MODEL ANSWER

SUMMER– 18 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 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.

1. (A) Attempt any THREE of the following: 12 marks

(a) Describe the different types of error present in data communication.

Ans: (Types of errors – 2 marks, description – 2 mks)

1. **Single bit error:**

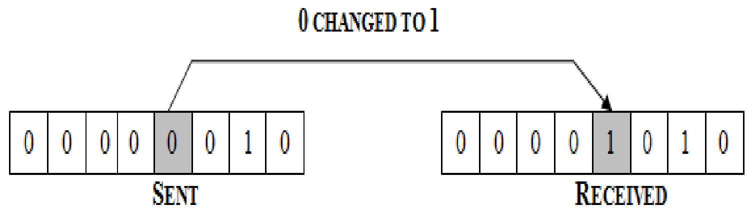
Single-bit error occurs when only one bit of a given data string is in error (changed from 0 to 1 or from 1 to 0).

2. **Burst error:**

A burst error or multiple-bit error occurs when two or more bits within a given data string are in error.

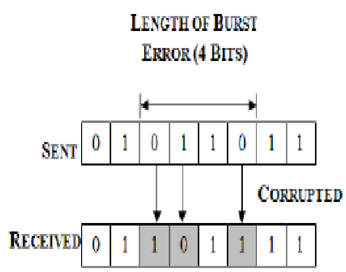
Example-

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

(b) State and explain Sampling Theorem with necessary waveforms.

Ans:

Statement: **02M**

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 $f_s \geq 2f_m$. Here f_s *is* the *sampling frequency*. This theorem is also known as the *Sampling Theorem for Baseband or Low-pass Signals.*

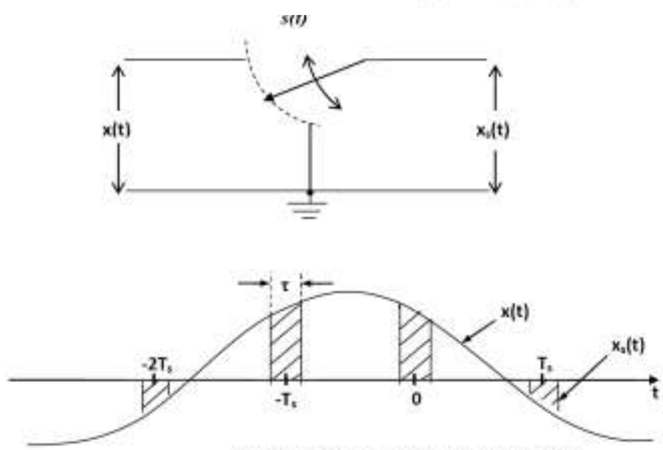
Description **01M**

Sampling process convert a continuous time varying signal to a discrete time varying signal. As shown in waveforms the x(t) is the modulating/information signal with frequency Fm ,which is sampled at a frequency Fs in such a way that Fs is greater or equal to 2Fm so that at the receiver the information is recovered from the sampled received with minimum distortion.

Diagram:- **01M**

The output $x_s(t)$ of the sampler consists of segments of $x(t)$. So, $x_s(t)$ can be represented as,

$$x_s(t) = x(t)s(t)$$



Switching Sampler and its output waveform

c) Define multiplexing & describe the need of multiplexing.

Ans: Definition- **02M**

Multiplexing:-

Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link for efficient and effective utilization of the link or channel bandwidth.

Need of multiplexing **02M**

In the application like telephony there are large numbers of users involved. It is not possible to lay a separate pair of wires from each subscriber to the other entire subscriber this is very expensive and practically impossible.

- In the Process of multiplexing two or more individual signals are transmitted over a single communication channel. Here we used medium as a coaxial cable or an optical fiber cable because of multiplexing bandwidth utilization is possible. As the data and telecommunications usage increases, so does the traffic. We can accommodate this increase by continuing to add individual lines each time a new channel is needed, or we can install higher capacity links and use each to carry multiple signals.
- Today’s technology includes high-bandwidth transmission media such as coaxial cable, optical fiber and terrestrial and satellite microwaves.
- Each of these has a carrying capacity (bandwidth) far in excess of that needed for the average transmission signal. If the bandwidth of the link is greater than the transmission needs of the devices connected to it, the excess capacity is wasted.
- An efficient system maximizes the utilization of all resources. Bandwidth is one of the most precious resources in data communications.

d) List the applications of spread spectrum modulation. (any four)

Ans:-

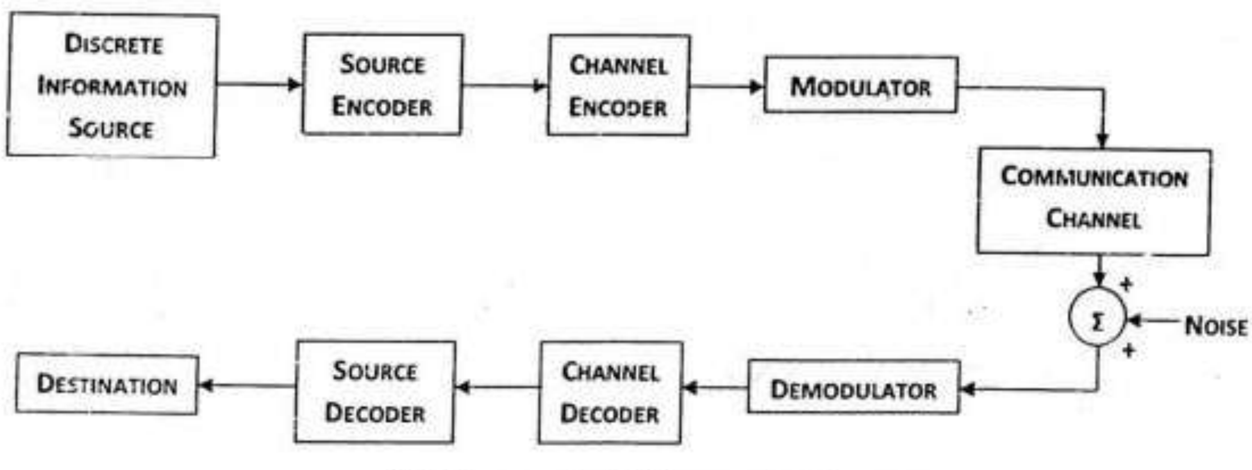
<u>Application of S. S. modulation:-</u> **01M each**

1. Military application- resistance to jamming.
2. Secure communication.
3. CDMA in satellite communication.
4. Police radar can employ spread spectrum to avoid detection by detectors employed by drives.
5. Low density power spectra for signal hiding.
6. Multipath rejection in a ground based mobile ration.
7. In local area network.
8. In global positioning system (GPS).

(B) Attempt any ONE of the following : **6**

(a) Draw the block diagram of the basic digital comm. system. State the function of each block in detail

Ans:- Diagram:- 2 mks



Block diagram of a digital communication system

Explanation: 04 mks

INFORMATION SOURCE: An *Information source* generates a message, examples of which include human voice, television picture, teletype data, atmospheric temperature and pressure.

The message signal can be of an *analog* or *digital* type. An analog signal can be converted into digital form through the process of *sampling*, *quantizing* and *encoding*.

In a digital signal, on the other hand, both amplitude and time take on *discrete values*. Computer data and telegraph signals are examples of digital signals.

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. It reduces the redundancy by performing a one-to-one mapping of its input bit stream into another bit stream at its output but with fewer digits.

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 themselves convey no information, they make it possible for the receiver to detect and/or correct some of the errors in the information bearing bits. This is needed because the data gets corrupted by the additive noise on the channel and this gives rise to the possibility of the channel decoder committing mistakes in decoding the data received from the channel.

MODULATOR: The modulator accepts a bit stream as its input and converts it to an electrical waveform suitable for transmission over the communication channel as they are basically analog in nature. Modulation can be effectively used to minimize the effects of channel noise, to match the frequency spectrum of the transmitted signal with channel characteristics, to provide the capability to multiplex many signals and to overcome some equipment limitations.

COMMUNICATION CHANNEL: The communication channel provides the electrical connection between the source and the destination. The channel may be a pair of wire or a telephone link or free space over which the information bearing signal is radiated. Due to physical limitations, communication channels have only finite bandwidth (*B* Hz) and the information bearing signal often suffers amplitude and phase distortion as it travels over the channel. While some of the degrading effects of the channel can be removed or compensated for, the effects of noise cannot be completely removed.

DEMODULATOR: Modulation is a reversible process and the extraction of the message from the information bearing waveform produced by the modulator is accomplished by the demodulator. There are a variety of techniques available for demodulating a given modulated waveform; the actual procedure used determines the equipment complexity needed and the accuracy of demodulation.

CHANNEL DECODER: The channel decoder recovers the information bearing bits from the coded binary stream. Error detection and possible correction is also performed by the channel decoder. The decoder operates either in a block mode or in a continuous sequential mode depending on the type of coding used in the system.

SOURCE DECODER: At the receiver, the source decoder converts the binary output of the channel decoder into a symbol sequence. The decoder for a system using fixed-length coding is quite simple, but the decoder for a system using variable-length coding will be very complex.

Decoders for such systems must be able to cope with a number of problems such as growing memory requirements and loss of synchronization due to bit errors.

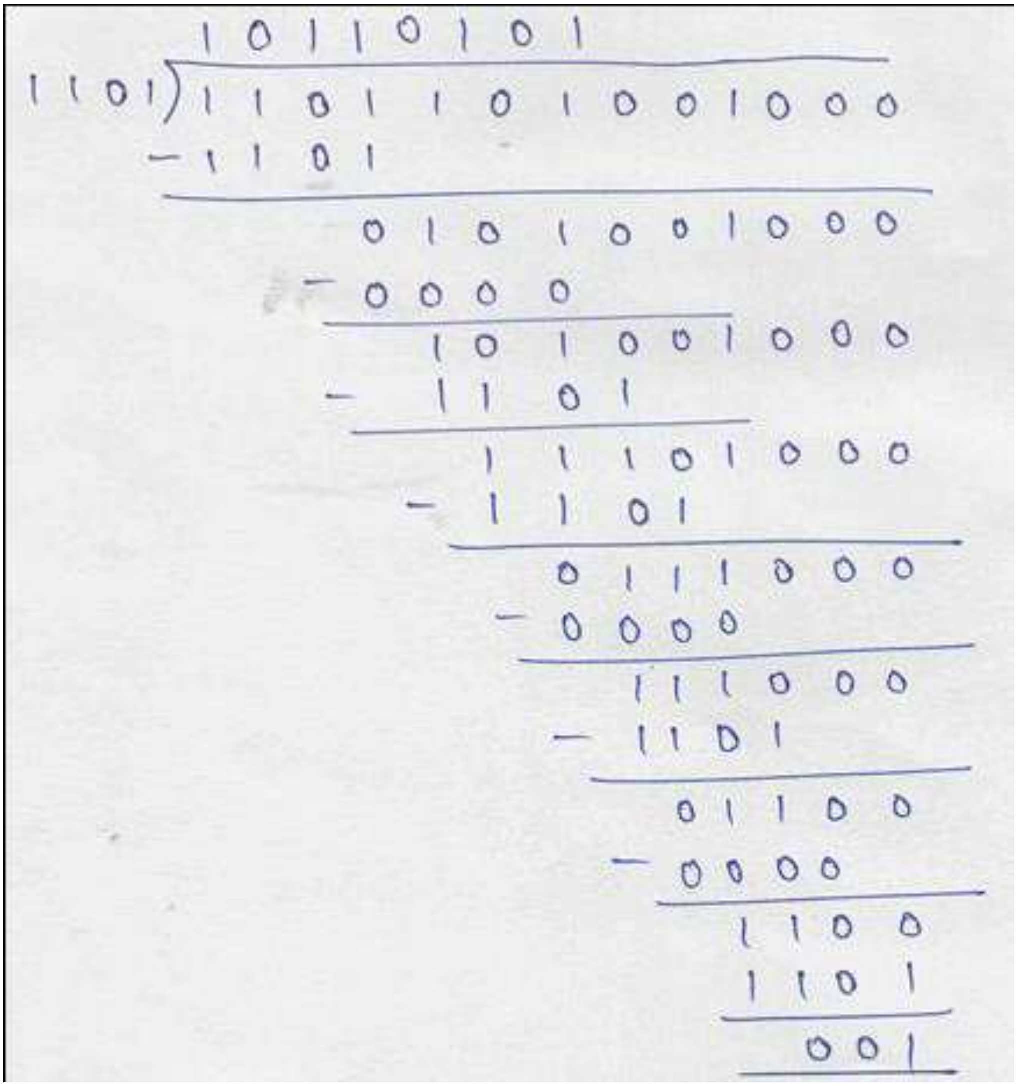
(b) Generate CRC code for data word 1101101001 by using divisor as 1101. State the two advantages of CRC method.

Ans. :-(Correctly solved Answer – 4 Marks, any 2 Advantages of CRC – 1 Mark Each)

Dividend: 1 1 0 1 1 0 1 0 0 1

Divisor: 1 1 0 1

No of zeros to be added to dividend: 3



Code word : 1 1 0 1 1 0 1 0 0 1 | 0 0 1

Advantages of CRC Codes

1. Implementation of encoding and error detection circuits is practically possible.

2. CRC codes are capable of detecting any kind of error bursts.

3. CRC can detect all burst errors of length less than or equal to degree of the polynomial.

2. Attempt any TWO of the following : **16**

(a) Draw the neat block diagram of PCM transmitter and receiver. Explain same with waveforms.

Ans. PCM Transmitter Diagram **(2 Marks)**

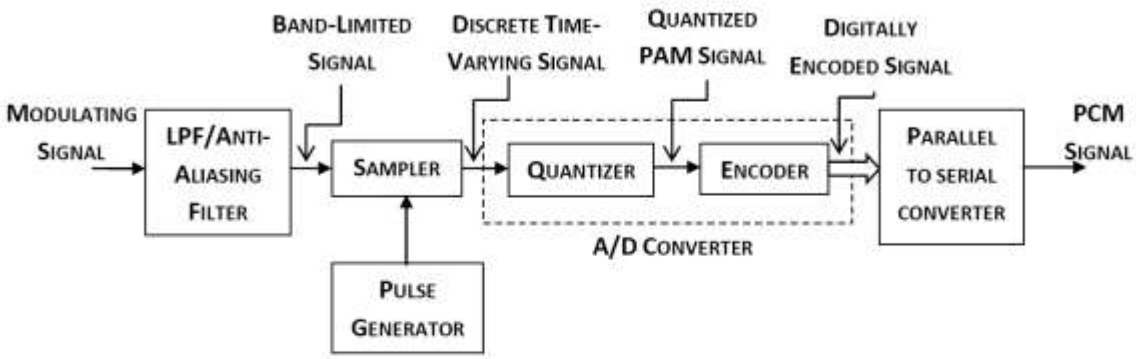


Fig. block diagram of PCM Transmitter

PCM Transmitter Explanation **(1 Mark)**

- The analog signal/modulating 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 sampled and hold 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 of sampled and hold block is a flat topped PAM signal.
- These samples are subjected to operation "quantization" in the "quantizer". Quantization is a process of approximation of the value of respective sample into 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.

PCM Receiver Diagram **(2 Marks)**

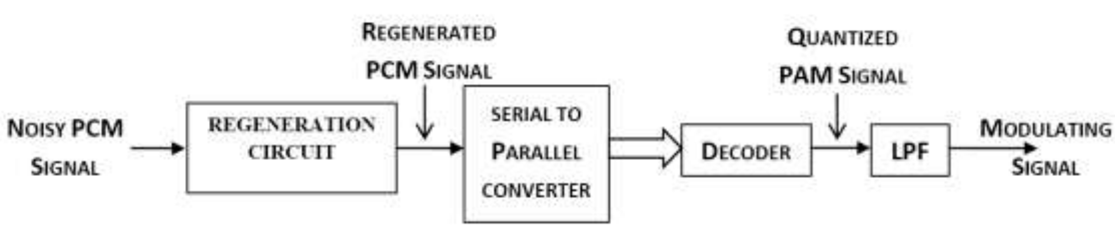
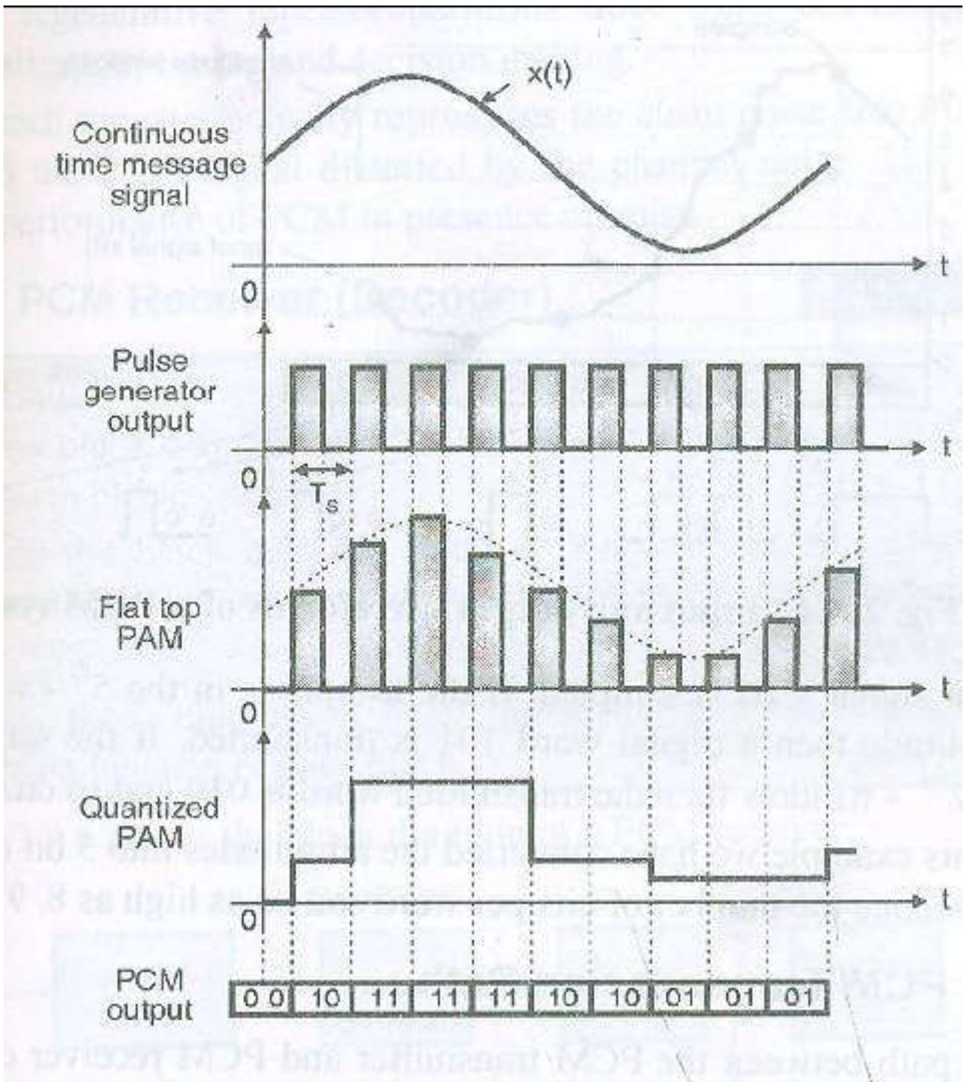


Fig. block diagram of PCM Receiver

PCM Receiver Explanation **(1 Mark)**

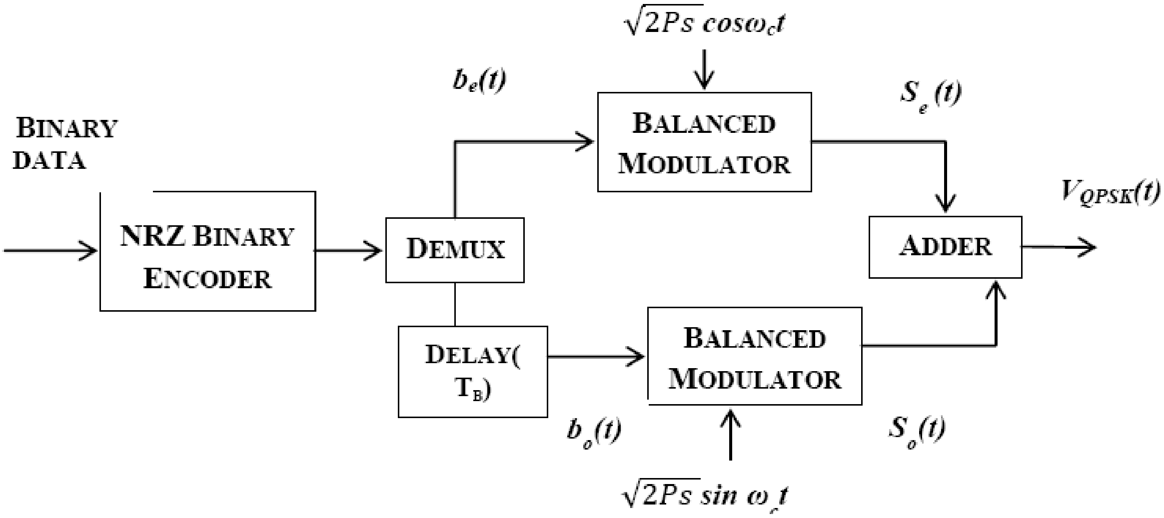
- A PCM signal contaminated with noise is available at the receive input.
- The regeneration circuit at the receiver will separate PCM pulses from noise and will reconstruct original PCM signal.
- Cleaned PCM is fed to a serial to parallel converter.
- Then applied to a decoder which converts each codeword into corresponding quantized sample value.
- This quantized PAM signal is passed through a low pass filter recovers the analog signal x (t).

Waveforms **(2 Marks)**

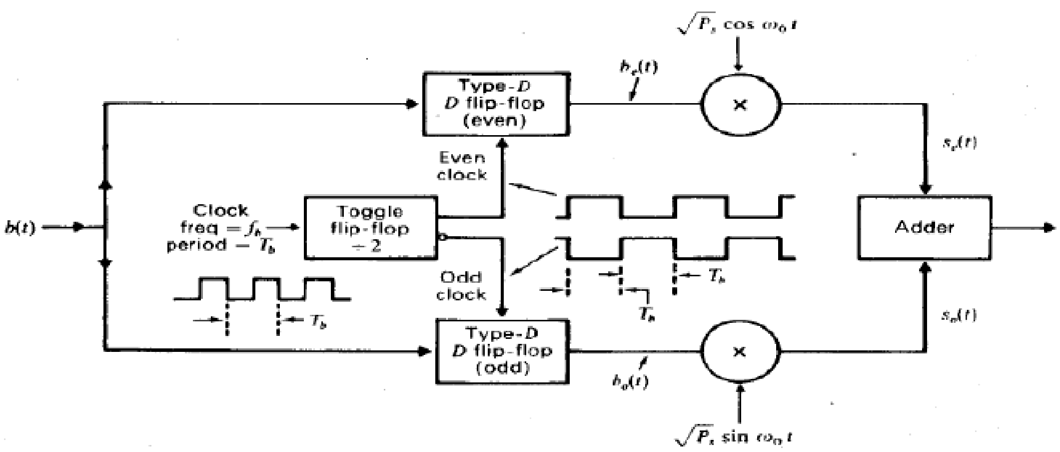


(b) Describe QPSK modulator and demodulator and draw it's constellation diagram.

Ans:-Transmitter and receiver diagram- 2 mks each, description – 1 mks each, constellation diagram-2 mks



QPSK Transmitter



(OR)

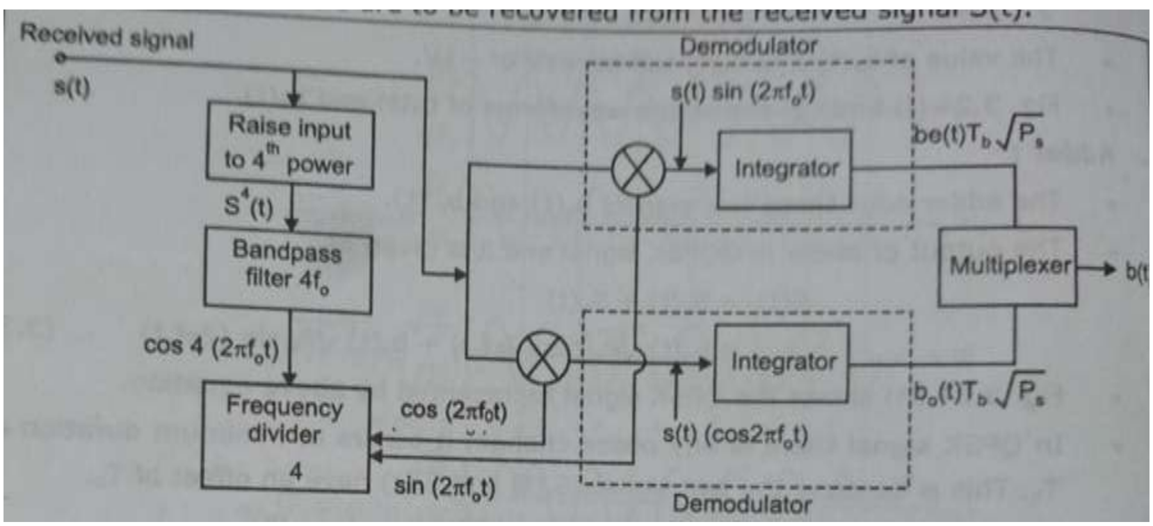
Working Principle

- QPSK is an expanded version from binary PSK where in a symbol consists of two bits and two orthonormal basis functions are used. A group of two bits is often called a 'dibit'. So, four dibits are possible. Each symbol carries same energy.

- The number of phase shifts in phase shift keying is not limited to only two states. The transmitted "carrier" can undergo any number of phase changes and, by multiplying the received signal by a sine wave of equal frequency, will demodulate the phase shifts into frequency-independent voltage levels which is nothing but the demodulated output.

- This is indeed the case in quadrature phase-shift keying (QPSK). With QPSK, the carrier undergoes four changes in phase (four symbols) and can thus represent 2 binary bits of data per symbol. Although this may seem insignificant initially, a modulation scheme has now been supposed that enables a carrier to transmit 2 bits of information instead of 1, thus effectively doubling the bandwidth of the carrier.

Receiver

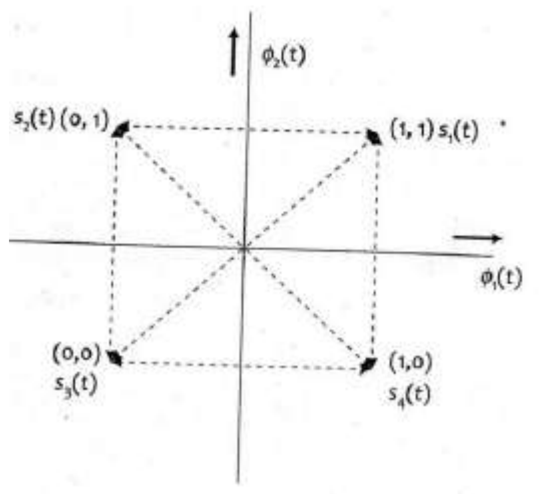


Explanation:- The received signal is raised to the power of 4 and passed through the bandpass filter centered around 4Fo ,so o/p is a coherent carrier which is further divided by frequency divider(% by 4) that gives two coherent carrier recovered frequencies Cos (2π Fot)and Sin (2 πFot).

The two carrier are applied to the two synchronous demodulators consisting of multiplier and integrator.. The o/p is product signals S(t)Sin(2 πFot) and S(t) Cos (2π Fot).The integrator integrates the product over two bit interval Ts= 2Tb, at the end of interval the o/p is sampled.upper integrator provides o/p= be(t)√P Tb and the lower o/p is= bo(t)√P Tb.

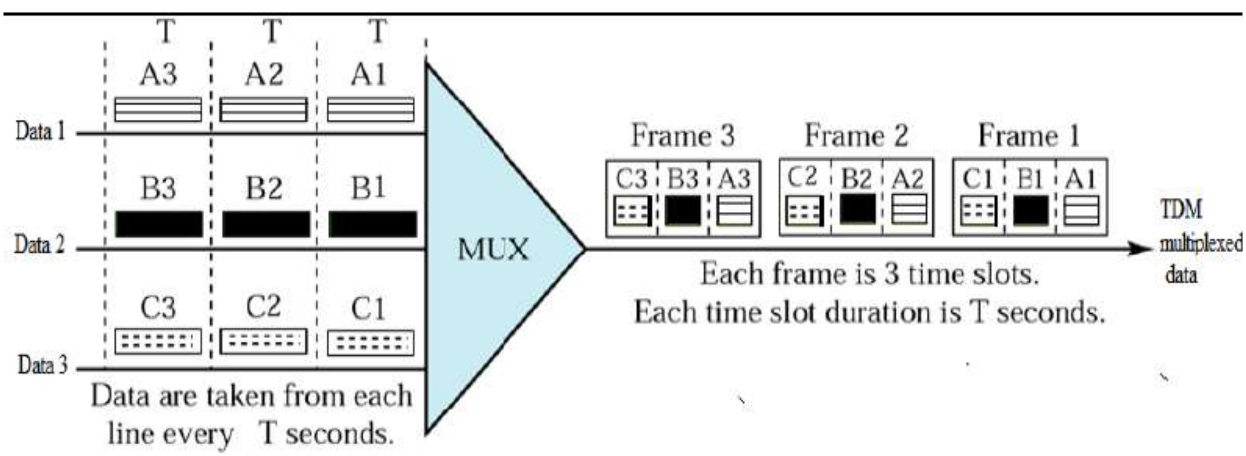
The odd and even i/ps are combined in the multiplier to provide odd and even bit sequence at its o/p.

Constellation diagram –



(c) Describe Time Division Multiplexing with block diagram and state it's two advantages and disadvantages.

Ans: (TDM transmitter block diagram – 3 marks, explanation – 3 marks, any two advantages – 1 marks , any 2 disadvantages- 1mks)
Diagram:



OR

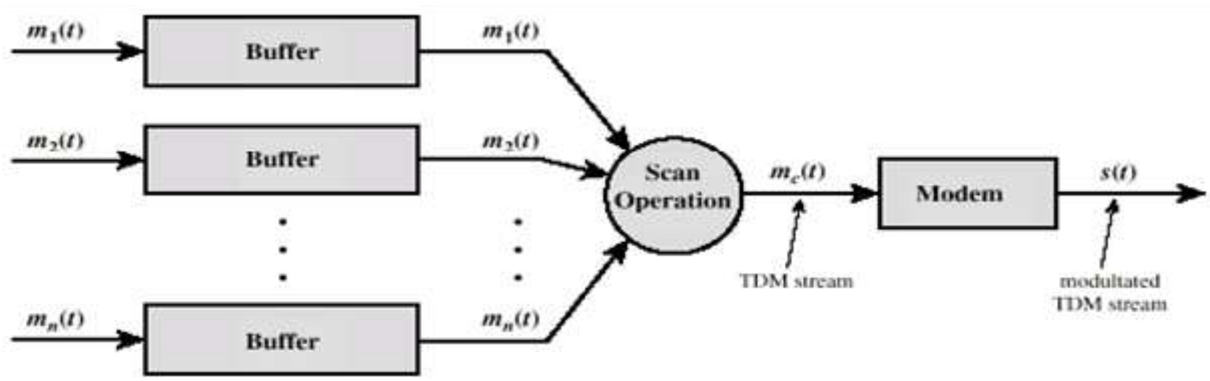


Fig: TDM Transmitter

Note: TDM Receiver not required to be explained

Explanation **3 mks**

- Process of combining digital signals from several sources whereby each connection occupies a portion of time in the link is called Time Division Multiplexing (TDM).
- Links are sectioned by time rather than frequency.
- Data flow of each connection is divided into units.
- In TDM data units from each input connection is collected in to a frame i.e link combines one unit of each connection to make a frame.
- If we have ¡§n¡¨ connection a frame is divided in to ¡§n¡¨ time slots and one slot is allowed for each unit. i.e. n input connections „³n time slots.
- One for each input line, if the duration of input is T, the duration of each slot is T/n and the duration of each frame is T.
- Data rate of link must be n times the duration of a time slot to guarantee flow of data.
- Time slots are grouped into frames; one complete cycle of time slots; each slot dedicated to one device.
- A simple TDM process for three different data transmission is shown above.
- Here, all three data are divided into equal timeslots also called as units.
- And each data unit from all three data are combined / multiplexed together to form TDM frames comprising of small units of all three data which is further transmitted.

Advantages:-
1. Full available channel bandwidth can be utilized for each channel.
2 .Intermodulation distortion is absent.
3. The problem of crosstalk is not severe.

Disadvantages
1. Requires synchronization for proper operation.
2. Due to narrowband fading,all TDM channels may get wiped out.

3. Attempt any FOUR of the following : **16**

(a) What are the limitations of DM ? Explain, how they overcome in Adaptive Delta Modulation.

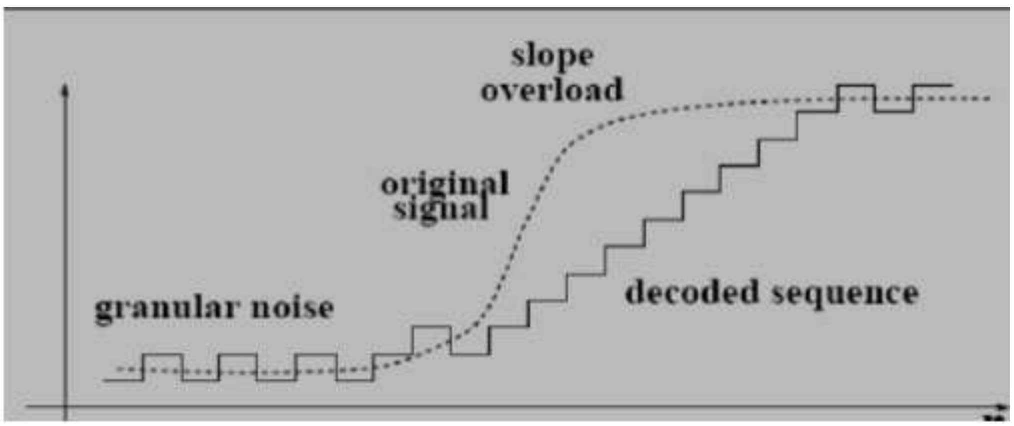
Ans:- Two Limitations- 1 mks (waveforms can be shown), ADM diagram-2 mks, explanation – 1 mks

In delta modulation, the step size is constant so that its 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.

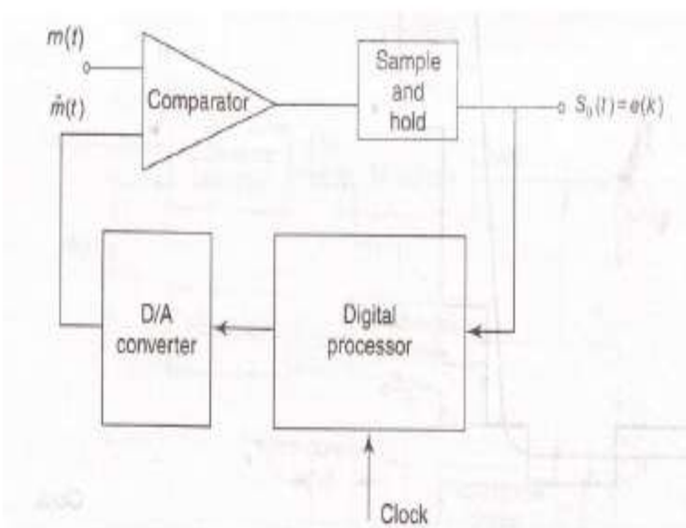
Thus the two limitations of DM are-

1) slope overload

2) granular noise



Block diagram of ADM



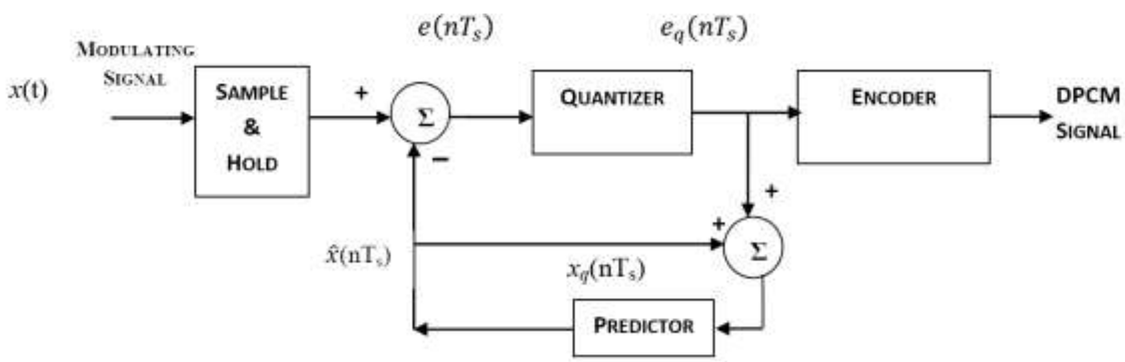
Explanation

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. 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.

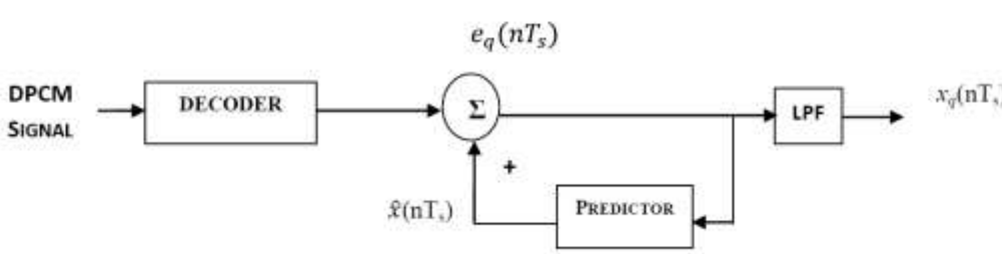
Thus with adaptive delta modulation the following are the advantages-
1. Slope overload distortion and granular noise problem in is reduced.
2. Improved signal to noise ratio.
3. Wide dynamic range is achieved with variable step size.
4. Better bandwidth utilization than delta modulation.

(b) Draw block schematic of DPCM transmitter and receiver.
Ans: Transmitter and receiver- 2 mks each

DPCM Transmitter:-



DPCM Receiver



c) Compare FDMA. TDMA & CDMA techniques based on (i) definition

(ii) bandwidth available (iii) code word & (iv) synchronization.

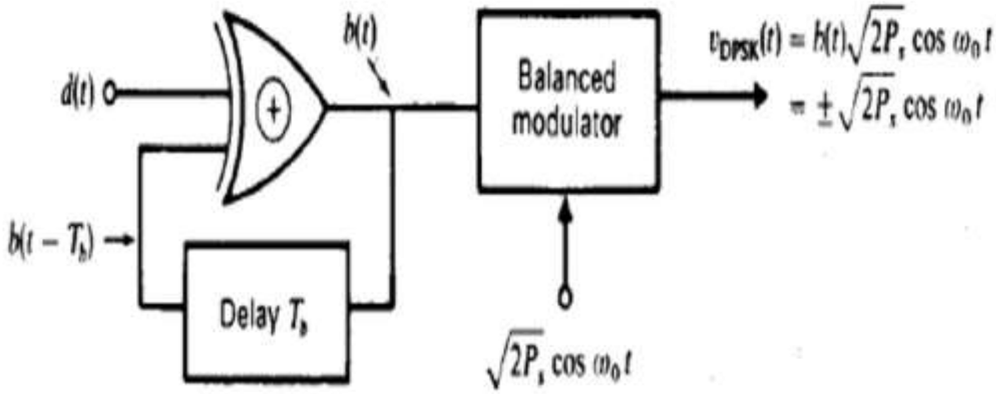
Ans: **(Each point 1Mark)**

PARAMETER	FDMA	TDMA	CDMA
Definition	Entire band of frequencies is divided into multiple RF channels/carriers. Each carrier is allocated to different users.	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.
Bandwidth available	Overall bandwidth is shared among many stations.	Time sharing of satellite transponder takes place	Sharing of bandwidth and time both takes place.
Synchronization	Synchronization is not necessary	Synchronization is essential	Synchronization is not necessary
Code word	Code word is not required	Code word is not required	Code words are required.

d) Describe DPSK transmitter with block diagram.

Ans:- Transmitter diagram- 2 mks, description – 1 mks, waveforms- 1 mks

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 -

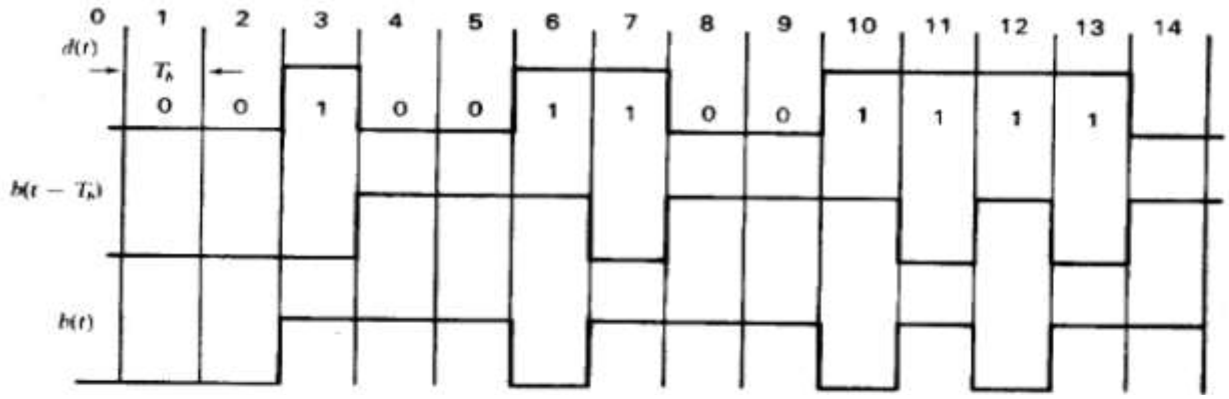


Explanation :- d (t) represents the data stream which is to be transmitted it is to one input of an EX-OR logic gate. The EX-OR gate output "b (t)" is delayed by one bit period the applied to the other input of EX-OR gate. The delayed represented by "b (t-Tb)". Depending on the values of "d (t)" and "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. 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

The o/p is given as-
VDpsk (t) = √ (2 Ps) cosωt
That Means no phase Shift has been introduced
But when b (t) = 0, b" (t) = -1 Hence
VDpsk (t) = √ (2 Ps) cosωt
Thus 180 Phase shift is introduced to represent b (t) =0

$d(t)$		$b(t-T_b)$		$b(t)$	
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

Waveforms:-



e) What is meant by M-ary encoding?Write the bandwidth requirement for ASK,BFSK and QPSK,

Ans:- Mary encoding- 1 mks, each correct bandwidth formula – 1 mks

M-ary encoding- In an M-ary signaling scheme, we can send one of the M possible signals such as s1, s2...sm (t) during each signaling interval of duration of t seconds.

The number of signals in an M is given as $M=2^N$.

The M-ary signals are obtained as follows.

Group of "N" bits together form N bit symbol.

These signals will extend over a period of NTb , where Tb is duration of one bit.

Due to grouping of n bit per symbols, we can have 2^N =M possible symbols.

These M possible signals are represented by sinusoidal signals of duration Ts = NTb which differ from one another by a phase of 2π/m radians .thus M-ary signal is produced at the output.

Bandwidth requirement-

1. $BW_{ASK} = 2\ F_b$
2. $BW_{BFSK} = 4\ F_b$
3. $BW_{QPSK} = F_b$

4. (A) Attempt any THREE of the following : **12**

(a) Differentiate the characteristics of communication channels with respect to bit rate, bandwidth, repeater distance and application, (any 4)

Ans:-Proper relevant comparison- 1 mks each

CHANNEL	BITRATE/ BANDWIDTH	REPEATER DISTANCE	APPLICATIONS
Unshielded twisted pair	64 kbps – 1 Gbps	Few km	Short haul PSTN, LAN
Co-axial cable	Few hundred Mbps	Few km	Cable TV, LAN
Optical fiber	Few Gbps	Few tens of km	Long haul PSTN,LAN
Free space broadcast	Few hundred KHZ to few hundred MHZ	No repeater	Broadcast radio /TV
Free space cellular	1 – 2 GHZ	No repeater up to base station	Mobile telephony, SMS
Wireless LAN	Up to 11 Mbps	No repeater up to access point	Wi-Fi, blue tooth
Terrestrial microwave link	2 – 40 GHZ	Every 10 – 100 km	Long haul PSTN, video transmission from playground to studio in a live telecast
Satellite	4/6 GHZ ,12/14 GHZ	Several thousand km	Transcontinental telephony, cable TV broadcast, DTH,GPS
Infrared	Few THZ	No repeater	Short distance LOS like TV remote.
Under Water Acoustic	Few KHZ	Few km	SONARS and all other under water communication

b) Explain quantization process with waveform.

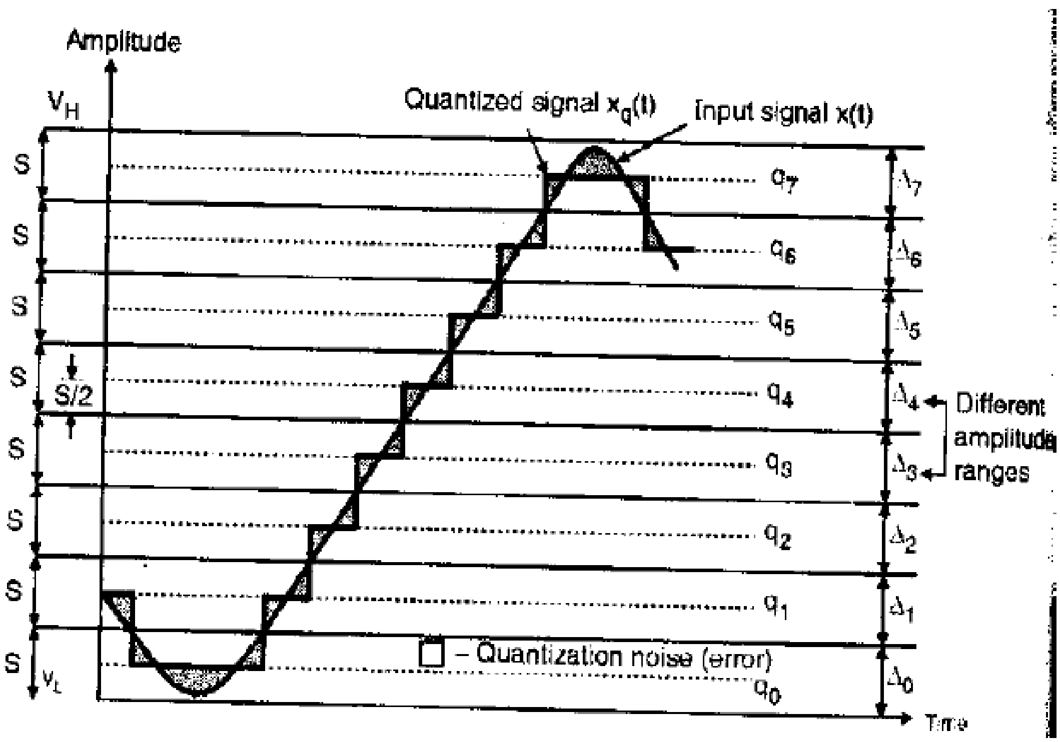
Ans: Explanation- 02M

Quantization is the process of approximation or rounding off the sampled signal. The quantizer converts sampled signal into approximated rounded values consisting of only finite no. of pre decided voltage levels called as quantization levels.

In the process of A to D conversion,,after sampling, quantization is the next step. The input signal x(t) is assumed to have a peak swing of VL to VH volts. This entire voltage range has been divided into Q equal intervals each of size "s". s is called as step size and its value is given as

$S= V_H-V_L /Q$

<u>Diagram of the Process quantization is as shown below</u>- **02M**



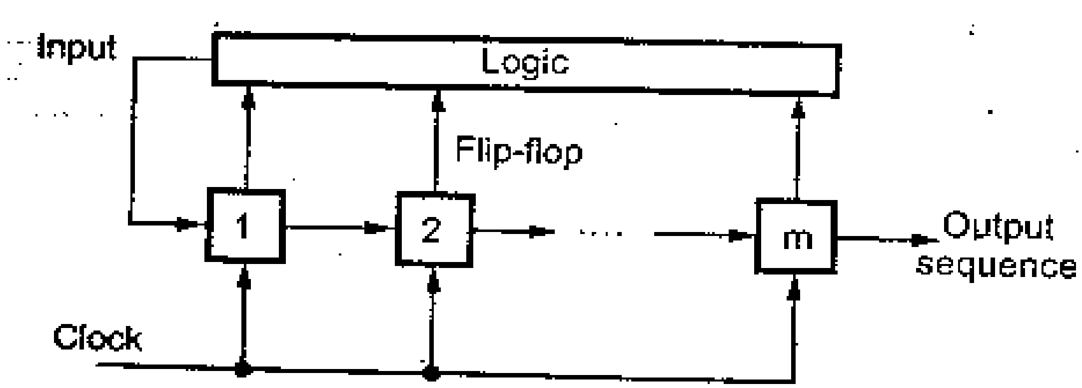
c) Define PN sequence. Draw the pseudo random sequence generator.

Ans: Definition:- 02M

A PN sequence is a noise like high frequency signal and defined as a pseudorandom coded sequence of 1s and 0s with certain auto correlation properties, generated by a well defined logic.

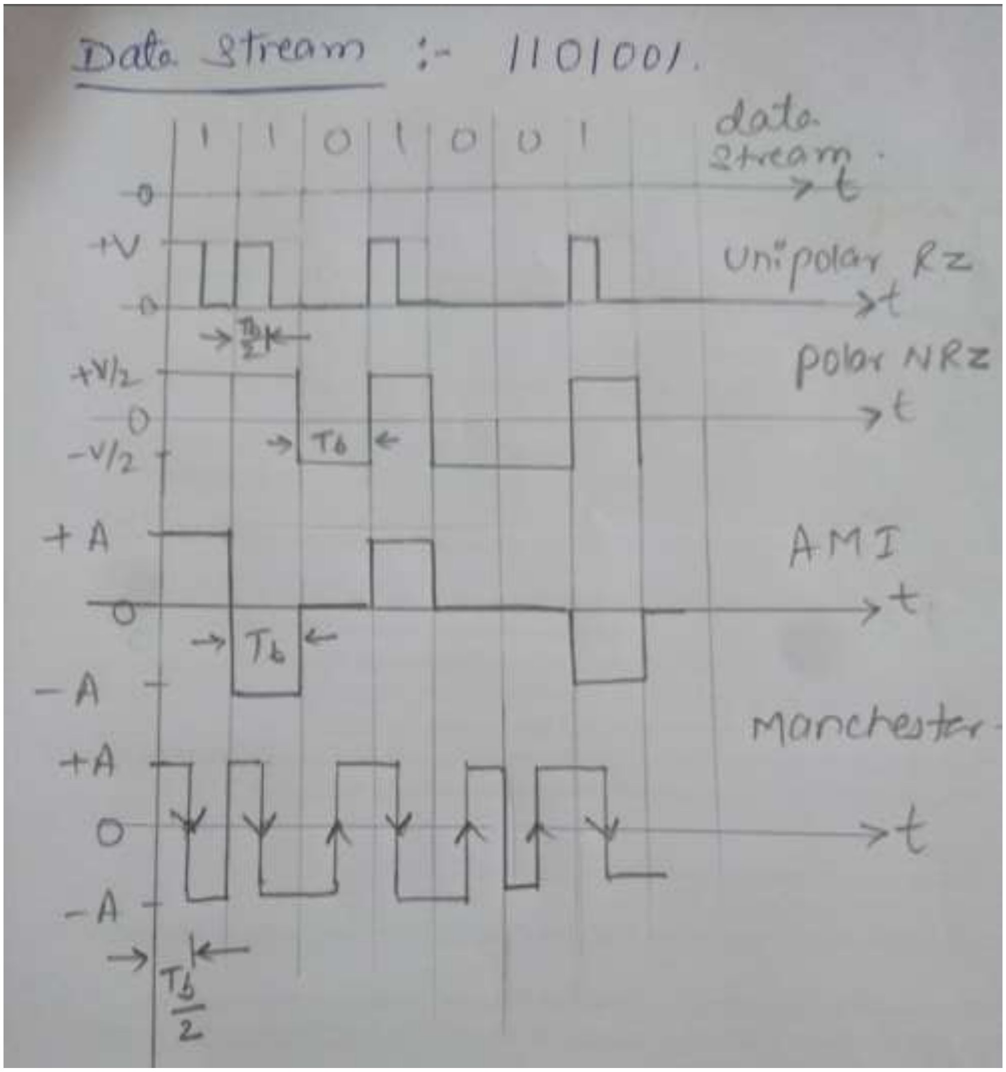
The pseudo random sequence generator:- 02M

. It can be generated by a shift register and the combinational logic circuit. The generalized block diagram of PN sequence generator scheme is as shown below.



d) For the binary data stream 1101001 draw unipolar RZ, Polar-NRZ, AMI and Manchester waveforms.

Ans: **Coded waveform- 01M each**



(B) Attempt any ONE of the following : **6**

(a) A discrete memory-less source has five message symbols S_0, S,, S_2, S_3 & S_4 with corresponding probabilities {0.08,0.2,0.12,0.4,0.15}

(i) Derive Huffman code for the above source & determine the average length of the code word.

(ii) Determine the code efficiency of the designed Huffman code

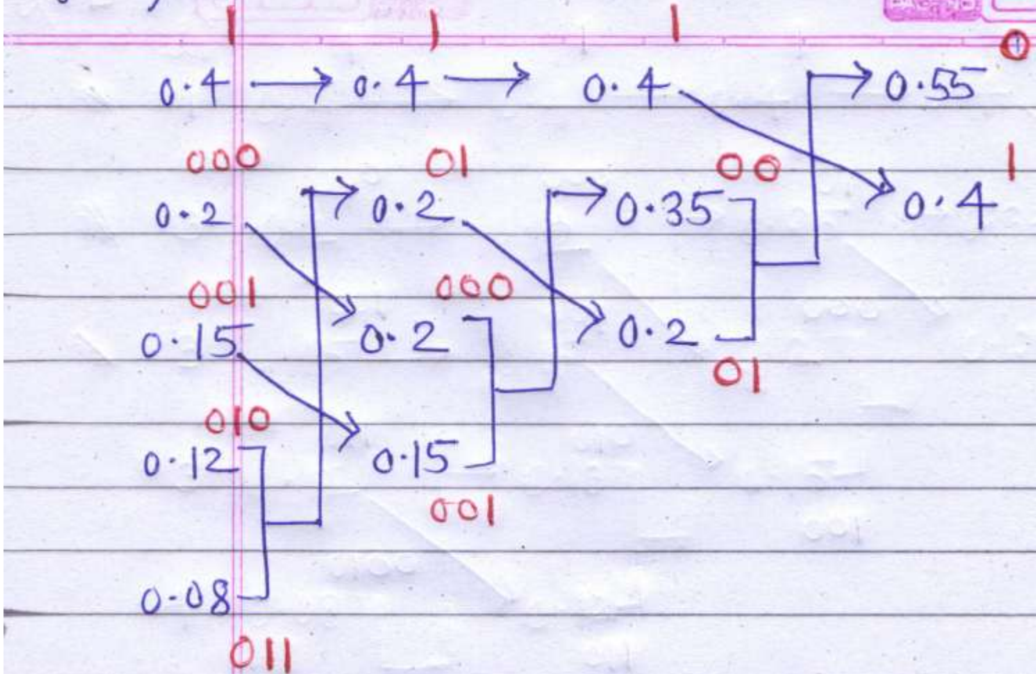
Ans:

i) Huffman code- (4 marks)
1) The average length of the code word (1 marks)
2) The coding efficiency of the Huffman code (1 marks)

In the given problem the addition of S_0, S,, S_2, S_3 & S_4

Is 0.95 where as it should be 1, give appropriate marks

Q4(B)(a)



$L_{avg} = P_i \times l_i$

$= 0.4\times1 + 0.2\times3 + 0.15\times3 + 0.12\times3 + 0.08\times3$

$= 2.05$ bits/symbol.

$H = \sum P_i \log_2\left(\frac{1}{P_i}\right)$

$= 2.0622$ bits/symbol.

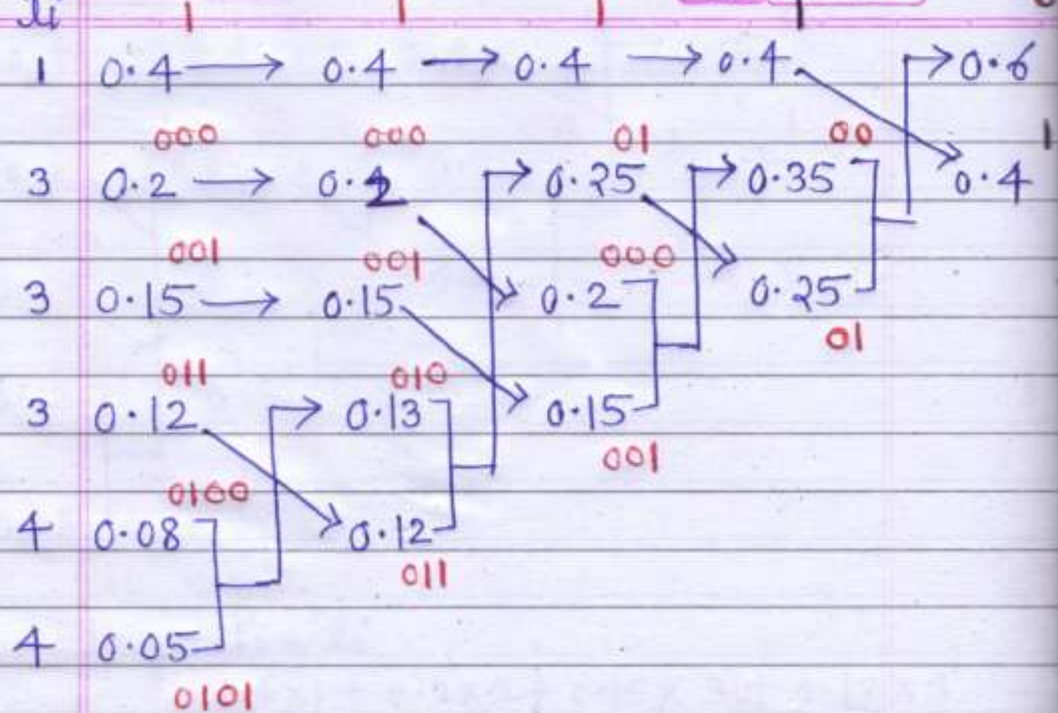
$\eta = \frac{H}{L_{avg}} \times 100 = \frac{2.0622}{2.05} \times 100$

$= 100.59\%$

Note:- But efficiency can not be more than 100%

Note:- By considering (Assuming) one more Probability 0.05 so that addition should be 1.

Q4B (a)



$L_{avg} = P_i \times l_i =$

$= 0.4\times1 + 0.2\times3 + 0.15\times3 + 0.12\times3 + 0.08\times4 + 0.05\times4$

$= 2.33$ bits/ symbol.

$$H = \sum P_i \log_2\left(\frac{1}{P_i}\right)$$

$$= 0.4\log_2\left(\frac{1}{0.4}\right) + 0.2\log_2\left(\frac{1}{0.2}\right) + 0.15\log_2\left(\frac{1}{0.15}\right) + 0.12\log_2\left(\frac{1}{0.12}\right) + 0.08\log_2\left(\frac{1}{0.08}\right) + 0.05\log_2\left(\frac{1}{0.05}\right)$$

$= 1.4037 + 0.8746 = 2.2783$ bits/symbol.

$$\eta = \frac{H}{L_{avg}} \times 100 = \frac{2.2783}{2.33} \times 100$$

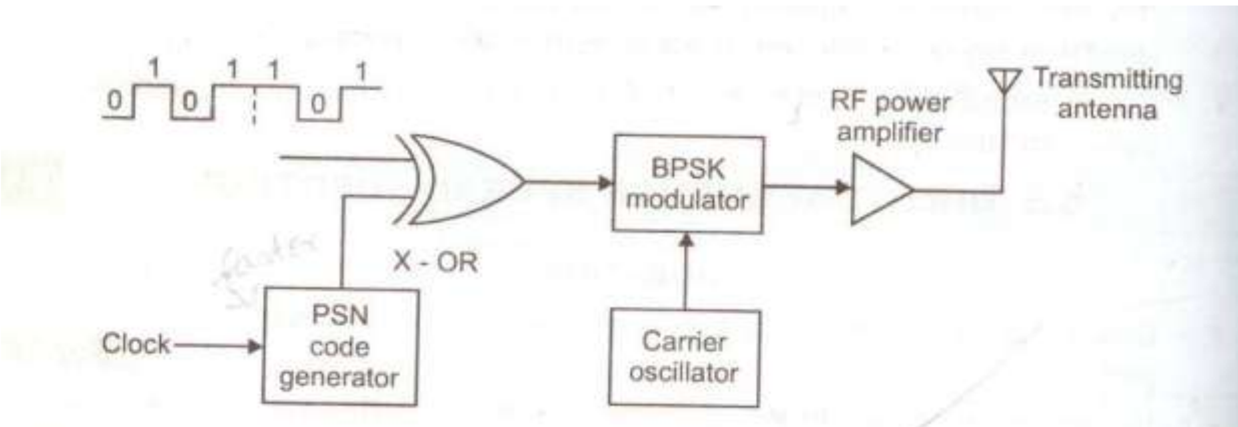
$= 97.78\%$

(b) Describe DSSS transmitter and receiver with block diagram.

Ans:- Transmitter and receiver block diagram – 2 mks each, explanation- 1 mks each

Direct sequence spread spectrum (DSSS): 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. Direct Sequence Spread Spectrum Coherent PSK Transmitter
The averaging system reduces the interference by averaging at over a long period the DSSS system is a averaging system. This technique can be used in practice for transmission of signal over a band pass channel (E.g. satellite channel). For such application the coherent binary phase shift (BPSK) is used in the transmitter and receiver.

Transmitter



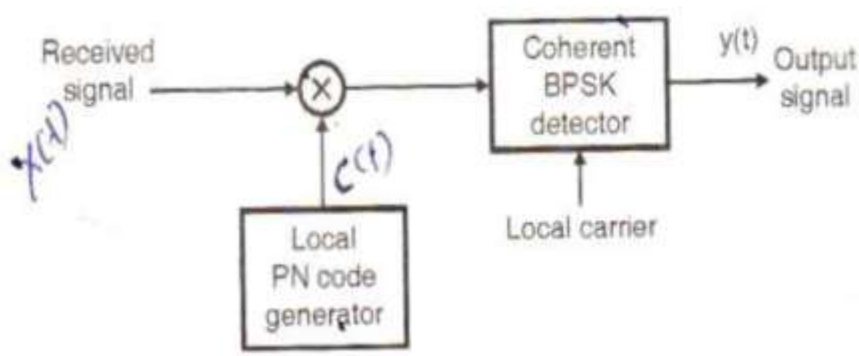
Explanation: The block diagram of DSSS transmitter is shown in figure above.The serial binary data is applied to an X- OR gate along with a serial pseudorandom code that occurs faster than binary data.
The signal developed at the output of the X-OR gate is then applied to a BPSK modulator. The carrier phase is switched between 0°and 180° by the 1"s and 0"s of X –OR output. The signal phase modulating carrier, being much higher in frequency than the data signal causes the modulator to produce multiple widely spaced sidebands whose strength is such that the complete signal takes up a great deal of the spectrum. Thus the signal is spread. Also because of its randomness, the resulting signal is appears to be nothing more than wideband noise to a conventional narrow band receiver.

One bit time for the pseudorandom code is called a chip and the rate of the code is called the chipping rate. The chipping rate is faster than the data rate.

Receiver:-

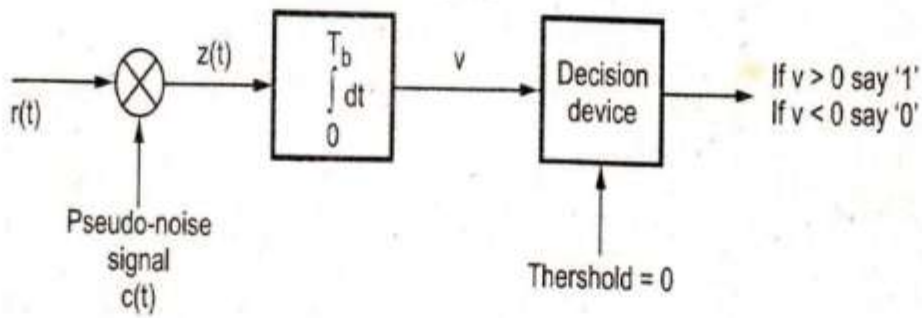
The receiver signal X(t) and the locally generated replica of the PN sequence generator are applied to a multiplier this is 1ST stage of multiplication. The multiplier performs the de-spreading operation output of multiplier is then applied to a coherent BPSK detector with local carrier applied to it.

At the output of coherent BPSK detector we get back the original signal d(t) data signal.



The DS-BPSK receiver

(OR)



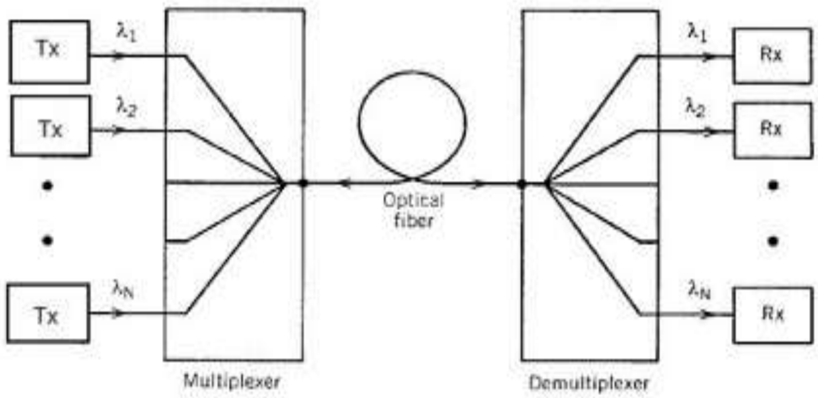
5.Attempt any TWO of the following : **16**

(a) Describe Wavelength Division Multiplexing with neat diagram. Compare FDM & TDM multiplexing (any 4 pts.).

Ans : WDM bock diagram- 2 mks, explanation- 2 mks, comparison on 4 points – 4 mks

Explanation-WDM is designed to use the high data rate capability of fibre optic cable. The optical fibre data rate is higher than the data rate of metallic transmission line. Using optical fibres and mulyiplexing technology , several lines can be connected. WDM involves the optical signals transmission through fibre optical cables where narrow bands of light of different wavelengths are combined to make a wide band of light. They are separated at the receiver by demultiplexing.

Block diagram-

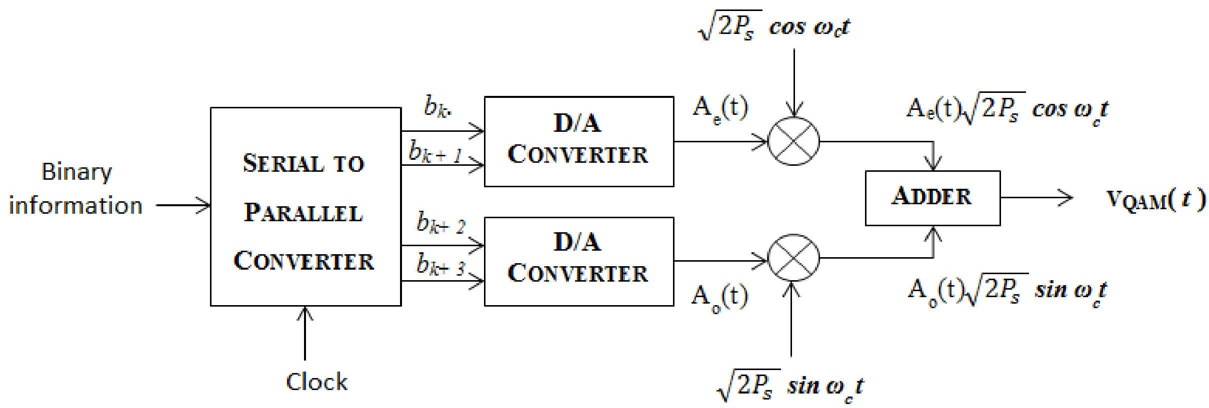


Compare TDM and FDM

Sr. No	Parameters	TDM	FDM
1	**Definition**	TDM divides and allocates certain time periods to each channel.	FDM divides the channel into the two or more frequency ranges that do not overlapped
2	**Synchronization**	Synchronization is required.	Synchronization is not required.
3	**Cross talk**	In TDM the problem of crosstalk is not present.	FDM suffers from the problem of crosstalk due to imperfect band pass filter
4	**fading**	Due to fading only a few TDM channel will be affected	Due to wide band fading in the transmission medium, all the FDM channel will be affected

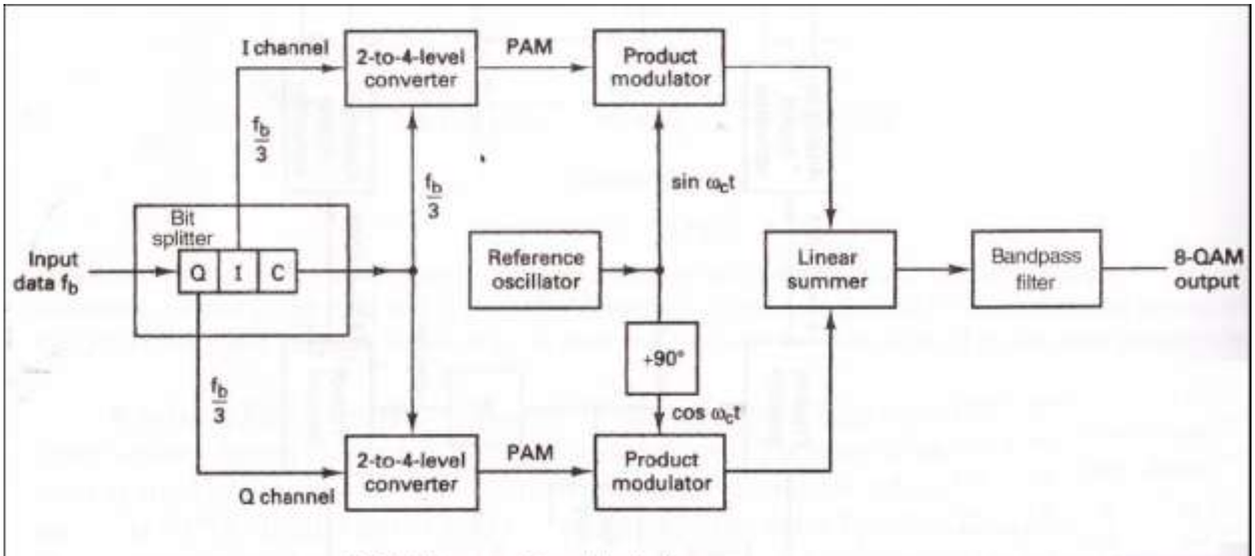
(b) With the help of block diagram explain the working of Transmitter and Receiver of QAM System.

Ans:- Diagram: 02M each



QAM transmitter

OR



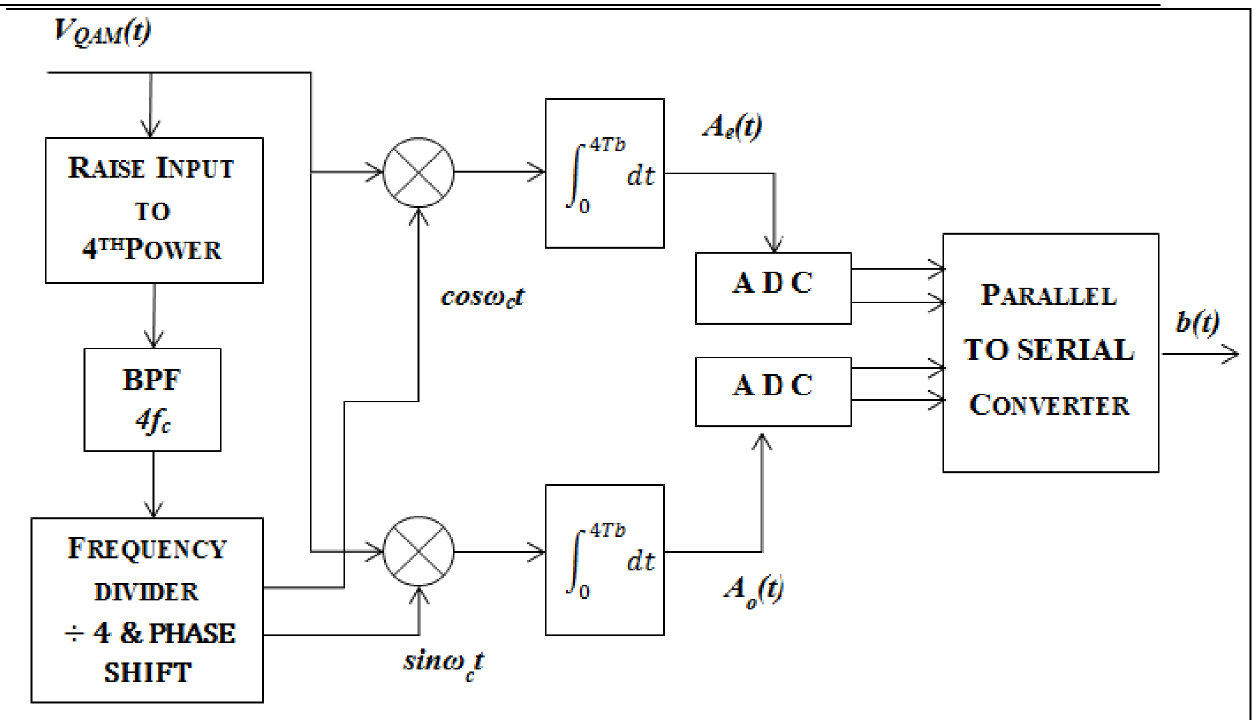
8-QAM transmitter: block diagram

QAM transmitter explanation 02M

- The bit stream $b(t)$ is applied to the serial to parallel converter, operating on a clock which has a period of T_s, which is the symbol duration. The bits $b(t)$ are stored by the converter and then presented in the parallel form. The four bit symbols are b_{k+3}, b_{k+2}, b_{k+1}, b_k.
- Out of these four bits, the first two bits are applied to a D/A converter and the other two bits are applied to the second D/A converter.
- The output of the first converter is $A_e(t)$, which is modulated by the carrier $\sqrt{2P_s}\ cos\omega_c t$ whereas the output of the second D/A converter, $A_o(t)$ is modulated by the carrier $\sqrt{2P_s}\ sin\omega_c t$ in the balanced modulators.
- $A_e(t)$, $A_o(t)$ are voltage levels generated by the convertor -3,-1,+1,+3 volts.
- The balanced modulator outputs are added together to get the QASK output signal which is expressed as,

$$v_{QASK}(t) = A_e(t)\sqrt{2P_s}\ cos\omega_c t + A_o(t)\sqrt{2P_s}\ sin\omega_c t$$

Receiver Diagram: 02M



QAM receiver

QAM receiver explanation **02M**

- The quadrature carriers are recovered from the received QAM signal. The input QASK signal is first raised to the 4th power and then by using a BPF, with a center frequency $4f_c$, along with a frequency divider (÷4), the required quadrature carriers are recovered.
- Then, two balanced modulators are used together with two integrators to recover the signal $A_e(t)$ and $A_0(t)$. Both the integrators integrate over one symbol interval T_s or $4T_b$. The symbol time synchronizer is used along with each integrator.
- Integrator output = $A_0(t)\sqrt{2P_s}\,2T_b$ and $A_e(t)\sqrt{2P_s}\,2T_b$
- Finally, the original bits are obtained from $A_e(t)$ and $A_0(t)$ by using two A/D converters. The outputs of the two A/D converters are then applied to the serial to parallel converter to obtain the sequence $b(t)$.

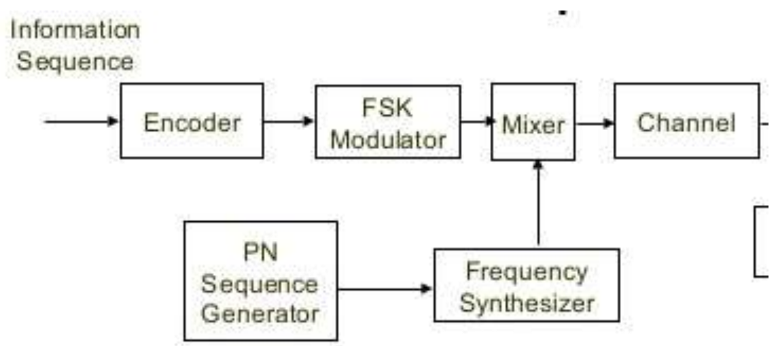
(c) Define slow and fast frequency hopping & describe BFSK/FHSS transmitter with block diagram.

Ans:- Each definition 2 mks, block diagram – 2 mks, explanation- 2 mks

Slow frequency hopping-In slow frequency hopping the symbol rate Rs of the MFSK signal is an integer multiple of the hop rate Rn that means several symbols are transmitted corresponding to each frequency hop.
Each frequency hop: → several symbols
Here frequency hopping takes place slowly and thus
Hop rate Rh > Symbol rate Rs

Fast frequency hopping-In fast frequency hopping, multiple frequencies or hops are used to transmit one symbol. That is each symbol-> several hops. So several frequencies changes for one symbol such that Symbol rate Rs > Hop rate Rh.

BFSK/FHSS Transmitter **2 marks**



Explanation (2 marks)

The binary data sequence is applied to the M-ary FSK modulator. The output of M-ary FSK is mixed with the frequency synthesizer output. The frequency synthesizer decides the hopping patters of the system.
The output of mixer is the stream of two frequencies. Sum and the difference of both the inputs to it.
The frequency of the mixer input obtained from MFSK modulator is changing continuously.
Other input to the mixer is obtained from the digital frequency synthesizer.
The synthesizer output at a given instant of time is the frequency hop.
Frequency hop at the output of synthesizer are controlled by the successive bit at the output of PN code generator.
The band pass filter is centered at the sum frequency band and rejects all other components. This sum components of the frequency are then retransmitted as FHSS signal.
In slow frequency hopping the symbol rate Rs of the MFSK signal is an integer multiple of the hop rate Rn that means several symbols are transmitted corresponding to each frequency hop.
Each frequency hop: → several symbols

6.Attempt any FOUR of the following : **16**

(a) **State any two advantages and disadvantages of PCM system**.

Ans: **Advantages: any 2** **(2 Marks)**

1. PCM has very high noise immunity.

2. Repeaters can be used between the transmitter and the receiver which can further reduce the effect of noise.

3. It is possible to store the PCM signal due to its digital nature.

4. It is possible to use various coding techniques so that only the desired receiver (user) can decode the message.

Disadvantages: any 2 **(2 Marks)**

1. The encoding decoding & quantizing circuitry of PCM is complex.

2. PCM requires a large BW as compared to other systems.

(b) State the principle of orthogonality and describe OFDM technique.

Ans: Orthogonality principle:- **01 M**

OFDM stands for Orthogonal Frequency Division Modulation. It is based on the principle of orthogonality . Two signals are said to be orthogonal if they are independent of each other in specified time interval & do not interact with each other. It is possible to transmit multiple signals over a common channel without interference & get detected on the receiving end without interference.

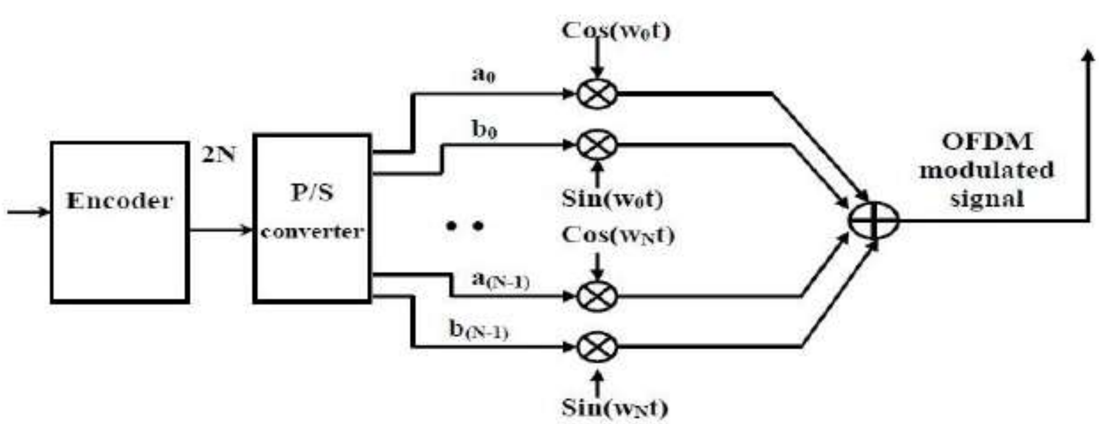
<u>OFDM technique</u>: **01M**

OFDM is a multicarrier system . In FDM we have different channels occupying different frequency band with a guard band in between to avoid interference between adjacent channels but this makes FDM a BW in-efficient system. The BW efficiency improves considerably if we use OFDM technique instead of simple FDM.

The subcarriers are placed at the null points of all other subcarriers this automatically eliminates interference among the adjacent subcarriers. Due to this total BW of OFDM system is much less than that of the conventional FDM system.

It has a high spectral efficiency. That means it can accommodate more number of users. It is multiplexing/multiple access scheme which has features suitable for the fourth generation of wireless communication systems .It is mainly based on the DSP techniques.

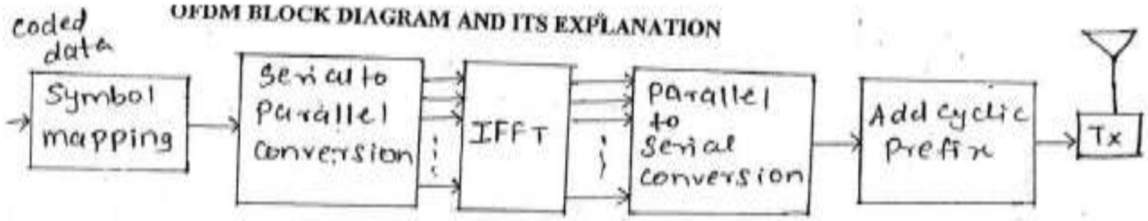
Block diagram **2 mks**



OR

OR

OFDM BLOCK DIAGRAM AND ITS EXPLANATION



OFDM - TRANSMITTER.

c) Compare between BPSK and QPSK w.r.t. variable character of carrier, type of modulation, bit rate/baud rate and application.

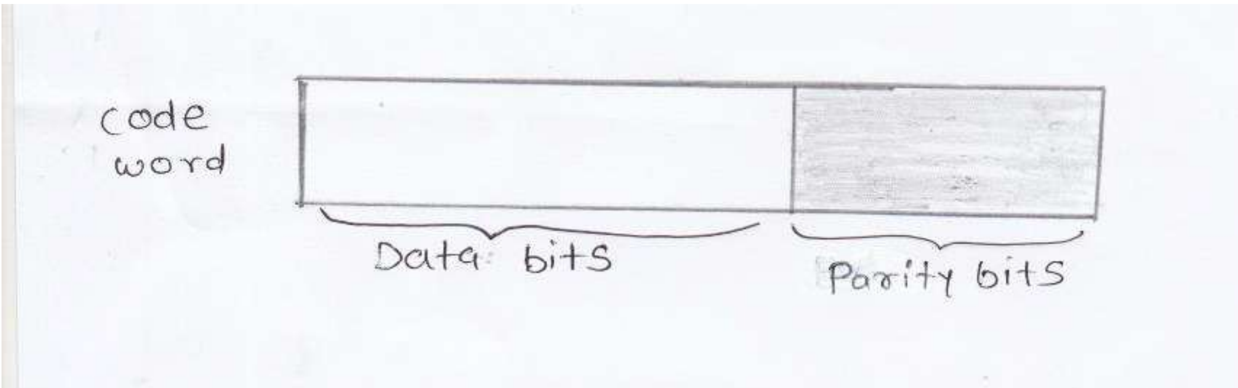
Ans: **01M each**

Sr..No	Parameter	BPSK	QPSK
1	Variable characteristics of the carrier	Variable characteristics of the carrier is phase.	Variable characteristics of the carrier is phase
2	Type of modulation	Binary modulation	M-ary encoding
3	Bit rate	Bit rate = baud rate	Bit rate = 2 baud rate
4	Application.	Satellite communication, DSSS	Satellite communication, DSSS

(d) Define the following : (i) Code word (ii) Code rate (iii) Hamming weight & (iv) Hamming distance.

Ans :**Code word: -** 01M

The code word is code consisting of data unit and party bits/redundant bits.



Code rate:- 01M

The code rate is defined as the ratio of the number of message bits (k) to the total number of bits (n) in a code word.

$$\text{Code rate (r)} = \frac{k}{n}$$

Hamming weight:- **01M**

The Hamming weight of a code word x is defined as the number of non-zero element in the code word. Hamming weight of a code vector (code word) is the distance between that code word and an all zero code vector (a code having all element equal to zero)
Eg : Data x =10010101 .
Hamming weight=4 (total non zero elements)

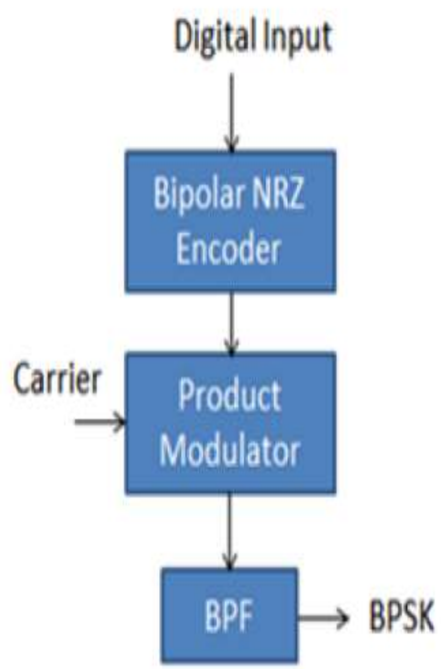
Hamming distance related to code:- 01M

The “Hamming distance” is the distance between two code word:

Code word No.1	:	1	1	0	1	0	1	0	0
		↕				↕		↕	
Code word No.2	:	0	1	0	1	1	1	1	0

The bits 2,4 and 8 are different from each other . Hence Hamming distance is three.

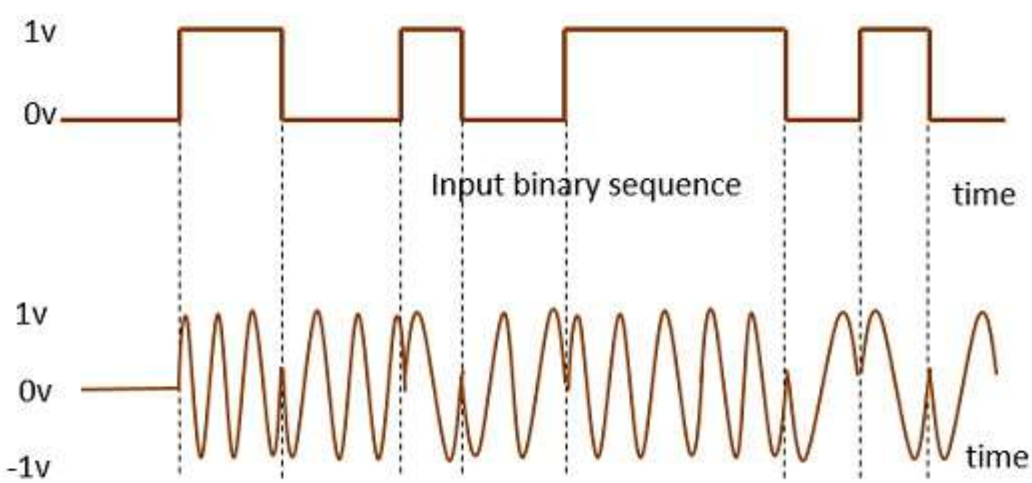
e) Describe BPSK generator with block diagram and waveforms.
Ans:- Block diagram- 2 mks, explanation- 1mks, waveforms- 1 mks



Explanation:-

- The digital information from the source may be a unipolar NRZ data which varies between values 1 and 0. But if digital data is unipolar then during 0 data output will be zero due to use of multipliers at transmitter. So to avoid this problem the unipolar data is first converted into bipolar data whose value varies between +1 and -1.
- The transmitter of BPSK signal is shown in the above figure.
- The digital signal from the information source is a unipolar NRZ signal which is first converted into bipolar signal. This signal acts as the modulating signal. The BPSK modulator is nothing but a multiplier followed by a band pass filter as shown in above figure.
- Due to multiplication, the BPSK output will be present with 0^0 phase shift when a binary '1' is to be transmitted and when the digital input is '0' then we get BPSK output with 180^0 phase shift as shown in the waveform above.
- From the waveform analysis we can conclude that when a binary '1' is to be sent the carrier is transmitted with 0^0 phase shift and when binary '0' is to be sent then the carrier is transmitted with 180^0 phase shift.
- The transmitted BPSK signal is s (t) = b (t) $\sqrt{2Ps}\cos\omega_0 t$.
- The phase of this signal changes depending on the time delay from transmitter to receiver. This phase change is generally fixed in the transmitted signal. Let the phase shift be Θ.
- Therefore the signal at the input of receiver is s (t) = b (t) $\sqrt{2Ps}\cos\omega_0 t + \Theta$.

Waveforms:-



BPSK Modulated output wave