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.

Q1. Attempt any five: **20M**

1) Define the following terms:

a) Entropy: **1M**

Entropy is defined as the average number of bits per symbol needed to encode long sequences of symbols emoted by the source. Its unit is bit/symbol.

Mathematically:

$$H = \sum_{i=1}^{M} p_i \log_2(1/p_i) \text{ bits / symbol}$$

$$= p_1 \log_2\left(\frac{1}{p_1}\right) + p_2 \log_2\left(\frac{1}{p_2}\right) + \cdots + p_m \log_2(1/p_m)$$

b) Noise : **1M**

Noise is defined as an unwanted introduction of energy tending to interface with the proper reception and reproduction of transmitted signals.

If the noise impulse occur at the time of sampling and has amplitude equal to or exceeding the minimum level the noise pulse will be interpreted as a data bit and thus affects the sampling process.

The effect of noise on the data channel can be reduced by increasing the signal to noise ratio.

[Note : any of the relevant answer should be accepted]

c) S/N ratio : **1M**

Ratio of total signal power to total random noise power at input of the receiver within the frequency limits of their channel i.e. over the bandwidth is S/N ratio for a noiseless channel $S/N \rightarrow \infty$

d) Information rate : **1M**

It is defined as the average number of bits per seconds needed to encode the source output. It is represented by R.

$$R = r * H$$

Where r = no. of messages or symbols per second

H = average information per message or symbol.

2) Differentiate DM and ADM.

Sr. no	DM	ADM
1.	Is a digital modulation in which one bit per sample is transmitted i.e. step size is constant.	A system which uses variable step size is known as a adaptive data modulation.
2.	Reduces the BW requirement to a great extent.	BW requirement is increased.
3.	There are two types distortion in DM: slope overload & granular Noise.	In ADM there is reduction in slope overload and distortion.

4.	Wide dynamic range of analog signal cannot be used	Wide dynamic range of analog signal can be used due to variable step size.
5.	S/N ratio is high	Improvement in S/N ratio.

[note: any four points 4 marks]

3) Explain quadrature amplitude modulation with block diagram.

Ans: Explanation **2M**

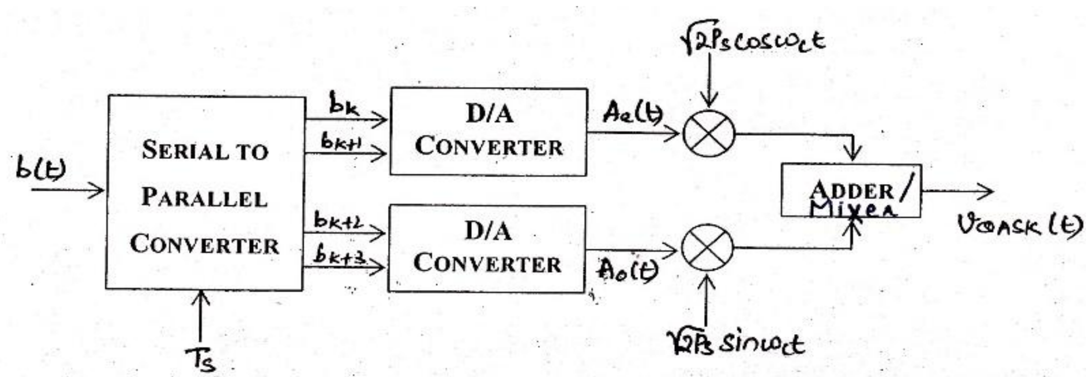
- QAM is a form of digital modulation where the digital information is contained in both the amplitude and phase of the transmitted carrier signal.
- Noise immunity of the system can be improved by allowing the signal to differ not only in phase but also in amplitude is called as QAM.
- The bit stream b(t) is applied to the series 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 convertor.
- 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, $A_0(t)$ is modulated by the carrier $\sqrt{2P_s}\sin\omega_c t$ in the balanced modulators
- The balanced modulator output 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 of QAM Modulator: **2M**



QAM Transmitter

4) How error detection and correction is done?

Ans : Error detection: **2M**

- is the process of monitoring data transmission and determining when error have occurred.
- It does not correct errors.
- The most common error detection techniques are redundancy checking, which includes:
  1. Vertical redundancy checking
  2. Checksum
  3. longitudinal Redundancy checking and
  4. Cyclic redundancy

- Error correction: **2M**

  - Two basic strategies are developed for handling transmission errors: error detecting codes and error correcting codes.
  - <u>error detecting codes:</u> parity bits, block and frame check characters and cyclic redundancy characters are example of error detecting codes.

- Error correcting codes: error correcting codes includes information to enable the receiver to determine when as error has occurred and which bit is in error.
- There are two primary methods used for error correction

1. Retransmission (ARQ)
2. Forward error correction (FEC).

5) Explain multiple access techniques and write its uses.

Ans: Explanation **3M**

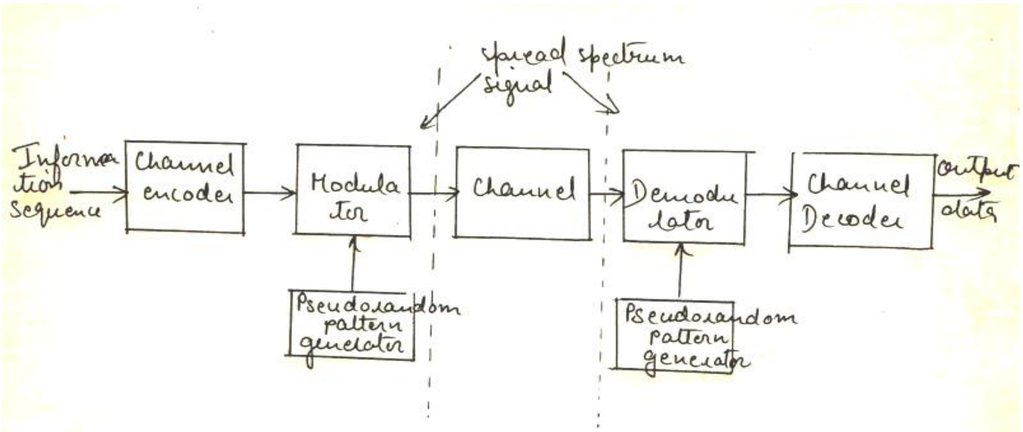
- Multiple access technique is the technique in which several users (separated geographically) share a common transponder to transmit and receive information, it represent the traffic feature of satellite communication.
- There are three different ways of sharing the transponder:
  1. Frequency division multiple access (FDMA)
  2. Time division multiple access (TDMA)
  3. Code division multiple access (CDMA)

Application of FDMA, TDMA and CDMA are: **1M**

i) Cell phone system.
ii) Satellite system.
iii) Digital system.
iv) Data networking.

6) Explain the block diagram of spread spectrum of digital communication system.

Ans: Diagram **2M**



Explanation **2M**

- The input of the system is a binary information sequence.
- Pseudo random pattern generator generates PN binary sequence.
- It is impressed on the transmitted signal at the modulation.
- The modulated signal along with the pseudo random sequence travels over communication channel.
- This sequence spreads the signal randomly over a wide frequency band.
- Output of the modulated signal is spread spectrum signal.
- The Pseudo random sequence is random from the receiver signal by the other 'Pseudo random' generator at the receiver.
- The two pattern generators operate in synchronization with each other.
- The receiver demodulated the transmitted signal if and only if a known pseudo – noise sequence has been transmitted along with the information signal.

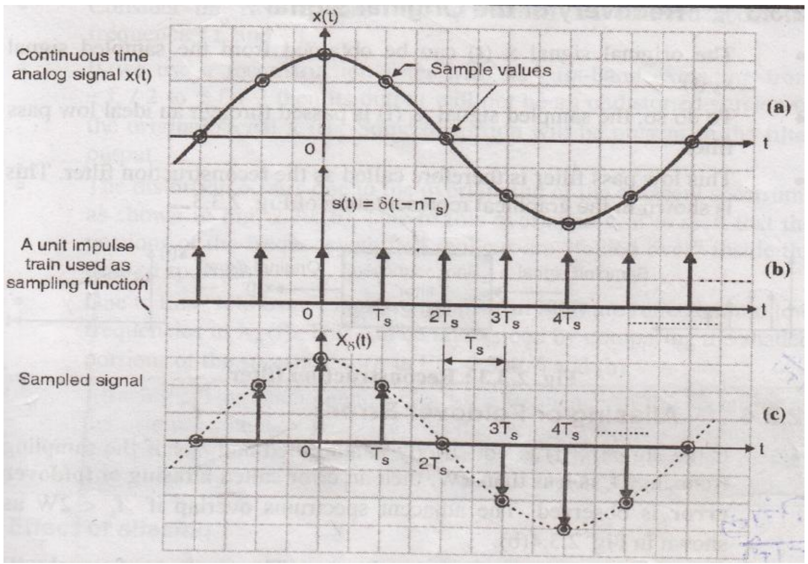
7) Explain sampling theorem.

Ans: Explanation **2M**

- Sampling process converts a continuous time varying signal to a discrete time varying signal.
- Sampling theorem states that a band limited signal of finite energy having the highest frequency component $F_m H_g$ can be represented and removed completely from a set samples taken at a rate of F_s sampling per seconds provides that $F_s \geq 2F_m$

Where F_s is the Sampling Frequency.

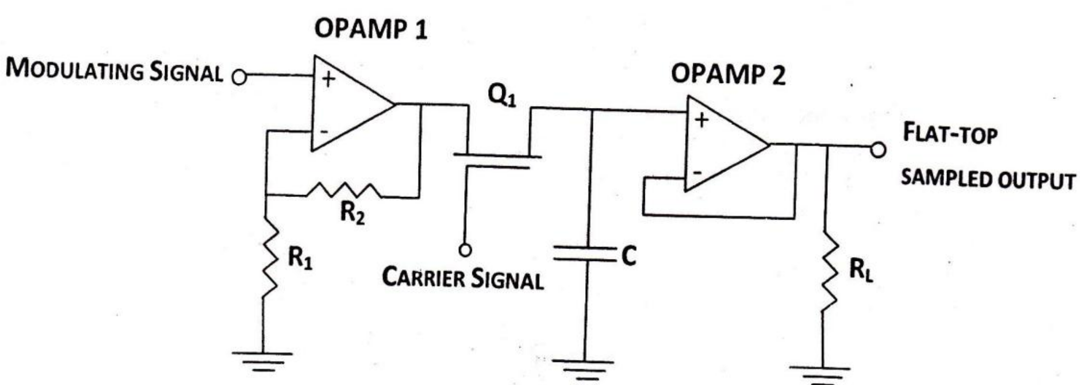
Diagram **2M**



Q2. Attempt any four: **16M**

1) Explain flat TOP sampling;

Ans: Diagram **1M**



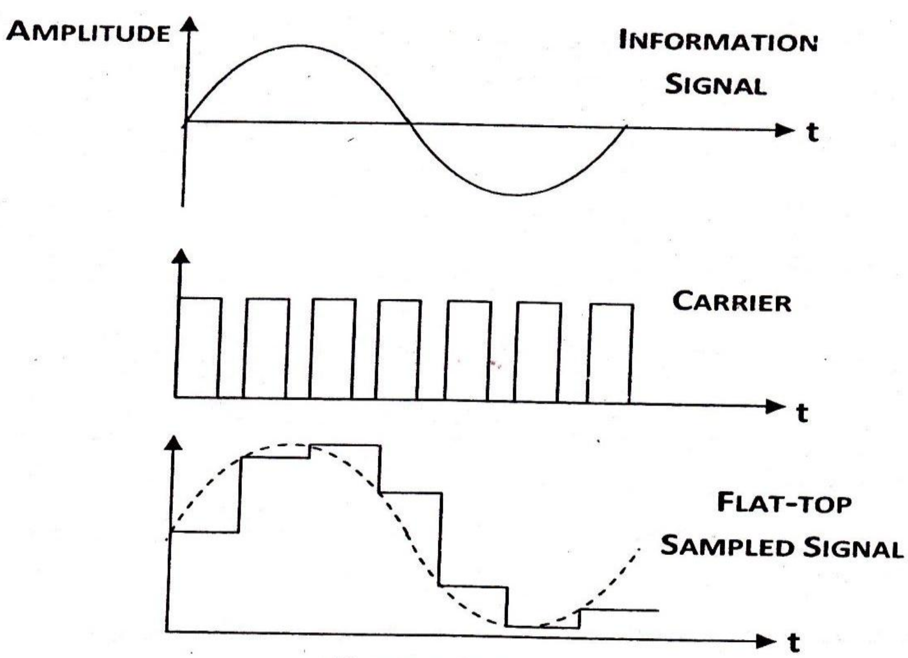
Flat-top Sample Generator

Explanation **2M**

- Flat TOP sampling is the sampling process in which the amplitude of each sampled pulse remains constant during that particular pulse duration.
- Flat top sampling uses the sample and hold circuit.

- The circuit samples the analog waveform and holds the amplitude value constant unit it is sample again.
- The top of the pulse does not follow the analog signal and remains flat, therefore the name flat-top sampling.
- Such a sampling metho9d can be applied to the PAM circuit of simply adding a capacitor to the nature sampling circuit.
- When the pulse is ON, the amplitude of the pulse does not follow the modulating signal but remains constant and is proportional to the sampled amplitude of the modulating signal.
- The flat-top sampled waveform is shown in figure.

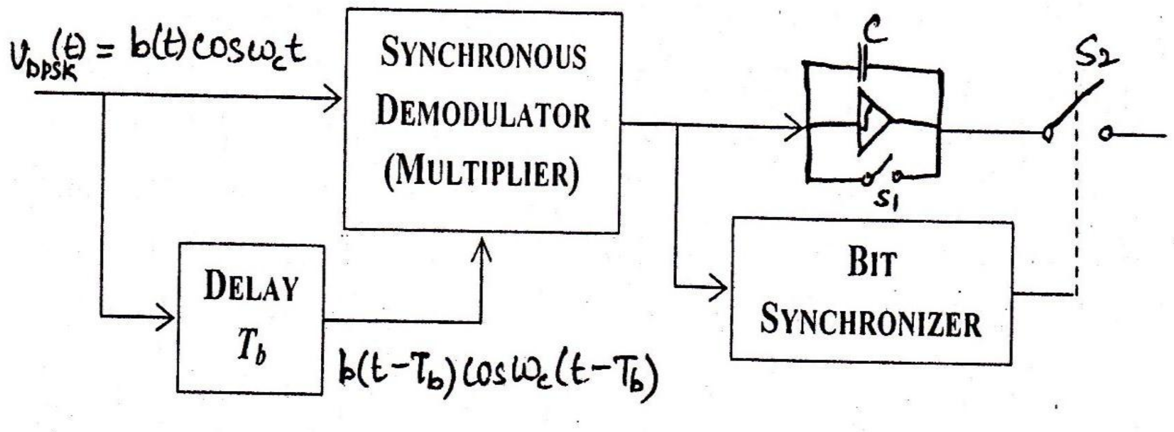
  Waveform: **1M**



Flat-top Sampled waveform

2) Draw and explain the DPSK receiver.

Ans: diagram **2M**



DPSK Receiver

Explanation: **2M**

- Here the received signal and the received signal delayed by the bit time Tb are applied to a balanced modulator. The balanced modulator output is given by,

  $b(t)b(t-T_b)(2P_s)\cos\omega_c(t+\theta)\cos[\omega_c(t-T_b) + \theta]$

  $= b(t)b(t-T_b)P_s\{\cos\omega_c T_b + \cos[2\omega_c(t-T_{b/2}) + 2\theta]\}$

- The output of the balanced modulator is applied to the integrator which suppressed the double frequency term.

- The first term $[b(t)b(t-T_b)P_s\cos\omega_c T_b]$ is the required signal and $\omega_c T_b$ is selected in such a way so that $\omega_c T_b = 2n\pi$ (where n is integer) so that $\cos\omega_c T_b = +1$ and the signal output will be as large as possible.

- Further with this selection, the bit duration encompasses an integer number of clock cycle of the double frequency term is exactly zero.

- The transmitted data bit d(t) can be determined from the product $b(t)b(t-T_b)$.

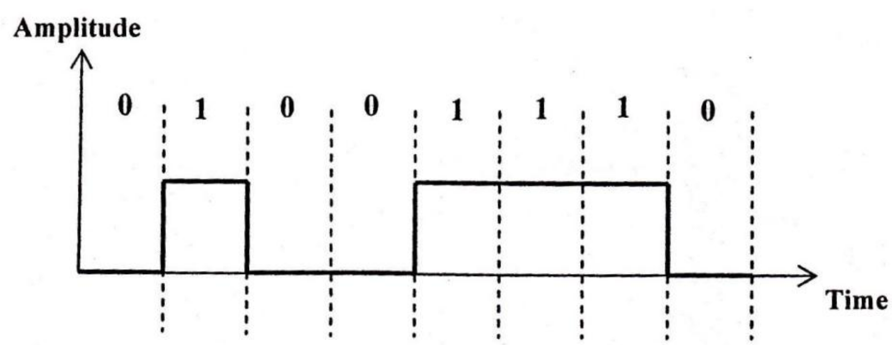
- If there is no phase change between b(t) and b(t-T_b) then d(t) = 0.
- In this case, b(t)b(t-T_b) = -1. Then d(t) = 1.

$$BW= \frac{2}{Ts} = \frac{2}{2Tb} = \frac{2}{2Tb} = f_b$$

3) Explain the unipolar, bipolar, NRZ and RZ with waveforms.

Ans: Unipolar Encoding: **1M**

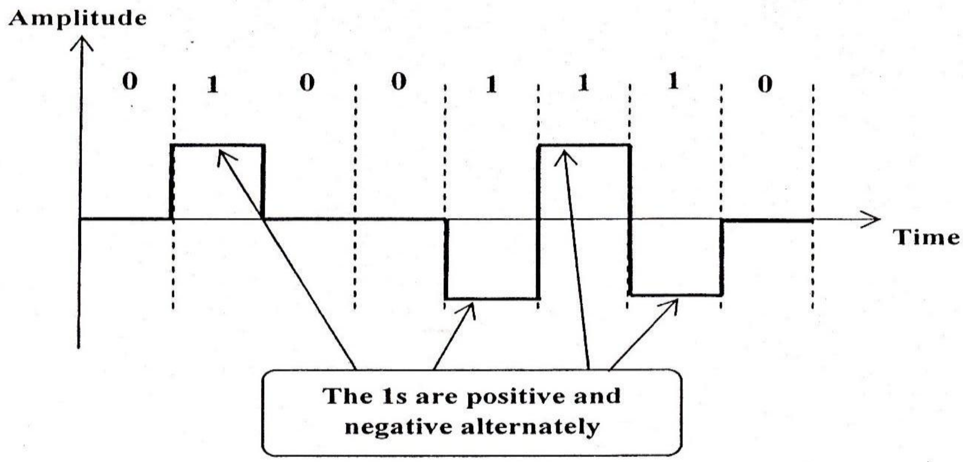
It uses polarity. Logic 0 is represented by 0V and logic 1 is represented by a constant signal level, say +v during its entire bit interval T_b.



Unipolar NRZ encoding

Bipolar encoding: **1M**

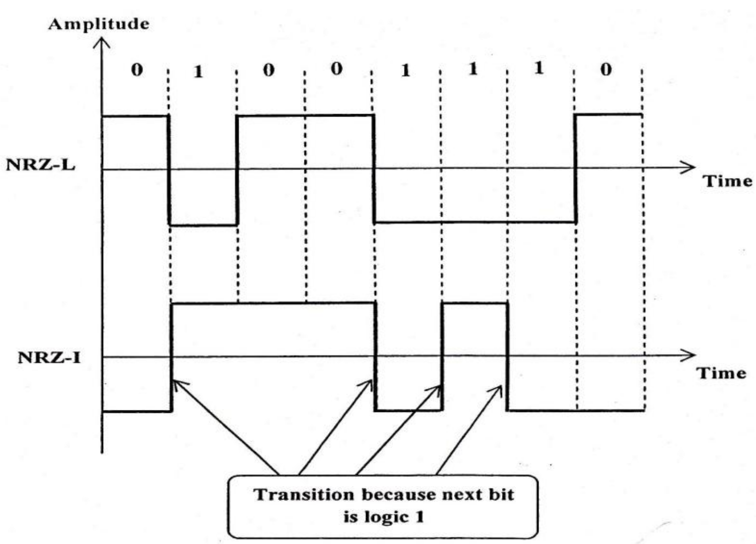
- It uses three voltage level: positive, negative and zero. The voltage level for logic 0 is at zero, while the voltage level for logic 1 alternates between positive and negative. i.e. if first logic1 is represented by positive voltage, the second will be represented by negative voltage and third by positive and so on.



Bipolar AMI encoding

NRZ: **1M**

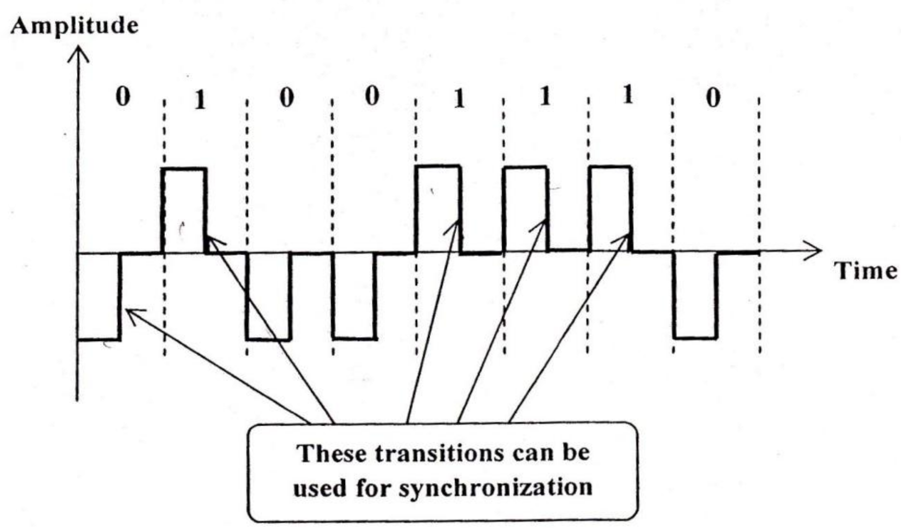
- In polar NRZ encoding the level of the signal is always either positive or negative. There are two methods : NRZ-L and NRZ-I
- In NRZ-L (NRZ-level) level encoding the level of voltage determines the value of bit positive voltage means bit is logic0 and the negative voltage means the bit is logic1.
- In NRZ-I (NRZ-Invert) an inversion of the voltage level represents logic1 i.e. the signal is inverted if logic is encountered. If there is no change the bit is logic0.



NRZ-L and NRZ-I encoding

RZ (Return to Zero): **1M**

- RZ encoding uses three values: positive, negative and zero. In RZ the signal changes not between bits but during each bit. Unlike NRZ-L, half way through each bit interval the signal returns to zero.
- Logic1 is represented by positive to zero and Logic0 by negative to zero.



RZ encoding

4) State the application of spread spectrum modulation.

Ans: Applications of spread spectrum are: **4M**

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 drivers.
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)

5) Write hamming code fir data bits 1010.

Ans: **4M**

D7	D6	D5			D3	
1	0	1	R4	0	R2	R1

1	0	1	R4	0	R2	0

D7	D5	D3
1	1	0

Therefore R1 = 0

1	0	1		0	1	0

D3	D6	D7
0	0	1

Therefore R2 =1

			R4		R2	R1
1	0	1	0	0	1	0

Completed code word

D5	D6	D7
1	0	1

Therefore R4 =0

6) Explain the slope overload and granular noise in delta modulation.

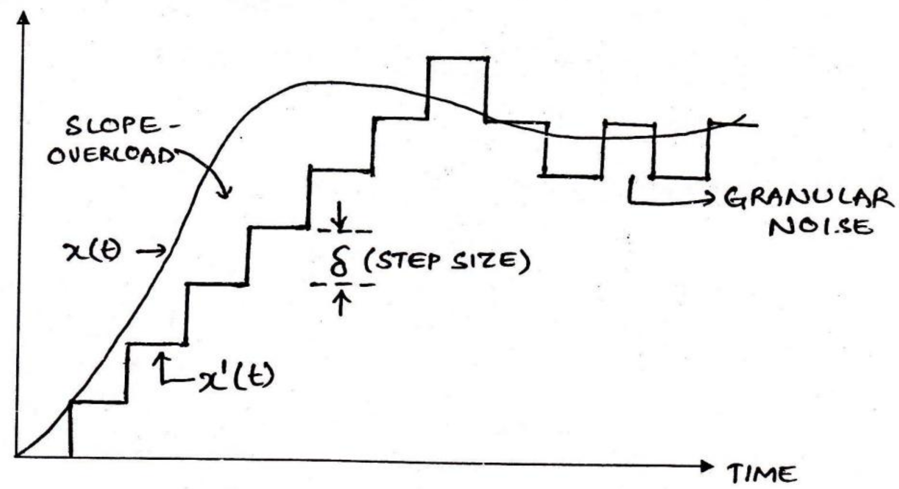
Ans:

1. Slope-overhead distortion: **2M**

- If the slope of the signal x(t) is much higher than that of the approximate signal x'(t) over will not follow x(t) at all as shown in figure.
- The difference between x(t) and x'(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'(t).
- The slope overload error can be reduced by increasing the step size δ or by increasing the sampling frequency.
- However with the bit rate and bandwidth requirement will increase.

2. Granular nose: 2M

- When the input signal x(t) is relatively constant in amplitude, the approximated signal x'(t) will hunt above and below x(t) .
- This leads to a noise called granular noise.
- It increases with in step size δ.
- To reduce granular slope-overload distortion.



Waveform showing Slope-Overload error and Granular noise in DM

Q .3 Attempt any four

1) Distinguish between baseband and band pass transmission.

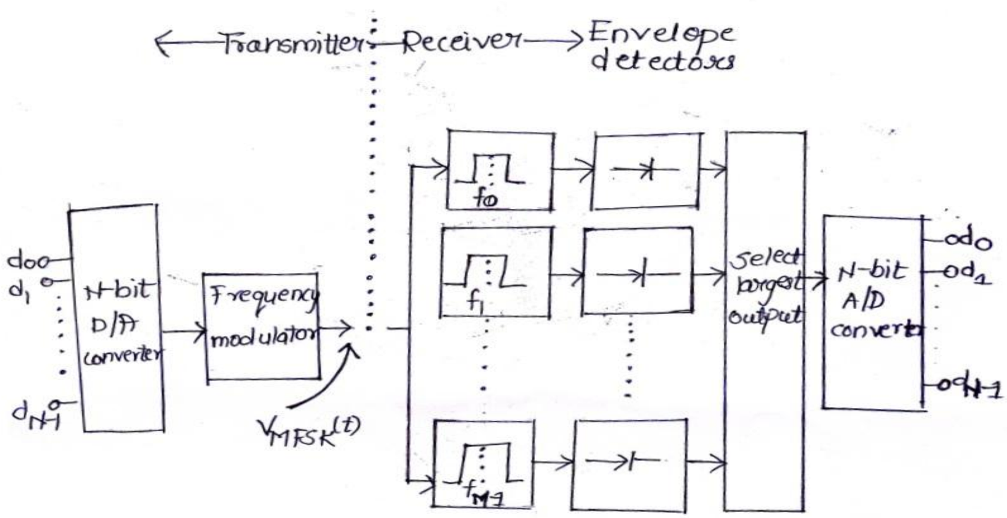
Ans :

Baseband	Bandpass
In baseband transmission original Information or baseband signal Are transmitted directly.	In bandpass transmission signal having frequency from F1 to F2 are transmitted where F1 not equal to Zero.
In baseband transmission signal Occupies lower order frequency of frequency spectrum.	In bandpass transmission signal occupies upper part of the frequency spectrum
Modulation is not necessary	Modulation is necessary
It is used for small distance Communication.	It is used for long distance communication.
Eg. telephone line , LAN	eg. Radio / TV broadcast, satellite communication.

(Note : Any four points 4 marks)

2) Draw labeled block diagram of M-ary FSK transmitter and receiver.

Ans**:** Diagram: *[Note: transmitter 1 mark, receiver 3 marks]*



3) List the steps involved in creating a checksum.

Ans: Following are the steps **4M**

After transmitting the block of deta bytes the checksum bytes are also transmitted. The checksum byte is re generated at receiver separately by adding the received byte.

a) The re-generated checksum byte is then compared with transmitted data if both are identical then there is no error and if they are different then the errors are present in the block of received data bytes.

b) Sometimes 2’s complement of checksum is transmitted instead of checksum itself. The receiver will accumulate all the bytes including 2’s complement of the checksum.

4) Why Pseudo noise sequence is used in spread spectrum modulation.

Following are the reasons of adding pseudo noise sequence used in spread spectrum modulation.

1) The spread spectrum signal is pseudo random in nature, this makes it like a random noise therefore normal receiver cannot demodulate spread spectrum signal.

2) To avoid the intentional interference or jamming pseudo noise sequence is used in spread spectrum modulation.

3) In rejecting unintentional interference from some other user also the pseudo noise sequence is added in spread spectrum modulation.
4) In obtaining message privacy by superimposing a pseudo random noise on the transmitted message *(1 reason 1 mark)*

5) Explain companding PCM (Pulse Code Modulation)

Ans: Explanation **2M**

For the input signal of smaller amplitude the signal power is very low hence in case of non-uniformed signal this weak signal are lost during the transmission if they are transmitted using uniform quantization techniques but by using commanding techniques we can improve signal to quantization noise ratio of weak signals .

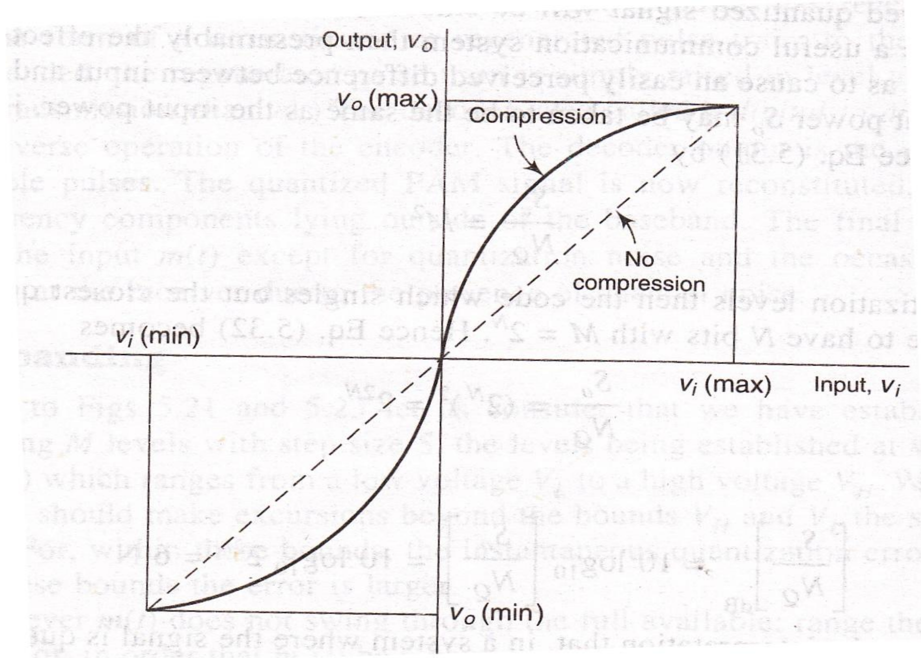
Companding is the process of compressing and expanding the non-uniform signal by using log and antilog amplifier at the start of the transmitter and at the end of receiver respectively.

Before taking the samples of non-uniform information it will convert it in to uniform nature where high amplitude signal gets compressed in to small value by using log amplifier which is called as compression. This uniform signal can be transmitted through PCM system using linear quantization.

At the receiver for getting the original information which is having non uniform nature the signal will be expanded using antilog amplifier at the end of receiver.

In this way compression provides higher gain to the weak signal and smaller gain to strong input signals. Thus the strong signals are artificially compressed and at the receiver using expander the compressed signals gets boosted the original amplitude which is shown in following characteristics.

Diagram of Companding 2M



6) Explain the term hamming weight and hamming distance.

Ans: Hamming weight : **2M**

The hamming weight of a code word " X " is defined as the no.of non zero elements in the code word. Eg. Code word " X " = 1011010

So hamming weight = 4

Hamming distance : **2M**

Hamming distance between the two code words is defined as the no.of locations in which their respective elements changes .

Eg. Codeword 1 = 1 0 1 1 0 1 1 1

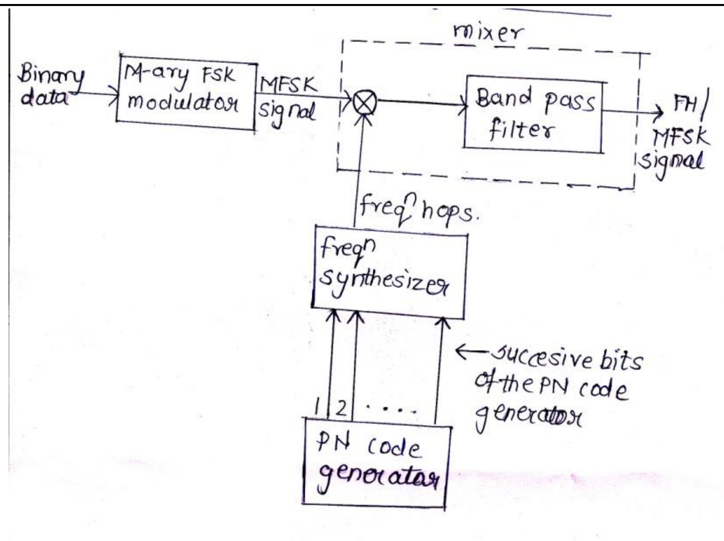
Codeword 2 = 1 1 0 0 1 1 1 1

Bits 2 , 3 , 4 , 5 are changes so hamming distance is 4.

Q4. Attempt any two: 16 M

1) Draw block diagram of slow frequency hopping transmitter and receiver. Also explain its working?

Ans:- Block Diagram of transmitter:- 02 M

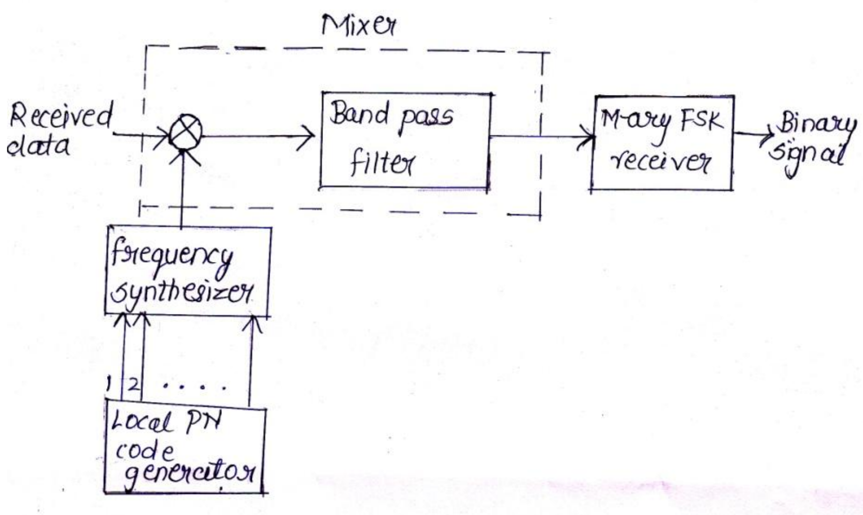


Explanation:- **02 M**

Slow frequency hopping transmitter as shown in diagram

- The binary data sequence b(1) is applied to the M - ary FSK modulator the o/p of which goes to the i/p of mixer.
- The other i/p to the mixer block is obtained from a digital frequency synthesizer. The mixer is consist of multiplier followed by a band pass filter.
- At the multiplier o/p, we get two frequencies their sum and their difference frequency components.
- The band pass filter is designed to select only the sum frequency component and rejecting all other components only sum components of frequency is transmitted.
- Each frequency hop is mixed with MFSK signal to produce the transmitted signal.
- Frequency hops at the o/p of synthesizer are controlled by successive bits at the o/p of PN code generator.
- The o/p bit of code generator change randomly.
- Frequency hops produced will vary in a random manner.
- The total BW of the transmitted FH/MFSK signal is equal to the sum of all the frequency hops.
- BW of transmitted signal is very large in GHz.

Block diagram of slow frequency hopping receiver:- **02 M**



Explanation:- **02M**

Slow frequency hopping Receiver

- The received signal is applied to a mixer.
- The other I/P to the mixer come from frequency synthesizer.
- The digital signal is driven by PN code generator which is synchronized with PN code generator at the transmitter and generates the same code sequence.
- At the o/p of multiplier, we get i/p signals their sum and difference.
- The o/p of mixer is then applied to a non- coherent MFSK demodulator.
- At the o/p of the MFSK detector we obtained the digital modulating signal.
- Each matched filter is matched to one of the tones of the MFSK signal
- The o/p of M - ary FSK receiver is binary signal.

2) Compare PAM, PWM

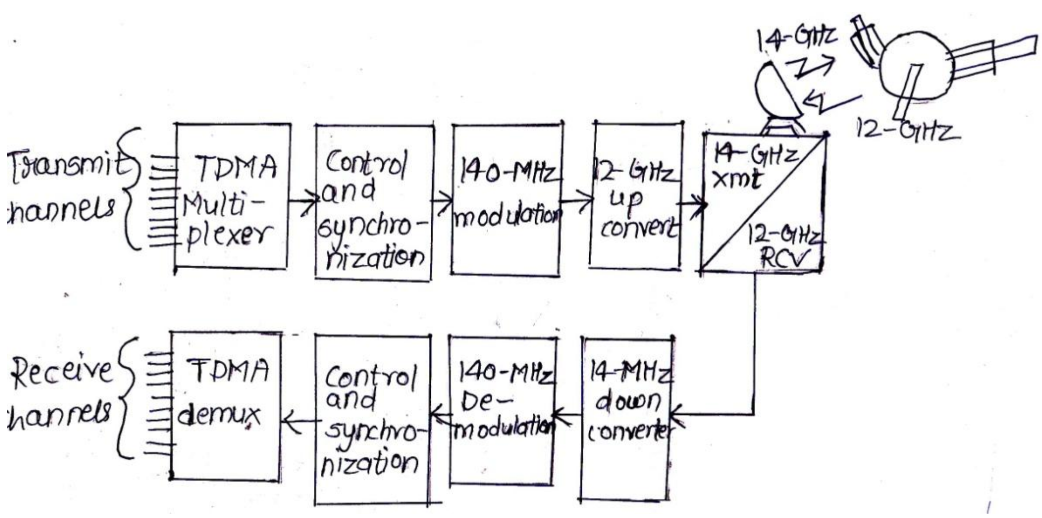
Ans :- (Any four point, each point 02 marks)

Sr. no	PAM(Pulse Amplitude Modulation)	PWM (Pulse Width Modulation)

1.	In PAM system, the amplitude of the pulsed carrier changed in proportion with instantaneous amplitude of modulating signal	In PWM, the width of pulse of the modulated pulses varies in proportion with amplitude of modulating signal
2.	Carrier signal are in train of pulses	Carrier signal are in train of pulses
3.	Bandwidth requirement is low	Bandwidth requirement is High
4.	Noisy immunity is low	Noisy immunity is high
5.	Generation and detection Circuit are complex	Generation and detection Circuit are easy
6.	No need of synchronization	No need of synchronization
7.	Transmitted power varies with amplitude of pulses	Transmitted power varies with width.

3) Draw block diagram of TDMA .State advantages of TDMA over FDMA

Ans:- Block diagram of TDMA:- **04 M**



- TDMA stands for Time Division Multiple Access.
- In TDMA the entire bandwidth can be used by every user.
- A station can use the entire bandwidth only for allocated time.

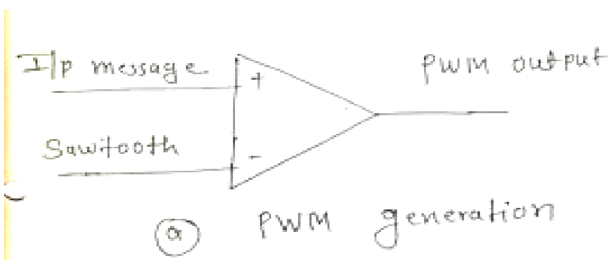
Advantages of TDMA over FDMA :- **(Any four points, each point carries 01 mark)**

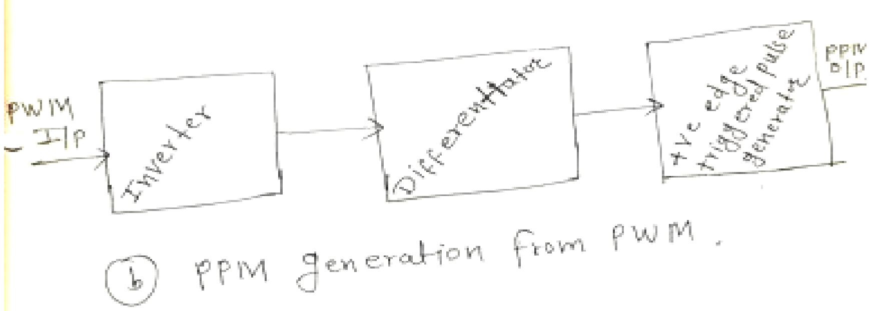
- In TDMA since only one station is present at any given time.
- The entire channel bandwidth can be allotted to a single channel at given instant of time. This is the advantages for digital channel which demands larger bandwidth
- The frequency, selective fading does not affect the TDMA to the extent if effect the FDMA.
- TDMA by default can well with the digital. It can be easily used for data transmission.
- As only one channel is being transmitted at a time it is not necessary to separate out varies channel at the receiver.

Q5. Attempt any two: 16M

1) Describe the pulse position modulation with neat waveform and suitable diagrams. And write advantages and disadvantages.

Ans**: Block diagram 2M**



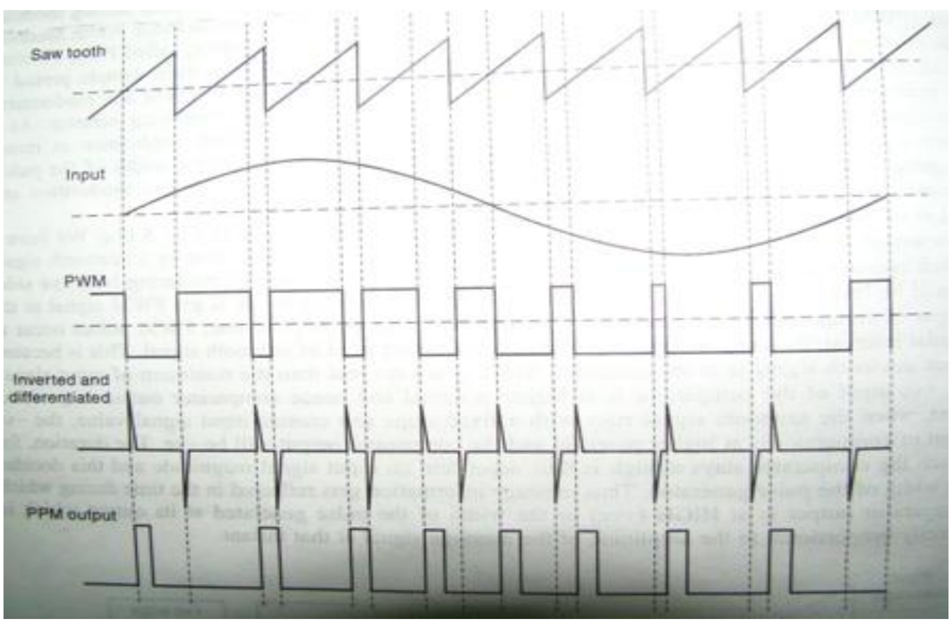


Explanation: 2M

When the position of pulse is varied in accordance with the instantaneous value of modulating signal called as PPM where the amplitude and width of pulse is constant.

The generation of PPM is as shown in block diagram. The PPM generation this is a post processing of PWM signal as shown in a diagram. This PWM signal is given to an inverter which is reverses polarity of the pulse. If is followed by differentiator we will have positive spikes original. PWM signal pulse was going high to low and negative spikes where low to high. This is shown in W/F in differentiator. These spikes are then fed to a positive edge triggered. Fixed with pulse generator which generates pulse of fixed width a positive spike appears coinciding with the falling edge of original PWM signal. These falling edges were dependent on input message and hence delay in occurrence of the fixed width pulses is proportional to the amplitude of input messages at that instant. The PPM outputs where positions of the pulses in sample period carry input message information as shown in wave form.

Waveform of PPM: **2M**

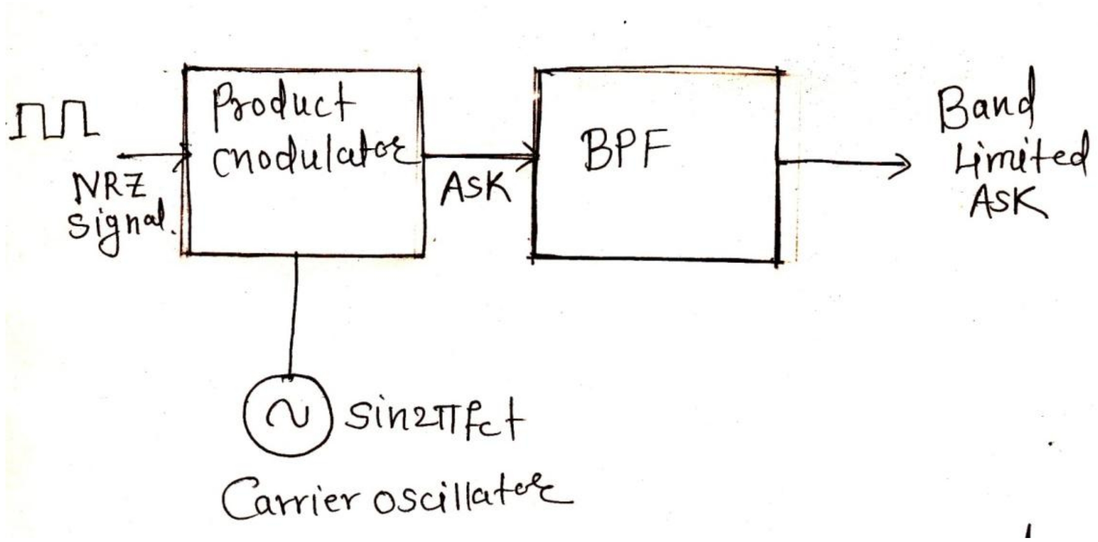


- Advantages of PPM **1M**
  1. Good noise immunity
  2. Requires constant transmitter power output
- Disadvantages of PPM **1M**
  1. Bandwidth is large

2. It requires synchronization between transmitter and receiver.

2) Draw and explain ASK and FSK transmitter and receiver block diagram

Ans : **ASK transmitter** **1M**



The ASK signal cab be transmitted

VASK(t) = dsin(2πfct)

VASK(t) = sin(2πfct) where d =1

VASK(t) = 0 where d =0

Where d = data bit (data bit either 0 or 1)

Explanation : **1M**

- ASK is the digital carrier modulation in in which the amplitude of sinusoidal carrier will take one of two predetermined values in response to 0 or 1 value of digital input signal.
- The bandwidth of ASK signal is dependent on the bit rate F_b where bit rate $F_b = 1/T_b$ as shown in diagram.
- The maximum bandwidth of ASK signal is

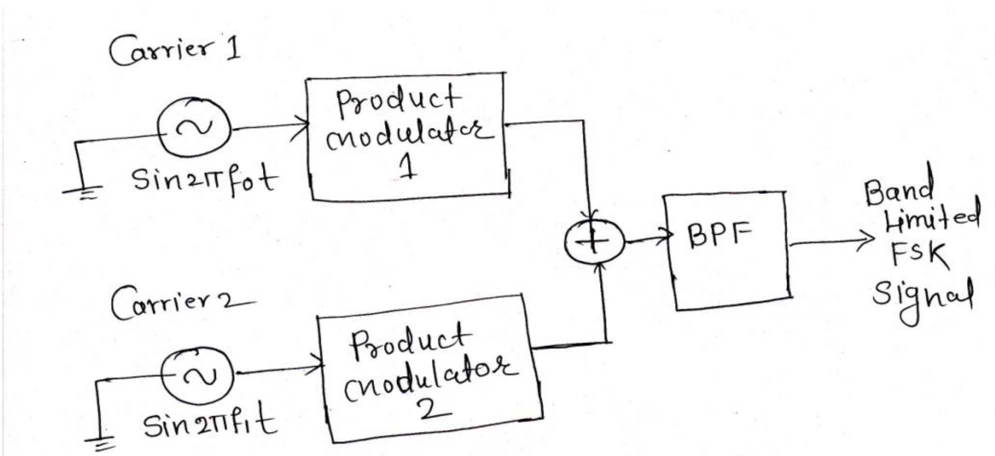
  Bandwidth = 2 F_b

There are two types of ASK receiver: **1M**

1. Coherent ASK receiver
2. Non Coherent ASK receiver

- ASK signal analog with noise is also applied to the multiplier
- The multiplier output is then applied to an integrator which integrator over one bit duration T_b
- The integrator output is sampled at a particular instant corresponding to the maximum passible value of output and the sampled value is held by the sample and hold circuit.
- The output of sample and hold circuit is compared with a reference voltage V by a comparator.
- If the sample and hold output less than V, then comparator output is high which indicates that the receiver ASK signal corresponds to 1.
- Thus at the receiver output we recover the original binary signal.

Diagram shows FSK transmitter **1M**



Explanation: **1M**

In frequency shift keying (FSK) the frequency of sinusoidal carrier is shifted two discrete values.

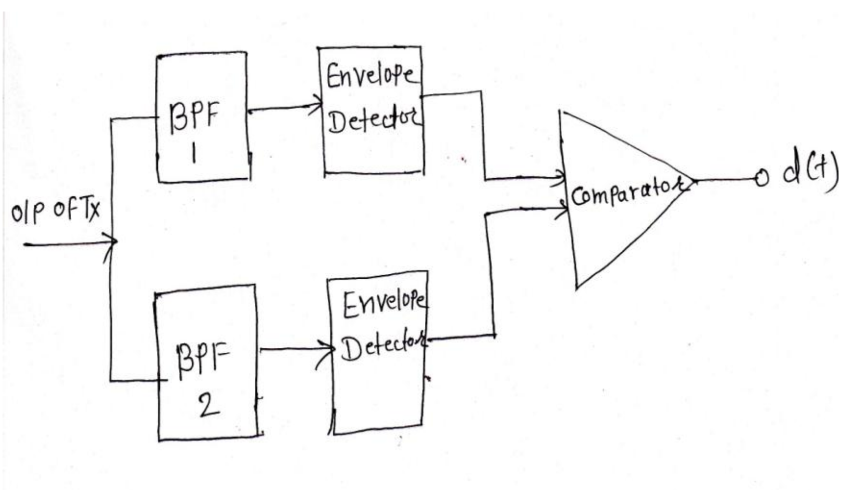
- One of these frequency f_1 represents a binary 1 and other values f_0 represents a binary 0. In FSK there is no change on amplitude of the carrier.

- It consist of two oscillator which produces sin ware at frequency f_1 and f_0 the oscillator output are applied to the input of two products modulators.

- When a binary 0 is transmitted $P_0 = 1$ and When a binary 1 is transmitted $P_1 = 0$.

- Bandwidth of FSK is $4f_b$

Diagram shows FSK receiver:- **1M**



Explanation:- **1M**

- Output of transmitter is applied to two band pass filter one with Centre frequency at F_H the other at F_L.
- We assumed that $F_H - F_L = 2(\Omega/2\pi) = 2F_b$

- One filter will pass higher frequency and other will pass lower frequency. The output of filter is applied to envelope detectors by a comparator. Comparator is a circuit that accepts two input signal.
- It generates a binary output which is at one level.

3) Why multiplexing is necessary and explain concept of WDM.

Ans :

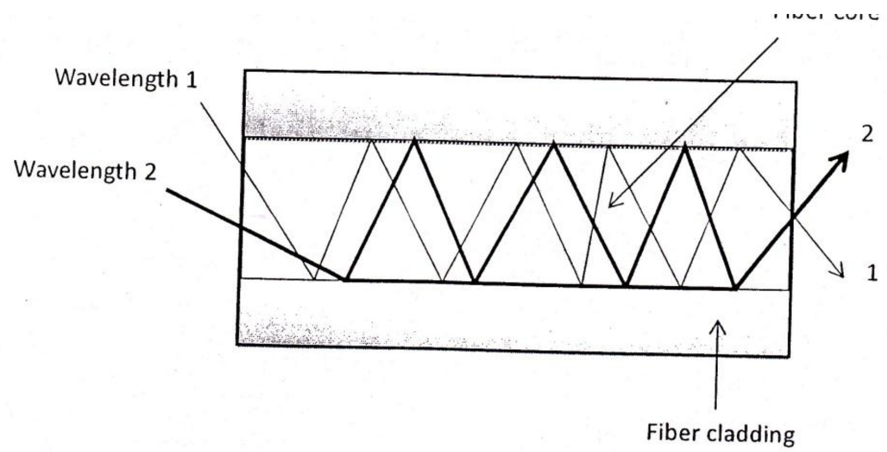
Multiplexing: **4M**

- As the data and telecommunication usage increases, so does the traffic we can accommodate this increase by continuing to add individual line each time a new channel is needed.
- Today's technology includes high bandwidth transmission media such as coaxial cable fiber optic and satellite microwaves. Each of these has a carrying capacity for increases far of that needed for the average transmission signal.
- If the bandwidth of link is greater than the transmission needs of the devices to it, the excess capacity is wasted.
- An efficient system maximizes the utilization of all resources.

Concept of WDM: **2M**

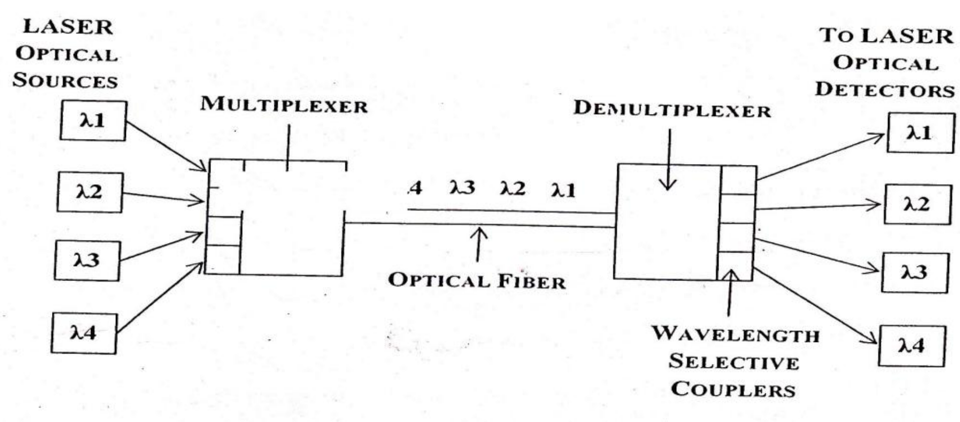
- Wavelength division multiplexing is an analog multiplexing technique in which different sources of information are propagated down an optical fiber on different Wavelength WDM is designed to use the high data rate capacity of optic cable.
- WDM is an analog multiplexing technique that combines optical signals.
- WDM is same as that of FDM except that the multiplexing and demultiplexing involve light signal transmitted to fiber optical channel
- In WDM the optical signal enter the fiber at the same time and travel through the same medium.
- It takes different transmission path and each signal arrives at the receiver at a slightly different time.

Diagram:- **1M**



Transmission of light waves through optical fiber

1M



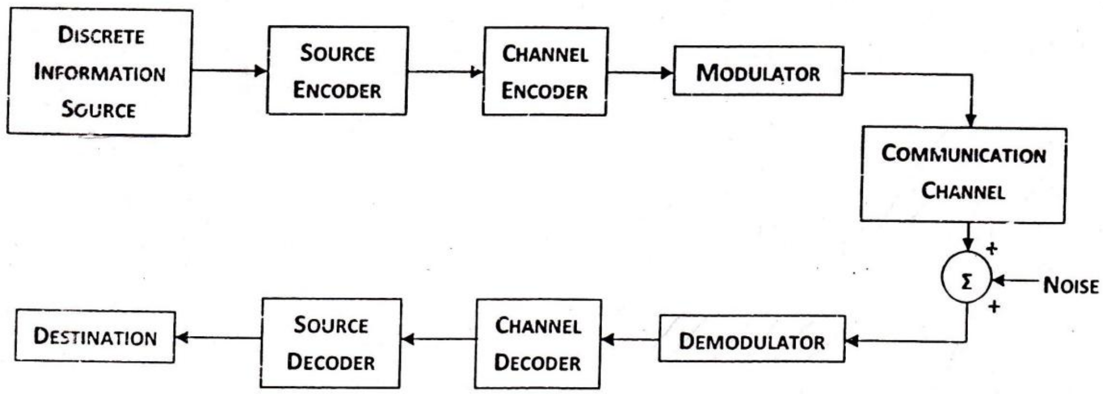
Conceptual view of WDM system

Q6. Attempt any four: 16M

1) Draw basic digital communication system and explain each block

Ans :

Diagram **2M**



Block diagram of a digital communication system

Explanation : **2M**

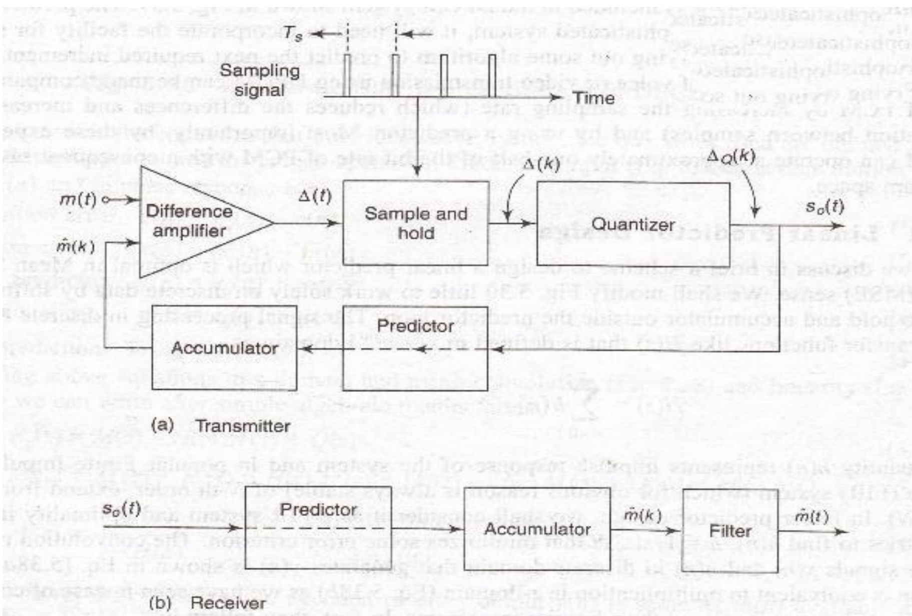
- The source of information is assumed to be digital. If it is analog thin it must be converted first to digital.
- Source coding :
  - Source coding consists of source encoder and source decoder.
  - The encoder generates the digital signal generated at source output into another signal in digital form.
  - The source encoder is used to eliminate and reduced redundancy for ensuring an efficient representation types of source techniques are PCM, DM and ADM
- Source decoding :
  - The source decoder is at receiver side. It functions as inverse to source encoder.
  - It delivers the destinations the original digital source output.
- Channel encoding :
  - Channel encoding consists of Channel encoder channel decoder.
  - It is used to minimize the effect of channel noise. This will reduce no. of errors in received data and make system more reliable.
- Channel decoding :
  - The channel decoder is at receiver side. It maps the channel output into digital signal.

  - In such a way that the effect of channel noise is reduced to a minimum by using channel decoder and encoder together provide a reliable communication over a noisy channel.
  - The channel decoder converts the code word into digital message.

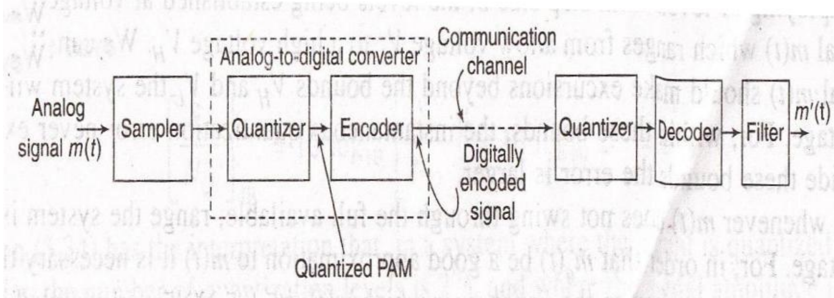
- Modulation :
  - Modulation is used to provide an efficient transmission of the signal over a channel,
- Discrete channel :
  - Discrete channel consists of modulator channel and demodulator.
  - It is called as discrete channel because its input as well as output are in discrete form.

2) Draw the block diagram of PPCM.

Ans : ***[NOTE : if student draw any one of these diagram they will get 4 marks.]***



Representation of the basic principle of differential PCM.



A PCM communication system.

3) Describe ASCII and EBCDIC code with example.

Ans :

ASCII (American standard code for information interchange):- 1M

- It is designed by American National Standards Institute.
- It is a 7 bit code with 2^7 = 128 possible combination and all of them have defined Delete characters and 32 control symbols.
- ASCII is a 7 bit code but the eight bit is often used. This bit is called as parity bit.
- Parity bit is used to detect the error introduced during transmission.
- Parity bit is generally added in the most significant bit(MSB)

Example : 1M

ASCII code word for character 'K'

Ans. Character "K" is (4B) therefore its code is 1001011.

EBCDIC (Extended binary coded decimal interchange code):- 1M

- This is an 8 bit code. However all the possible 256 combinations are not used.
- Therefore is no parity bit used to check errors in the basic code set.

Example : 1M

The EBCDIC code for representing 'SP' is

B_0	B_1	B_2	B_3	B_4	B_5	B_6	B_7
1	0	1	0	0	0	0	0

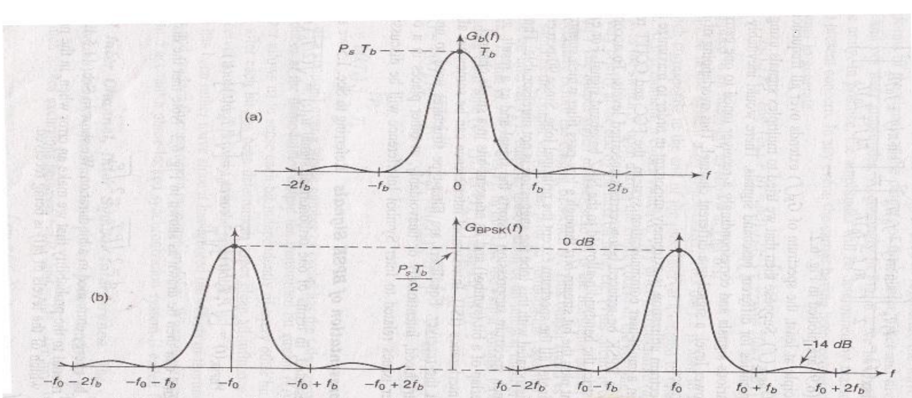
4) Compare between FHSS(frequency hopping spread spectrum modulation) and DSSS (direct sequence spread spectrum)

Ans:- ***[Note: any four points. 1 mark for each point.]***

DSSS	FHSS
1) PN Sequence of large bandwidth is multiplied with narrow band data signal.	Data bits transmitted in different frequency slot which are changed by PN Sequence.
2) Chip rate, it is fixed $Rc = \frac{1}{Tc}$	$R_c = \max(R_h, R_s)$
3) Modulation technique is used BPSK	M-ary FSK modulation technique is used
4) Processing gain $P_G = \frac{Tb}{Tc}$	$P_G = 2^t$
5) Acquisition time is long	Acquisition time is short.
6) This system is distance relative	Effect of distance is less

5) Draw and explain power spectral of BPSK.

Ans : Diagram:- **3M**



(a) Power spectral density of NRZ data $b(t)$. (b) Power spectral density of binary PSK.

Explanation : 1M

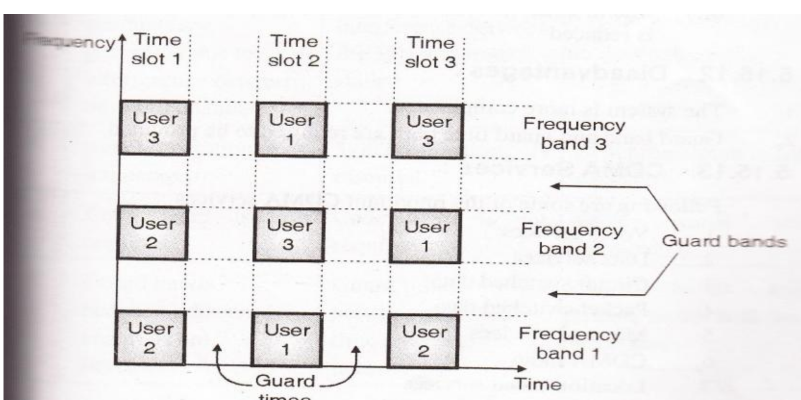
- In BPSK the distance d between signals is
  $2\sqrt{P_s T_b} = 2\sqrt{E_b}$
- Where E_b is $P_s T_b$ is the energy contained in a bit duration.
- The distance d is inversely proportional to the probability that we make an error, in the presentation noise.
- At end of receiver b(t) is being received.

6) Explain the concept of CDMA technology.

Ans : Explanation:- **2M**

- 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.

Diagram:- **2M**



Structure of CDMA