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. a) Attempt any three: **12M**

1)State advantages of digital communication.

Ans. (Any 4 points) **1M each**

Advantages of digital communication

1) Immunity to transmission noise and interference.
2) Regeneration of the coded signal along the transmission path is possible (Repeater can be used).
3) Digital signals are better suited than analog signals for procession and combining using technique called multiplexing.
4) Communication can be kept "private" and "secured" through the use of encryption.
5) Digital transmission systems are more resistant to analog systems to additive noise because they use signal regeneration rather than signal amplification.
6) Digital signals are simpler to measure and evaluate.
7) It is possible to store the signal and process it further.
8) In digital systems transmission errors can be corrected and detected more accurately.
9) Using data encryption only permuted receivers can be allowed to detect the transmission data.
10) Wide dynamic range.
11) Techniques such as data compression and image enhancement can be used.
12) Because of the advances of IC technologies and high speed computers, digital communication systems are simpler and cheaper.

2) What is meant by Quantization error? Describe quantization process in brief?

Ans: (**Quantization error-2 M, Quantization Process -2 M)**

Quantization error:-.

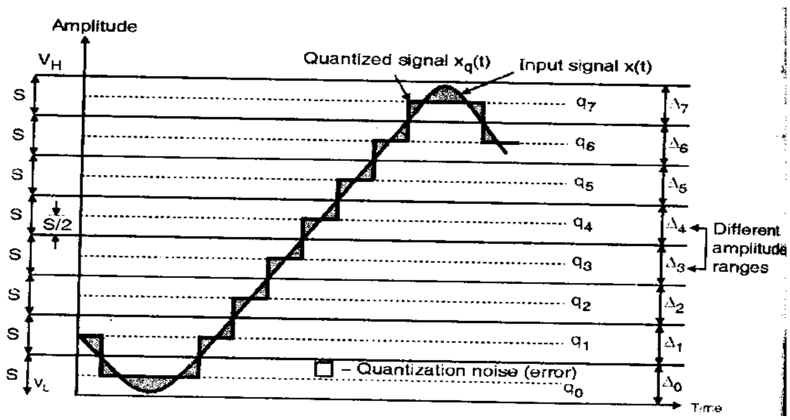
- The signal with discretized amplitude value is termed as quantized signal.
- The difference between the sampled signal and its quantized version which is measured and is represented in terms of quantization noise or quantization error.

Quantization process:

- 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= VH-VL / Q

Diagram of the Process quantization is as shown below-



3) Write any four specification of T-carrier system.

Ans: **(Any four specifications)** **1M Each**

<u>Specification of T-carrier system</u>

i) Leased lines come in two configurations T1 and T3. A T1 line offers a data transfer rate of 1.54 million bits per second.
ii) A T1 line is a dedicated connection meaning that it is permanently connected to the internet.
iii) This is useful for web server or other computers that need to be connected to the internet all the time.
iv) It is possible to lease only a portion of a T1 line using one of two systems fractional T1 or Frame relay.
v) You can lease them in blocks ranging from 128 kbps to 1.5 Mbps.
vi) The differences are not worth going into in detail but fractional T1 will be more expensive at the slower available speeds and frame relay will be slightly more expensive as you approach the full T1 speed of 1.5 Mbps.
vii) AT3 line is significantly faster at 45million bits per second.
viii) Leased lines are expensive and are generally used only by companies whose business is built around the internet or need to transfer massive amounts of data.

4) List out application of spread spectrum system.

Ans: **(Any 4)** **1M each**

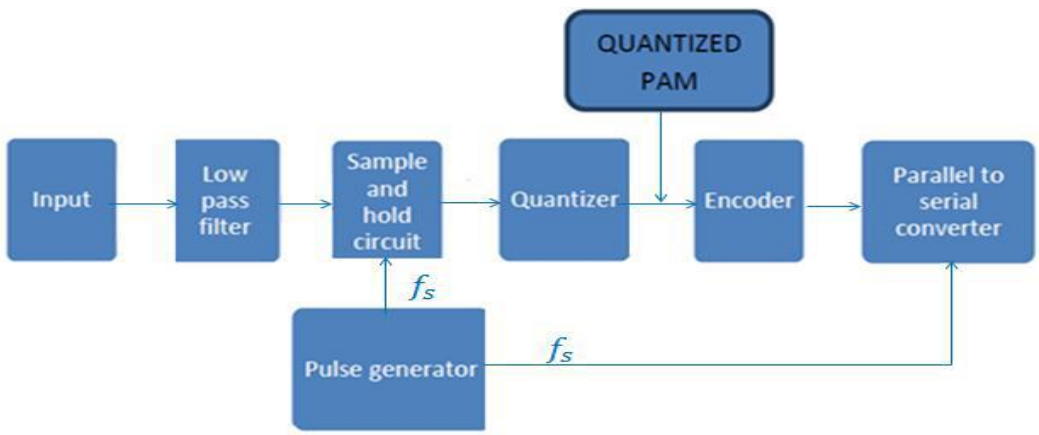
Applications of spread spectrum are:

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

b) Attempt any one: **06M**

1) Draw & explain PCM transmitter Block diagram

Ans: **(03M Transmitter diagram, 03M Explanation)**



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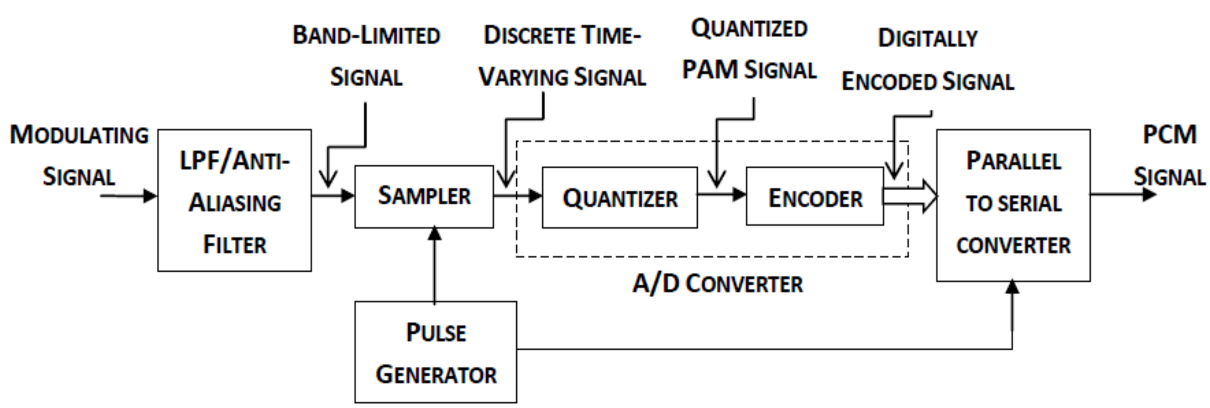


- The analog signal x (t) is passes through band limiting / low pass filter, which has a cut-off frequency fc=W Hz. This will ensure x (t) will not have any frequency component higher than "W". In other words, suppresses high frequency components and passes only low frequency signal to avoid 'aliasing error'.

- The band limited analog signal is then applied to sample and 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 sample and hold block is a flat topped PAM signal.
- These samples are subjected to operation "quantization" in the "quantizer". The quantizer is used to reduce effect of noise. Quantization is a process of approximation of the value of respective sample in to 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.

2) Compare between FDMA, TDMA and CDMA system (any six points)

Ans: (Any Six points) 1M Each

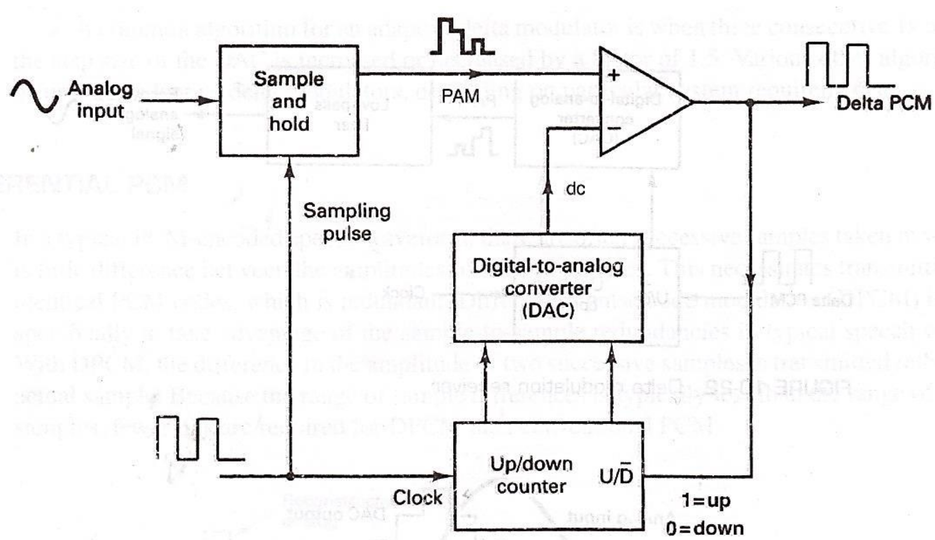
Sr No	**PARAMETER**	**FDMA**	**TDMA**	**CDMA**
1	Full Form	Frequency Division Multiple Access	Time Division Multiple Access	Code Division Multiple Access
2	Definition	Entire band of frequencies is divided into multiple RF channels/carriers. Each carrier is allocated to different users.	Entire bandwidth 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.
3	Bandwidth available	Overall bandwidth is shared among many stations.	Time sharing of satellite transponder takes place	Sharing of bandwidth and time both takes place.
4	Synchronization	Synchronization is not necessary	Synchronization is essential	Synchronization is not necessary
5	Interference	Due to nonlinearity of devices Intermodulation products are generated due to interference between adjacent channels.	Due to incorrect synchronization there can be interference between the adjacent time slots.	Both type of interference will be present.

6	Code word	Code word is not required	Code word is not required	Code words are required.
7	Guard bands	Guard bands between adjacent channels are necessary.	Guard times between adjacent timeslots are necessary.	Guard bands and Guard times both are necessary
8	Active terminals	All terminals active on their specified frequencies	Terminals are active in their specified slot on same frequency	All terminals active on same frequency
9	Signal separation	Filtering in frequency	Synchronization in time	Code separation
10	Near Far Problem	No	No	Yes
11	Handoff	Hard handoff	Hard handoff	Soft handoff
12	Variable Transmission Rate	Difficult	Easy	Easy
13	Application	GSM , PDC(pacific digital cellular), Radio, TV	Advanced mobile phone, system(AMPS), Cordless telephone	IS95 Wide band, CDMA 2000,2.5G and 3G

Q2. Attempt any two:- **16M**

1. **Draw and explain block diagram of delta modulation.**

Ans:- **(04M diagram Tx , 04M explanation Tx)**

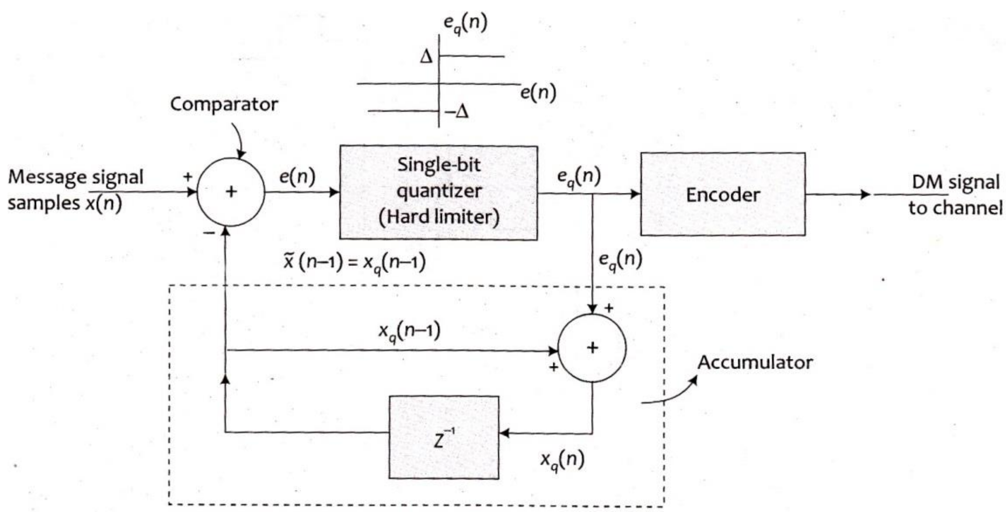


In fig(1) the input analog is sampled and converted to PAM signal, which is compared with the output of DAC.

- The output of DAC is a voltage equal to regenerated magnitude of the previous sample, which was stored in the up-down counter as a binary number.

- The up-down counter is incremented or decremented depending on whether the previous
- sample is larger or smaller than the current sample.
- The up-down counter is clocked at a rate equal to the sample rate. Therefore up-down counter is updated after each comparison.
- Initially, the up-down counter is zeroed, and the DAC is outputting 0V. The first sample is taken, converted to a PAM signal, and compared with zero volts.
- The output of comparator is a logic 1 condition (+V), indicating that the current sample is larger in amplitude than the previous sample.
- On the next clock pulse, the up-down counter is incremented to count of 1. The DAC now outputs a voltage equal to the magnitude of the minimum step size (resolution).
- With the input signal shown, the up-down counter follows the input analog signal up until the output of the DAC exceed the analog sample; then the up-down counter will begin counting down until the output of DAC drops below the sample amplitude.

OR



Explanation:-

DM TRANSMITTER

➢ It consists of a comparator , single but quantizer and an accumulator connected together as shown in in fig 2.12(a)

➢ In delta modulation we compare the present sample x(n) of the message signal x(t) with an approximation to the previous sample and the difference between x(n) and is applied to a single bit quantizer.

➢ If the output of the comparator is positive whatever may be the actual magnitude quantizer gives an output of + and if he comparator output is negative quantizer output will be -.

➢ When = , it is binary 'one' and whenever = it is binary 'zero'.

➢ This sequence of is encoded by the encoder.

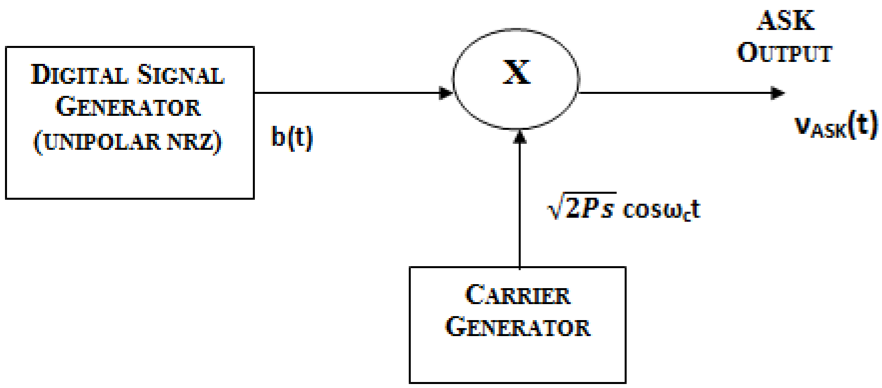
2. **List out different digital modulation technique and explain amplitude shift keying with suitable circuit diagram and wave forms.**

Ans:- **(list 2M, explain ask 2M , Tx diagram 2M, wave form 2M)**

List of Different Digital Modulation technique (ANY 4) :-

(i) Amplitude shift keying -ASK
(ii) Phase shift keying - PSK
(iii) Frequency shift keying - FSK
(iv) Quadrature Phase shift keying - QPSK
(v) Differential Phase shift keying -DPSK
(vi) Quadrature amplitude modulation- QAM

Transmitter Diagram:-



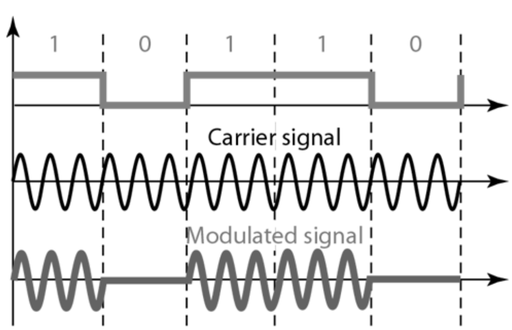
ASK Transmitter

EXPLANATION

ASK MODULATOR:

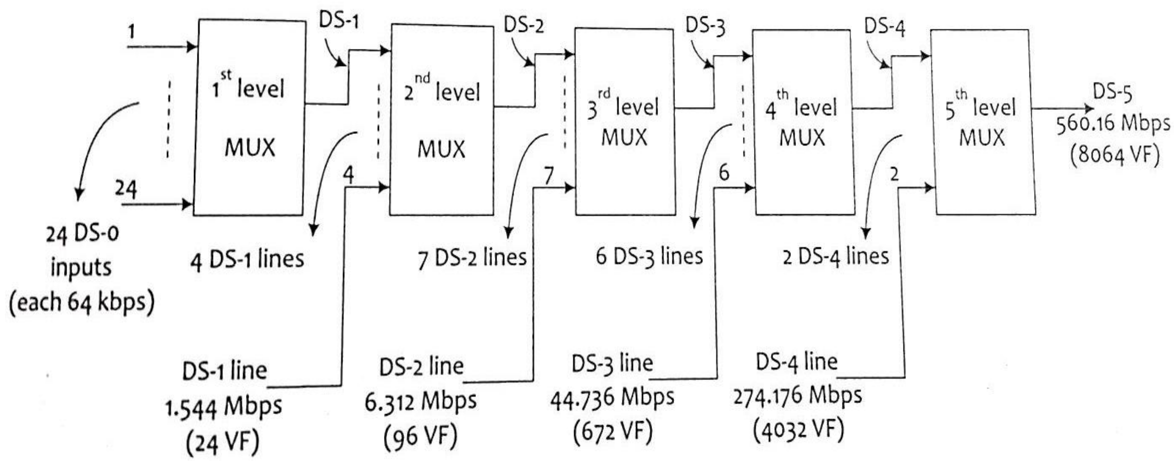
- The ASK technique of binary modulation is illustrated in Figure where modulating signal consists of unipolar pulses.
- Because in this case the carrier is switched ON and OFF, this method is also known as *ON-OFF keying*.
- For the entire time the binary input is high, the output is a constant amplitude, constant frequency signal and for the entire time the binary input is low, the carrier is off.
- Ps is signal power given by $(\text{Amplitude})^2/2$
- ASK is given by:
    - $v_{ASK}(t) = b(t)\cos\omega_c t$

WAVEFORM :-



3. Explain North American digital multiplexing hierarchy with neat diagram.

Ans:- **(diagram 4M, Explanations 4M)**



Explanation:-

The first digital signal in true sense is the PCM voice signal. A PCM voice signal represents 64kbits/sec. i.e. 8000 sample /second* 8 bits per samples. Such a signal is called as digital signal at level zero (DS0). It is also called as T1 signal. Due to 8ooo sample/second, sampling rate, the time duration between adjacent samples will be 125 µsec. But practically DS0 signal is not transmitted because most of the telephone lines are analog. Hence in telephone central office, the subscriber analog line is passed through an anti-aliasing filter. The band limited signal is applied to a codec, which convert it into DS0 signal.24 DS0 lines are multiplexed into a DS1.The telephone companies implement TDM through the hierarchy of digital signals. This is called as digital signal service. Multiplexed signal is converted into a frame at the DS1 or T1 level.

- In this hierarchy the first level of multiplexing involves 24 numbers of 64 kbps PCM-ed voice channels.
- This gives a 1.544 Mbps digital signal. Four such signals are multiplexed in the second –level multiplexing to obtain an 6.312 Mbps digital signal.
- The third involves seven inputs to give a 44.736 Mbps multiplexed signal. Six such signals are multiplexed in the fourth –level multiplexer to obtain a 274.176 Mbps digital signal.
- Again two such signals are multiplexed in the 5^{th} level to get a 560.16 Mbps signal.

Q**3. Attempt any four:** **16M**

1) Compare analog and digital communication (any four points).

Ans: **(Any four points)** **1M Each**

Sr No	Parameters	Analog Communication	Digital communication
1	Nature of signal	The information signal is continuous / analog in nature	The information is in digital form.

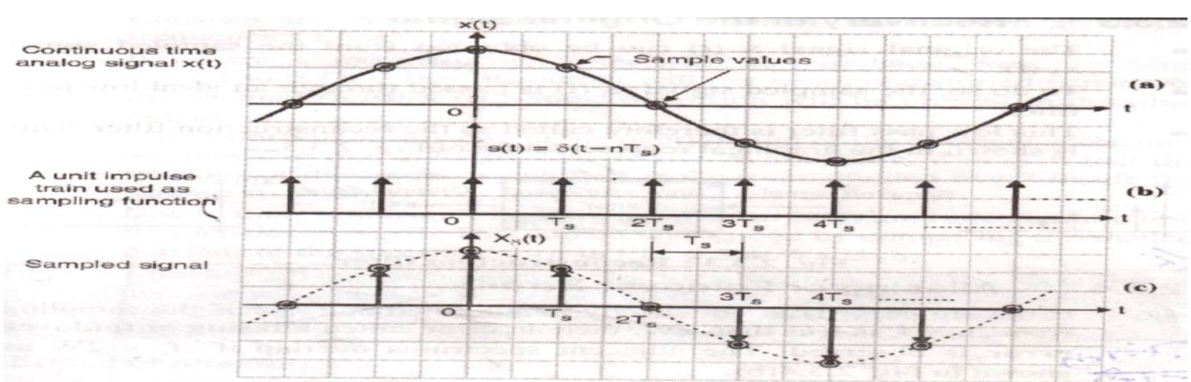
2	Noise immunity	Poor as coding is not possible	Very good due to coding
3	Coding	Not possible	Possible
4	Bandwidth	Requires less bandwidth	More bandwidth
5	Use of repeaters	Not possible	Possible
6	Type of multiplexing	FDM	TDM
7	Complexity	Complex and difficult to built	Simple and less complex
8	Flexibility	Low	High
9	Long distance communication	Restricted	Possible because repeater can be used.
10	coding	Not possible.	Possible
11	Secrecy of communication	Not possible	Possible due to coding and encryption technique
12	Cost	Low	High in the earlier days but now cost has been reduced
13	Storage and retrieval	Not possible	Easily possible to store and retrieve voice, data and video information
14	Example systems	AM, FM, PM, PAM, PWM	PCM, DM, ADM, DPCM

2) State sampling theorem. Describe different types of sampling techniques.

Ans: **(Sampling Theorem 1 Mark, Description 1 Mark – Each Type)**

Statement:

Sampling theorem states that a band-limited signal of finite energy having the highest frequency component fm Hz can be represented and recovered completely from a set of samples taken at a rate of fs samples per second provided that **fs ≥ 2fm**. Here fs is the sampling frequency and fm is the maximum frequency of the continuous time signal x(t). This theorem is also known as the Sampling Theorem for Baseband or Low-pass Signals.

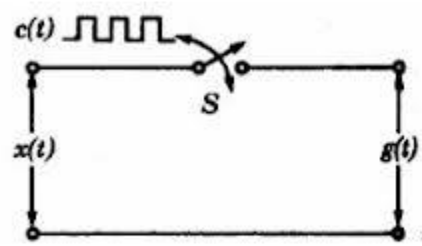


There are basically three types of Sampling techniques, namely:

1. Natural Sampling
2. Flat top Sampling
3. Ideal Sampling

1. Natural Sampling or Chopper Sampling

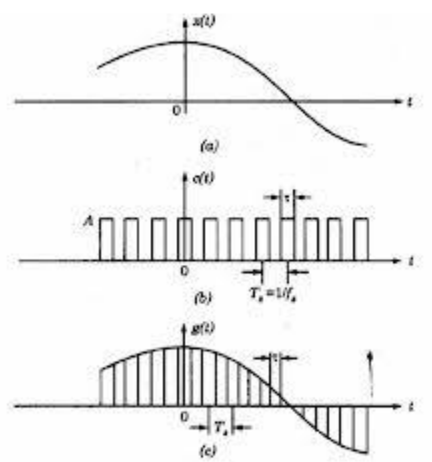
Natural Sampling is a practical method of sampling in which pulse have finite width equal to T. Sampling is done in accordance with the carrier signal which is digital in nature.



With the help of functional diagram of a Natural sampler, a sampled signal g(t) is obtained by multiplication of sampling function c(t) and the input signal x(t).

Spectrum of Natural Sampled Signal is given by:

$$G(f) = A\tau / T_S . [\Sigma \sin c(n f_s . \tau) X(f - n f_s)]$$



Natural Sampled Waveform

2. Flat Top Sampling or Rectangular Pulse Sampling

Flat top sampling is like natural sampling i.e; practical in nature. In comparison to natural sampling flat top sampling can be easily obtained. In this sampling techniques, the top of the samples remains constant and is equal to the instantaneous value of the message signal x(t) at the start of sampling process. Sample and hold circuit is used in this type of sampling.

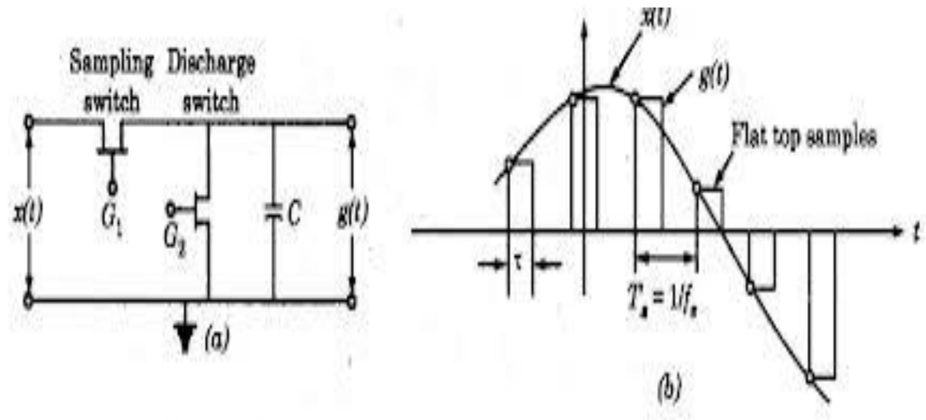


Figure (a), shows functional diagram of a sample hold circuit which is used to generate fat top samples. Figure (b), shows the general waveform of the flat top samples. It can be observed that only starting edge of the pulse represent the instantaneous value of the message signal x(t)

Spectrum of Flat top Sampled Signal is given by: $G(f) = f_s .[\ \Sigma\ X(f\text{-}n\ f_s).\ H(f)]$

3. **Ideal Sampling:**

Ideal Sampling is also known as Instantaneous sampling or Impulse Sampling. Train of impulse is used as a carrier signal for ideal sampling. In this sampling technique the sampling function is a train of impulses and the principle used is known as multiplication principle.

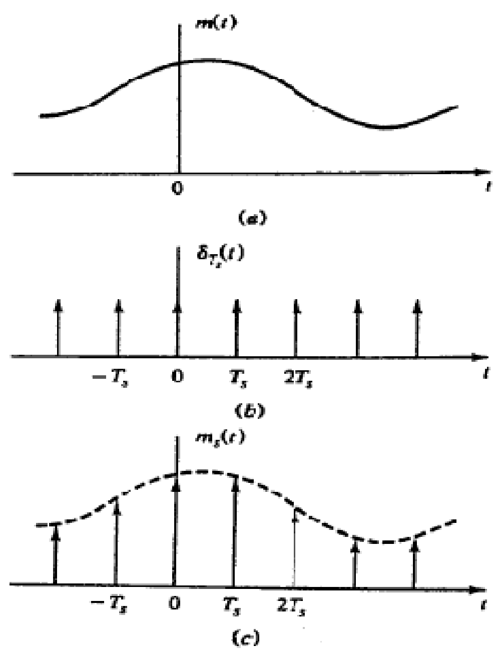


Fig.Ideal signal sampling

Here, Figure (a), represent message signal or input signal or signal to be sampled. Figure (b), represent the sampling function. Figure (c), represent the resultant signal.

Spectrum of Ideal Sampled Signal is given by: $G(f) = f_s .[\ \Sigma\ X(f\text{-}n\ f_s)]$

3) List different types of error and their causes.

Ans: (Types of errors – 2 M, Any two relevant causes of errors – 2 M)

Types of errors:

The errors introduced in the data bits during their transmission can be categorized as,
1. Content errors
2. Flow integrity errors
Depending on the number of bits in the error, the errors can be classified into two types as:
1. Single bit error
2. Burst error

Causes of errors:

Due to addition of noise in transmission & reception of data following errors occur.
1. If data block is lost in the network as it has been delivered to wrong destination.
2. If two or more bits from data unit such as a byte change from 1 to 0 or 0 to 1.

4) Explain M-ary Encoding Technique.

Ans: (Description 2 M; Any relevant Description shall be considered, Example 2 M)

- 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=2n.
- The M-ary signals are obtained as follows.
- Group of "N" bits together form N bit symbols.
- 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 2n =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.

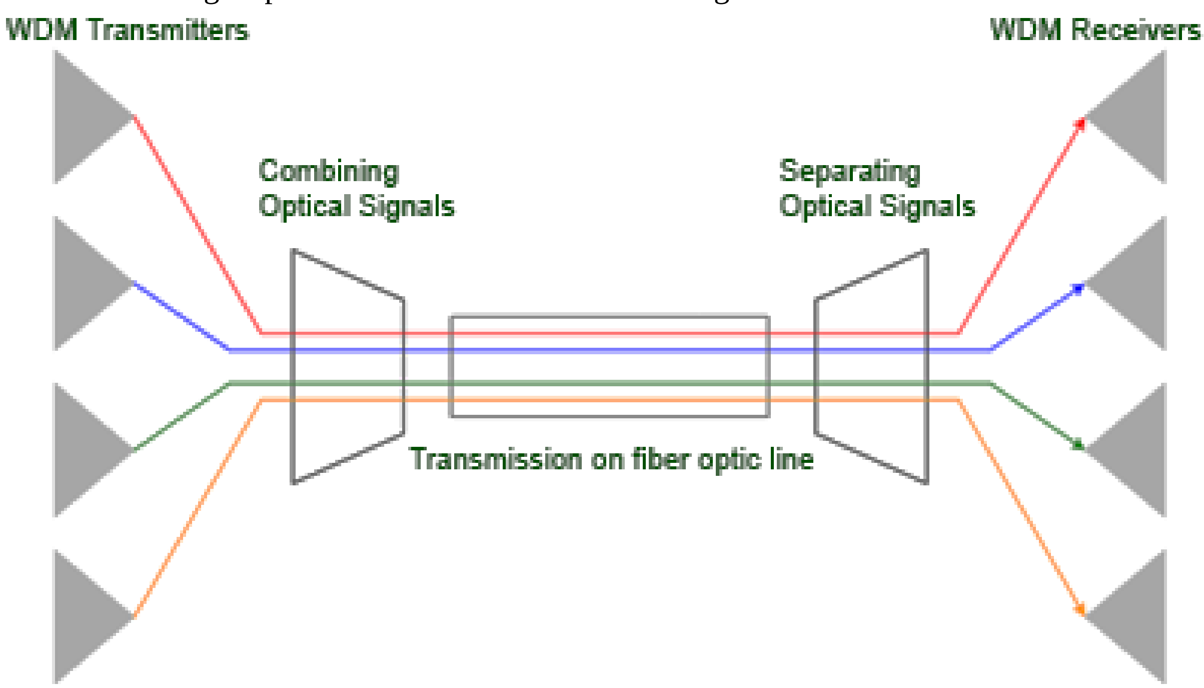
  Example: QPSK

Value of M = 4 also called 4aryPSK.The number of bits in one symbol are = 2.
Hence there are 22 = 4 possible symbols.
Symbol1: 00
Symbol2: 01
Symbol3: 10
Symbol4: 11

5) State WDM techniques and write its two advantages.

Ans. (WDM technique- 2 M, Any 2 Advantages- 2 M; Any relevant diagram shall be considered)

i) WDM is referred to as wave-division multiplexing.
ii) WDM involves the transmission of multiple digital signals using several wavelengths without their interfacing with one another
iii) WDM is a technology that enables many optical signals to be transmitted simultaneously by a single fiber cable.
iv) WDM is accomplished by modulating injection laser diodes that are transmitted highly concentrated light waves at different wavelengths (i.e., at different optical frequencies).
v) WDM is coupling light at two or more discrete wavelengths into and out of an optical fiber Each wavelength is capable of carrying vast amounts of information in either analog or digital form, and the information can already be time or frequency-division multiplexed.
vi) The carrier with WDM is in essence a wavelength rather than a frequency.
vii) The wavelength spectrum used for WDM is in the region of 1300 or 1500 nm.



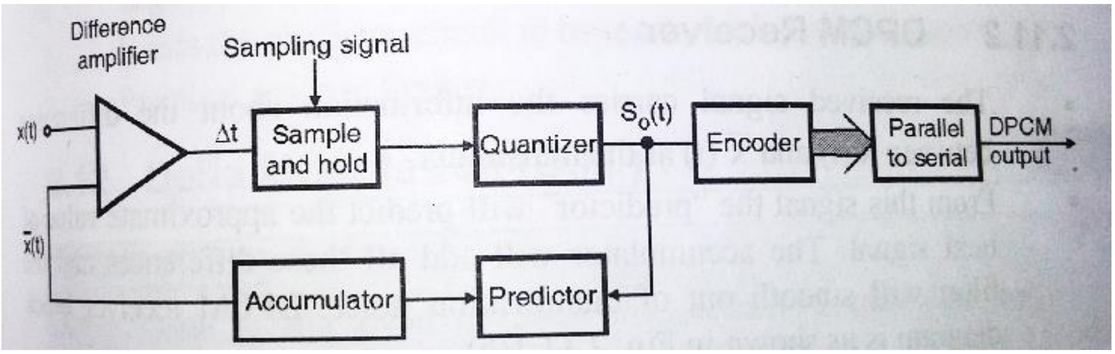
Advantages:

i) A wavelength division multiplexed (WDM) system is one where, typically in order to increase system capacity, multiple optical carriers, operating a different wavelengths, share a common optical fiber.
ii) A WDM system can be simple, operating at two very different wavelengths such as 1310 nm and 1550 nm, or more complex whereby four, eight, or even more optical transmitters operate with very small wavelength separations.
iii) The amount of data being transmitted can be significantly increased without increasing the number of fibers or repeaters. Hence capacity is more.

Q.4. a) Attempt any three: 12M

1) Explain DPCM transmitter with neat block diagram.

Ans: (Block diagram-2M, Explanation-2M)



The above figure shows the block diagram of DPCM transmitter x(t) is the analog input signal and x^(t) is its approximated signal. What is important to know is whether x^(t) is larger or smaller than x(t) and by how much.

At each sampling instant the difference amplifier compares x(t) and x^(t) and the sample and hold circuit will hold the result of this subtraction.

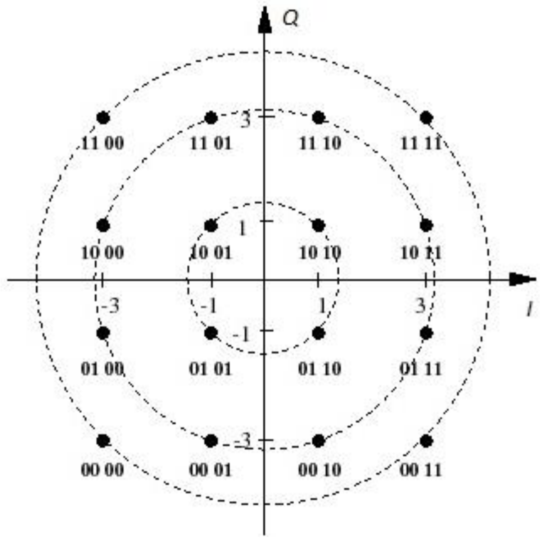
The difference signals at the output of sample and hold circuit is quantized by the quantizer. The quantizer output $S_0(t)$ is the transmitted as it is or it is encoded into a stream of bits as explained in conventional PCM system.

The quantizer output is also used to produce the approximated signal x^(t)by passing the quantizer output through a predictor and accumulator.

2) What is constellation diagram and draw constellation diagram for 16-QAM modulator?

Ans: (Definition-2M, Diagram-2M)

A constellation diagram is a representation of a signal modulated by a digital modulation scheme such as quadrature amplitude modulation or phase-shift keying. It displays the signal as a two-dimensional X-Y plane scatter diagram in the complex plane at symbol sampling instants. It represents the possible symbols that may be selected by a given modulation scheme as points in the complex plane. Measured constellation diagrams can be used to recognize the type of interference and distortion in a signal.



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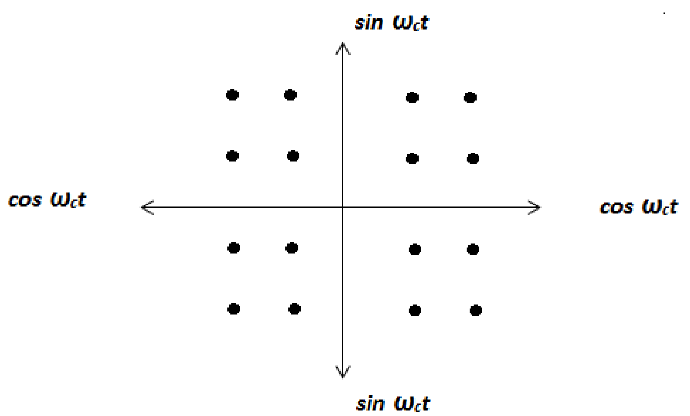


3) Explain CDM technique with its block diagram.

Ans: **(Explanation-2M, Diagram-2M)**

Code division multiplexing (CDM) is a networking technique in which multiple data signals are combined for simultaneous transmission over a common frequency band.

In CDM separation is achieved by assigning each user channel its own code. Guard spaces are realized by using codes with necessary distance in code spaces, orthogonal codes. Above fig shows CDM scheme.Good protection against unauthorized reception is the main advantage of CDM.
When CDM is used to allow multiple users to share a single communications channel, the technology is called code division multiple access (CDMA).

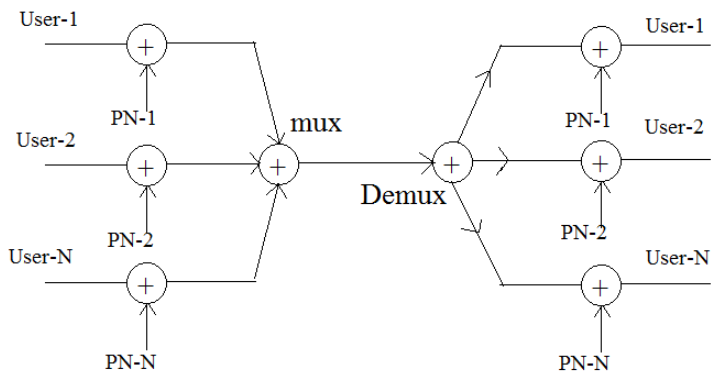


Fig: code division multiplexing for N- users

4) Why pseudo noise sequence is used in spread spectrum modulation?

Ans: **4M**

In spread-spectrum systems, the receiver correlates a locally generated signal with the received signal. Such spread-spectrum systems require a set of one or more "codes" or "sequences" such that

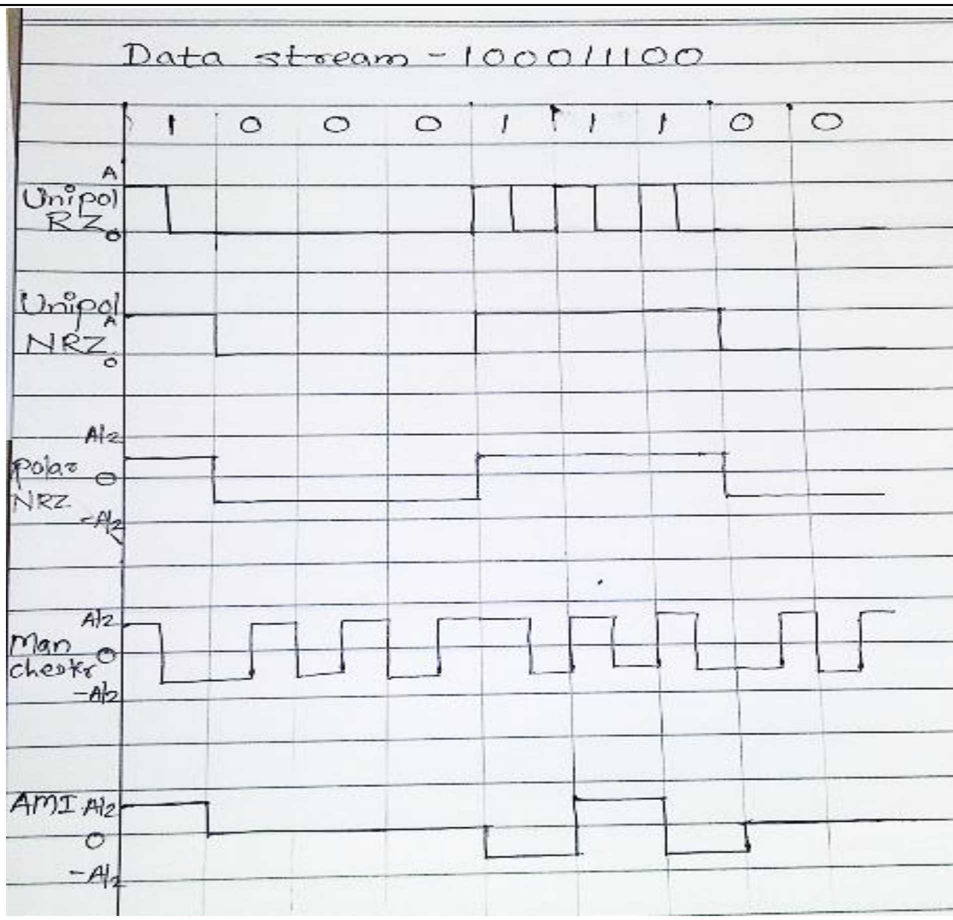
i) Like random noise, the local sequence has a very low correlation with any other sequence in the set, or with the same sequence at a significantly different time offset, or with narrow band interference, or with thermal noise.

ii) Unlike random noise, it must be easy to generate exactly the same sequence at both the transmitter and the receiver, so the receiver's locally generated sequence has a very high correlation with the transmitted sequence.

b) Attempt any one: **06M**

1. Draw unipolar RZ, NRZ, Manchestor and Alternate Mark inversion Line code waveform for data stream 100011100.

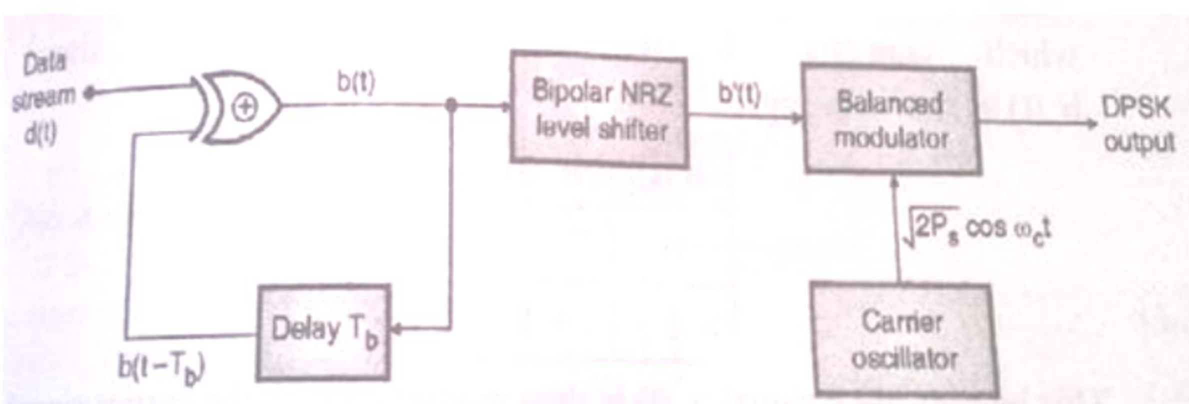
Ans: **1M Each**



2. Draw the block diagram of DPSK transmitter and explain its working.

Ans: **(Diagram-2M, Explanation-2M)**

The DPSK transmitter is as shown –



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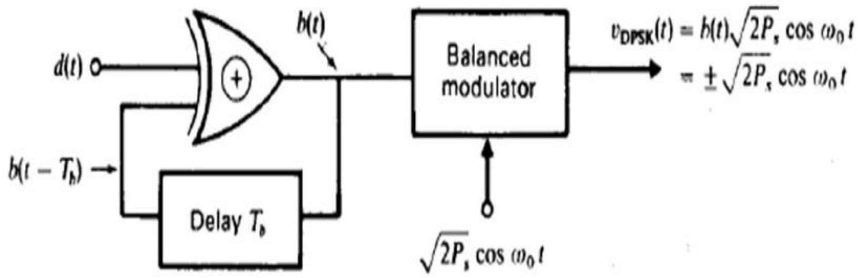


d(t)		b(t − Tb)		b(t)	
logic level	voltage	logic level	voltage	logic level	voltage
0	−1	0	−1	0	−1
0	−1	1	1	1	1
1	1	0	−1	1	1
1	1	1	1	0	−1

Explanation

1) d (t) represents the data stream which is to be transmitted it is to one input of an EX-OR logic gate.
2) The EX-OR gate output 'b (t)' is delayed by one bit period the applied to the other input of EX-OR gate. The delayed represented by 'b (t-Tb)'.
3) Depending on the values of 'd (t)' and 'b (t-Tb)' the EX-OR produces the output sequence 'b (t)'.the waveform for the generator .the waveform drawn by arbitrarily assuming that in the first interval b (0) = 0.
4) Output of EX-OR gate is the applied to a bipolar NRZ level which converts 'b(t)' to a bipolar level "b(t) as shown

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

5) 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.

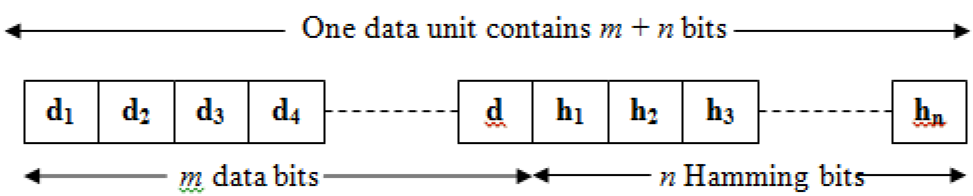
Q5. Attempt any two:- **16M**

1. **Explain hamming code with suitable example. Find the hamming weight of the code vector X= 11010100**

Ans:- **(Explain 2M, example 4M , finding code 2M)**

HAMMING CODE:

- The *Hamming code* is an error-correcting code used for correcting single-bit errors. It cannot correct multiple-bit errors or burst errors and it cannot identify errors that occur in the Hamming bits themselves.
- Hamming bits are inserted into the message at random locations. The combination of the data bits and the Hamming bits is called the *Hamming code*. Both the sender and receiver must know where the Hamming bits are placed.
- To calculate the number of redundant Hamming bits necessary for a given data length, a relationship between the data bits and the Hamming bits must be established. As shown in Figure a data unit contains *m* character bits and *n* Hamming bits. Therefore, the total number of bits in one data unit is $m + n$.



- Since n bits can produce 2^n different codes, 2^n must be equal to or greater than $m + n + 1$. Therefore, the number of Hamming bits is determined by the following expression:

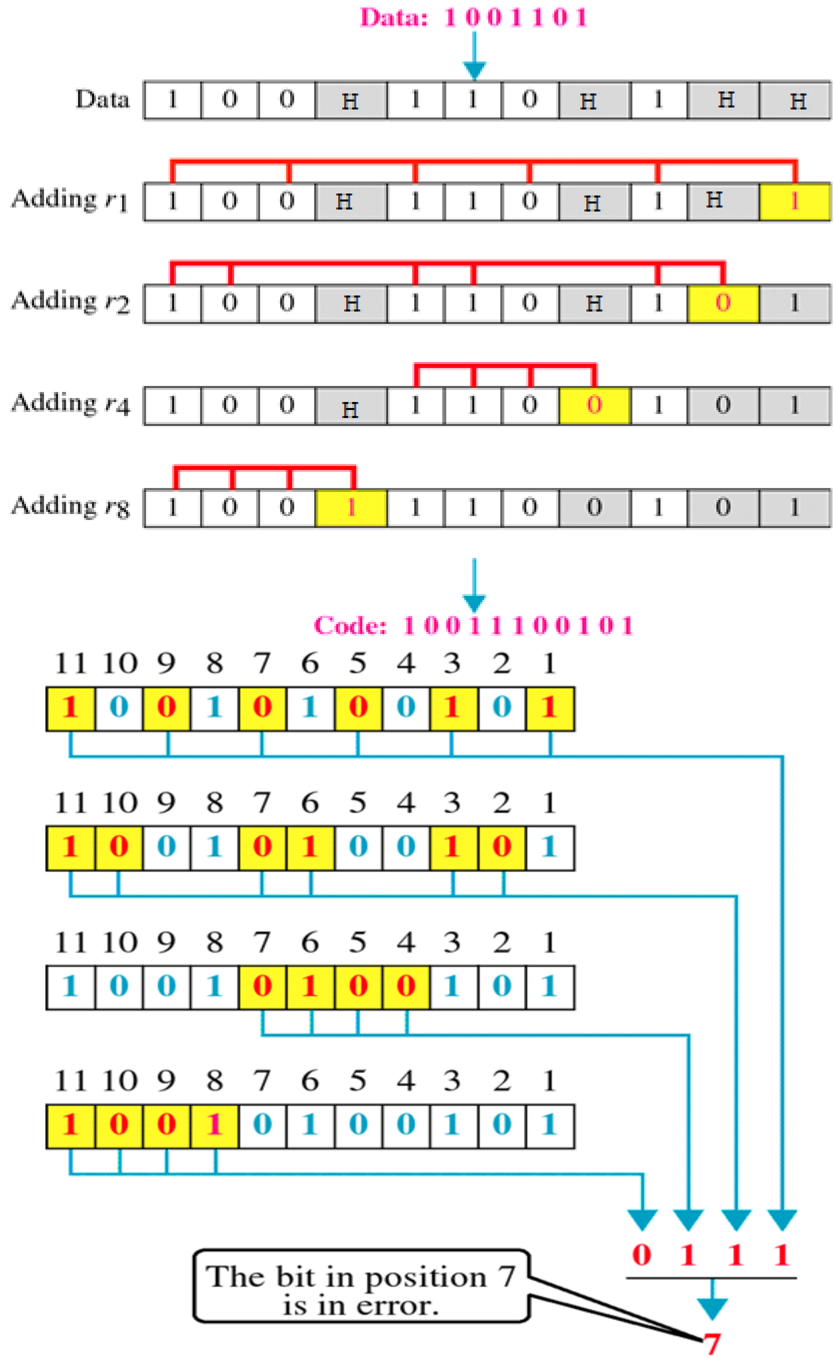
$$2^n \geq m + n + 1$$

where, n = number of Hamming bits

m = number of bits in each data character

- Hamming bits can be placed at the beginning of the character bits, at the end of the character bits or interspersed (inserted randomly) throughout the character bits.

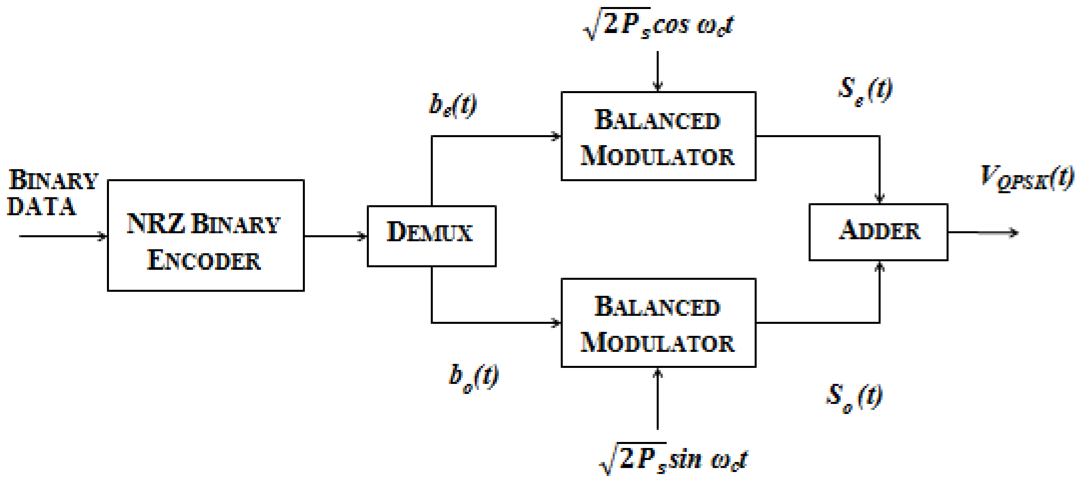
Note:- Any relevant example can be considered



- **The hamming weight of the code vector x= 11010100 is 4**

2. Draw QPSK modulator block diagram. Explain with constellation diagram and phasor diagram.

Ans:- **(diagram 2M, explain 2M, constellation 2M, phasor 2M)**



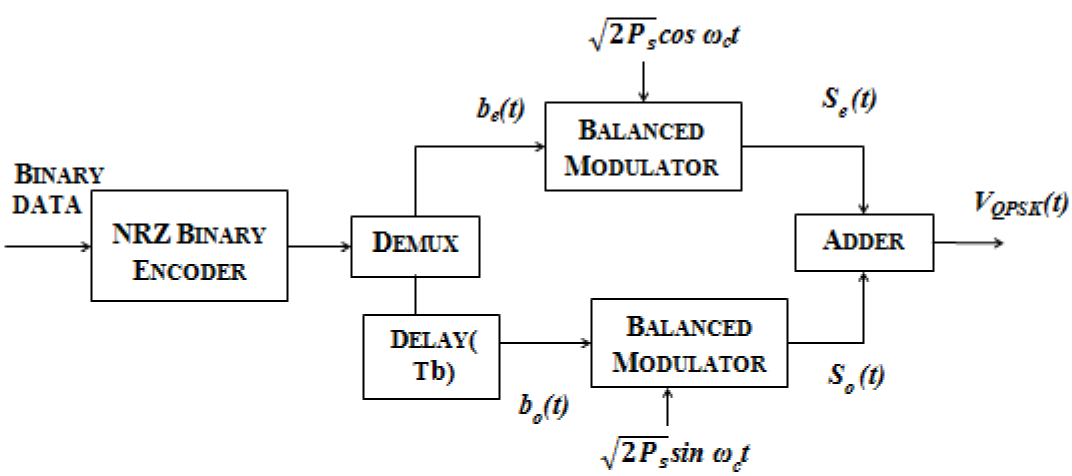
QPSK Transmitter (non - offset)

Operation:

- 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 twoseparate bit streams $b_o(t)$ and $b_e(t)$. The bit stream $b_e(t)$ consists of only the even numbered bits 2, 4, 6, 8, ... whereas $b_o(t)$ bit stream consists of only the odd numbered bits i.e., 1, 3, 5, ... as shown in Figure 3.18.
- Each bit in the even and odd stream will be held for a period of $2T_b$. This duration is called as symbol duration T_s. Thus, every symbol contains two bits.
- The bit stream $b_e(t)$ is superimposed on a carrier $\cos\omega_c t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sin\omega_c t$ 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,

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

OR



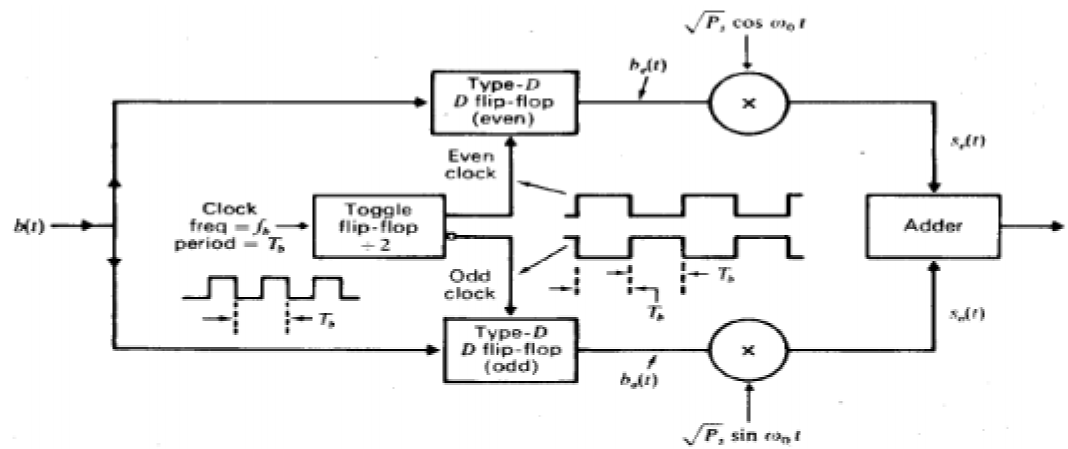
Offset QPSK Transmitter

OFFSET QPSK:

The block diagram of offset QPSK transmitter is shown in Figure

- In QPSK the transition for odd and even bit streams occur simultaneously. This leads to sudden phase changes of the carrier 90^0 or 180^0 depending upon whether the sign change occurs for only for one of the two components or both.
- Such sudden phase changes in the carrier can result in reduction of the amplitude of the QPSK signal when it is filtered. So if the QPSK signal during the course of transition is passed through a filter before the signal is detected, the resulting amplitude reduction of the signal can lead to errors in detection. Carrier phase changes by 180 in particular, cause considerable reduction in the envelope amplitude and so are to be avoided.
- As shown in the Figure 3.18, the even bit will occur before the 1st odd bit. Therefore, the odd bit stream $b_o(t)$ will start with a delay of one bit period T_b. This delay is called as Offset and therefore, this system is called as Offset QPSK. Because of this offset, the bit streams $b_o(t)$ and $b_e(t)$ cannot change their levels at the same instant of time. Only one component can make transition at one time, changes are limited to 0^0,90^0.Hence 180^0 phase shift is been avoided
- Here, the odd bit stream $b_o(t)$ has been delayed by one bit period T_b by adding a delay block as shown in Figure 3.
  As a result of this additional time delay, two bits which occur in time sequence (serially) in the input bit stream $b(t)$, will appear at the same time (in parallel). Therefore, $b_o(t)$ and $b_e(t)$ can change at different time after each time $2T_b$and there can be a phase change in the output signal.

OR

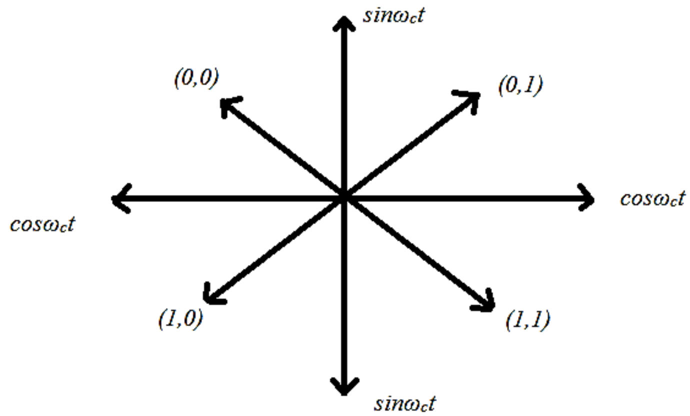


- QPSK allows the bits to be transmitted using half the bandwidth that is required in BPSK. With QPSK, the modulated output signal is shifted by four phases in accordance with the input binary data as when two bits are combined four symbols are obtained. Thus, the QPSK method requires two input bits for each phase shift.
- QPSK modulation exhibits better spectral efficiency but at the expense of more complex circuitry and more critical performance requirements.
- Mathematical representation of QPSK signal is given by,

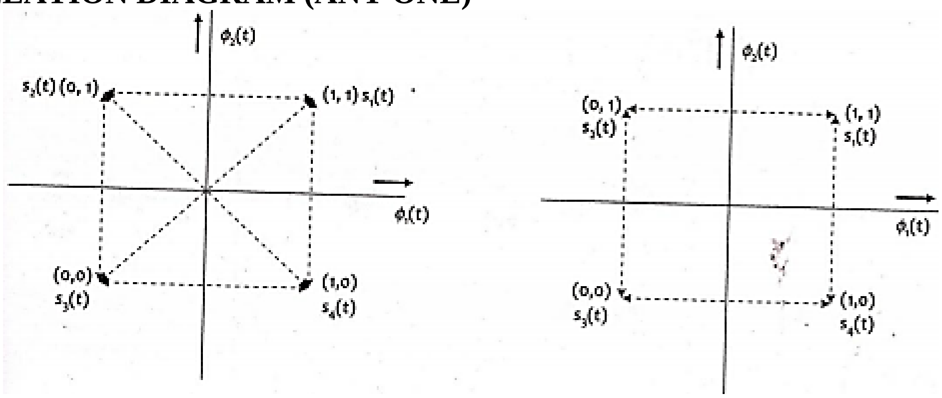
$$V_{QPSK}(t) \cos [\, \omega_c t + (2i - 1)\pi/4 \,]$$

Where, $i = 1, 2, 3, 4$

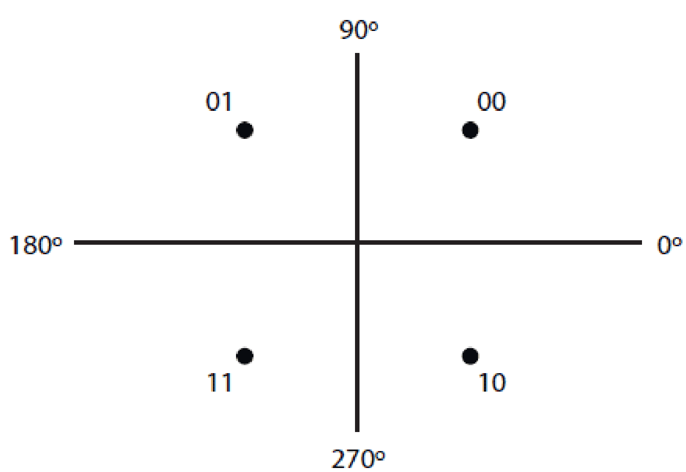
PHASOR DIAGRAM:-



CONSTELLATION DIAGRAM (ANY ONE)



OR



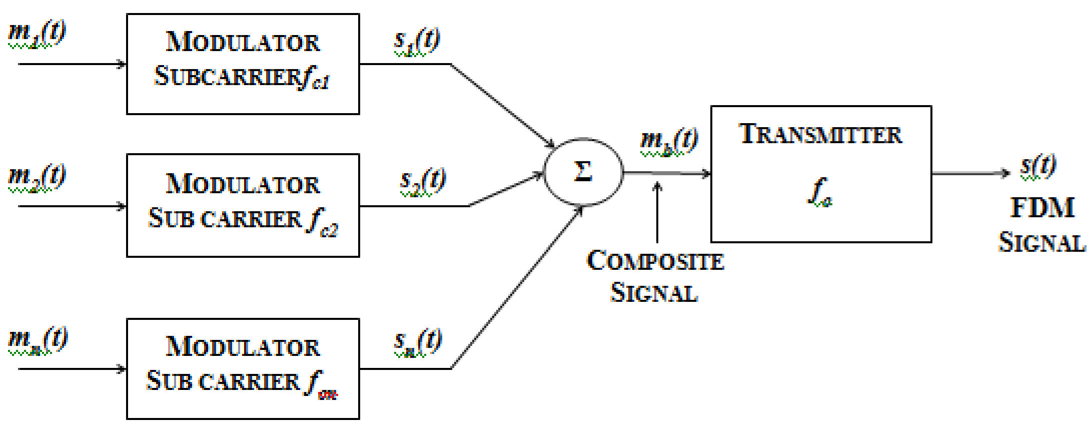
3. Define FDM and explain frequency division multiplexing with block diagram.

Ans:- **(define 2M , explain FDM 2M, diagram 4M)**

Frequency Division Multiplexing:

FDM is an analog multiplexing technique in which many signals are sent simultaneously at the same time but are separated from each other in the frequency domain.

Diagram:-



FDM Transmitter

Explanation:

- 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.

Q.6 Attempt any four: **16M**

1) State Shannon's Hartley theorem. What is Shannon's information rate theoretically?

Ans: **4M**

Shannon's Hartley theorem:

The channel capacity of a white, band limited Gaussian channel is given by,

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

Where, B = Channel Bandwidth S = Signal Power N = Noise within the channel bandwidth

According to Shannon's Hartley theorem, the maximum rate of transmission is equal to the channel capacity (2M)

$$R_{max} = C_{\infty} = 1.44(S/No)$$

This is Shannon's information rate theoretically.

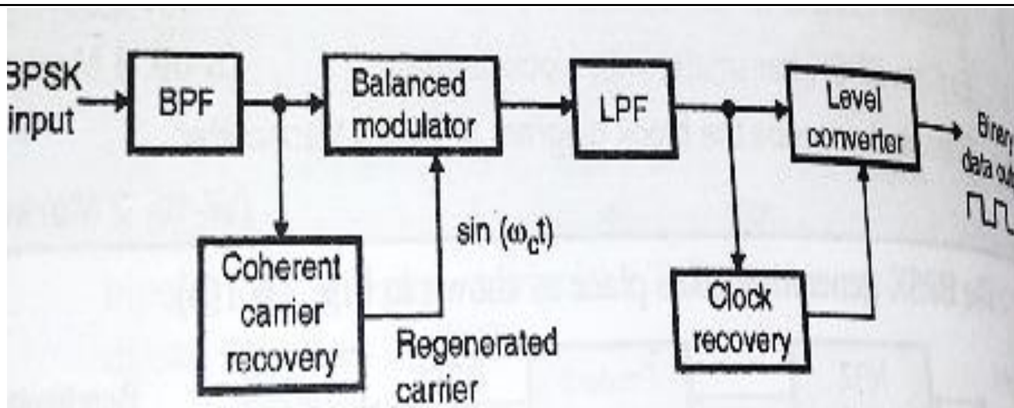
2) Compare between QAM and QPSK (any four point).

Ans: (any four) **01M each**

Sr No	Parameter	QASK / QAM	QPSK
1	Type of modulation	Quadrature amplitude and phase modulation	Quadrature phase modulation
2	Location of signal points	Equally spaced and placed symmetrically about origin.	On the circumference of a circle
3	Noise immunity	better than QPSK	Comparatively less than QASK
4	System complexity	More complex than QPSK	Less complex than QASK
5	Probability of error	less than QPSK	More than QASK
6	Performance of the system	Better than QPSK	Less than QASK

3) Draw and explain PSK receiver block diagram.

Ans: **(Diagram-2M, Explanation-2M)**



Operation:

i) The coherent carrier recovery circuit detects and regenerates a carrier signal sin ω_ct. This regenerated carrier has the same frequency and phase as the carrier used at the transmitter.

ii) So the regenerated carrier is known as coherent carrier.

iii) The filtered BPSK signal along with the regenerated carrier is applied to a balanced modulator which acts as a product detector.

$$\therefore \quad \text{B.M. output} = \text{BPSK} \times \text{Regenerated carrier}$$
$$= (\pm \sin \omega_c t \times \sin \omega_c t = \pm \sin^2 \omega_c t)$$
$$\text{But} \quad \sin^2 \theta = \frac{1}{2} - \frac{1}{2} \cos 2\theta$$
$$\therefore \quad \text{B. M. output} = \pm \frac{1}{2} \mp \frac{1}{2} \cos 2\omega_c t$$

iv) The BM output consists of a dc term and a term having frequency twice the carrier frequency.

v) The BM output is passed through LPF which allows only the second term to pass through.

$$\therefore \quad \text{LPF output} = \mp \frac{1}{2} \cos 2\omega_c t$$

vi) The LPF is applied to the level detector and clock recovery circuit at the output of level detector we get the following output.

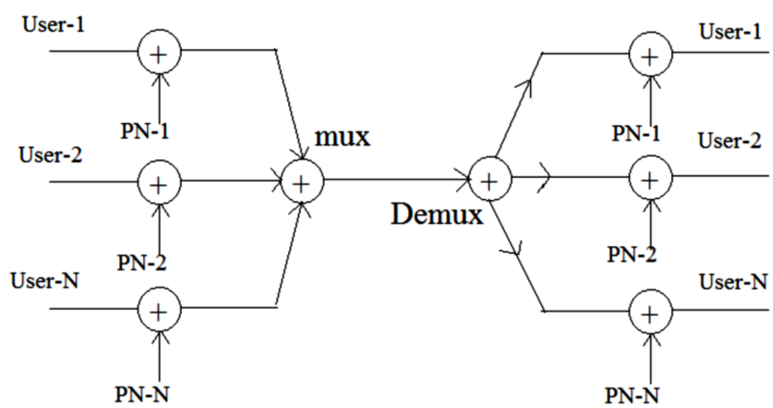
$$-\frac{1}{2}\cos\omega_c t \rightarrow \frac{1}{2} V\text{ (logic 1)}$$

$$+\frac{1}{2}\cos\omega_c t \rightarrow -\frac{1}{2} V\text{ (logic 0)}$$

vii) Thus the binary signal is obtained at the output.

4) Explain basic principle involved in CDMA technology.

Ans: **04M**



Description:

1. 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.
2. 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.
3. All the users in CDMA use same carrier and may transmit simultaneously. Each user has its own pseudo random 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.

5) List out advantages and disadvantages of FHSS system.

Ans: **(Advatages-2M, Disadvatages-2M)**

Advantages of FHSS system:

1) The serial search system with FH-SS needs shorter time for acquisition.
2) The synchronization is not greatly dependent on the distance.

3) The Processing gain is higher than that of DS-SS system

Disadvantages of FHSS system

1) Complex and costly digital frequency synthesizers are required to be used.
2) Bandwidth of FHSS system is too large (in GHz)