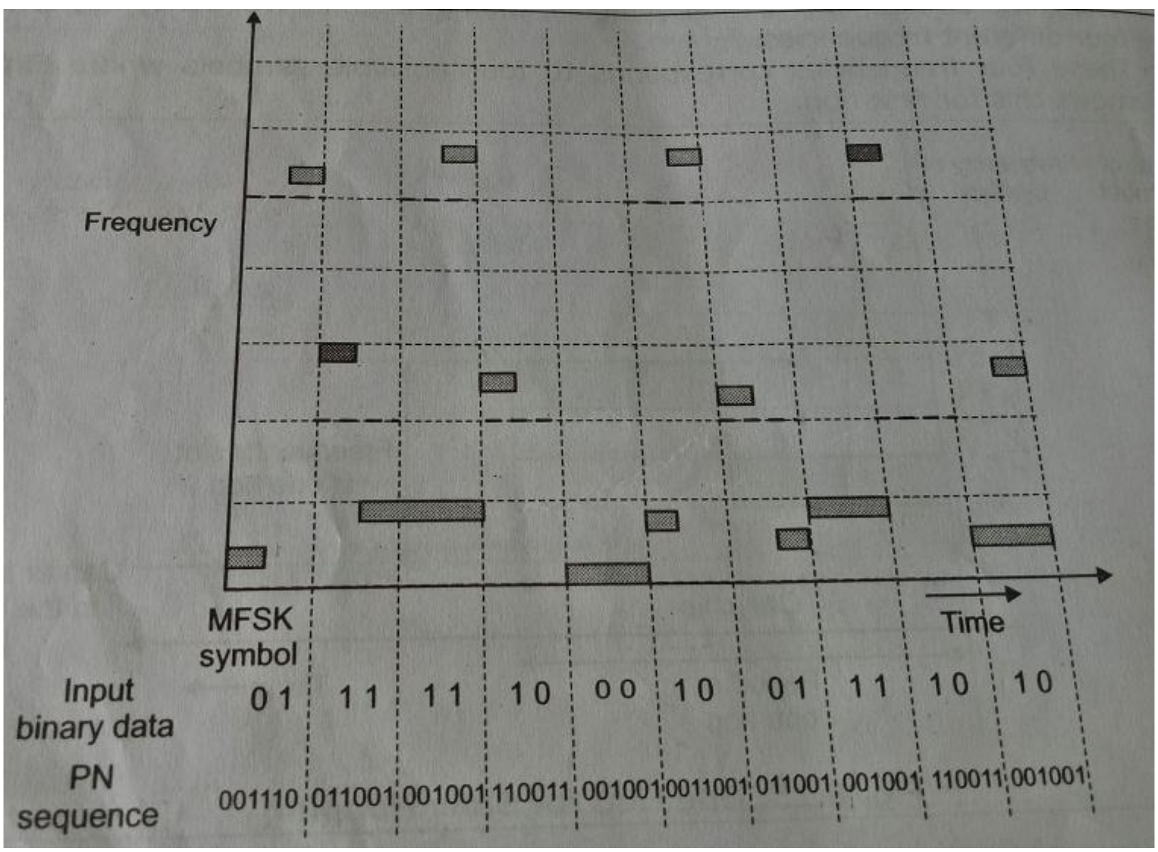


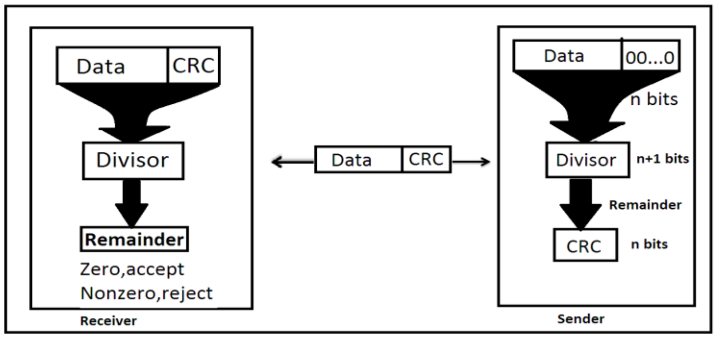
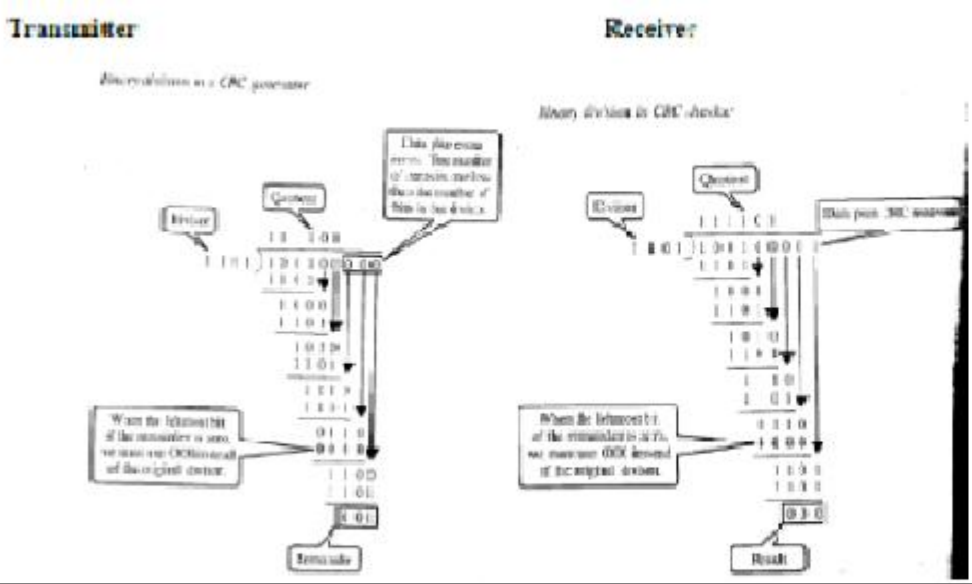
Explanation:

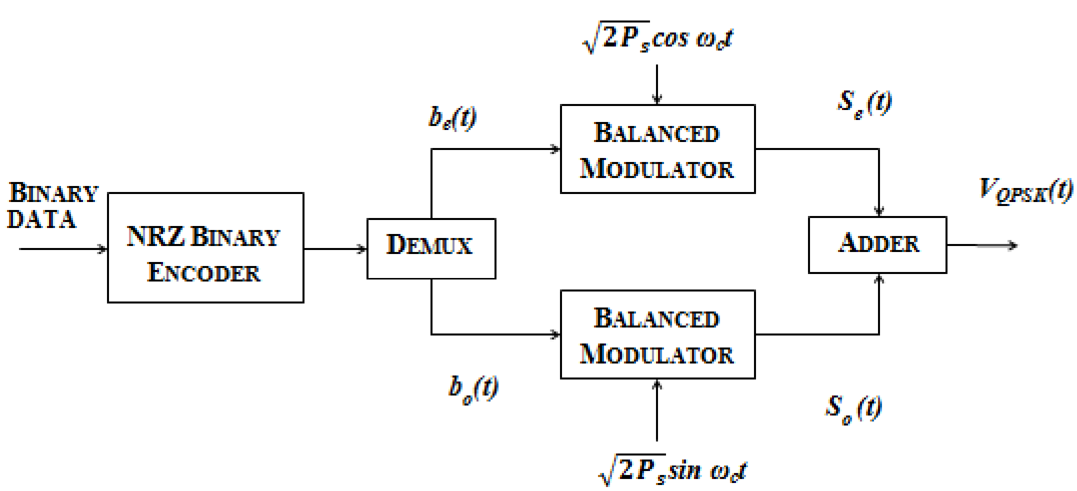
As shown, X (t) is the analog input signal & x' (t) is the quantized version of x(t). Both these signal are applied to comparator. Comparator output is goes high if x(t) >x'(t) & it goes low if x(t)<x'(t). Thus the comparator output is either 1 or 0.Sample & hold circuit will hold this level for entire clock cycle.

2M

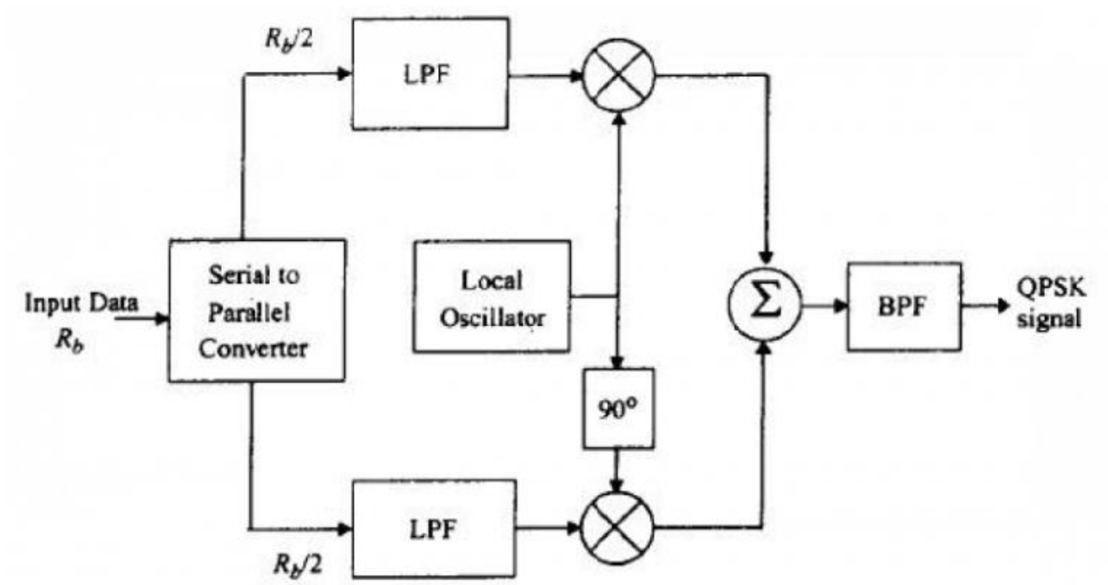
In response to k^{th} clock pulse trailing edge, a processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. (k-1)th clock edge. If the

		direction of both the step is same then the processor will increase the magnitude of present step by delta. If the direction is opposite then the processor will decrease the magnitude of present step by delta.	
	(c)	**Using Shannon Hartley theorem, calculate channel capacity for a channel having BW of 15KH$_Z$ and signal to noise ratio of 20dB.**	**4M**
	Ans :	Given: B = 15KHz S/N = 20 dB $C = B \log_2 (S/N)$ $C = 15 * 10^3 * \log_2 (20)$ C = 64.83 Kbps	**(Formula-1M)** **(Final Answer3M)**
	(d)	**Explain fast frequency hopping with diagram.**	**4M**
	Ans :	**<u>Fast frequency hopping</u>-**In fast frequency hopping, multiple frequencies or hops are used to transmit one symbol. That is each symbol, several hops takes place. So several frequencies changes for one symbol such that Symbol rate Rs < Hop rate Rh.  	**2M** **2M**
	(B)	**Attempt any ONE of the following :**	**6M**
	(a)	**Explain the working of CRC generator and checker.**	**6M**
	Ans :	**<u>Cyclic Redundancy Check (CRC):</u>** With CRC the entire data stream is treated as long continuous binary number. In this method, a sequence of redundant bits, called the CRC or the CRC remainder, is appended to the end	**(CRC Generator**

		of the unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number. At its destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit assume to be correct and is accepted, otherwise it indicate that data unit has been damaged in transmission and therefore must be rejected The redundancies bits are used by CRC are derived by dividing the data unit by a predetermined divisor. The remainder is the CRC.   	**3M)** **(CRC Checker :3M)**
	(b)	**Differentiate between Direct sequence spread spectrum and frequency hopped spread spectrum.**	**6M**

	Ans :	<table><tr><th>DSSS</th><th>FHSS</th></tr><tr><td>• Definition: PN sequence of large bandwidth is multiplied with a narrow band information signal.</td><td>• Definition: Data bits are transmitted in different frequency slots which are changed by PN sequence.</td></tr><tr><td>• Chip rate (R_c) = $\frac{1}{Tc}$</td><td>• Chip rate (R_c) = $max(R_b, R_s)$</td></tr><tr><td>• Applications with large multipath delays: DS represents a reliable mitigation method as such signals render all multipath signal copies that are delayed by more than one chip time from direct signal as invisible to the receiver.</td><td>• FH systems can provide the same mitigation only if the hopping rate is faster than the symbol rate and if the hopping bandwidth is larger.</td></tr><tr><td>• For commercial applications implementation of DSSS radios with large gap can also be costly due to the need of high speed circuits.</td><td>• Implementation of FHSS radio can be costly and complex due to the need of high speed frequency synthesizers.</td></tr><tr><td>• DSSS radios encounter more randomly distributed errors that are continuous and lower level.</td><td>• SFH suffers from strong burst error.</td></tr><tr><td>• Modulation technique: BPSK.</td><td>• Modulation technique: M-ary FSK</td></tr><tr><td>• Long acquisition time.</td><td>• Short acquisition time.</td></tr><tr><td>• DSSS is distance dependent.</td><td>• In FHSS, effect of distance is less.</td></tr><tr><td>• Processing gain is less.</td><td>• Processing gain is higher.</td></tr><tr><td>• Bandwidth required is less than FHSS system.</td><td>• Bandwidth of FHSS system is too high.</td></tr></table>	1M For Each Point (Any 6 Points)
Q.5		**Attempt any TWO of the following :**	**16M**
	(a)	**Draw the block diagram of QPSK transmitter and receiver. Explain its working principle. Draw its construction diagram.**	**8M**
	Ans:	**Diagram:**  QPSK Transmitter (non - offset)	**(QPSK transmitter & Explaination: 4M)**

OR



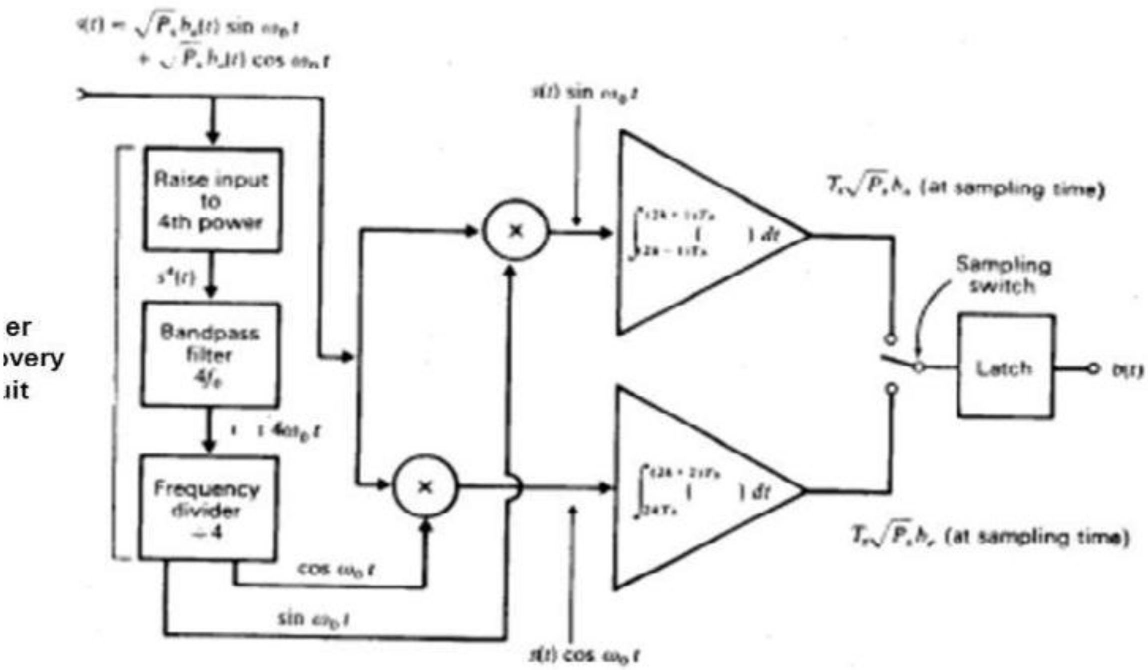
QPSK Transmitter Explaination:

- 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 two separate bit streams $b_o(t)$ and $b_e(t)$. The bit stream $b_e(t)$ consists of only the even numbered bits 2, 4, 6, 8, ... whereas $b_o(t)$ 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 $2T_b$. This duration is called as symbol duration T_s. Thus, every symbol contains two bits.
- The bit stream $b_e(t)$ is superimposed on a carrier $\sqrt{2Ps}\, cos\omega_c t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sqrt{2Ps}\, sin\omega_c t$ by using two balanced modulators (or multipliers) to generate $s_e(t)$ and $s_o(t)$. These two signals are basically BPSK signals.
- These signals are then added to generate the QPSK output signal $v_{QPSK}(t)$ given by,

$$v_{QPSK}(t) = b_o(t)\sqrt{2Ps}\, sin\omega_c t + b_e(t)\sqrt{2Ps}\, cos\omega_c t$$

QPSK receiver:

(QPSK receiver & Explanation: 4M)



QPSK Receiver Explaination:

- Let the received QPSK signal be $v_{QPSK}(t)$. The received QPSK signal $v_{QPSK}(t)$ is raised to the fourth power i.e., $v^4_{QPSK}(t)$.
- This signal is then filtered by using a BPF with a center frequency of $4\omega_c$. The output of the BPF is $-cos4\omega_c t$.
- A frequency divider divides the frequency at the filter output by 4 and generates the two carrier signals $sin\omega_c t$ and $cos\omega_c t$.
- The incoming signal $v_{QPSK}(t)$ is applied to two synchronous demodulators consisting of multipliers followed by an integrator. Each integrator integrates over a two-bit interval $T_s = 2T_b$.
- One synchronous demodulator uses $cos\omega_c t$ as the carrier signal and the other synchronous demodulator uses $sin\omega_c t$ as the carrier signal. The input to the upper integrator is given by,

$$v_{QPSK}(t)\times sin\omega_c t= b_o(t)\sqrt{2Ps}\,sin^2\omega_c t + b_e(t)\sqrt{2Ps}\,sin\omega_c t\,cos\omega_c t$$

- The upper integrator output is given by,

$$= b_o(t)\sqrt{2Ps}\int_0^{2Tb} sin2\omega ct + b_e(t)\sqrt{2Ps}\int_0^{2Tb} sin\omega ct\,cos\omega ct$$

We know that, $sin^2\omega_c t= \frac{1}{2}[1 - cos2\omega_c t]$

$$sin\omega_c t\,cos\omega_c t= \frac{1}{2}sin2\omega_c t$$

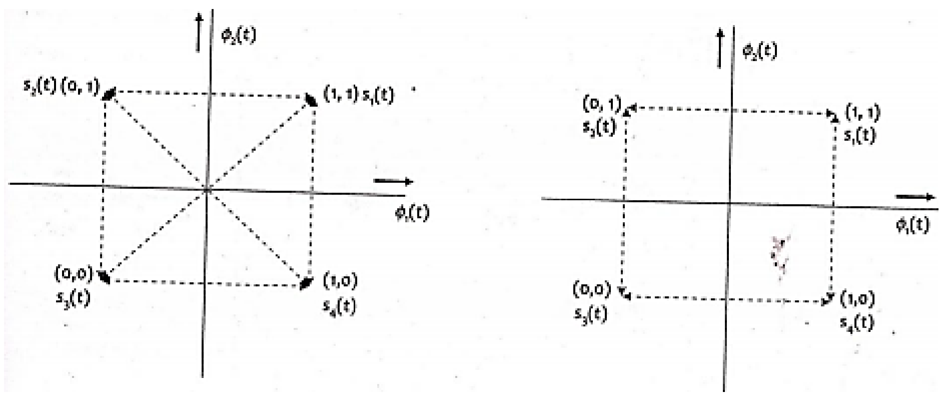
$$\int_0^{2Tb}\frac{1}{2}sin2\omega ct = 0$$

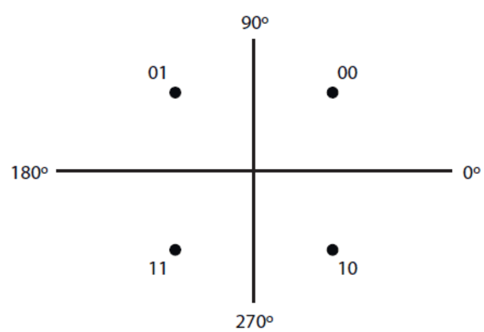
Integrator output = $\frac{1}{2}b_o(t)\sqrt{2Ps}\int_0^{2Tb} dt$

$= b_o(t)\sqrt{2Ps}\ T_b$

- Similarly, the output of the lower integrator is given by $b_e(t)\sqrt{2Ps}\ T_b$
- Thus, at the output of the two integrators we obtain the bit streams $b_e(t)$ and $b_o(t)$.
- Bit synchronizer is used to establish the beginning and end of the bit intervals of each bit stream. It is also used to operate the sampling switch.
- The integrator output is sampled at the end of each integration time for each integrator. The samples are taken alternately from the two integrator outputs at the end of each bit time T_b and these samples are then held in the latch for the bit time T_b. Each integrator output is thus sampled at intervals $2T_b$. At the output of the latch we get the signal $b(t)$.

Constellation diagram:



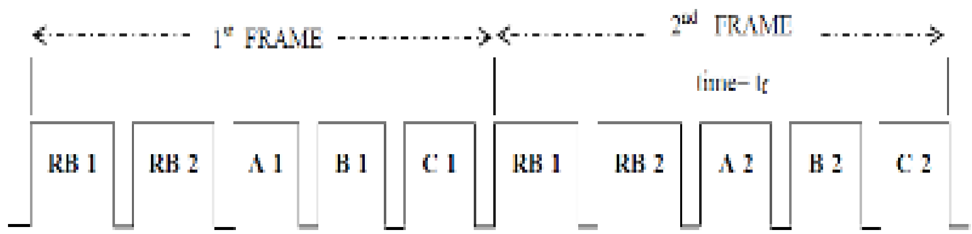


(b)	**Draw block diagram of TDMA and explain it. State advantages of TDMA over FDMA.**	**8M**
Ans:	**TDMA technology :** • In TDMA, each user has all the bandwidth, all the power and part of the time. It is frequently used with data and digital voice transmission. TDMA sends data in buffer and hence it is bursty communication. It is non-continuous. TDMA cannot send an analog signal directly due to buffering required. It is used for digital data. • In this method, all the earth stations share transponder time. Each earth station in the network is allocated a time slot in a periodic sequence. • It is a method of time division multiplexing, digitally modulated carrier between participating earth stations within the satellite network through a common satellite transponder. With TDMA, each earth station transmits a short burst of digitally modulated carrier during a precise time slot (called epoch) within a TDMA frame.	**(Diagram:3M,Explanation:3M)**

- Each earth station‟s burst is synchronized so that it arrives at the satellitetransponder at a different time. Consequently, only one earth station‟s carrier is present in the transponder at any given time thus avoiding collision with another station‟s carrier.
- The transponder is an RF to RF repeater that simply receives the earth stations transmissions, amplifies them and retransmits them in a downlink beam that is received by all participating earth stations. Each earth station receives the bursts from all other earth stations and must select from them the traffic destined only for itself.

TDMA FRAME:

- A TDMA frame consists of one or two reference bursts and several traffic bursts. A new frame starts with fresh reference bursts. A set of two TDMA frames is illustrated in Figure for three stations



TDMA FRAME STRUCTURE

RB are the reference bursts and A, B and C are the traffic bursts. A guard band is used between bursts. There is no transmission during the guard time. It prevents overlapping that may occur between various bursts.

- The frame time tf is the time interval from the start of the reference burst RB-1 to the end of the last traffic burst (TB) of the frame. Typical frame time lies between 0.75 ms to 20 ms.
- The bursts transmitted from the earth stations in their respective slots are received at a receiving station as shown in Figure 5.17. RB will enable the correct bursts to be recognized by the concerned station while the other bursts will be ignored.

Figure: Diagram of TDMA system:

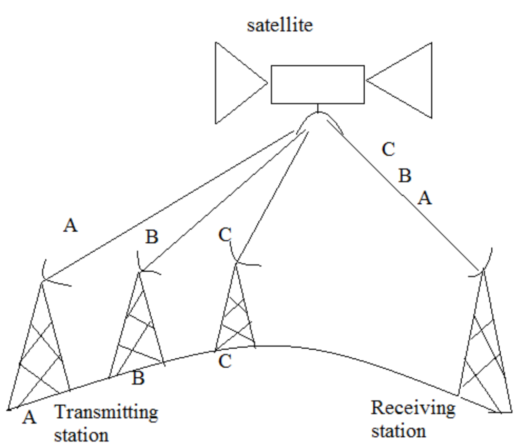


Figure: Diagram of TDMA system

As the transmission is done in burst mode, prior to transmission, input bits are temporarily stored in the transmitter"s memory storage and then sent during the assigned slot of time as burst signals.

Advantage of TDMA over FDMA:

2M

1. Intermodulation products are absent as there is one carrier only in all time slots.
2. Due to the absence of intermodulation products, TWT can be operated with maximum power output or saturation level.
3. It is easier to change the capacity between nodes by simply changing the duration and position of each burst in the TDMA frame. It is very flexible.
4. Transmission bit rate in TDMA is higher in FDMA due to burst mode of operation.
5. As the transmission is taking place in bursts, its interception by unauthorized elements is difficult. Hence it is more secure than FDMA.
6. TDMA adapt to transmission of data as well as voice communication.

(c)	**Draw the block diagram of Direct sequence spread spectrum and state the function of each block.**	**8M**

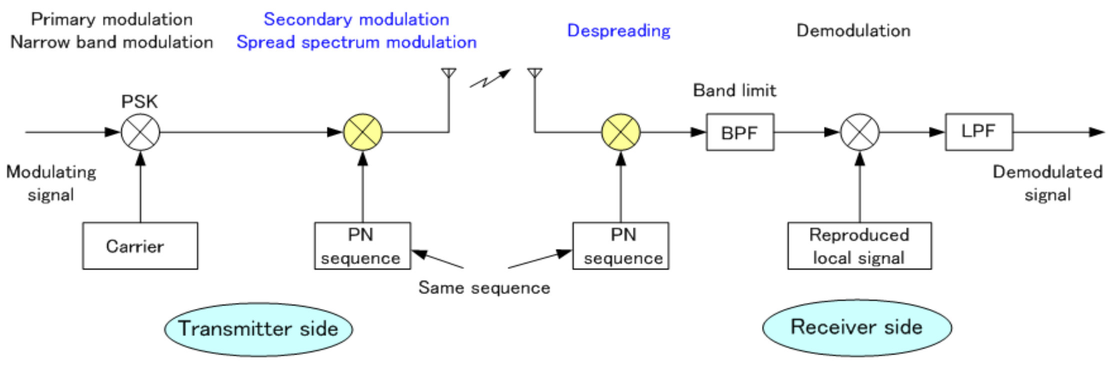
Ans:

Spread spectrum (DSSS):

(Diagram:4 M, Explanatio n n:4M)

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.

DSSS :



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 of spread spectrum modulation and demodulation using PSK for primary modulation.

Receiver:

If despreading is applied to the received diffuse wave, it returns to the PSK or FSK modulated wave resulting from primary modulation. Then, as with narrowband

demodulation, if the despread wave and local signal are multiplied, and appropriate low pass processing is applied, the information signal is obtained. Despreading involves multiplying the same PN code as that used at the transmitting end for the receiving wave. At this time, it's necessary to synchronize the receiving wave and PN code.
There are two processing methods on the receiving side, demodulation of the information signal after despreading, and obtaining a positive and negative PN code by multiplying the local signal by the receiving wave and despreading using correlation detection. With the former there is process gain but the problem of synchronization remains. With the latter, the spectrum density of the receiving wave itself is low, and regeneration of the local carrier for performing synchronous detection is a problem. Commercial SS radio equipment uses the latter, but it requires considerable power and has a short communication range.

Despreading:

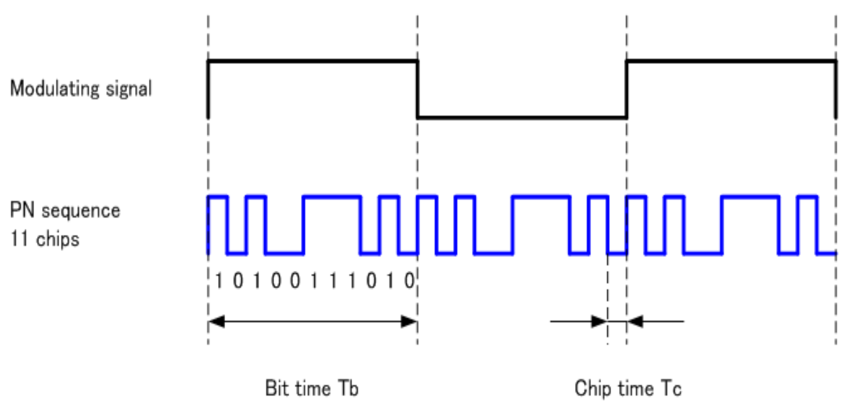
The signal that enters the antenna of the receiver includes outside interference waves and noise. If this signal is despread, the signal component returns to a narrowband modulated wave and the interference components are diffused, expanding the spectrum infinitely so that its power density falls. Therefore, by inputting the signal with frequency band restricted using a BPF, the interference component power that falls into the demodulation frequency band is reduced. The occurrence of errors is calculated using a stochastic process, so ultimately, using a spread spectrum results in fewer errors, and this is why spread spectrum communication is resistant to interference.

Demodulation:

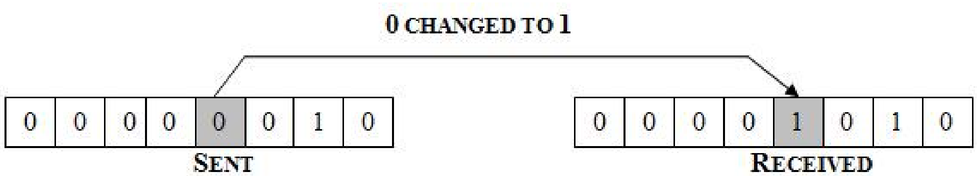
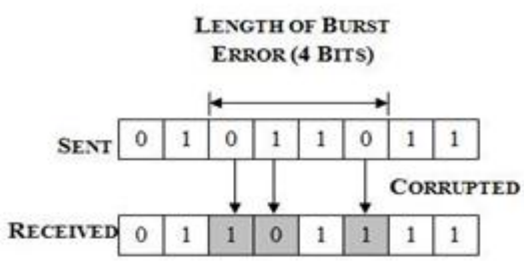
Demodulation is normal narrowband demodulation. The local signal is regenerated from the receiving wave and after multiplication by the receiving wave, unnecessary components are eliminated with an LPF. Primary modulation uses PSK, so synchronous detection is necessary.

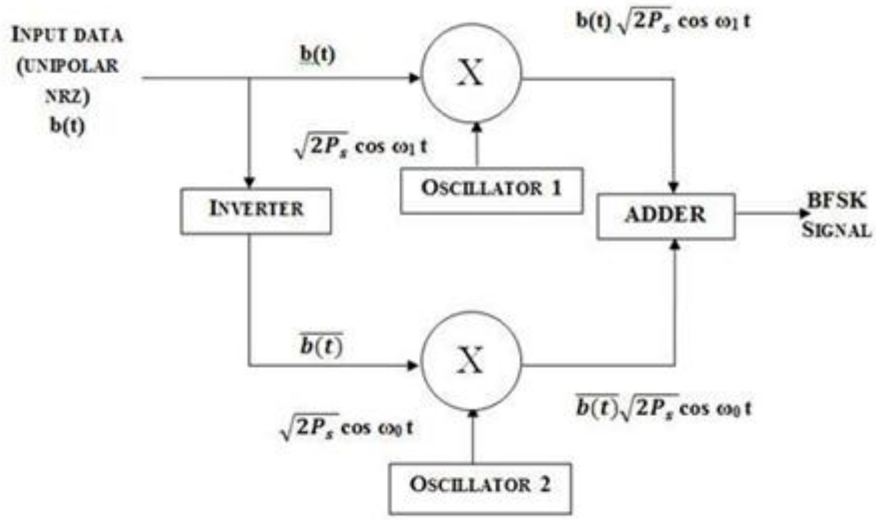
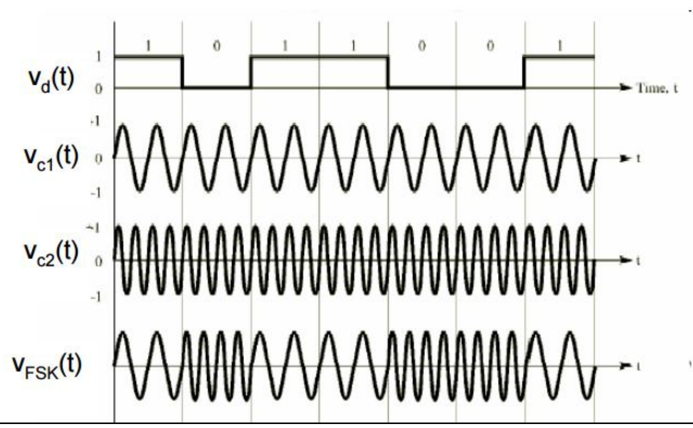
PNsequence :

The PN sequence is switched at a far faster speed than the symbol rate of the information signal and its spectrum covers a wide band. For this reason, the spectrum of the modulated wave after primary modulation also covers a wide band. We won't go into detail here, but PN sequences must meet the conditions required for spread spectrum modulation such as the relationship of the numbers 1 and 0.

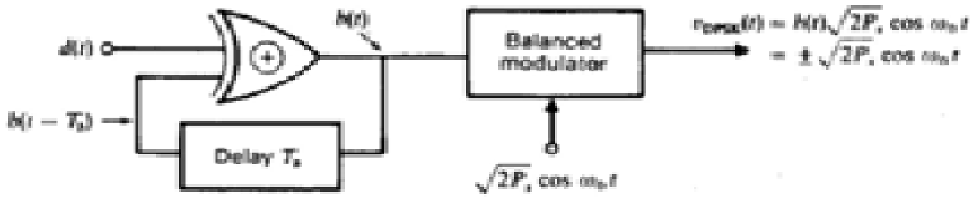


Q.6		**Attempt any FOUR of following :**	**16M**
	(a)	**Explain the role of predictor in differential pulse code modulation.**	**4M**
	Ans:	In DPCM if the redundancy is reduced, then the overall bit rate will decrease and the number of bits required to transmit one sample will also reduce. This type 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.	
	(b)	**State the types of errors present in the digital communication system. Also explain the causes and effects of errors.**	**4M**
	Ans:	**Types of error:** 1. Single bit error: Single-bit error occurs when only one bit of a given data string is in error (changed from 0 to 1 or from 1 to 0). 2. Burst error: A burst error or multiple-bit error occurs when two or more bits within a given data string are in error. **Example**: 1. Single-bit errors affect only one character within a message. The following figure illustrates single-bit error.  Single-bit error 2. Burst errors can affect two or more characters within a message. The length of the burst is measured from the first corrupted bit to the last corrupted bit. Some bits in between may not have been corrupted as shown in Figure .  Burst error of length 4 **Causes of errors:** Due to addition of noise in transmission & reception of data following errors occur. 1. If data block is lost in the network as it has been delivered to wrong destination. 2. If two or	**(2 M types of error, 1 M for Causes 1 M for effects of error.)**

		more bits from data unit such as a byte change from 1 to 0 or 0 to 1.	
	(c)	**Draw and explain the block diagram of FSK with suitable waveform.**	**4M**
	Ans:	**Block diagram of FSK :**  **Explanation :** In FSK, the frequency of the carrier is changed with respect to the input bits 1 & 0. In case of binary data, two carrier frequencies are used. The carrier frequency corresponding to logic 0 or binary 0 is called as *space frequency* and the carrier frequency corresponding to binary 1 is called as *markfrequency*. As shown in Figure, the input binary data is given directly to the multiplier and is inverted and given to second multiplier. Two different carriers have different frequency generated by the two oscillators and applied to the multipliers. The output of both the multipliers is an ASK signal which is added by the summer. Thus, the output of the adder is the BFSK wave. **Waveforms:**  	**2M** **1M** **1M**

(d)	**Explain the generation of DPSK with block diagram.**	**4M**
	In BPSK receiver, the carrier recovery is done by squaring the received signal. Hence, when the received signal is generated by negative data bit, it is squared and thus we cannot determine if the received bit is –b (t) or b (t) .Hence DPSK is used to eliminate the ambiguity of the received bit. The DPSK transmitter is as shown	**PSK block Diagram: 2M**



$d(t)$		$b(t-T_b)$		$b(t)$	
logic level	voltage	logic level	voltage	logic level	voltage
0	−1	0	−1	0	−1
0	−1	1	1	1	1
1	1	0	−1	1	1
1	1	1	1	0	−1

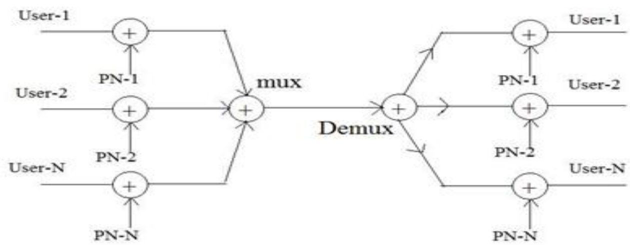
2M

Explaination:

1) d (t) represents the data stream which is to be transmitted it is to one input of an EX-OR logic gate.
2) The EX-OR gate output ,,b (t)" is delayed by one bit period the applied to the other input of EX-OR gate. The delayed represented by ,,b (t-Tb)".
3) Depending on the values of ,,d (t)" and ,,b (t-Tb)" the EX-OR produces the output sequence ,,b (t)".the waveform for the generator .the waveform drawn by arbitrarily assuming that in the first interval b (0) = 0.
4) Output of EX-OR gate is the applied to a bipolar NRZ level which converts ,,b(t)" to a bipolar level "b(t) as shown

b(t)	b(t)
0	-1
1	+1

(e)	**Explain the concept of CDMA technology.**	**4M**
Ans:	**Diagram:**	**2M**



CDMA

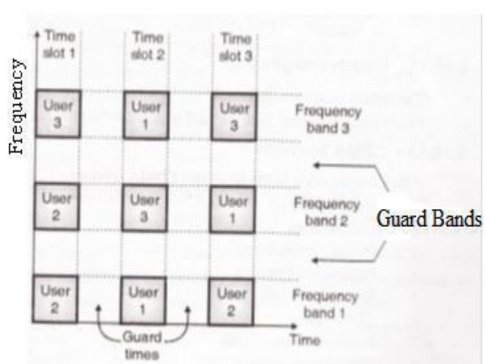
Explanation:

- CDMA system uses same frequency band and transmit simultaneously. They can use the whole available bandwidth for all the time. The transmitted signal is recovered by co-relating the received signal with the PN code used by the transmitter.
- CDMA allows all the users to occupy all channels at the same time. Transmitted signal is spread over the whole band and each voice or data call is assigned a unique code to differentiate it from other calls carried over the space spectrum.
- All the users in CDMA use same carrier and may transmit simultaneously. Each user has its own pseudorandom code word which is orthogonal to all other code words. For detection of message signal the receiver needs to know the code word use by transmitter. Each user operates independently with no. of knowledge of other users.

2M

OR

- In CDMA more than one user is allowed to share a channel or sub channel with the help of DSSS signals.
- In CDMA each user is given a unique code sequence. This sequence allows the user to spread the information signal across the assigned frequency.
- At the receiver the signal received from various users are separated by checking the cross correlation of the received signal with each possible user sequence.
- In CDMA as the bandwidth as well as time of the channel is being shared by users. In CDMA the users access the channel in a random manner.
- Hence the signals transmitted by multiple users will overlap both in time and in frequency. CDMA does not need any synchronization.



Structure of CDMA