WINTER– 17 EXAMINATION

Subject Name: Digital Communication System **Model Answer** Subject Code: 17535

Important Instructions to examiners:

1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given more Importance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in the figure. The figures drawn by candidate and model answer may vary. The examiner may give credit for any equivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constant values may vary and there may be some difference in the candidate's answers and model answer.
6) In case of some questions credit may be given by judgement on part of examiner of relevant answer based on candidate's understanding.
7) For programming language papers, credit may be given to any other program based on equivalent concept.

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

(a) Define entropy and state its unit.

Ans:- (Definition-2 mks,1 mk formula, unit- 1 mks)

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

Mathematically

$$H = \sum_{i=1}^{M} p_i \log_2 \frac{1}{p_i}$$

Where H is entropy , pi is the probabilities of the occurrence of the ith symbol

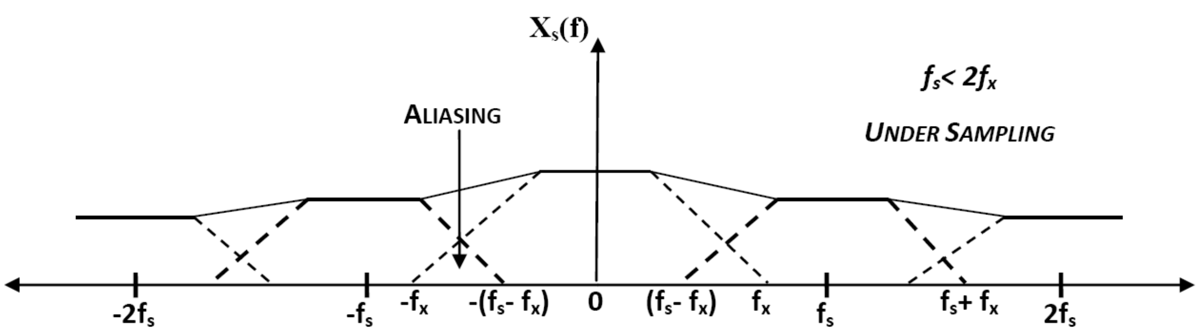
(b) State sampling theorem & explain aliasing effect with neat diagram.

Ans.**(Theorem – 1 Mark, Aliasing Effect Diagram –1 1/ 2 Marks, Explanation – 1 1/2Mark)**

Sampling theorem states that a band-limited signal of finite energy having the highest frequency component f_mHz can be represented and recovered completely from a set of samples taken at a rate of $f_s$samples per second provided that $\boldsymbol{f_s \geq 2f_m}$.

Where, fs = sampling frequency
fm = maximum frequency of continuous original signal

Aliasing Effect
If the sampling rate $f_s < 2f_x$ *(Under Sampling)*, then the sidebands of the signal overlap and information signal *x(t)* cannot be recovered without distortion from sampled signal, $X_s(f)$. This distortion is referred to as ***Aliasing or Fold-over distortion***. Here the sideband frequency from one harmonic will fold-over or overlap with the sideband frequency of another harmonic as shown in



(c) Define multiplexing & describe it's need in communication.

Ans:- (Definition- 1mks, need three points – 3 mks)

Definition-

Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel.
Due to multiplexing it is possible to increase the number of communication channels so that more information can be transmitted.
The typical applications of multiplexing are in telemetry and telephony or in the satellite communication.

Need of multiplexing:

- In telephone systems, there are large numbers of users involved.
- It is not possible to connect separate wires from each subscriber to all other subscribers.
- It is very expensive and increases complexity.
- Instead we can use a communication medium such as a coaxial cable or optical fiber cable to carry many telephone signals from different sources together.
- This can be achieved by 'Multiplexing'.

(d) List the advantages of SS modulation over the fixed frequency modulation.

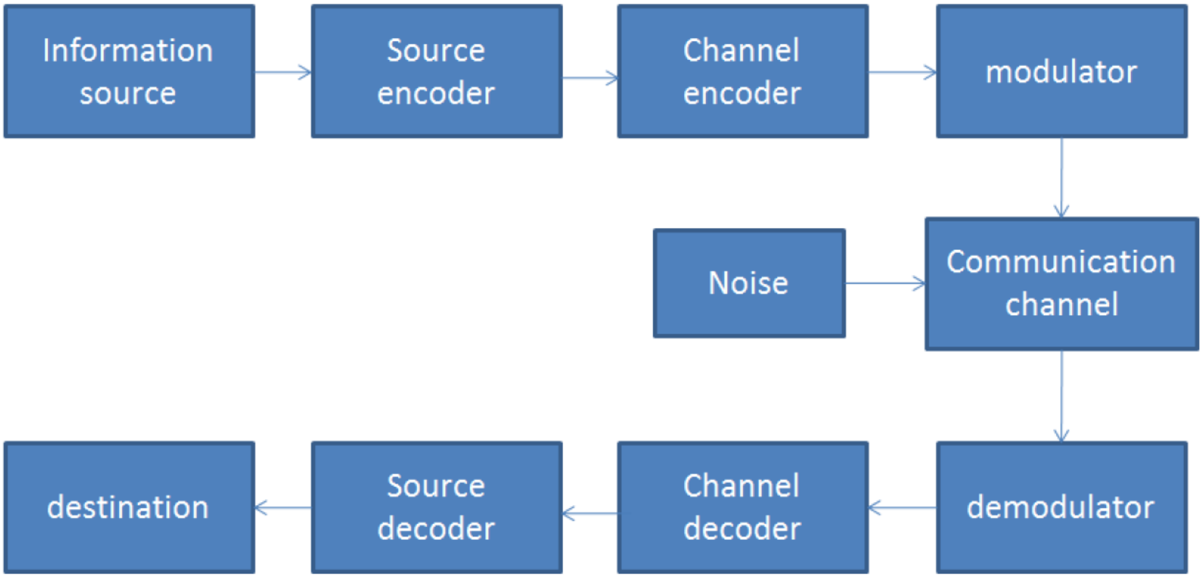
Ans:- (Any 4 advantages-4 mks)

1. Unauthorized listening is prevented.
2. SS signals are highly resistant to the jamming.
3. Unintentional interference occupying the same band is greatly minimized and in most cases virtually eliminated.
4. Many users can share a signal band with no interference.
5. With SS, more signals can use a band than with any other type of modulation and multiplexing.
6. Resistant to fading.
7. The pseudorandom code makes it possible to accurately determine the start and end of a transmission.
8. Superior method for radar.

(B) Attempt any <u>ONE</u> of the following: Marks 6

(a) Draw the block diagram of digital communication system & explain it in detail.

Ans: **(Diagram – 3 marks, explanation – 3 marks)**



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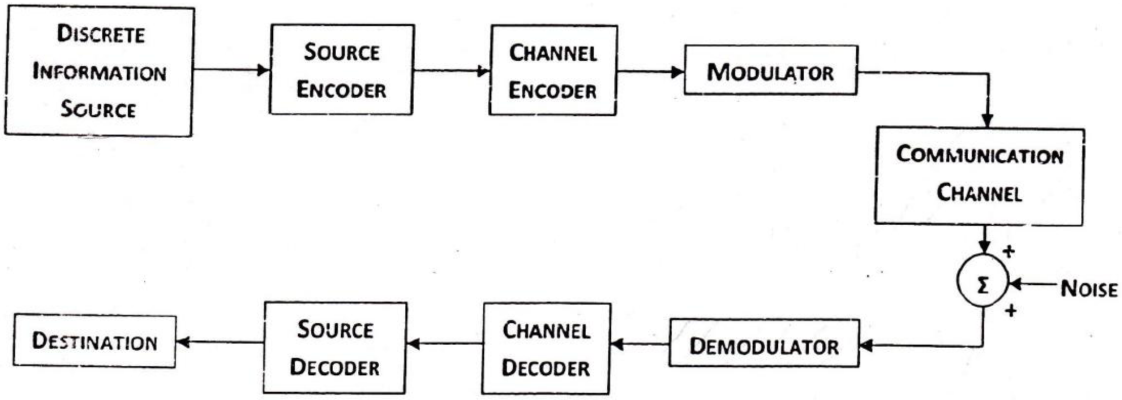


Block diagram of a digital communication system

Explanation : **2M**

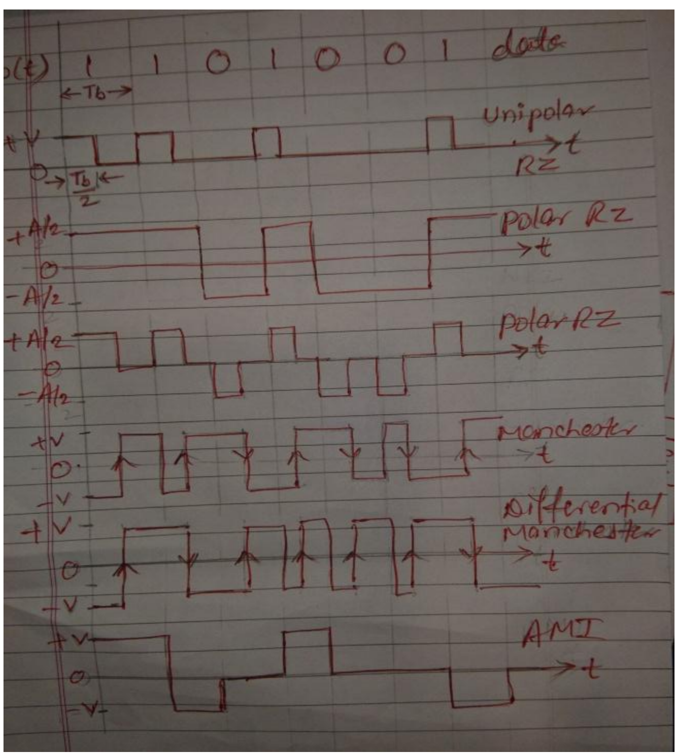
- The source of information is assumed to be digital. If it is analog thin it must be converted first to digital.
- Source coding :
  - Source coding consists of source encoder and source decoder.
  - The encoder generates the digital signal generated at source output into another signal in digital form.
  - 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.

(b) Draw unipolar RZ, Polar NRZ, polar Rz, Manchester, differential Manchester and AMT waveform of line codes for data stream : 1101001.

Ans:-(Each correct line code -1 mks)



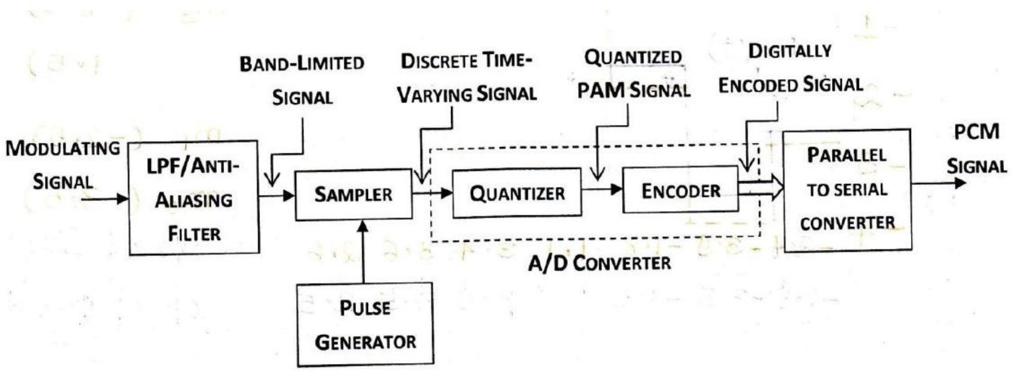
2. Attempt any Two of the following: Marks 16

(a) Draw the block schematic of PCM transmitter. Explain the same with waveform.

Ans:- (Block diagram- 4 mks, waveforms- 2 mks,explanation- 2 mks)

Ans:- **(Transmitter diagram 1 ½ M, Explanation 1 ½ M, Receiver Diagram 1 ½ M, Explanation 1 ½ Waveform 02m)**

Transmitter :-

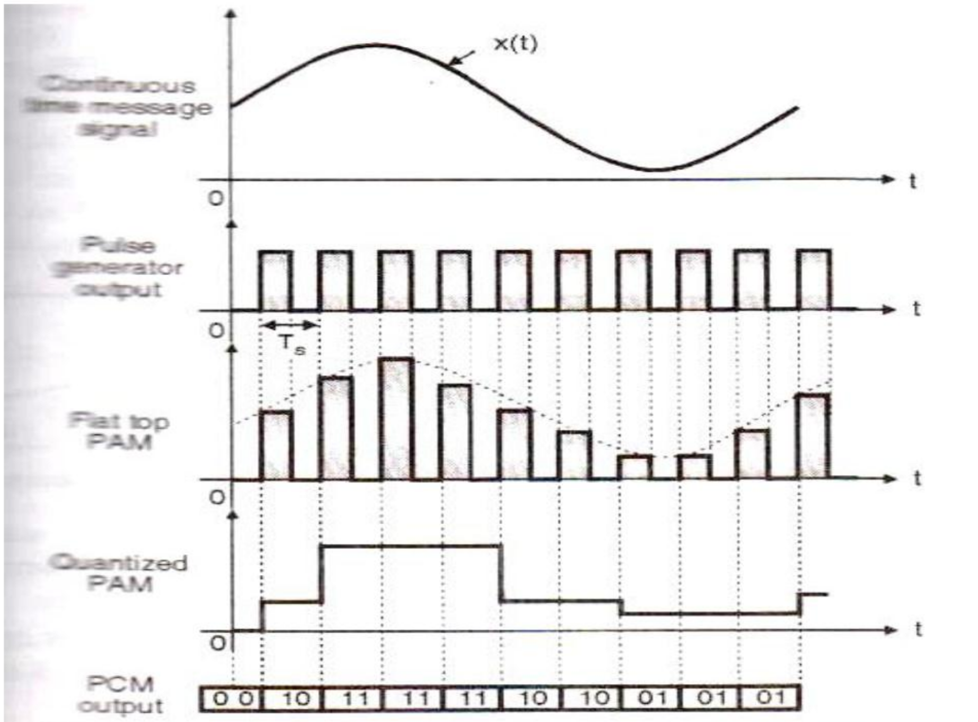


PCM Transmitter

The analog signal X(t) is passed through a band limiting low pass filter, which has a cut off freq. F_C = W Hz. This will ensure that X(t) will not have any freq. component higher than W. This will Eliminate the possibility of aliasing.

- The band limited analog signal is then applied to a sample & hold circuit where it is sampled at adequately high sampling rate.
- These samples are then subjected to the operation called "Quantization" in the " Quantizer". Quantization process is the process of approximation. The quantization is used to reduce the effect of noise.
- The quantized Pam pulses are applied to an encoder which basically an A to D convertor. Each quantized level is converted into an N bit digital word by the A to D convertor.
- The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses.

Waveform :-



Waveforms at different points in PCM transmitter

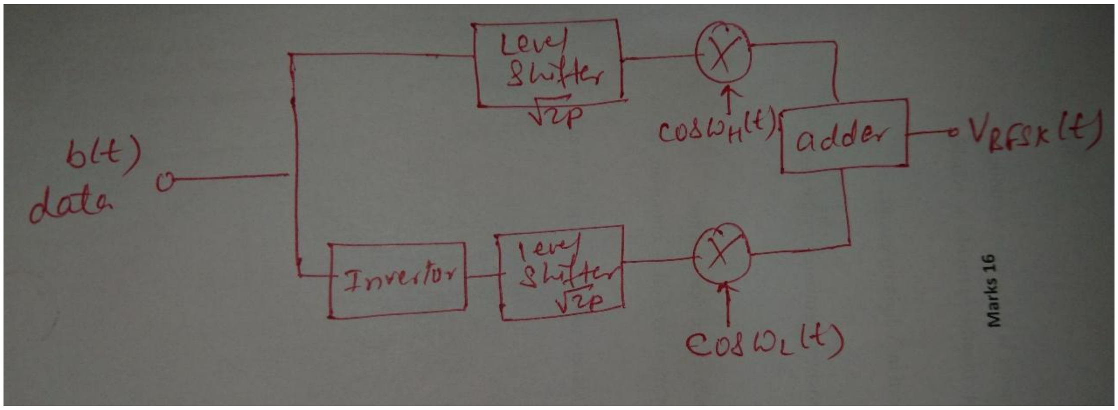
(b) List the different types of digital modulation techniques and explain FSKmodulation in detail.

Ans :- (Different techniques - 4 mks, FSK transmitter- 2 mks, description or waveforms- 2 mks)

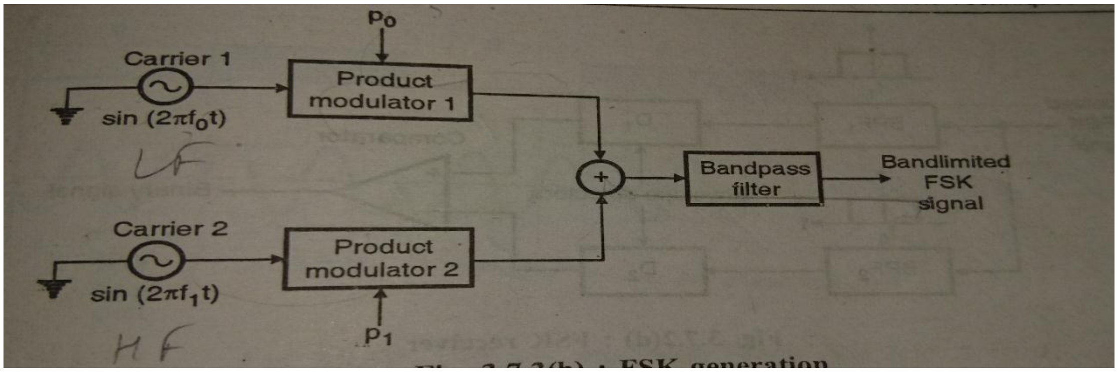
List of Different Digital Modulation technique (ANY 4) :-

(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

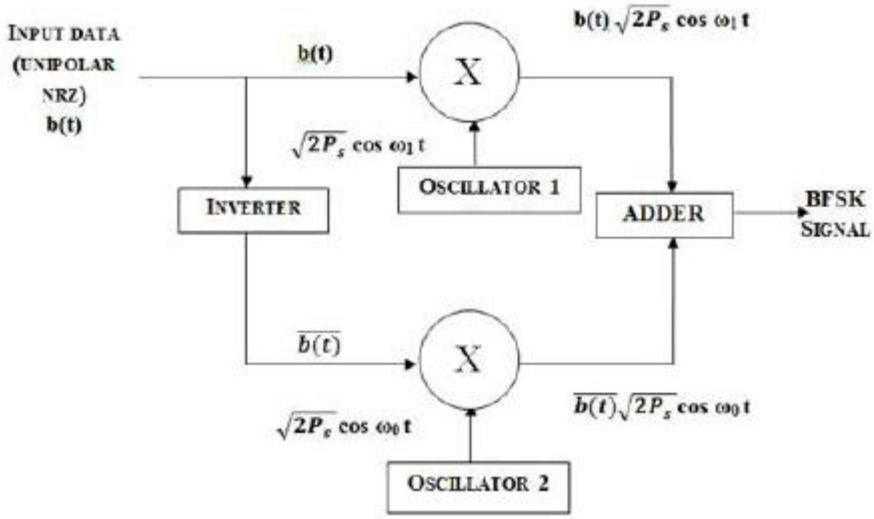
FSK TRANSMITTER-


Level Shifter $\sqrt{2P}$
$\cos\omega_H(t)$
adder
$V_{BFSK}(t)$
b(t) data
Invertor
level Shifter $\sqrt{2P}$
$\cos\omega_L(t)$
Marks 16

OR


P_0
Carrier 1
sin ($2\pi f_0 t$)
LF
Product modulator 1
Bandpass filter
Bandlimited FSK signal
Carrier 2
sin ($2\pi f_1 t$)
HF
Product modulator 2
P_1

OR



BFSK generator

- 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.

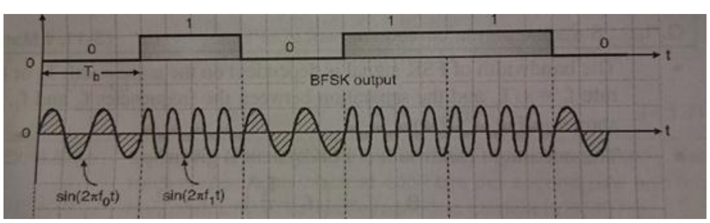
Mathematical equation:

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

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

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

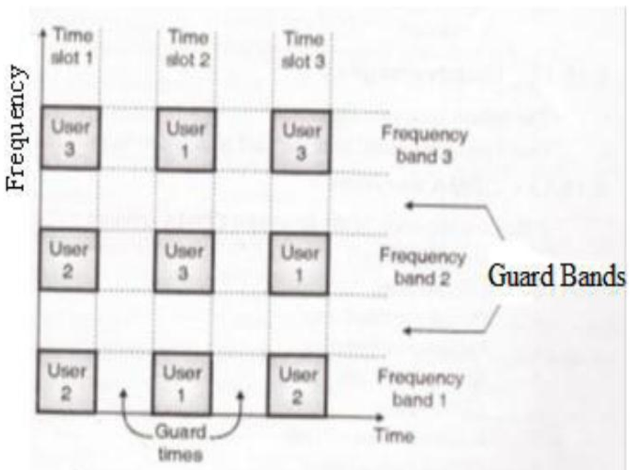
Waveforms-



(c) Describe the basic principle involved in CDMA technology with neat sketch. State it's any four advantages.

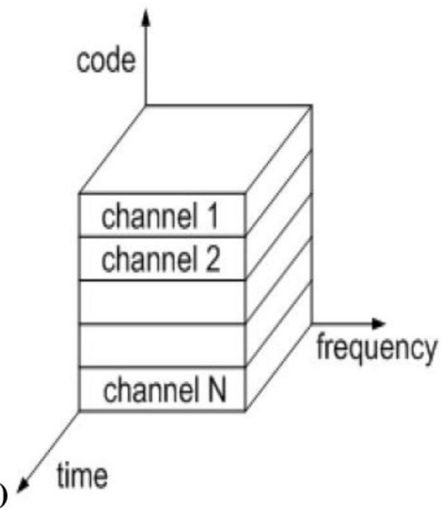
Ans:- (Concept-2 mks , diagram- 2mks, any four advantages- 4 mks)

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



Structure of CDMA

(OR)



Advantages – any 4

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

3. Attempt any <u>FOUR</u> of the following: Marks 16

(a) State limitations of DM. Explain how they overcome in ADM.

Ans:- (Limitations- 2 mks, elimination method explanation-2 mks)

The delta modulation has two major drawbacks as under:

(i) Slope overload distortion

(ii) Granular or Idle Noise:

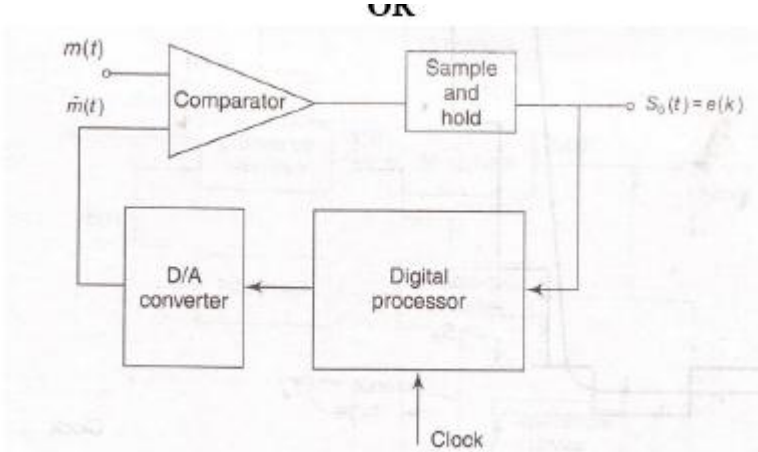
Overcoming of DM drawback by ADM-

In delta modulation, the step size is constant so that its slope overload distortion and granular noise both cannot be controlled. These drawbacks can be controlled by using adaptive delta modulation wherein the step size is variable.

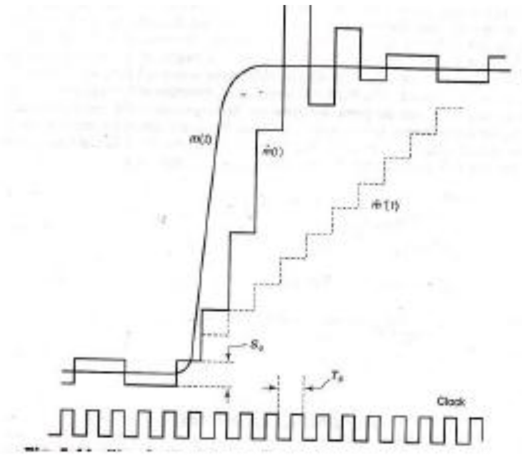
Thus with adaptive delta modulation the following are the advantages-

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

ADM Transmitter



Waveforms-



(b) Compare digital pulse modulation with analog pulse modulation.(4 points)

Ans:- (Any 4 relevant comparison – 4 mks)

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

(c) Give the advantages of TDMA over FDMA. (any 4)

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

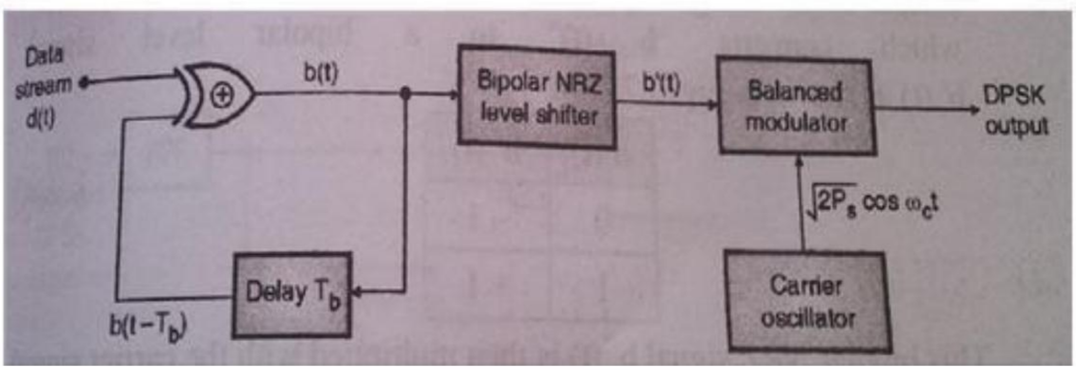
- In TDMA since only one station is present at any given time.
- The entire channel bandwidth can be allotted to a single channel at given instant of time. This is the advantages for digital channel which demands larger bandwidth
- The frequency, selective fading does not affect the TDMA to the extent 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.

(d) Draw the block diagram of DPSK transmitter and state the function of each block.

Ans: **(Block diagram- 2 marks, explanation-2 marks)**

In BPSK receiver, the carrier recovery is done by squaring the received signal.

Hence, when the received signal is generated by negative data bit, it is squared and thus we cannot determine if the received bit is –b (t) or b (t) .Hence DPSK is used to eliminate the ambiguity of the received bit. The DPSK transmitter is as shown -



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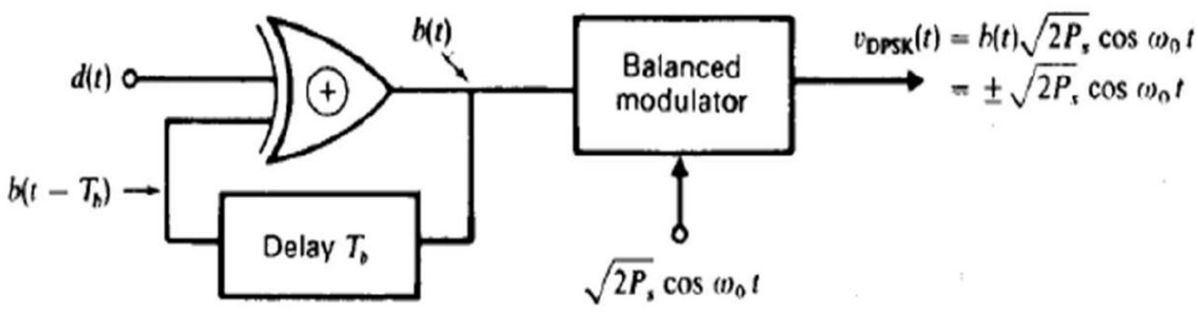


$d(t)$ logic level	$d(t)$ voltage	$b(t-T_b)$ logic level	$b(t-T_b)$ voltage	$b(t)$ logic level	$b(t)$ voltage
0	−1	0	−1	0	−1
0	−1	1	1	1	1
1	1	0	−1	1	1
1	1	1	1	0	−1

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-

$V_{Dpsk}(t) = \sqrt{(2 Ps)} \cos\omega t$

That Means no phase Shift has been introduced

But when b (t) = 0, b" (t) = -1 Hence

$V_{Dpsk}(t) = \sqrt{(2 Ps)} \cos\omega t$

Thus 180 Phase shift is introduced to represent b (t) =0.

(e) Write the bandwidth requirement for ASK, FSK, BPSK and QPSK.

Ans:- (Each formula – 1 mks)

Bandwidth requirement for

BASK: $2f_b$, BFSK: $4f_b$, BPSK: $2f_b$,QPSK: f_b ,Where f_b is Bit Frequency

Winter 2017 DCO Solution -17535

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

(a) State the advantages and disadvantages of digital communication system.

Ans. **Advantages of digital communication** **(Any 2 points -2 marks)**

- Noise Immunity is good.
- Digital signals are better suited than analog signals for procession and combining using technique called multiplexing.
- Digital transmission systems are more resistant to analog systems to additive noise because they use signal regeneration rather than signal amplification.
- Digital signals are simpler to measure and evaluate than analog signals.
- In digital systems transmission errors can be corrected and detected more accurately.
- Using data encryption only permuted receivers can be allowed to detect the transmission data.
- Wide dynamic range.
- Because of the advances of IC technologies and high speed computers, digital communication systems are simpler and cheaper.
- Digital communication is adaptive to other advance branches of data processing such as digital.

Disadvantages of Digital Communication **(Any 2 points- 2 marks)**

- The transmission of digitally encoded analog signals requires significantly more bandwidth.
- Digital transmission requires precise time synchronization between the clocks in the transmitter and receiver.
- Digital transmission systems are incompatible with older analog transmission systems.

(b) Describe the process of quantization with neat diagram.

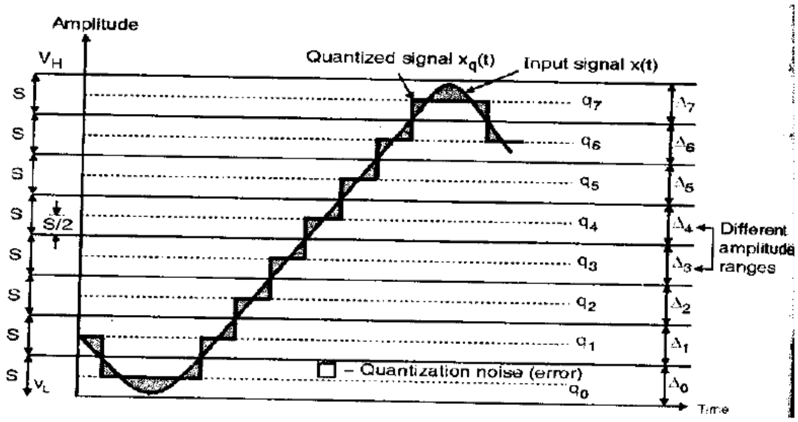
Ans:- (Diagram-2 mks, description – 2 mks)

Quantization process:

- Quantization is the process of approximation or rounding off the sampled signal. The converts sampled signal into approximated rounded values consisting of only finite n decided voltage levels called as quantization levels.
- In the process of A to D conversion, after sampling, quantization is the next step. The inp x(t) is assumed to have a peak swing of VL to VH volts. This entire voltage range has bee 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-



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

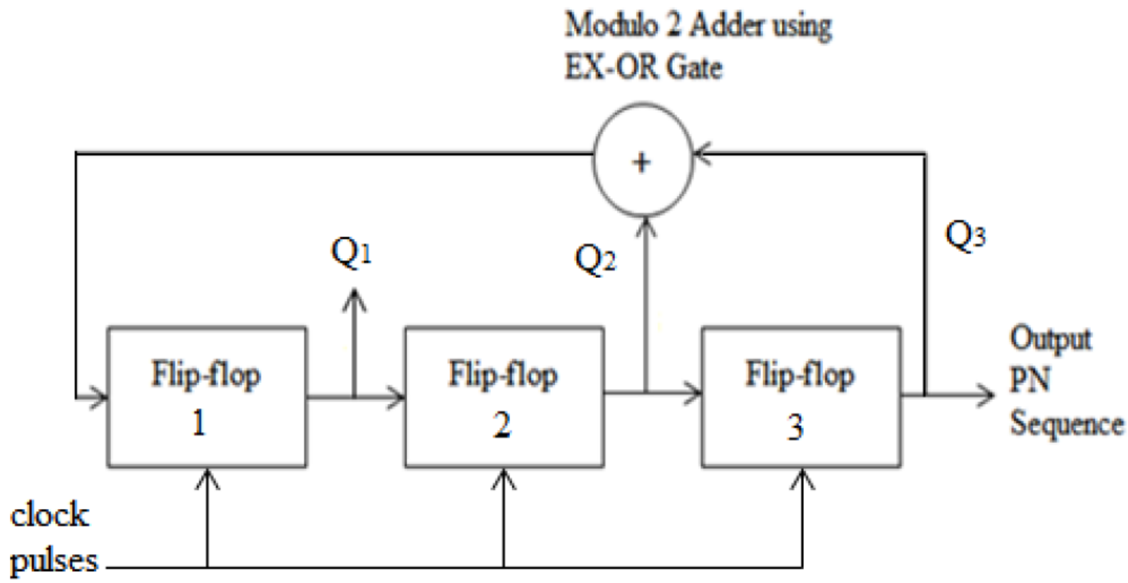
Ans:- (Definition- 2 mks, any suitable PN sequence generator- 2 mks)

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

Maximum length of PN Sequence 'L' is the no. of bits in a PN sequence and it depends upon the number of flip-flops 'n' used for the PN Sequence generator and given as

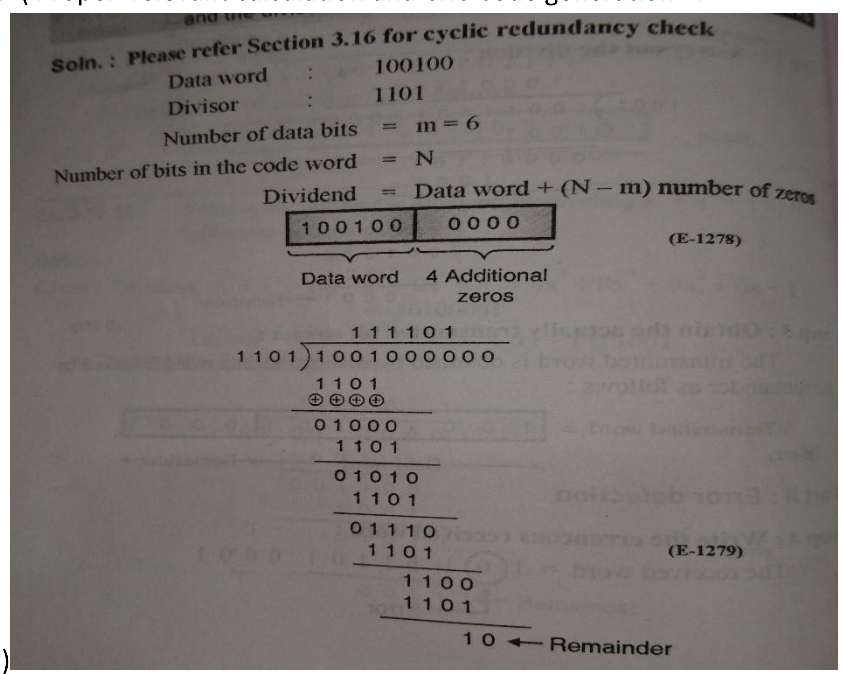
$L=2^n - 1$

The block diagram for 3 bit that is 7 bit length of PN sequence generator is as shown with feedback taps [3, 1]



(d) Calculate CRC code for data word 100100 to be transmitted and divisor is 1101.

Ans:- (Proper relevant calculation and CRC Code generation- 4



Soln. : Please refer Section 3.16 for cyclic redundancy check

Data word : 100100

Divisor : 1101

Number of data bits = m = 6

Number of bits in the code word = N

Dividend = Data word + (N – m) number of zeros

100100	0000

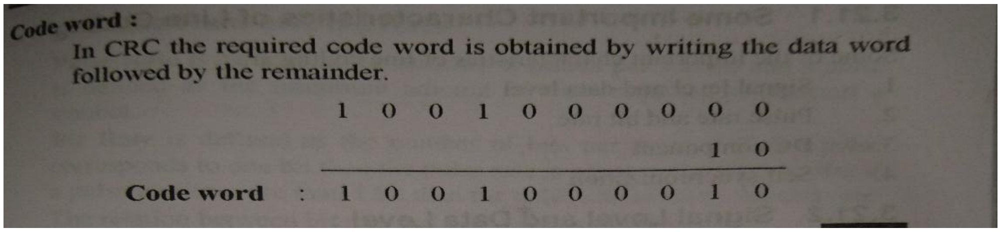
Data word | 4 Additional zeros

(E-1278)

```
          111101
1101)1001000000
     1101
     ⊕⊕⊕⊕
     01000
      1101
      01010
       1101
       01110
        1101
         1100
         1101
           10 ← Remainder
```

(E-1279)

mks)



Code word :

In CRC the required code word is obtained by writing the data word followed by the remainder.

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

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

(a) A discrete memory less source has the letters A, B, C & D with corresponding probabilities {0, 08, 0.2, 12, 0.4}

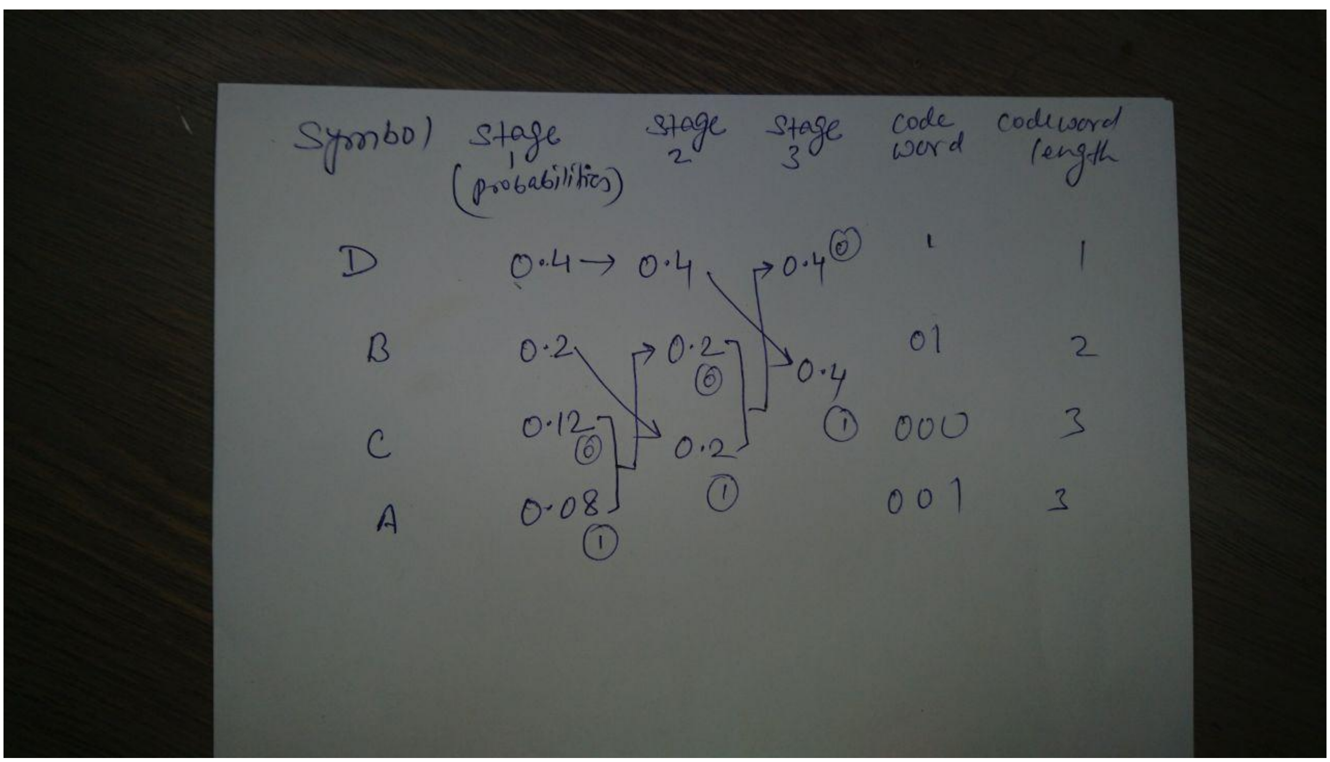
(i) Derive Huffman code for the above source.

(ii) Determine the average length of the code word.

(iii) Determine the coding efficiency of the Huffman code design.

Ans:- (Huffman code derivation- 4 mks, average length- 1 mks,coding efficiency-2 mks)

Hamming code-



Average length of code word

Hence, the average length of a code word is

$$\bar{n} = \sum_{i=0}^{M-1} p(x_i) n_i = 0.08 \times 4 + 0.2 \times 3 + 0.$$

$$\bar{n} = \sum_{i=0}^{M-1} p(x_i) n_i =$$

= (0.4*1) +(0.2*2) + (0.12*3) +(0.08*3)

= 1.4 bits/symbol

$H(S)$ = Entropy of the source

$$= -\sum_{i=0}^{M-1} p(x_i) \log_2 p(x_i)$$

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

$$= 0.52 + 0.46 + 0.367 + 0.292$$

$$= 1.639 \text{ bits/message.}$$

Codeword efficiency= H/L

= 1.637/1.4

117 %

Note: addition of Probabilities is not 1 for Huffman coding in the given problem.Numerical is solved based on data given & Efficiency ans is 117% in the solution but efficiency can not be more than 100%.

(b) Compare FHSS and DSSS system (any 6 points).

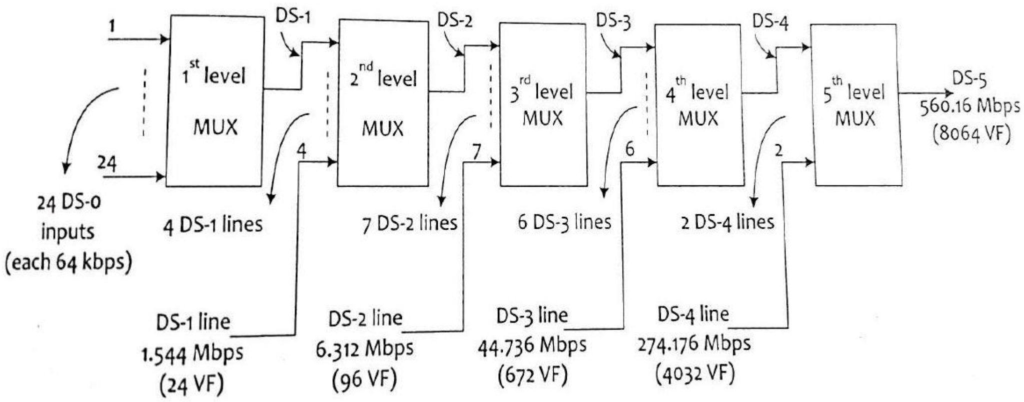
Ans:- (Any 6 points- 6 mks)

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.

5. Attempt any <u>TWO</u> of the following: Marks 16

(a) Describe the North American digital multiplexing hierarchy with neat diagram.

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



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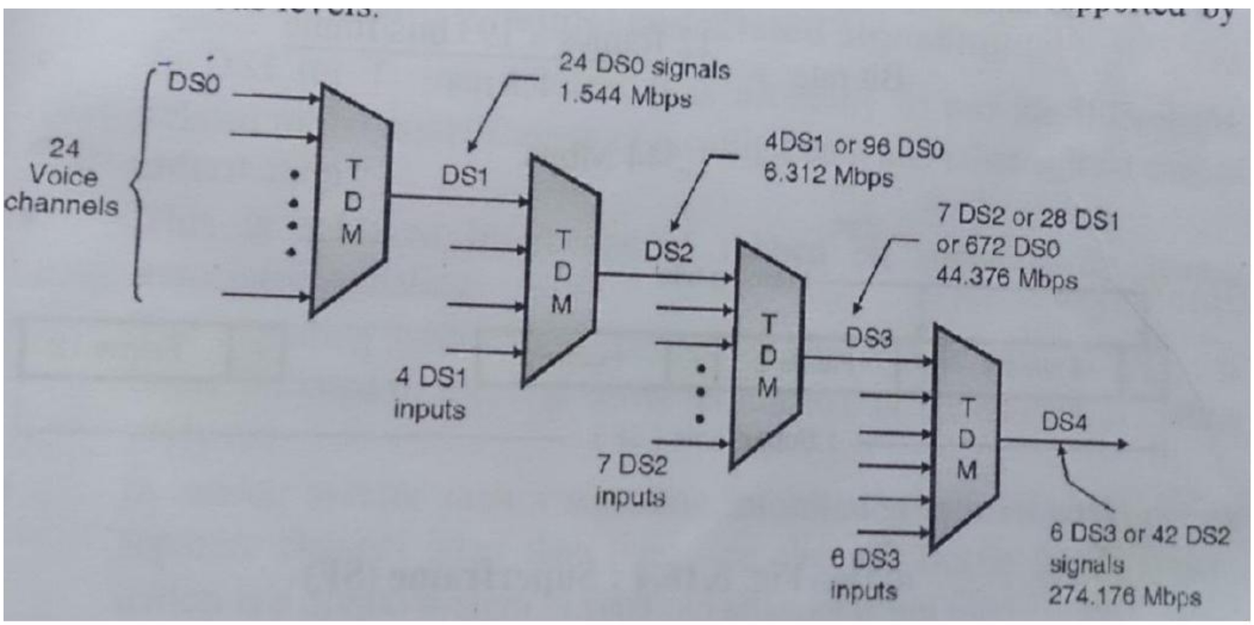


Explanation:-

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 5th level to get a 560.16 Mbps signal.

(b) Draw the block diagram of QAM generation system & explain it with waveform.

Ans:- (Block diagram- 4 mks, explanation- 2 mks, waveforms-2 mks)

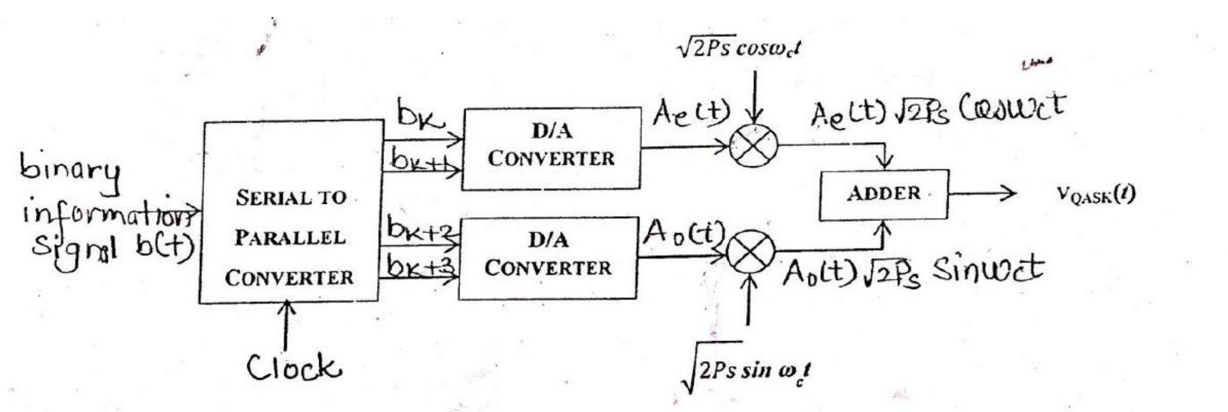
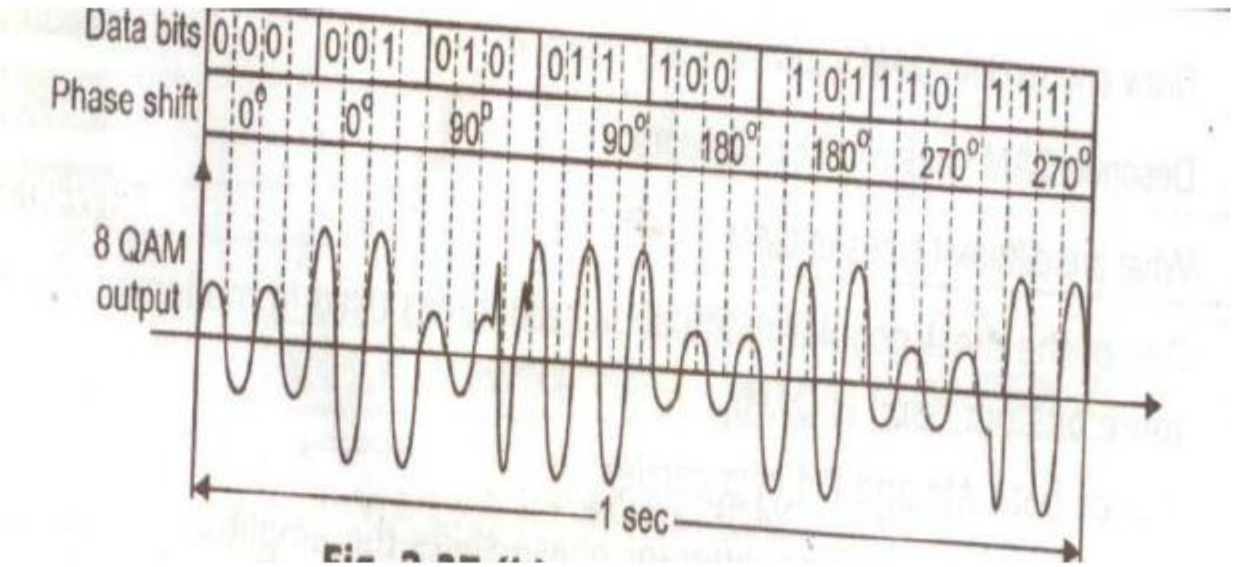


Figure shows transmitter for 4 bit QAM system. The input bit stream is applied to a serial to parallel converter. Four successive bits are applied to the digital to analog converter. These bits are applied after every Ts second. Ts is the symbol period & Ts=4Tb.Bits Bk & Bk+1 are applied to upper digital to analog converter. & Bk+2, Bk+3 are applied to lower D to A converter. Depending upon the two input bits, the output of D to A converter takes four output levels. Thus Ae (t) & Ao (t) takes 4 levels depending upon the combination of two input bits. Ae (t) modulates the carrier $\sqrt{Ps}\cos(2\pi f_0 t)$ and Ao (t) modulates $\sqrt{Ps}\sin(2\pi f_0 t)$.

The adder combines two signals to give QAM signal. It is given as,

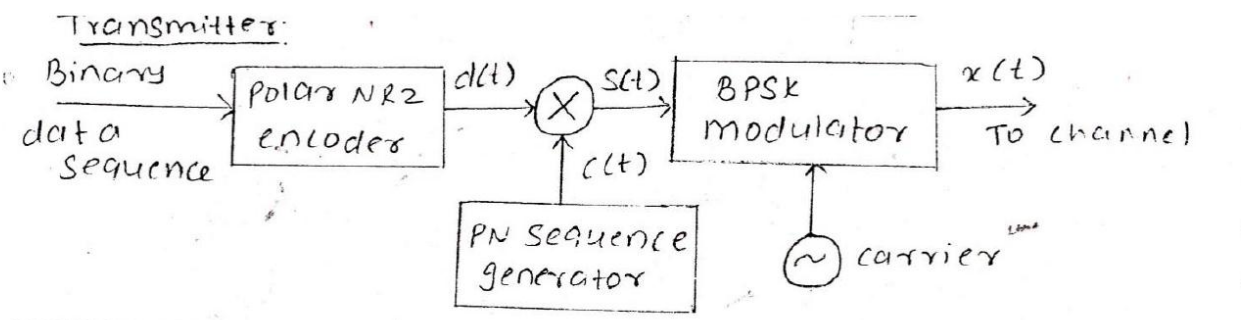
$$S(t) = A_e(t)\sqrt{Ps}\cos(2\pi f_0 t) + A_o(t)\sqrt{Ps}\sin(2\pi f_0 t).$$

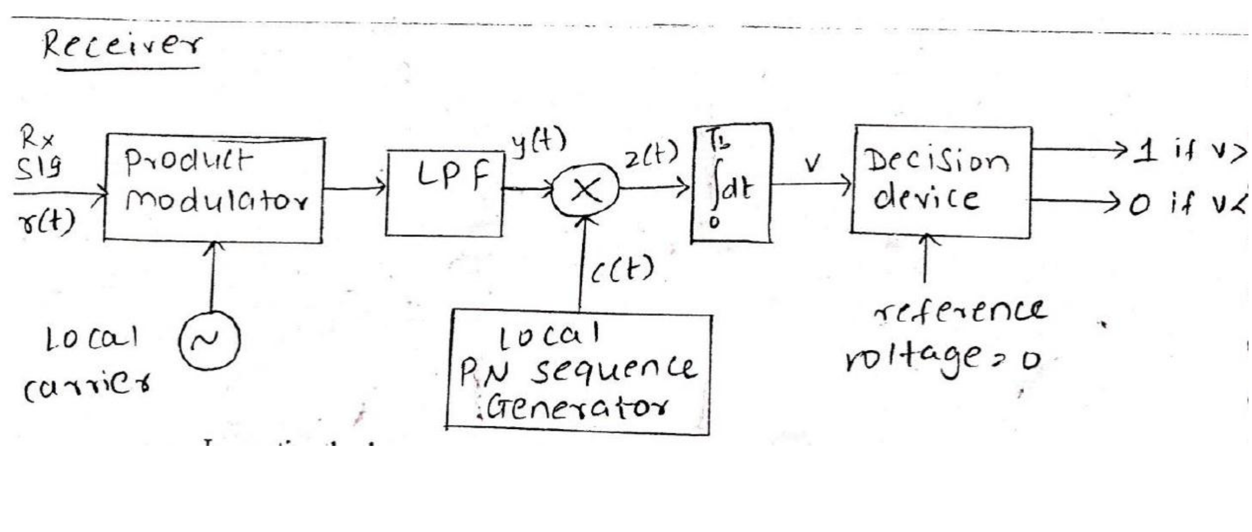
WAVEFORMS-



(c) Describe the direct sequence spread spectrum technique with the help of block diagram.

Ans:- (Block diagram -2 mks each , description- 4 mks)





The modulated signal at the output of product modulator or multiplier i.e. s (t) is used modulate tl carrier for BPSK modulation.
The transmitted signal x (t) is thus DSSS signal.
Product modulator output = s (t)
s (t) = d (t) * c (t)

The BPSK carrier signal is given by $\sqrt{2Ps}\sin 2\pi fC\, t$.
The output of BPSK modulator x (t) is transmitted x (t) = s (t) * $\sqrt{2Ps}\, 2\pi fCt$.
But m (t) = ± 1
Therefore x (t) = $\pm\sqrt{2Ps}\sin 2\pi fC\, t$
The phase shift of x (t) of x (t) is 0^0 to + m (t) at is 180^0 corresponding to a negative m (t).

At the receiver, the signal is coherently demodulated by multiplying the received signal by a replica of the carrier.

The signal *r(t)* at the input of the detector LPF is given by,

$$r(t) = d(t)c(t)\cos\omega_c t\,(2\cos\omega_c t)$$
$$= d(t)c(t) + m(t)c(t)\cos 2\omega_c t$$

The LPF eliminates the high frequency components at $2\omega_c$ and retains only the low frequency component $y(t) = d(t)c(t)$.

This component is then multiplied by the local code $c(t)$ in phase with the received code. $c^2(t) = 1$.

At the output of the multiplier, this gives,

$$z(t) = d(t)c(t)c(t) = d(t)$$

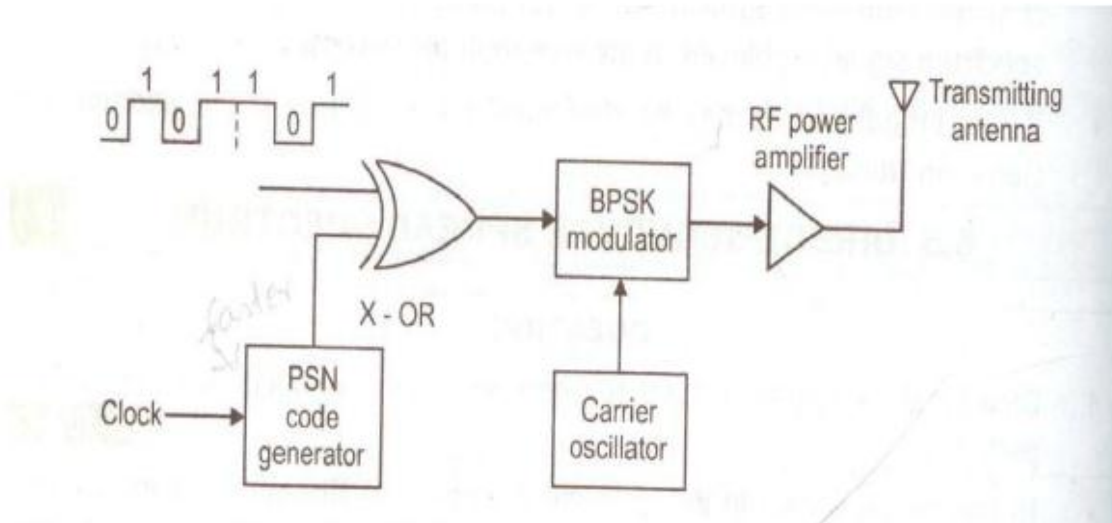
OR

(Block diagram -4mks, explanation- 4 mks)

Direct sequence spread spectrum (DSSS):

In direct sequence, the serial binary data is mixed with a higher frequency pseudorandom binary code at a faster rate and the result is used to phase-modulate a carrier.

DSSS Transmitter:



Explanation:

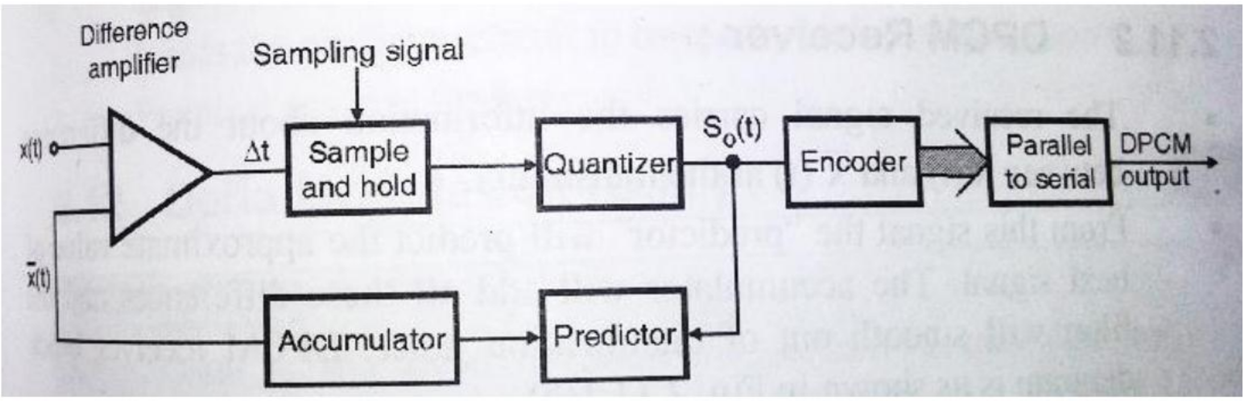
- A block diagram of DS transmitter is shown in figure
- The serial binary data is applied to an X- OR gate along with a serial pseudorandom code that occurs faster than binary data.
- The signal developed at the output of the X-OR gate is then applied to a BPSK modulator.
- The carrier phase is switched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

6. Attempt any <u>FOUR</u> of the following: **Marks 16**

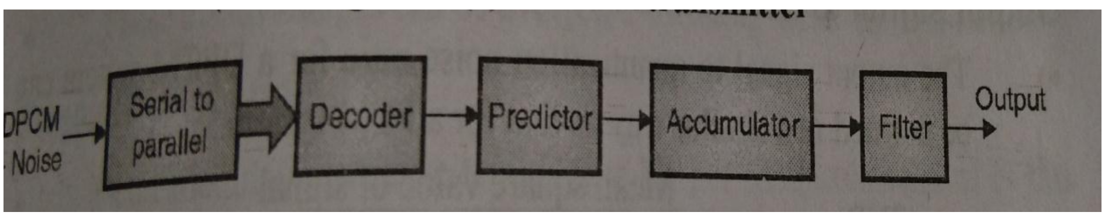
(a) Draw block schematic of DPCM transmitter and receiver.

Ans:- (Block diagram of transmitter- 2mks,receiver-2 mks)

Transmitter



Receiver



(b) Compare TDM, FDM & CDM (3 points)

Ans:- (Three relevant comparison points- 4 mks)

FDM	TDM	CDM
Divides the channel into the two or more frequency ranges that do not overlap.	Divides and allocates certain time periods to each channel.	Sharing of both bandwidth and time takes place.
Synchronization not required	Synchronization not required	Synchronization not required
Code word is not required	No coding	Code words are required
Needs guard bands	Needs guard time	Needs both guard band and guard time.
Problem of crosstalk	No problem of crosstalk	Secured communication

(c) Compare ASK with FSK modulation (any 4 points).

Ans: **(Any 4 relevant correct points - 4 marks)**

Parameters	ASK	FSK
Variable characteristics	Amplitude	Frequency
Bandwidth (Hz)	f_b	$\|f1 - f0\| + f_b$
Noise immunity	Low	High
Error probability	High	Low
Performance in presence of noise	Poor	Better than ASK
Complexity	Simple	Moderately complex
Bit rate	Suitable up to 100bits/sec	Suitable up to about 1200 bits/sec
Detection method	Envelope	Envelope

(d) Define the following terms:

(i) Code Word

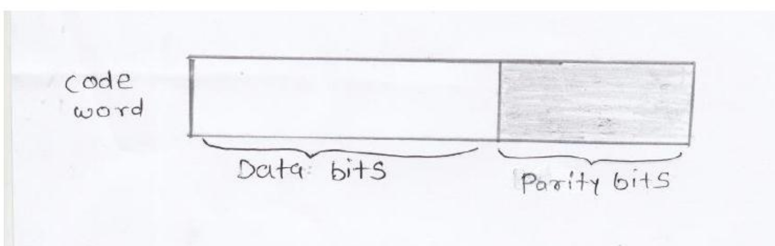
(ii) Code Rate

(iii) Hamming weight

(iv) Hamming distance ,related to code.

Code word: -

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**

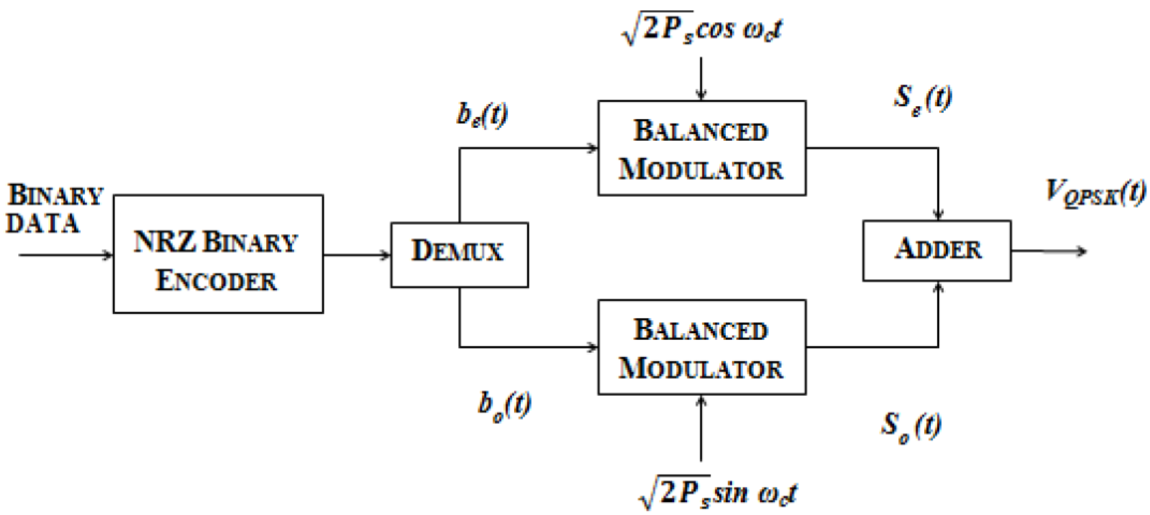
The Hamming weight of a code word x is defined as the number of non-zero element in the code word. Hamming weight of a code vector (code word) is the distance between that code word and an all zero code vector (a code having all element equal to zero)

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

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

(e) Describe QPSK generator with waveform.

Ans:- QPSK transmitter/generator- 2 mks, description – 1mks, waveforms- 1 mks)



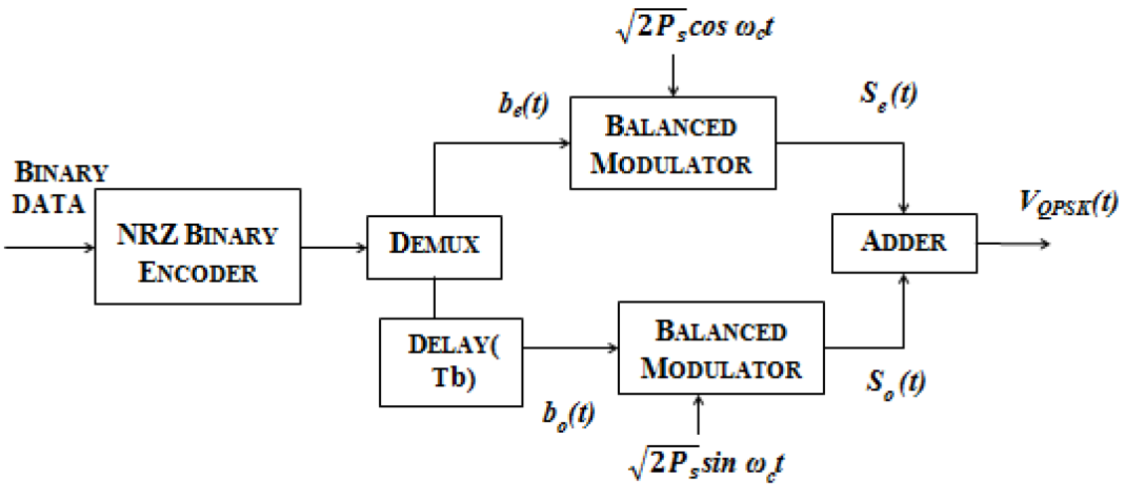
QPSK Transmitter (non - offset)

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



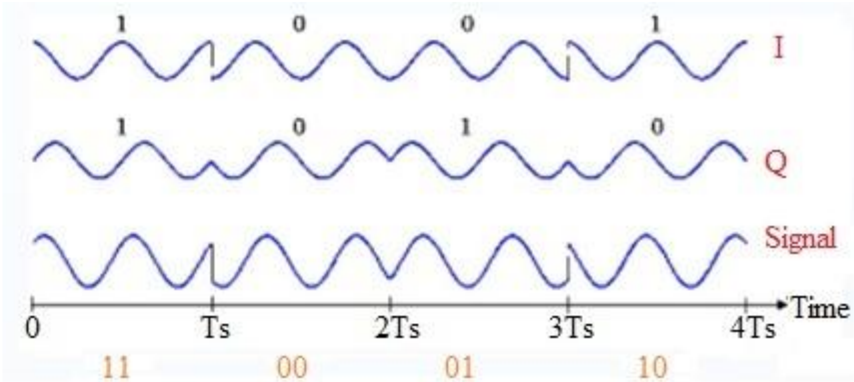
Offset QPSK Transmitter

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.

Waveforms



QPSK Time Diagram

OR