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 judgment 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.

Q1. Attempt any THREE of the following: **12M**

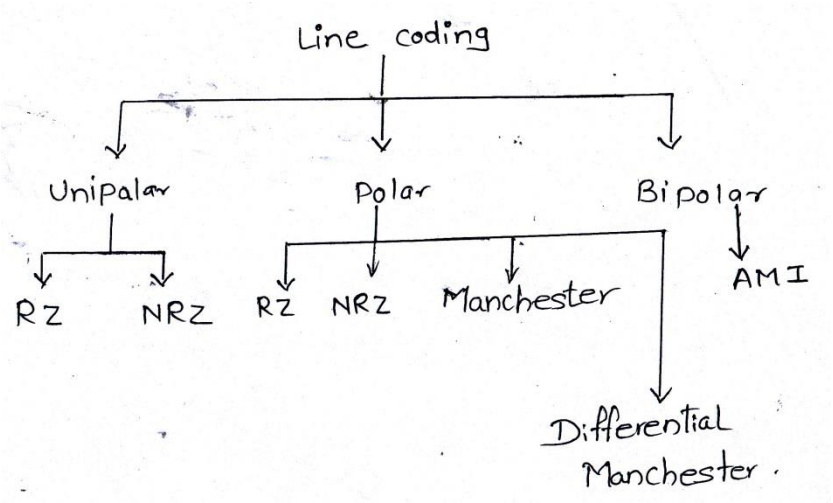
a) Define line coding. Give the classification of line coding.

Ans:

Definition- **01M**

Line coding-It is a process of converting binary data (a sequence of bits) to a suitable format for transmission through the channel with minimum probability of errors.

Classification:- **03M**



b) State and describe the sampling theorem with neat waveform.

Ans:

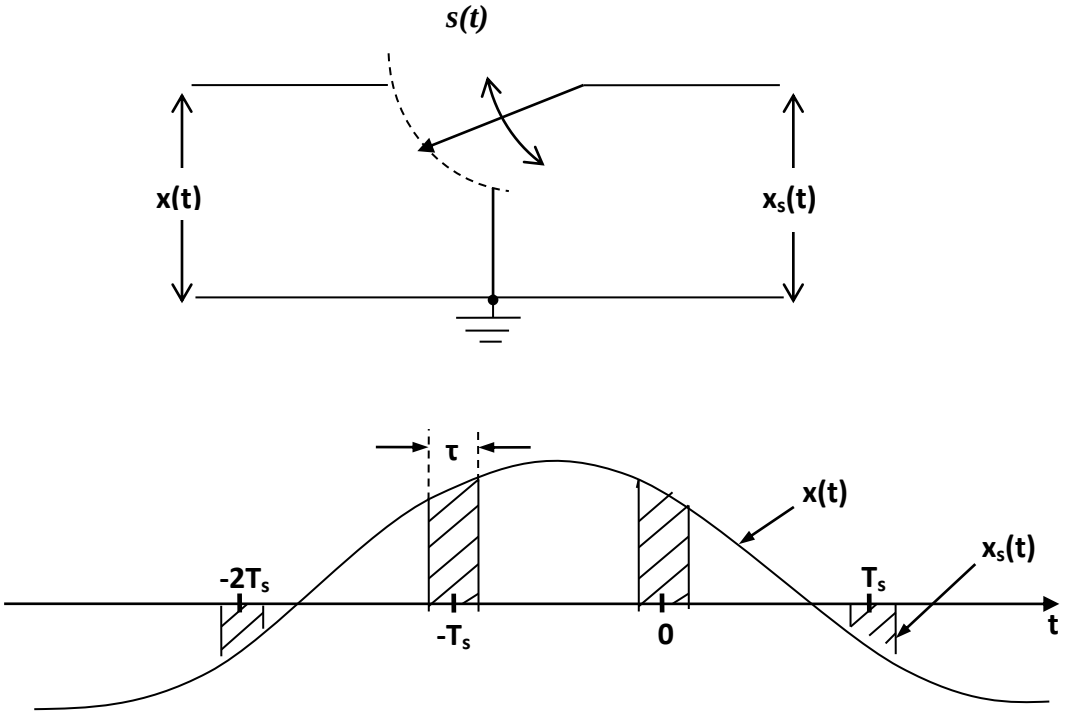
Statement: **02M**

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

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**



Switching Sampler and its output waveform

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

$$\mathbf{x_s(t) = x(t)s(t)}$$

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.

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 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:-

Application of S. S. modulation:- **01M each**

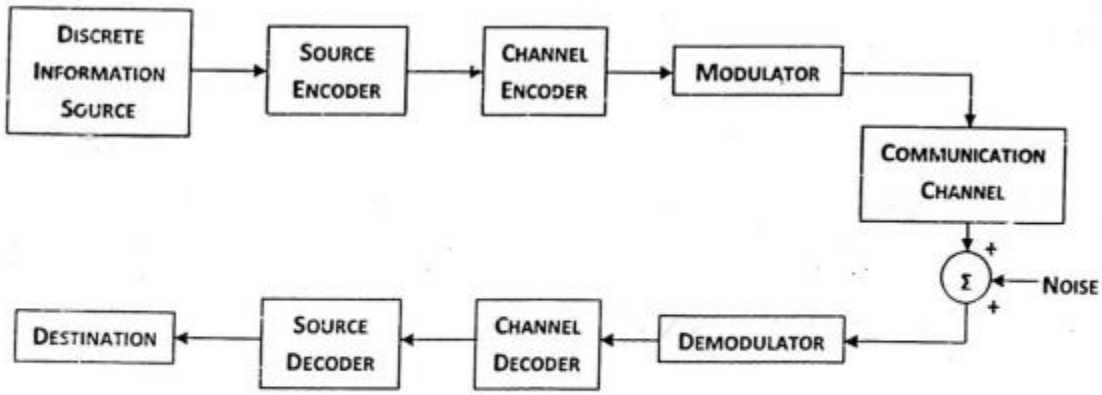
1. Military application- resistance to gainming.
2. Secure communication.
3. CDMA in satellite communication.
4. Police radar can employ spread spectrum to avoid detection by detectors employed by drives.
5. Low density power spectra for signal hiding.
6. Multipath rejection in a ground based mobile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

B) Attempt any ONE of the following: **06M**

a) Draw the block diagram of the basics digital communication systems. State the function of each block in detail.

Ans:-

Diagram:- **02M**



Block diagram of a digital communication system

Explanation: **04M**

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 s_y*mbols/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) Describe working of CRC generator and checker with an example.

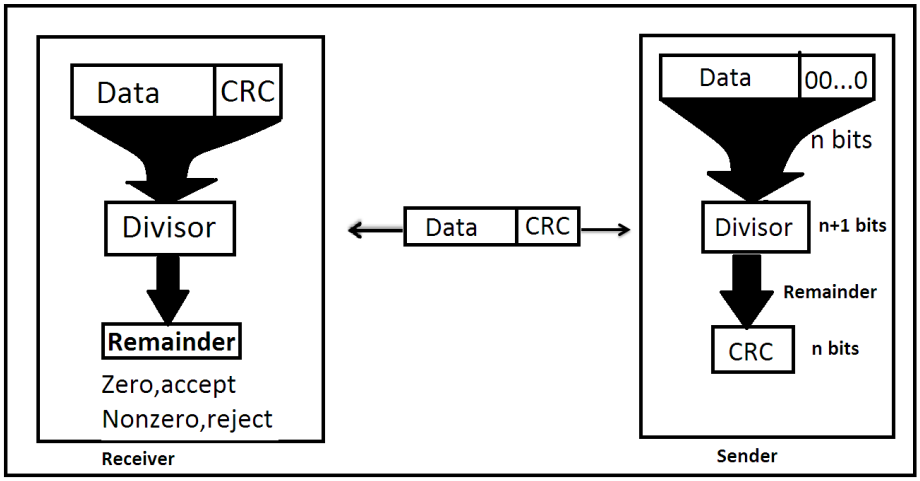
Ans:-

Cyclic Redundancy Check (CRC): **02M**

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 of the unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number.

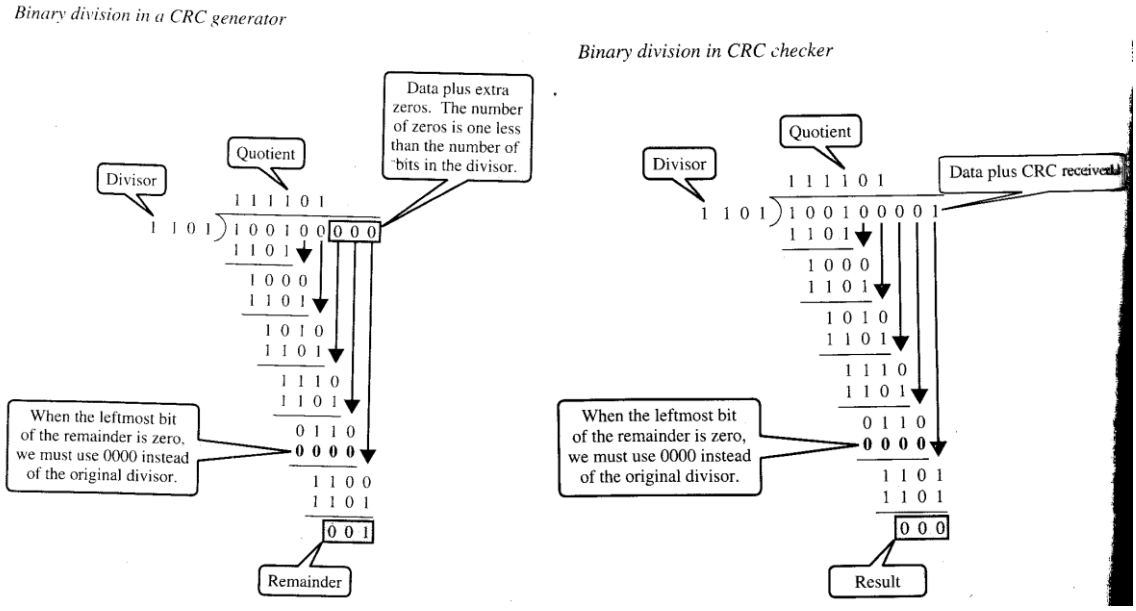
At its destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit assume to be correct and is accepted, otherwise it indicate that data unit has been damaged in transmission and therefore must be rejected

The redundancies bits are used by CRC are derived by dividing the data unit by a predetermined divisor. The remainder is the CRC.



Example-

Transmitter **Receiver** **02M each**

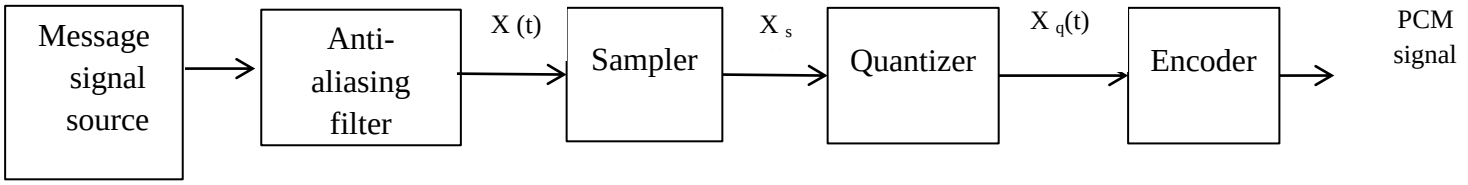


Q.2) Attempt any TWO of the following: **16M**

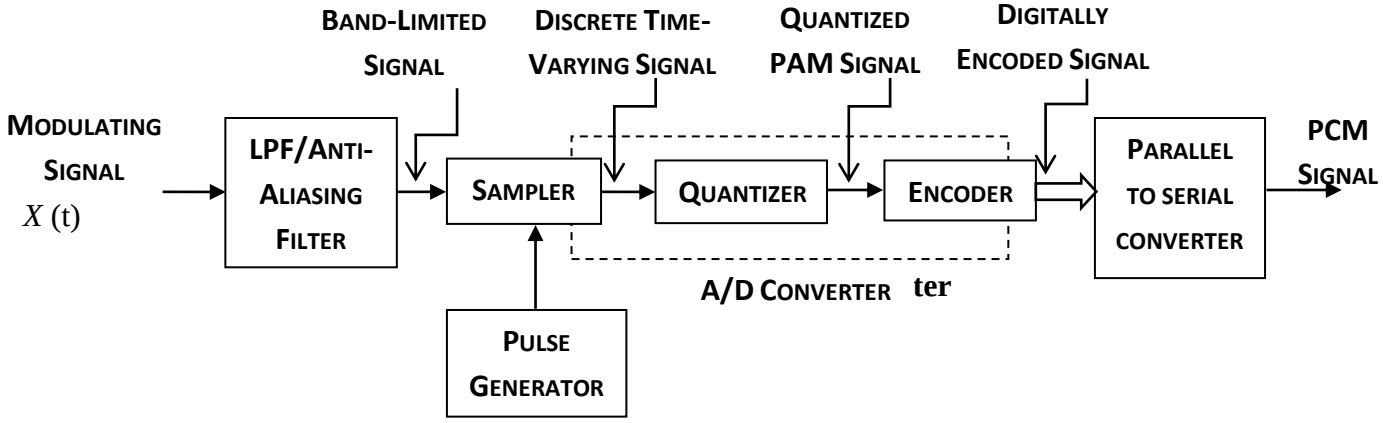
a) Draw the block schematic of PCM transmitter. State the function of each block.

Ans:

Diagram **04M**



OR



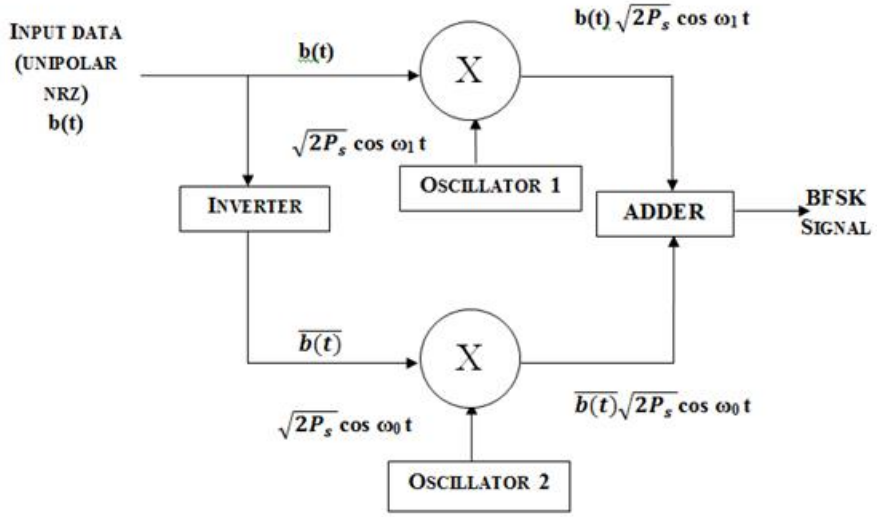
Function of block: **04M**

- The analog signal *x (t)* is passed through a LPF (anti-aliasing filter). The LPF band-limits the signal to f_m band-limiting is necessary to avoid the aliasing effect in the sampling process.
- The pulse generator generates a train of pulses at a frequency of f_s such that $f_s > 2\ f_m$. Thus, the *Nyquist criterion* is satisfied.
- The sampler block carries out flat-top sampling process on the modulating signal at adequately high frequency. Then these samples are subjected to the operation called Quantization in the Quantizer.
- 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.
- The quantized PAM pulses are applied to an encoder. The encoder converts each quantized level into an *N*-bit digital word (binary pattern) such that $\boldsymbol{Q = 2^N}$ where *Q* is the total number of quantization levels.
- The combination of the Quantizer and the Encoder is called as an Analog-to-Digital Converter (A/D Converter). Thus, the signal transmitted over the communication channel is a digitally-encoded signal.

b) Describe the generation of BFSK with block diagram. State the mathematical equation. Draw power density spectrum.

Ans:

Diagram: **02M**



BFSK generator

Description : **02M**

- 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 *mark frequency*.
- 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.

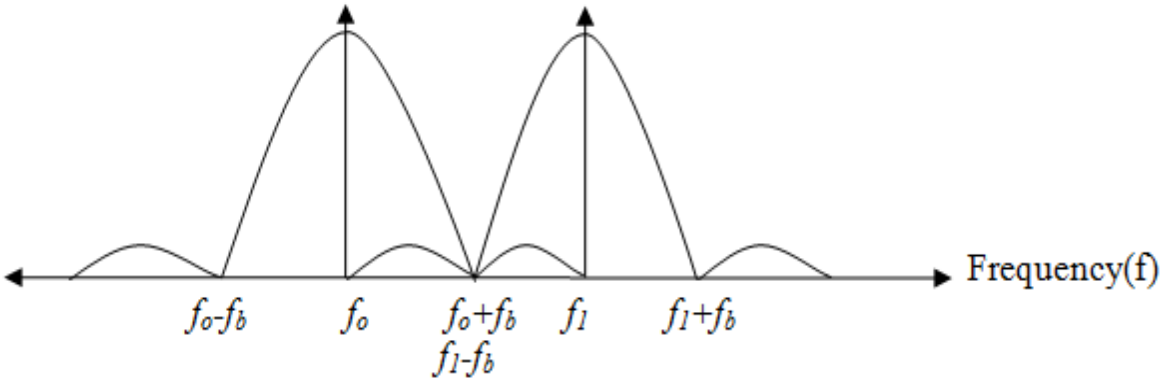
Mathematical equation: **02M**

- In general, for binary FSK, the carriers can be represented by:

For binary 0, $V_0(t) = \sqrt{2P_s} \cos 2\pi f_0 t = \sqrt{2P_s} \cos \omega_0 t$

For binary 1, $V_1(t) = \sqrt{2P_s} \cos 2\pi f_1 t = \sqrt{2P_s} \cos \omega_1 t$

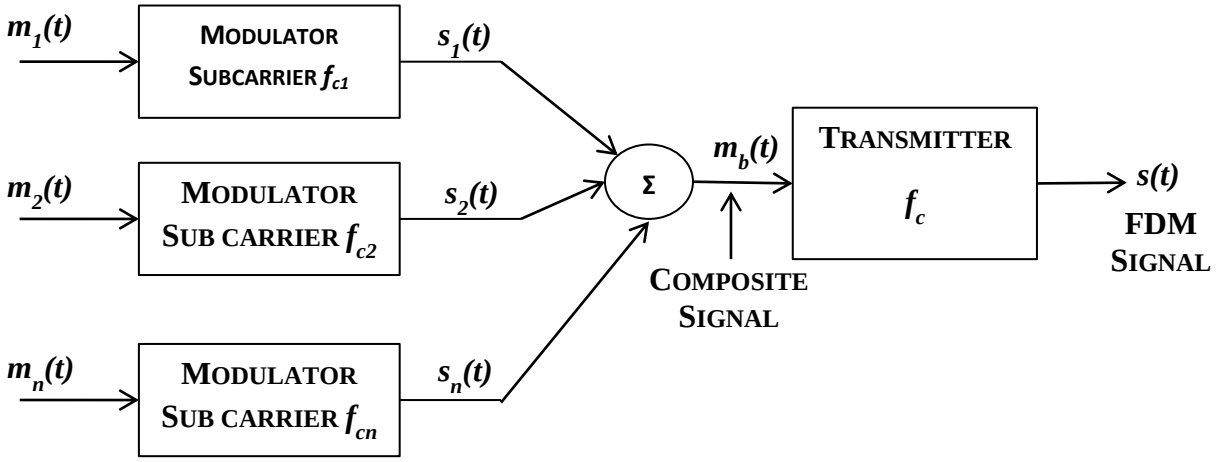
Power density spectrum: **02M**



c) Describe FDM technique with block diagram. Compare it with TDM with respect to definition, synchronization, cross talk and fading.

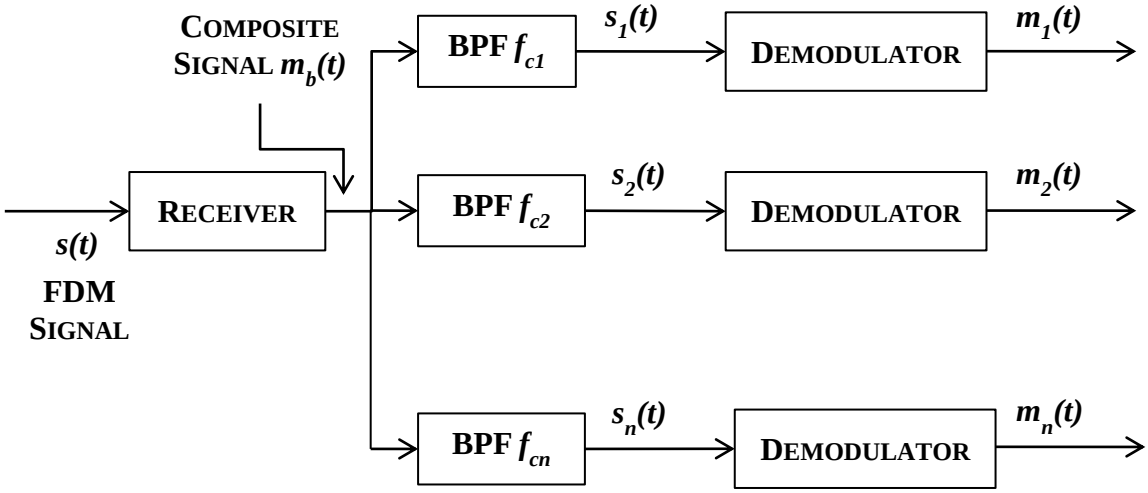
Ans:-

Diagram: **02M**



FDM transmitter

Diagram: **02M**



FDM receiver

Description:

FDM transmitter **01M**

- In FDM, the transmission channel is shared by multiple signals, each being allotted a portion of the spectrum of the bandwidth. A generalized block diagram of FDM transmitter is depicted in Figure.

- A number of analog signals (or digital signals converted into analog) $m_i(t)$, $i = 1, 2, \ldots, n$ are multiplexed onto the same transmission medium. Each signal $m_i(t)$ is modulated onto a carrier f_{ci}.
- As multiple carriers are to be used, each is referred to as a *subcarrier*. Any type of analog modulation may be used. The resulting analog signals are summed together to produce a composite signal, $m_b(t)$.
- The composite signal may be shifted, as a whole, to another carrier frequency by an additional modulation step.
- This second modulation step need not use the same modulation technique as the first. Thus, the FDM signal generated may be transmitted over a suitable medium.

FDM receiver **01M**

- The FDM receiver is shown in Figure. The FDM signal is received by the receiver and demodulated to retrieve the composite signal $m_b(t)$ which is further amplified.
- This composite signal is passed through n band pass filters, each filter having a center frequency equal to the subcarrier frequency f_{ci}.
- In this way the composite signal is split into its component signals. Each component signal is further demodulated to obtain the analog outputs that were originally transmitted. If required, these signals are stored and displayed.

Compare TDM and FDM **½ M each**

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

Q3. Attempt any FOUR of the following: **16M**

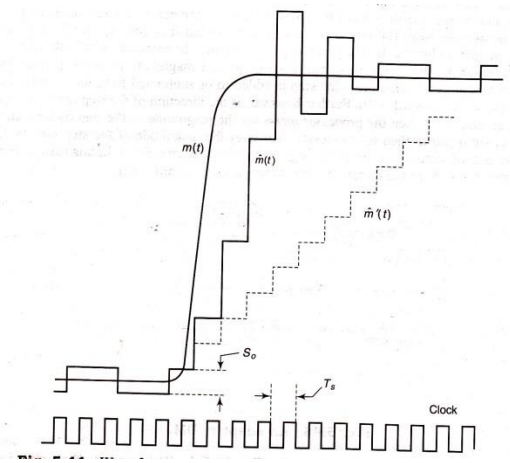
a) What is slope overload? Draw the schematic of adaptive delta modulation technique.

Ans:-

Slope overload error:- **02M**

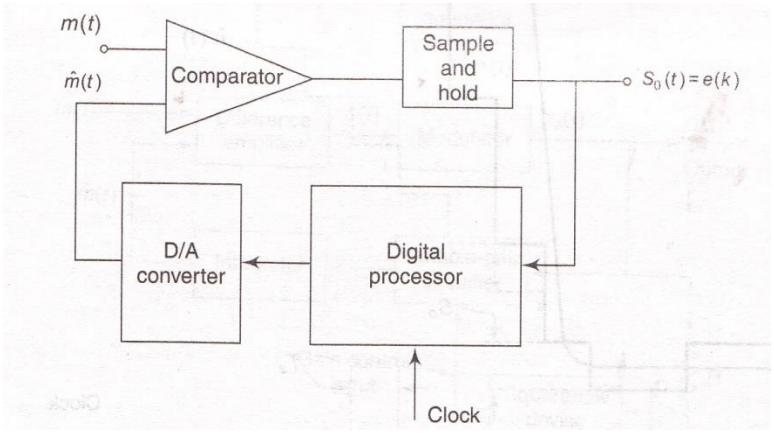
If slope of analog signal m(t) is much higher than the approximated signal m'(t) over a long duration then m(t) will not be able to follow m'(t) at all. The difference between m(t) and m'(t) is called slope overload distortion. Thus the slope overload error occurs when slope of the m(t) is much larger than slope of m'(t).

Diagram:- **02M**

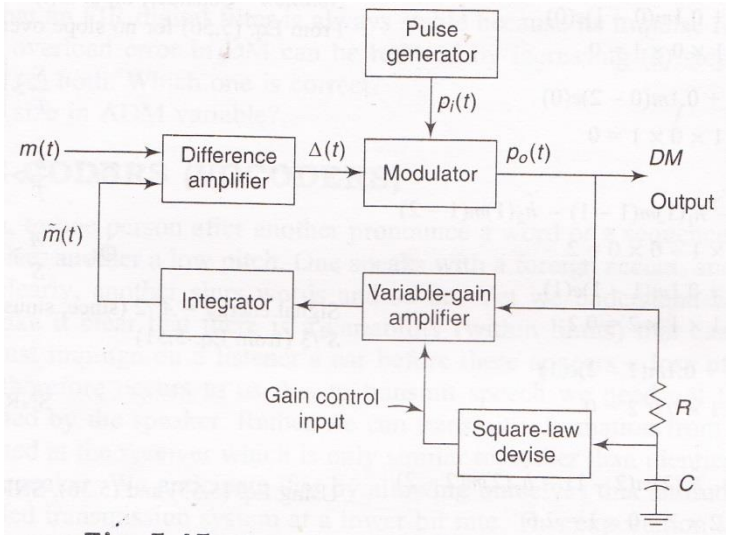


Waveforms of ADM

OR



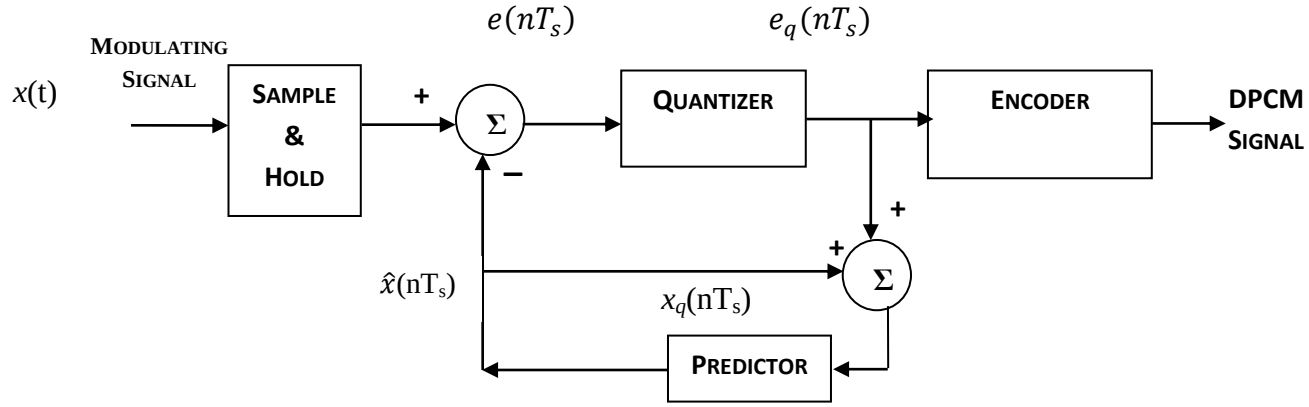
OR



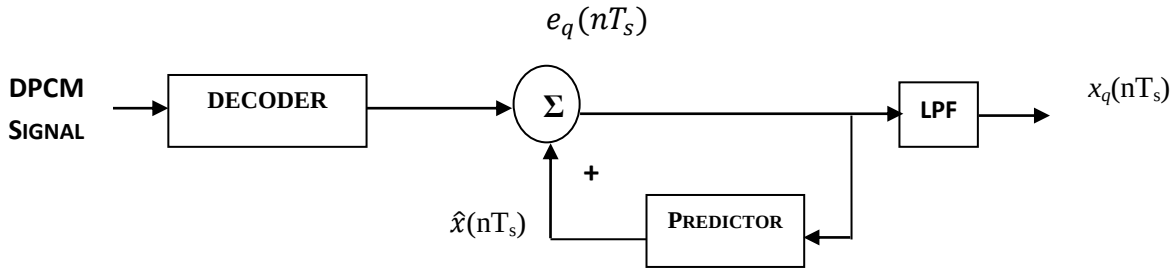
b) Draw block schematic of DPCM transmitter and receiver.

Ans:

DPCM Transmitter:- **02M**



DPCM Receiver **02M**



c) Give the advantages of TDMA over FDMA (any four).

Ans:-

Advantages:- (any four) **01M each**

- In TDMA since only one station is present at any given time the generation of intermodulation products will not take place.
- 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.
- The frequency selective fading does not affect the TDMA to extent it affect of FDMA.
- As only one channel is being transmitted at a time it is not necessary to separate out various channels at the receiver.
- TDMA by default can work well with the digital therefor it can be easily used for data transmission.

d) What is M-ary encoding? Compare the bandwidth requirement for BPSK, QPSK, QAM, and M-ary PSK

Ans:-

M-ary encoding **02M**

M-ary modulation is a technique of modulation in which *N* bits are combined together to form *M* symbols

(*2N = M*) and a signal is transmitted corresponding to each symbol for a duration of *NTb = Ts*. the signal is generated by changing the amplitude, phase or frequency of a sinusoidal carrier in discrete steps.

Bandwidth requireuirement for : **1/2 M each**

BPSK=$2f_b$

QPSK= f_b

QAM= $\frac{2f_b}{N}$

M-ary PSK= $= \frac{2f_b}{N}$

e) What are the different types of QAM ? Draw constellation diagram of 4 QAM.

Ans:-

Types of QAM **02M**

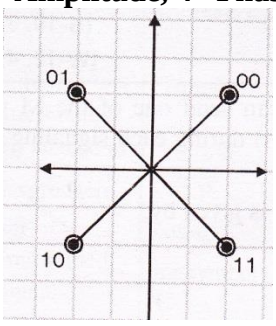
This is depending on the number of bits per message the QAM signals are classified as follows:

4 QAM
8 QAM
16 QAM
32 QAM
64 QAM

Constellation diagram of 4 QAM **02M**

The constellation diagram of 4 QAM is shown below. All the symbols have same amplitude and different phases.

1 Amplitude, 4 Phase



Q4. (A) Attempt any THREE of the following: **12M**

a) **Compare between analog and digital communication with respect to nature of signal, noise immunity, coding, bandwidth.**

Ans:

01M each

Parameters	Analog Communication	Digital communication
Nature of signal	The information signal is continuous / analog in nature	The information is in digital form.
Noise immunity	Poor as coding is not possible	Very good due to coding
Coding	Not possible	Possible
Bandwidth	Requires less bandwidth	More bandwidth

b) Describe the process of quantization with neat sketch.

Ans:

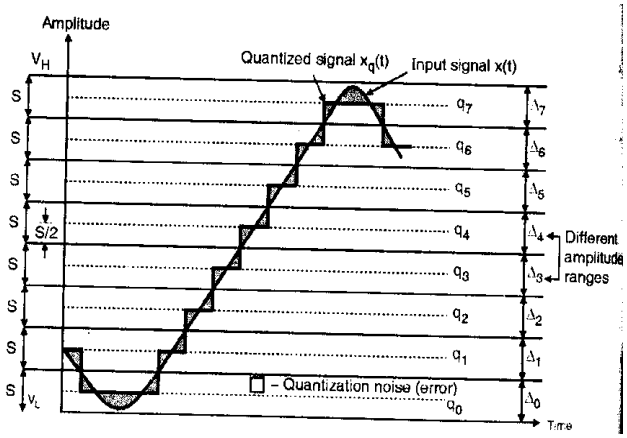
Explanation- **02M**

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 V_L to V_H 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

$$S = V_H - V_L / Q$$

Diagram of the Process quantization is as shown below- **02M**



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

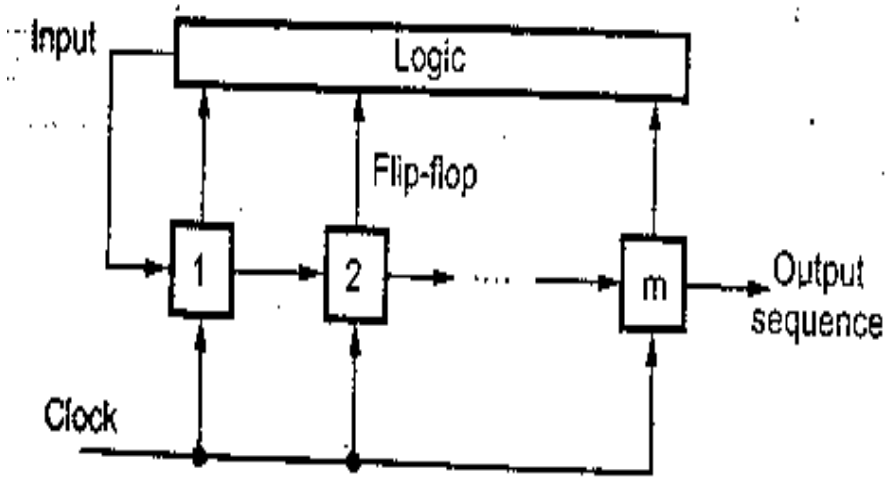
Ans:

Definition:- **02M**

- A PN sequence is defined as a pseudorandom coded sequence of 1s and 0s with certain auto correlation properties.

The pseudo random sequence generator:- **02M**

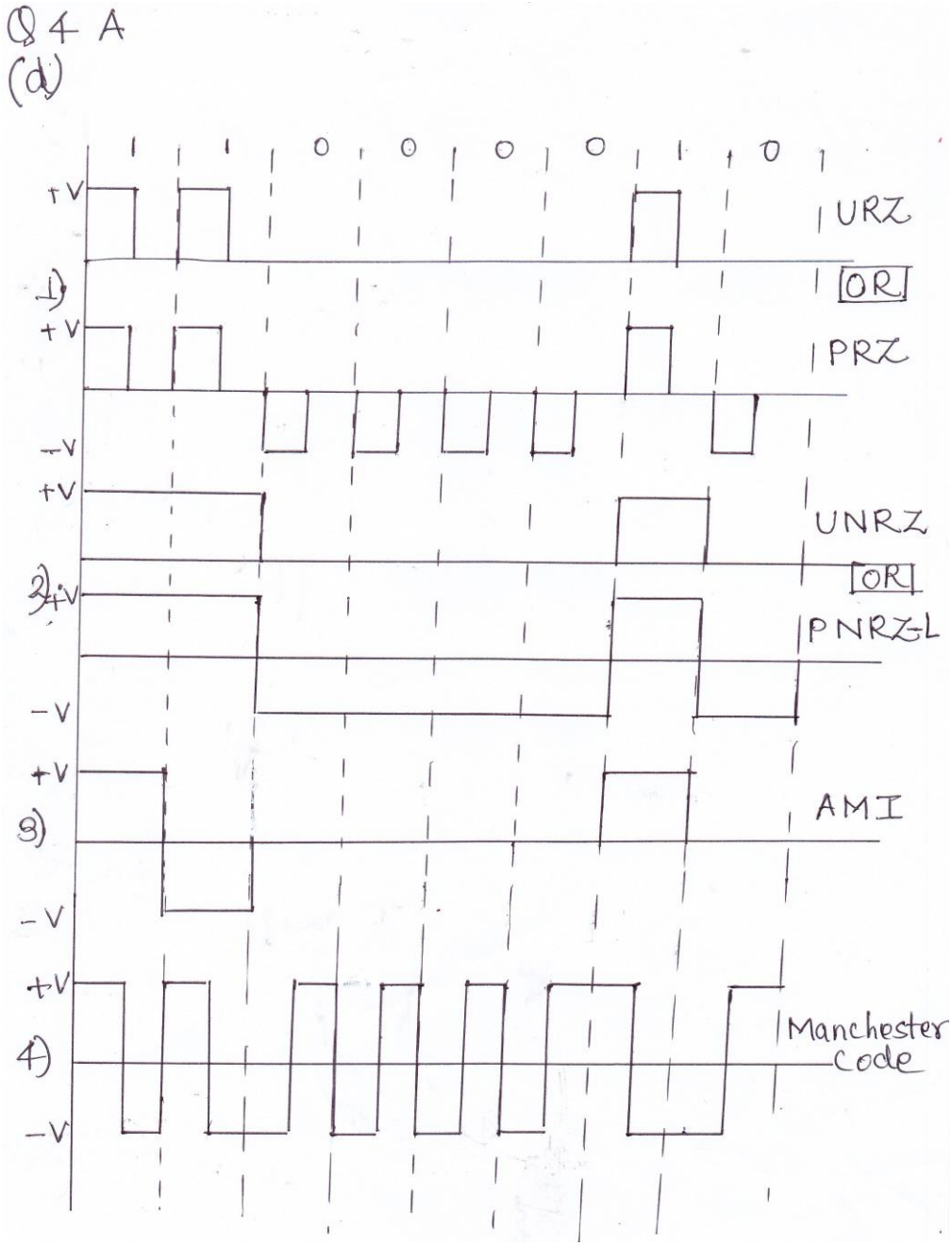
. It can be generated by a shift register and the combinational logic circuit. The generalized block diagram of PN sequence generator scheme is as shown below.



d) For the binary data stream 11000010 draw the Return to zero, non- Return zero, AMI and Manchester codes.

Ans:

Coded waveform- **01M each**



(B) Attempt any ONE of the following: **06M**

(a) State the different types of errors in digital communication. Describe each with example.

Ans:

Types of error: **01M each**

1. Single bit error:

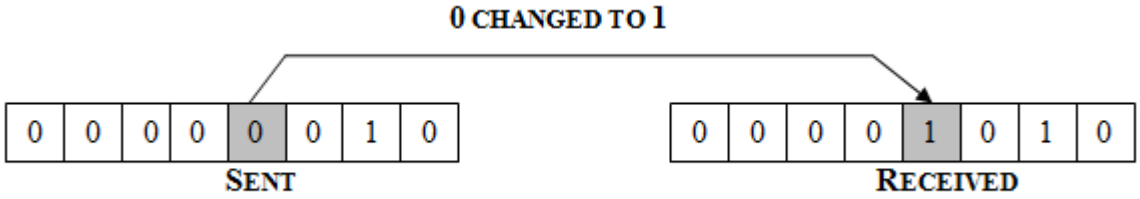
Single-bit error occurs when only one bit of a given data string is in error (changed from 0 to 1 or from 1 to 0).

2. Burst error:

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

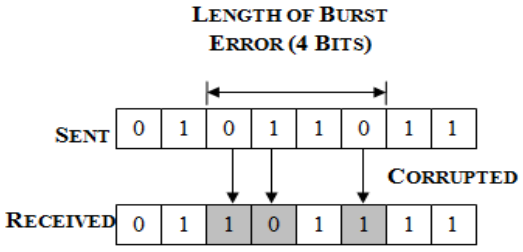
Example **02M each**

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) Compare FHSS and DSSS system. (any four points)

Ans:

Comparison **01M each**

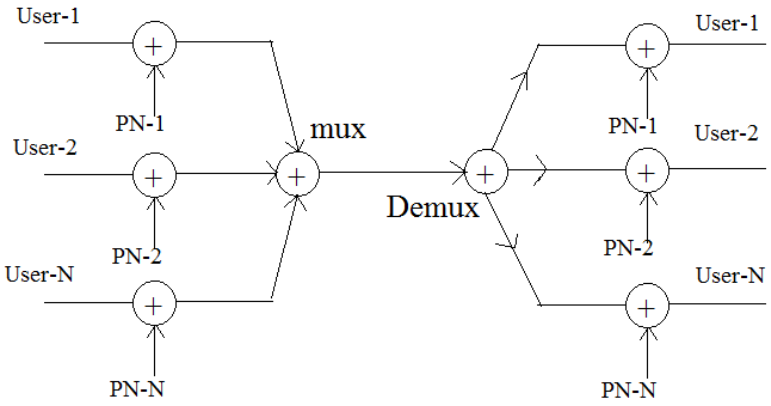
DSSS	FHSS
• Definition: PN sequence of large bandwidth is multiplied with a narrow band information signal.	• Definition: Data bits are transmitted in different frequency slots which are changed by PN sequence.
• Chip rate (R_c) = $\frac{1}{Tc}$	• Chip rate (R_c) = $max(R_h, R_s)$
• Applications with 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.	• 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.
• For commercial applications implementation of DSSS radios with large gap can also be costly due to the need of high speed circuits.	• Implementation of FHSS radio can be costly and complex due to the need of high speed frequency synthesizers.
• DSSS radios encounter more randomly distributed errors that are continuous and lower level.	• SFH suffers from strong burst error.
• Modulation technique: BPSK.	• Modulation technique: M-ary FSK
• Long acquisition time.	• Short acquisition time.
• DSSS is distance dependent.	• In FHSS, effect of distance is less.
• Processing gain is less.	• Processing gain is higher.
• Bandwidth required is less than FHSS system.	• Bandwidth of FHSS system is too high.

Q5.Attempt any TWO of the following: **16M**

a) Describe the basic principle involved in CDMA technology with neat sketch. State its any four advantages.

Ans:-

Diagram: **02M**



CDMA

Description: **02M**

- 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.

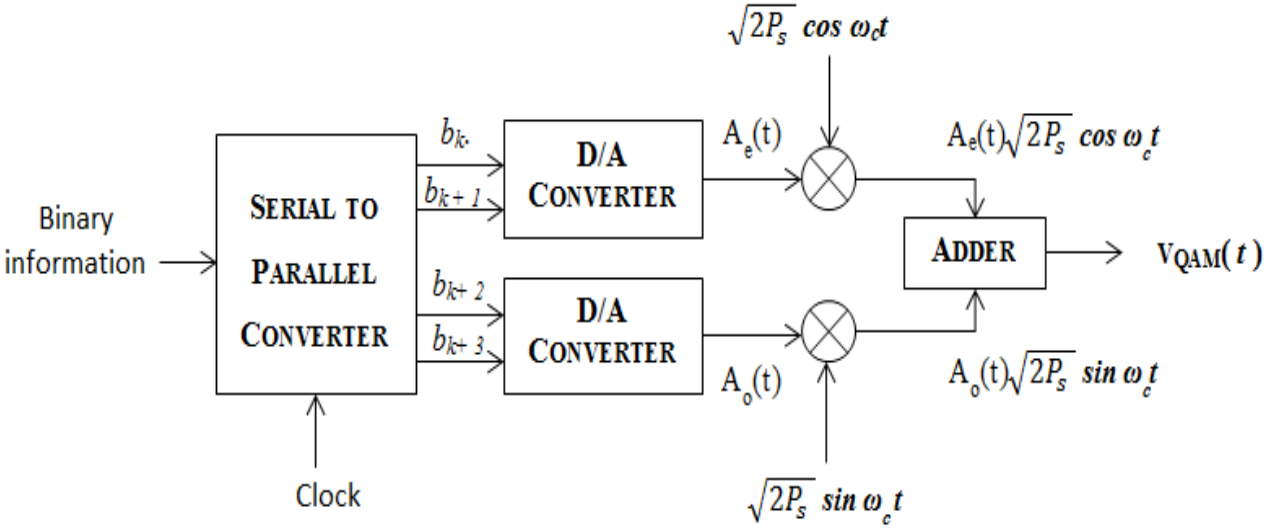
- **CDMA advantages (Any 4)** **01M each**
  1. It combats the intentional interference (jamming) and the unintentional interference from some other user
  2. To avoid the self-interference due to multipath propagation
  3. Hides a signal by transmitting it at a low power and thus making it difficult for an unintended listener to detect in the presence of background noise.

4. Achieves message privacy in the presence of other listeners
5. High bandwidth available
6. Non-interference with existing system
7. Anti-jam and interference rejection
8. Information security
9. Accurate ranging
10. Multipath tolerance

b) Draw and describe QAM transmitter and receiver.

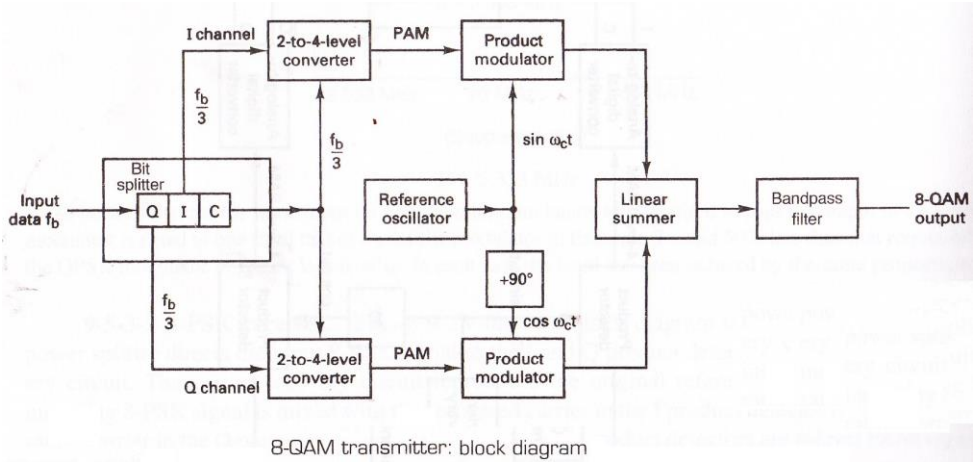
Ans:-

Diagram: **02M**



QAM transmitter

OR



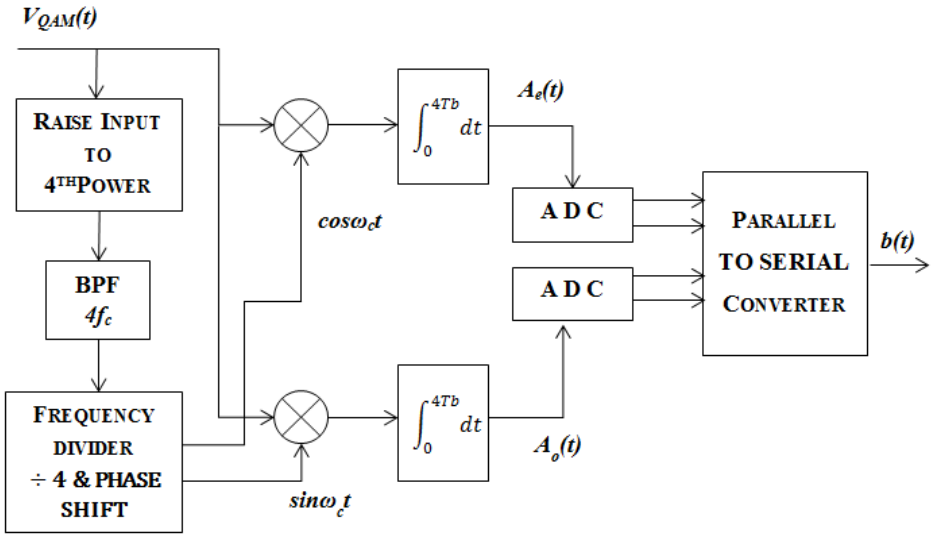
8-QAM transmitter: block diagram

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,

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

Diagram: **02M**



QAM receiver

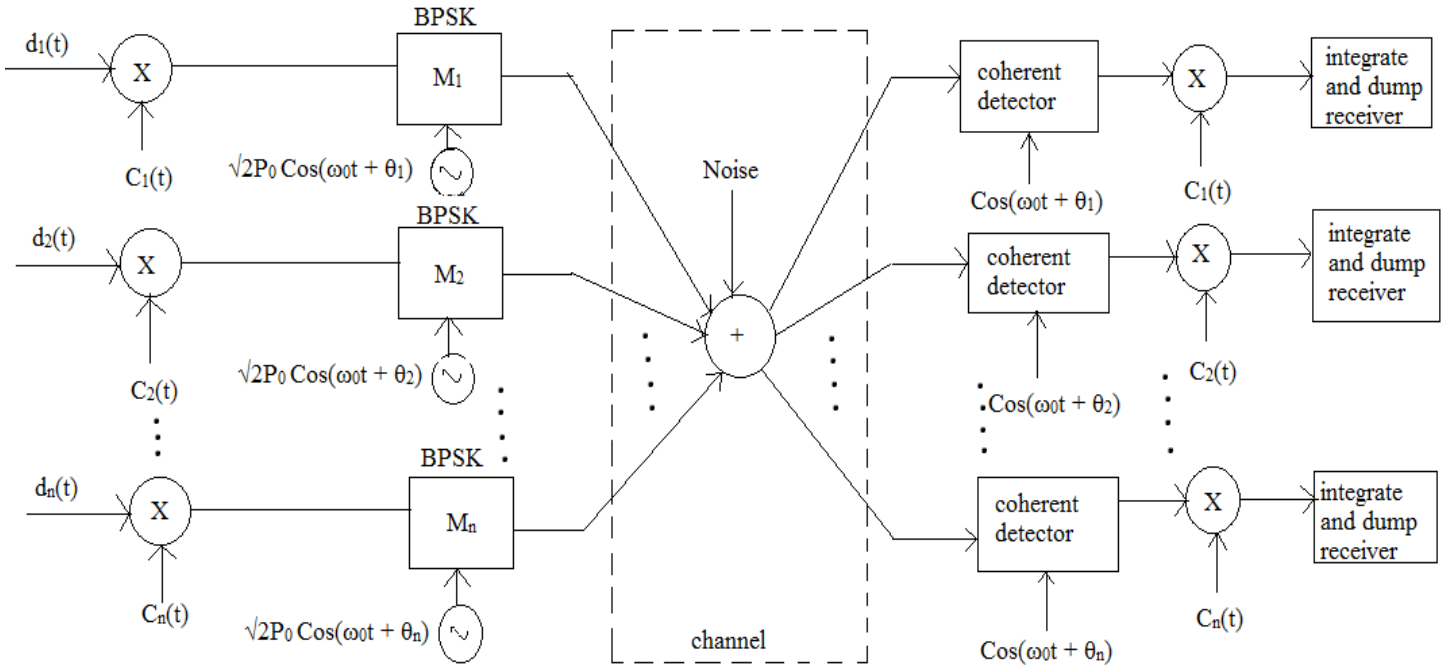
QAM receiver explanation **02M**

- 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 $A_o(t)$. 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 = $A_o(t)\sqrt{2P_s}\ 2T_b$ and $A_e(t)\sqrt{2P_s}\ 2T_b$
- Finally, the original bits are obtained from $A_e(t)$and $A_o(t)$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) Draw and explain the block diagram of DSSS based CDMA system.

Ans:

Diagram **04M**



DSSS based CDMA system

Explanation: **04M**

- If with all its huge transmission bandwidth if a DSSS system can serve only one user, it will be indeed a waste of bandwidth.
- DSSS systems can in fact provide multiple access i.e. allow multiple users.
- This multiple access facility is called the Code Division Multiple Access (CDMA).
- It has certain advantages over the other multiple access facilities like TDMA, FDMA.
- It does not require any bandwidth allocation as in FDMA or any time allocation as in TDMA.
- In CDMA using DSSS each user is provided with a unique PN code and the PN codes given to different users are almost uncorrelated
- 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 knowledge of other users.
- Fig above illustrates the principle of CDMA based on DSSS.

Q6. Attempt any FOUR of the following: **16M**

a) State the advantages and disadvantages of PCM (any two each).

Ans: **Advantages: (Any 2)** **02M**

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.
5. It is possible to integrate various baseband signals like audio, video etc into a common format for easy multiplexing using TDM

Disadvantages : (Any 2)

02M

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 techniques.

Ans:

OFDM Principle **02M**

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.

OFDM technique: **02M**

- OFDM is 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.

c) Compare between BPSK and QPSK w.r to variable characteristics of the 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 terms:

(i) Code word

(ii) Code rate

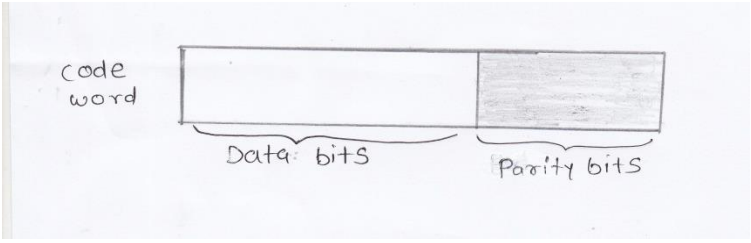
(iii) Hamming weight

(iv) Hamming distance related to code.

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.

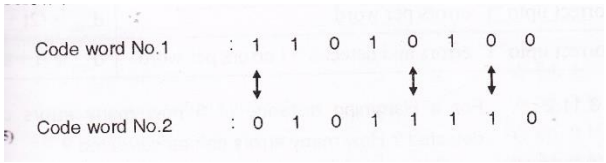
$$\text{Code rate (r)} = \frac{k}{n}$$

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)

Hamming distance related to code:- **01M**

The "Hamming distance" is the distance between two code word:



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

(e) Describe QPSK generator with waveform.

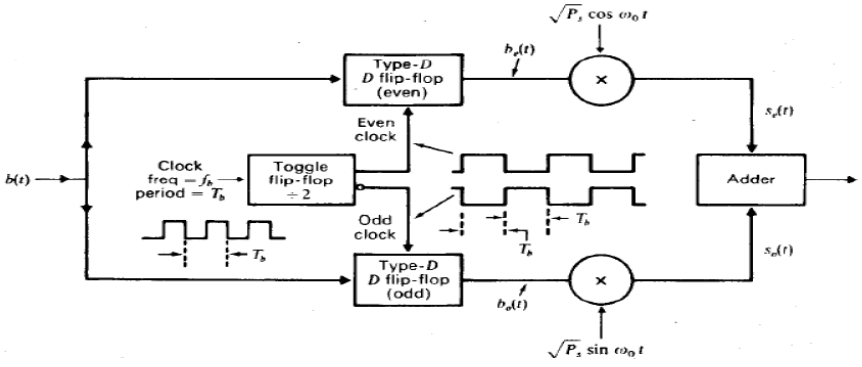
Ans:

Operation: **01M**

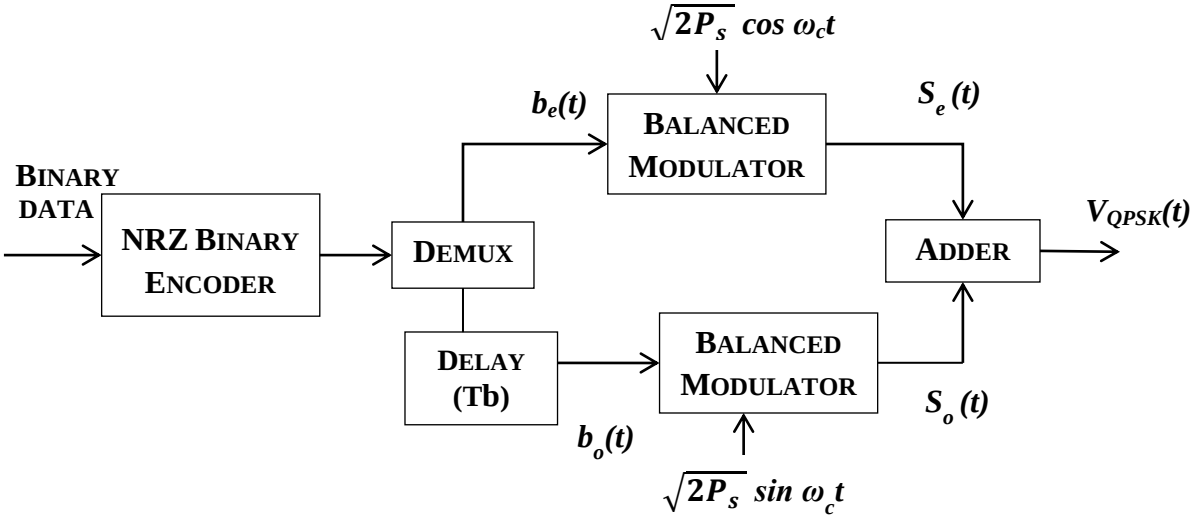
- The input data sequence is first converted into a bipolar NRZ signal *b(t)*. The value of *b(t) = + 1* for logic 1 input and *b(t) = - 1* when the binary input is equal to 0.
- The Demultiplexer (DEMUX) will divide *b(t)* into two separate bit streams $b_o(t)$ and $b_e(t)$. The bit stream $b_e(t)$ 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, .
- 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 $\sqrt{2P_s}\ cos\omega_c t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sqrt{2P_s}\ sin\omega_c t$ 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,

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

Diagram:- **02M**



OR



Waveform of QPSK: **01M**