WINTER– 14 EXAMINATION

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.

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

a) State Shannon Hartley Theorem. What are its implications?

Ans. **(Theorem – 2 Marks, Implications – 2 Marks)**

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

The Implications of the Shannon Hartley Theorem is as follows,

1. It gives us an upper limit that can be reached in the way of reliable data transmission rate over Gaussian channels. Thus, a system designer always tries to optimize his system to have data rate as close to channel capacity *C*, given in the equation, as possible with an acceptable error rate.
2. The second implication of the Shannon-Hartley theorem has to do with the exchange of signal-to-noise ratio *(S/N)* for bandwidth i.e. tradeoff between *S/N* and Bandwidth *B*.

b) State Sampling Theorem. Explain aliasing effect with neat diagram.

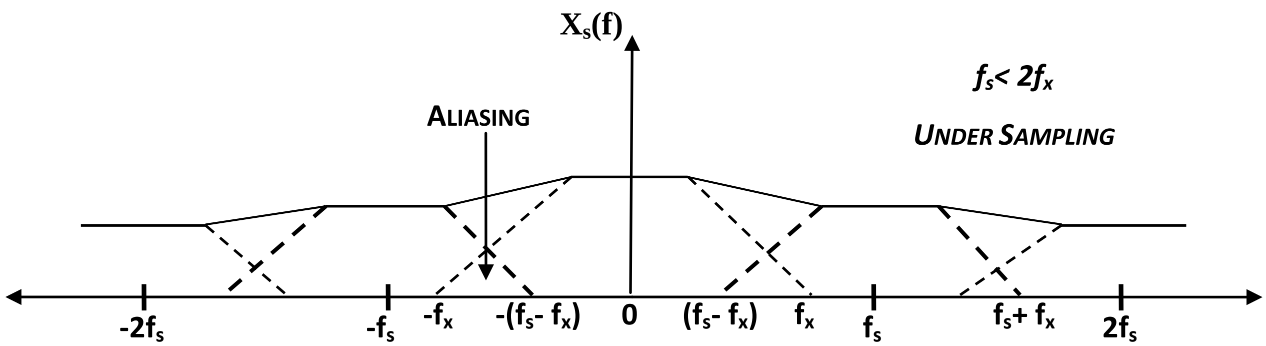
Ans.**(Theorem – 1 Mark, Aliasing Effect Diagram –1 1/ 2 Marks, Explanation – 1 1/2Mark)**

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

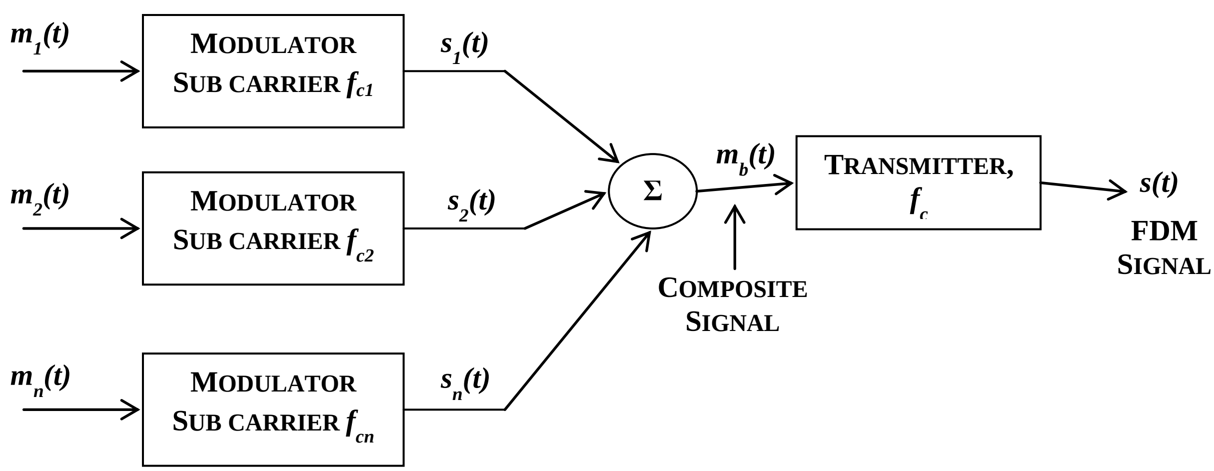


c) How does FDM technique combines multiple signals into one?

Ans.**(FDM Diagram – 2 Marks, Explanation – 2 Marks)**

Frequency Division Multiplexing (FDM) is based on the idea that number of signals can share the bandwidth of a common communication channel. The multiple signals to be transmitted over this channel are each used to modulate a separate carrier. Each carrier is on a different frequency. The modulated carrier are then added together to form a single complex signal that is transmitted over the single channel.

Fig below shows a general block diagram of FDM system. Each signal to be transmitted is fed to modulator circuit. The carrier for each modulation fc is on a different frequency. The carrier frequencies are equally spaced from one another over a specific frequency range. Each input signals given portion of the bandwidth. Another standard modulation like AM, SSB, FM or PM can be done.



The modulator output having side band information is added together in a linear mixer in which all the signals are simply added together algebraically. The resulting output signal is composite of all carriers containing their modulation. This signals transmitted over single communication channel.

d) Compare DSSS and FHSS system w.r.t

- **i. Definition**
- **ii. Chip Rate**
- **iii. Modulation Technique**
- **iv. Effect of Fading**

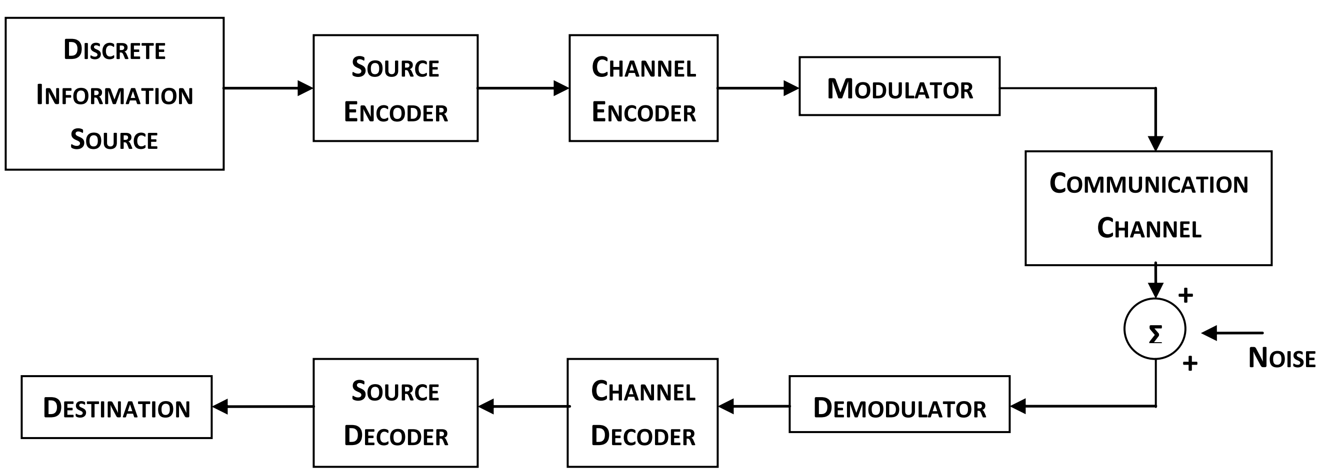
Ans. **(Each Correct Point - 1 Mark)**

Sr.No	Parameter	DSSS	FHSS
1	Definition	PN Sequence of large bandwidth is multiplied with narrow band data signal.	Data bits are transmitted in different frequency slots which are changed by PN Sequence.
2	Chip Rate	It is fixed Rc = 1/Tc	Rc = max (Rh , Rs)
3	Modulation Technique	BPSK or M-ary PSK	BFSK or M-ary FSK
4	Effect of Fading	More	Less

Q.1 B) Attempt any ONE of the following (6 Marks)

a) Draw the block diagram of digital communication system. What is the need of channel modeling? Explain any one in detail.

Ans.**(Diagram – 3 Marks, Need – 1 Marks, Any 1 of below listed model – 2 Marks)**



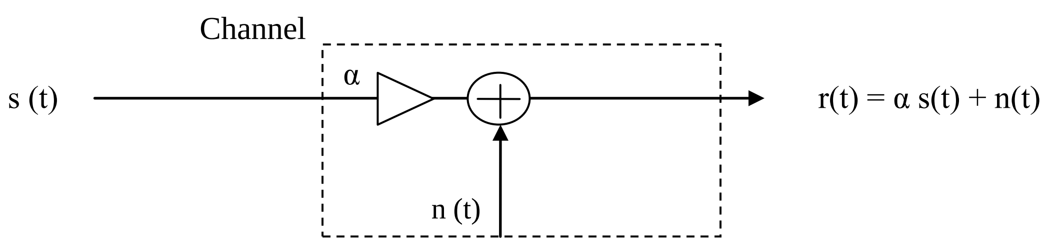
Need of Channel Modeling

In the analysis and design of communication system, it will be necessary to model the channel as system and incorporate into that model as many details of electrical behavior of the channel as possible, so as to make it represent the actual situation as accurately as possible.

Types of Channel Modeling

1. **Additive Gaussian noise channel**

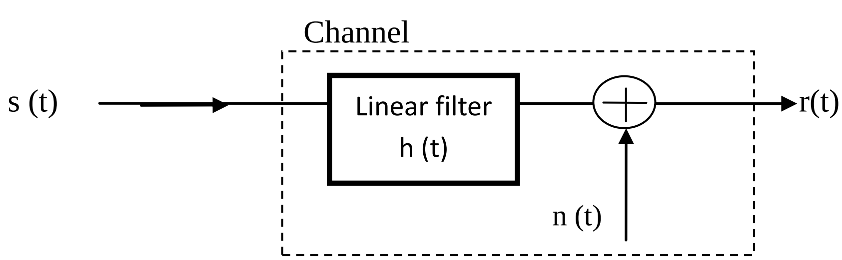
   - It is the most extensively used channel model which portrays the channel as shown below



   - It simply attenuates the signal by a factor α ($0 < \alpha < 1$) and introduces additive noise
   - The model is extremely simple and can be used to represent a large number of physical channels, and hence it is very widely used.

2. **Bandwidth limited linear channel**

   - Certain channels like telephone channel are linear, but bandwidth limited. Such channels may be modeled as shown

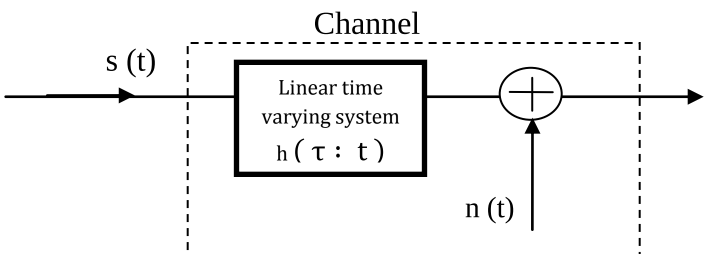


   - These channels are time –invariant and so the filter shown in the above fig is an LTI system with an impulse response function h(t).
   - Thus r(t) = s(t) * h(t) + n(t)

$$=\int_{-\infty}^{\infty} s(t-\tau)h(\tau)d\tau + n(t)$$

3. **Linear time-variant channels**

- Channels like the underwater acoustic channels, some mobile communication channels and ionospheric scatter channels, in which the transmitted signal reaches the receiver through more than one path, and were these path length are varying in time, have what is generally termed as 'time – varying' multipath propagation.
- Such channels are modeled using time varying system as shown in fig below

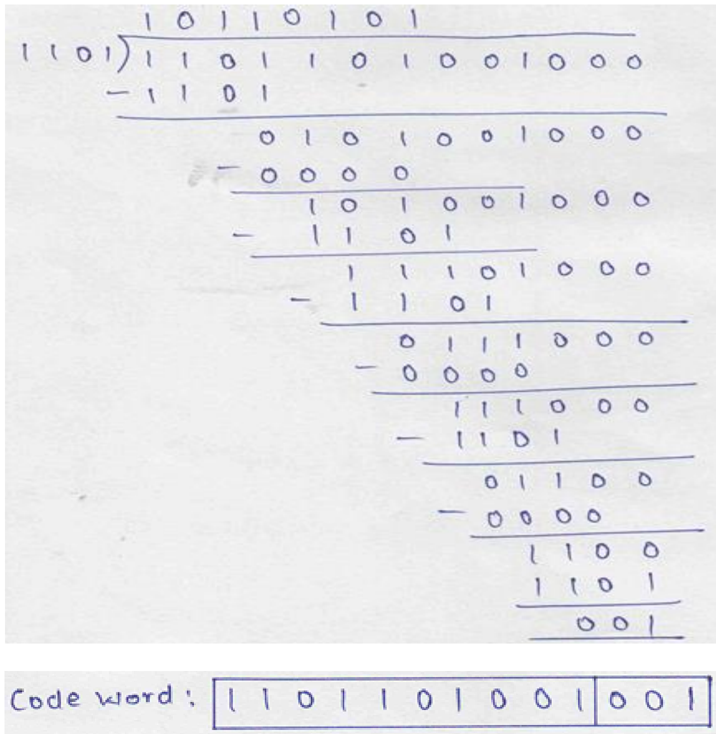


- In this model, h(τ : t) is the impulse response function of the time variant linear system and represents the output time t, of the system which is at rest, when an impulse of unit strength is applied to it as input at time (t – τ) thus ,
- $\int_{-\infty}^{\infty} s(t-\tau)h(\tau : t)d\tau + n(t)$

b) Generate CRC code for data word 1101101001 by using divisor as 1101. State 2 advantages of CRC Method.

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

Dividend: 1 1 0 1 1 0 1 0 0 1
Divisor: 1 1 0 1
No of zeros to be added to dividend: 3



Advantages of CRC Codes

1. Implementation of encoding and error detection circuits is practically possible.
2. CRC codes are capable of detecting any kind of error bursts.
3. CRC can detect all burst errors of length less than or equal to degree of the polynomial.

Q.2 Attempt any TWO of the following (16 Marks)

a) Draw the neat block diagram of PCM Transmitter and Receiver. Explain the same with waveforms.

Ans.

PCM Transmitter Diagram **(2 Marks)**

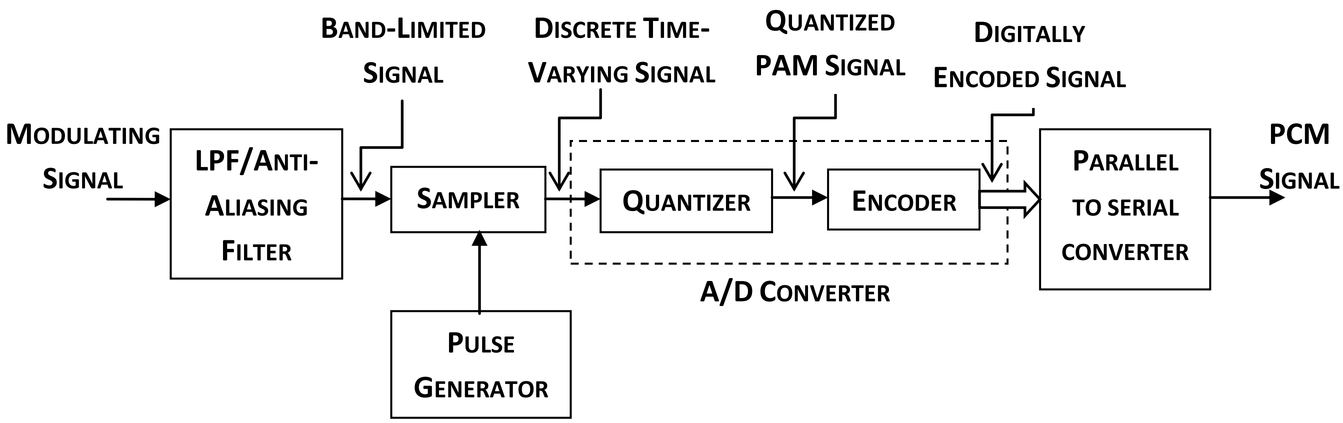


Fig. block diagram of PCM Transmitter

PCM Transmitter Explanation **(1 Mark)**

- The analog signal/modulating signal x (t) is passes through band limiting / low pass filter, which has a cut-off frequency fc=W Hz. This will ensure x (t) will not have any frequency component higher than “W”. In other words, suppresses high frequency components and passes only low frequency signal to avoid ‘aliasing error’.
- The band limited analog signal is then applied to sampled and hold circuit where this circuit acts as modulator and both modulating input signal and sampling signal with adequately high sampling rate are inputs to this circuit. Output of sampled and hold block is a flat topped PAM signal.
- These samples are subjected to operation “quantization” in the “quantizer”. Quantization is a process of approximation of the value of respective sample into a finite number that will reduce data bits. The combined effect of sample and quantization produces is ‘Quantized PAM’ at the quantizer output.

- The Quantized PAM output is analog in nature. So to transmit it through digital communication system the quantized PAM pulses are applied to an encoder which is basically A to D convertor. Each quantized level is converted into N bit digital word by A to D converter.
- The communication system is normally connected to each other using a single cable i.e. serial communication. But the output of ADC is parallel which cannot be transmitted through serial communicating links. So this block will convert the parallel data into serial stream of data bits.
- A pulse generator produces train of rectangular pulses of duration “t” seconds. This signals acts as sampling signals for the sample and hold block. The same signal acts as “clock” signals for parallel to converter .the frequency “f” is adjusted to satisfy the criteria.

PCM Receiver Diagram **(2 Marks)**

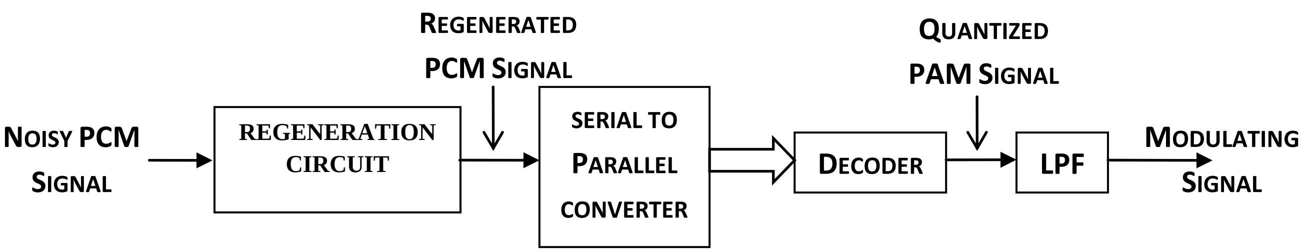
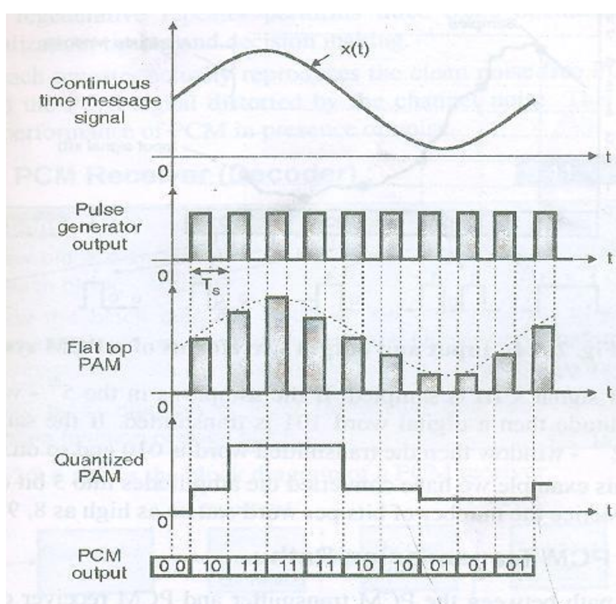


Fig. block diagram of PCM Receiver

PCM Receiver Explanation **(1 Mark)**

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

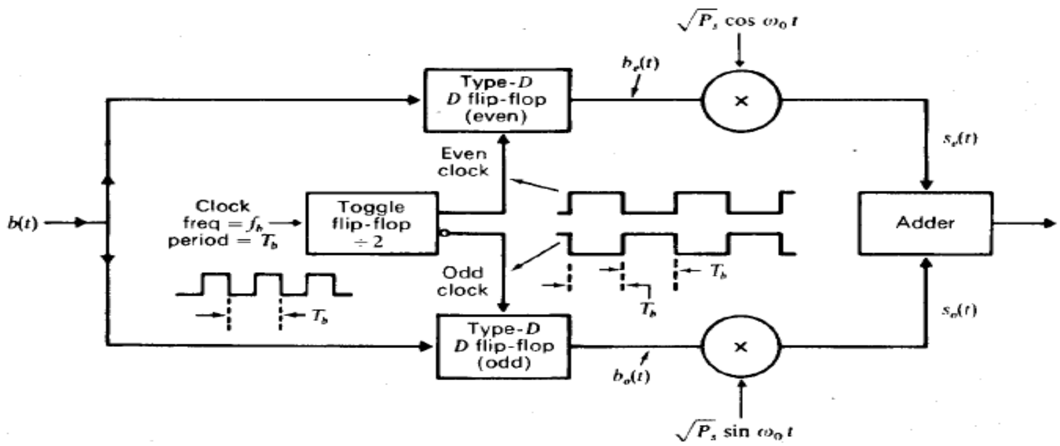
Waveforms **(2 Marks)**



b) Draw the block diagram of QPSK Transmitter and Receiver. Explain its Working Principle. Draw its Constellation Diagram.

Ans.**(QPSK Transmitter & Receiver Diagram – 2 Marks each, Working Principle – 2 Marks, Constellation Diagram – 2 Marks)**

QPSK Transmitter



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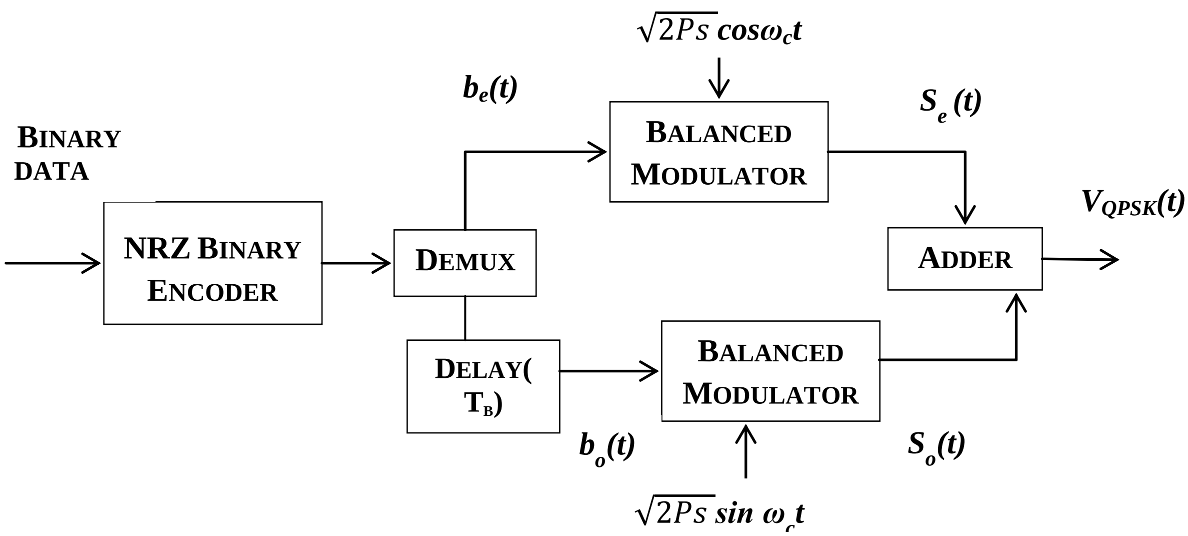
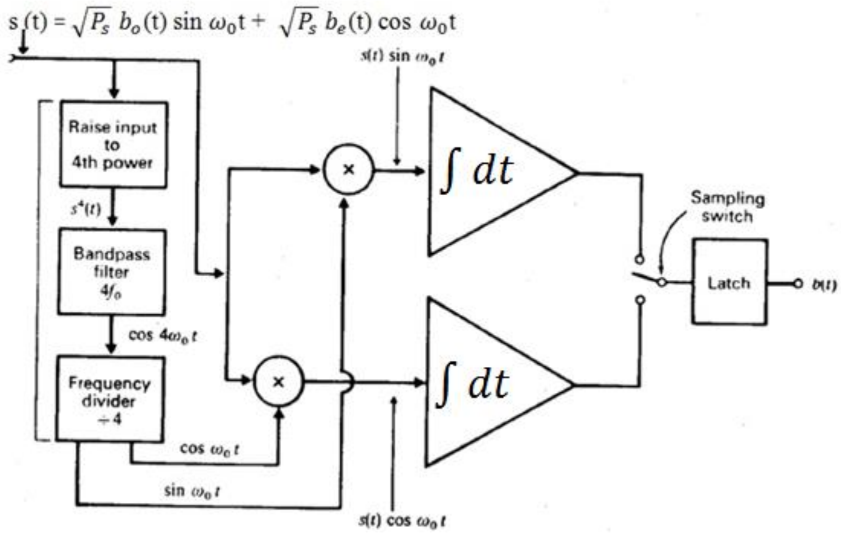


QPSK Receiver

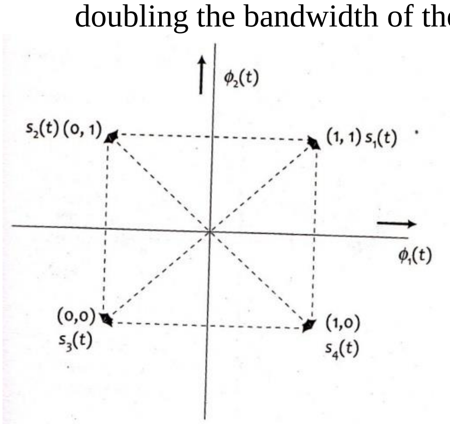


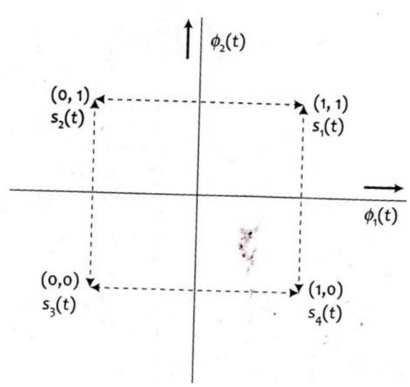
Working Principle

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

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

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





Symbol	Phase
00	0^0
01	90^0
10	270^0
11	180^0

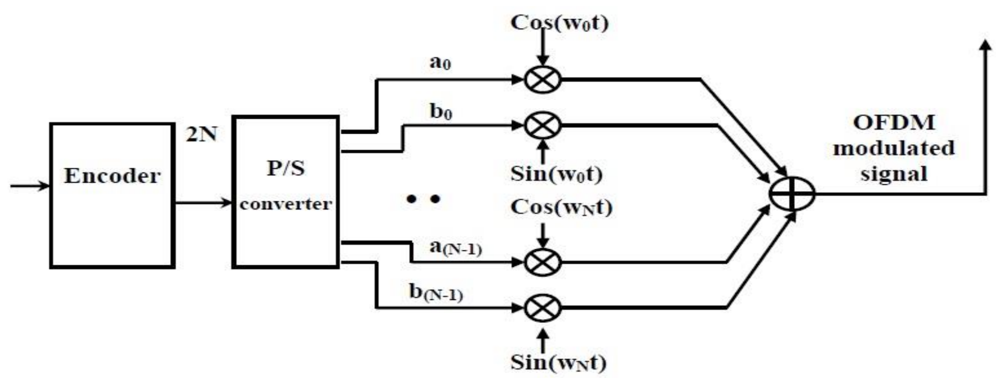
c) With a neat sketch describe the working of OFDM multi carrier system.

Ans. **(Diagram – 4 Marks , Explanation – 4 Marks)**

Figure below shows the conceptual diagram highlighting the orthogonal (OFDM) multiple carrier modulation scheme. The a_i – s in the diagram indicates the modulating signal in the I – path and the b_i – s are the modulating signals in the Q – path.

The 'encoder' in a practical system performs several operations but if of no special significance at the moment. All the cosine modulated signals are added algebraically and similarly are the sine modulated signals.

The overall I – phase and Q – phase signals together form a complex baseband OFDM signal. At this point, one may interpret the scheme consisting of a bank of N parallel QPSK modulators driven by N orthogonal sub carriers.



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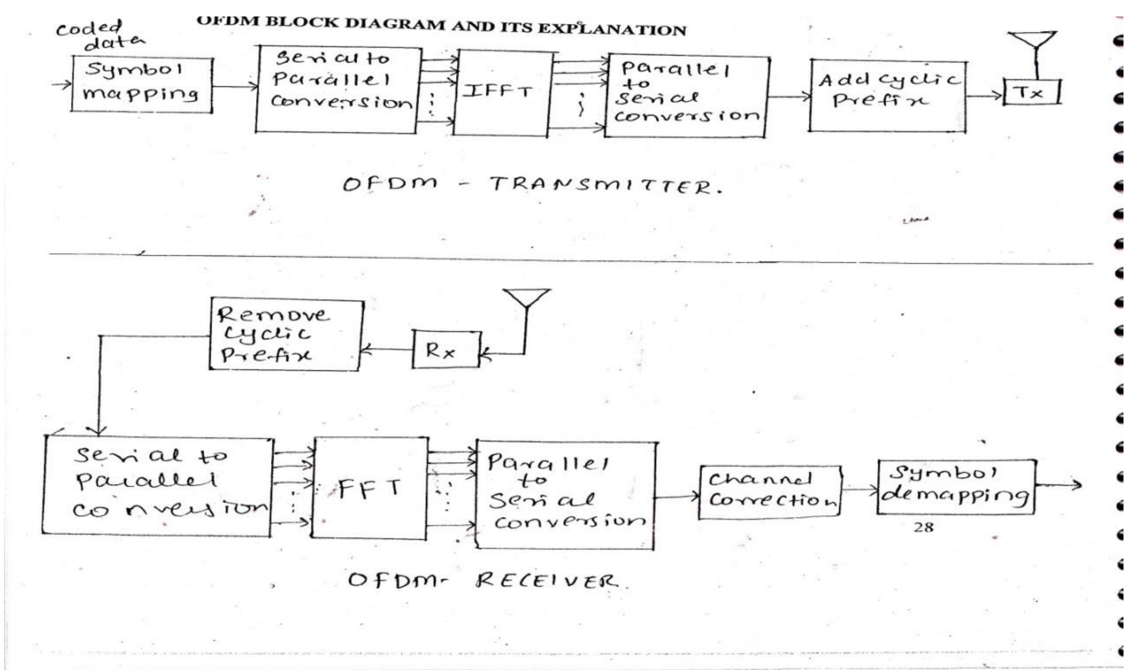


A useful feature of OFDM modulation scheme is that pulse shaping is not necessary for the modulating signals because a bunch of orthogonal carriers, when modulated by random pulse sequences, have a orderly spectrum as shown below.

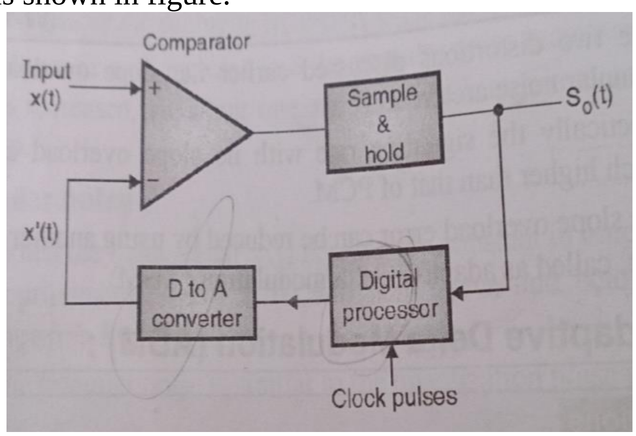
As indicated, the orthogonal sub-carriers occupy the spectral zero crossing positions of sub- carriers. This ensures that a sub carrier modulated signal with seemingly infinite spectrum does not interfere with the signals modulated by other sub carriers.

Q-3 Attempt any Four **16 Marks**

a) With the help of relevant block diagram, explain the working principle of adaptive delta modulation transmitter.

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

The ADM transmitter is shown in figure.



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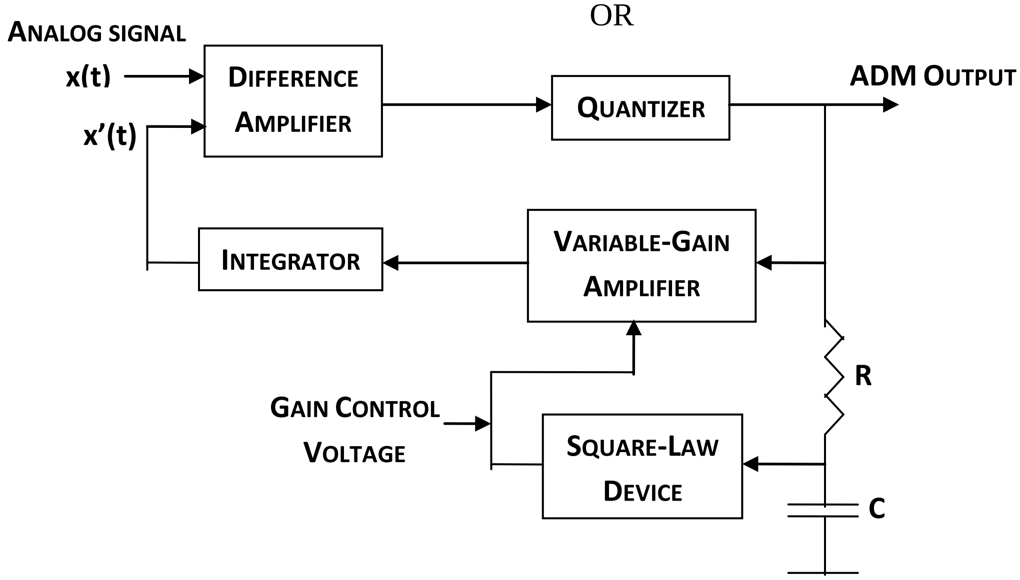


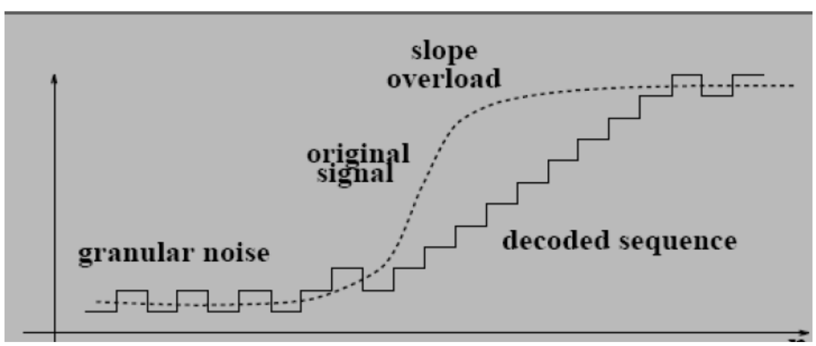
Explanation

As shown, X (t) is the analog input signal & x' (t) is the quantized version of x(t). Both these signal are applied to comparator. Comparator output is goes high if x(t) >x'(t) & it goes low if x(t)<x'(t). Thus the comparator output is either 1 or 0.Sample & hold circuit will hold this level for entire clock cycle. In response to k^{th} clock pulse trailing edge, a processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. (k-1)th clock edge. If the direction of both the step is same then the processor will increase the magnitude of present step by delta. If the direction is opposite then the processor will decrease the magnitude of present step by delta.

b) Describe the concept of slope-overload distortion in a DM system. Draw neat waveform. How it can be avoided.

Ans: **(Slope overload error 1 marks, waveform -2 marks, How to avoid -1 mark)**

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



Avoidance of slope overload-

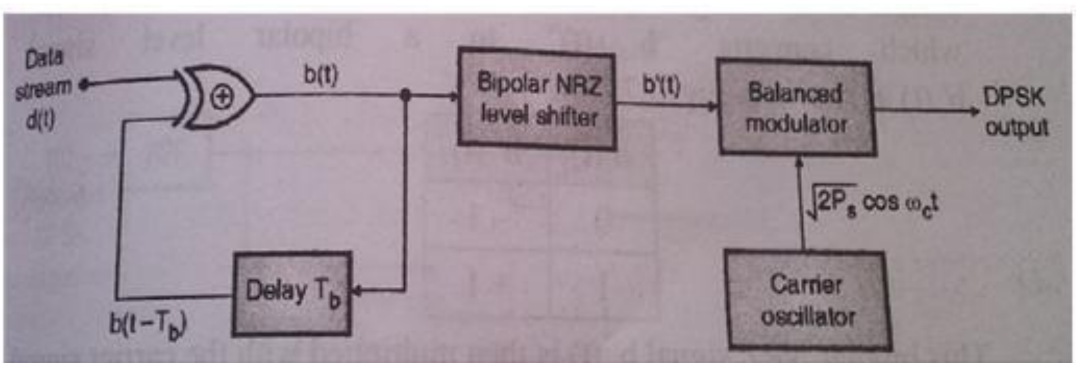
The slope overload error can be reduced by increasing slope of the approximated signal x‟ (t). Slope x‟ (t) can be increased and hence the slope overload error can be reduced by either increasing the step size or by increasing the sampling frequency.

The slope overload error can be reduced by increasing slope of the approximated signal X‟ (t). If slope of X‟ (t) can be increased and hence the slope overload error can be reduced by either increasing the step size δ or by increasing sampling frequency fs.

c) Compare TDMA, FDMA and CDMA techniques based.
i) definition ii) bandwidth available iii) synchronization iv) application.
Ans: **(Each point 1Mark)**

PARAMETER	TDMA	FDMA	CDMA
Definition	Time Division Multiple Access, here entire bandwidth is shared among different subscribers at fixed predetermined or dynamically assigned time intervals/slots.	Frequency Division Multiple Access, here entire band of frequencies is divided into multiple RF channels/carriers. Each carrier is allocated to different users.	Code Division Multiple Access, here entire bandwidth is shared among different users by assigning unique codes.
Bandwidth available	Time sharing of satellite transponder takes place	Overall bandwidth is shared among many stations.	Sharing of bandwidth and time both takes place.
Synchronization	Synchronization is essential	Synchronization is not necessary	Synchronization is not necessary
application	Advanced mobile phone, system(AMPS), Cordless telephone	GSM , PDC(pacific digital cellular)	IS95 Wide band, CDMA 2000

d) Draw the DPSK transmitter and outline its working principle.
Ans: **(Explanation- 2M, Diagram- 2M)**



Principle
It combines, differential encoding and phase shift keying.
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.

Operation

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.
3] The delayed represented by "b (t-Tb)". Depending on the values of "d (t)" and "b (t-Tb)" the EX-OR produces the output sequence "b (t)".The waveform for the generator .the waveform drawn by arbitrarily assuming that in the first interval b (0)=0
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

$V_{Dpsk}(t)= \sqrt{(2Ps)} \cos \omega t$
That Means no phase Shift has been introduced
But when b (t) = 0, b (t) = -1 Hence
$V_{Dpsk}(t)= -\sqrt{(2Ps)} \cos \omega t$
Thus 180^0 Phase shift is introduced to represent b (t) =0

e) Write the bandwidth requirement for BASK, BFSK, BPSK, QPSK.

Ans: (1 marks each for proper answer)
Bandwidth requirement for

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

Q4 (a) Attempt any three (12 Marks)

a) Discuss the characteristics of communication channels w.r.t.
i) bit rate ii) bandwidth iii) repeater distance iv) application

Ans: one mark for each channel (any 4)

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

WINTER– 14 EXAMINATION

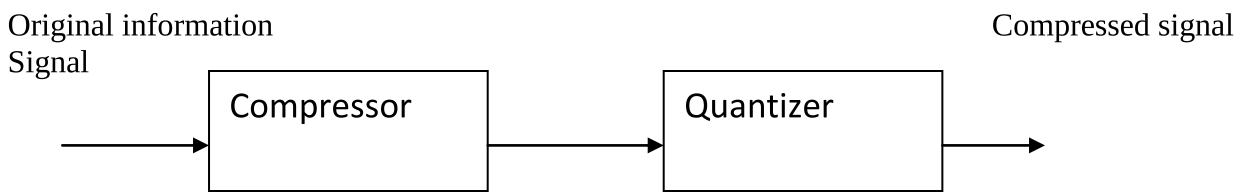
Subject Code: **17535** **Model Answer** Page No: 17/29

b) Explain Companding. Sketch the input-output characteristics of a compressor and an expander.

Ans: **(Explanation- 2 marks, characteristics with explanation -2 marks**)

The combination of compressor and expander is known as compander which performs the Companding process. It is used to increase the signal to quantization error ratio for weak signal. The figure shown below is the schematic block diagram of Companding.

Compression at the transmitter side:



Expansion at the Receiver side:

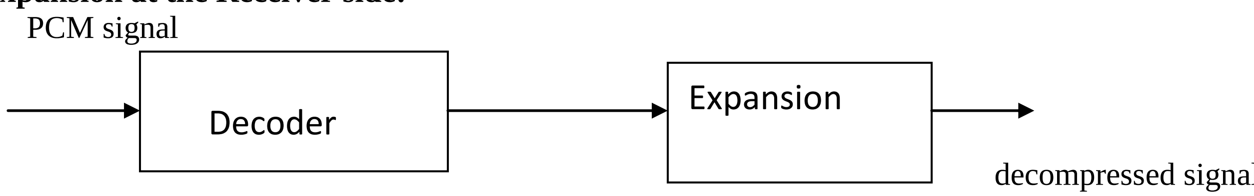
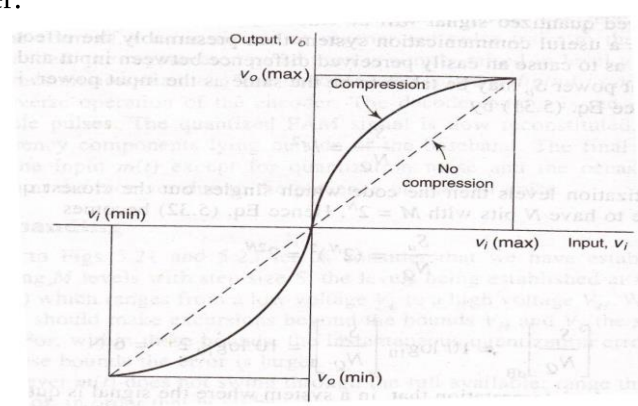


Fig. schematic block diagram of Companding

At the transmitter end the information signal is passed through compressor where the signal is amplified more at low amplitude than at high amplitude.

At the receiver side, an inverse operation is performed to recover the original information signal. This is achieved by an expander.



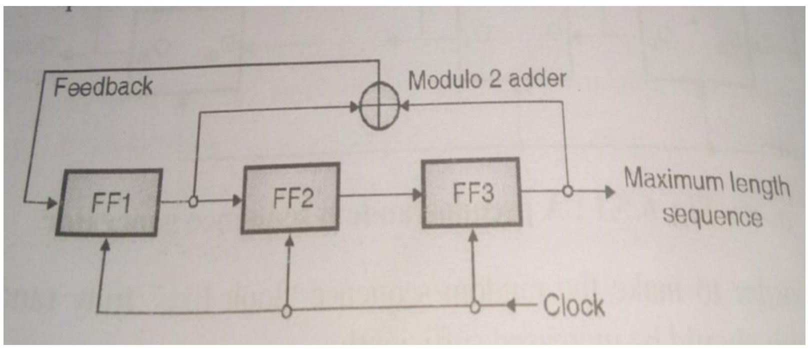
c) What is maximum-length sequence? Generate maximum-length sequence of length 7 with feedback taps = [3, 1].

Ans: 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

$L=2^n - 1$ **(1 marks)**

The block diagram for 3 bit that is 7 bit length of PN sequence generator is as shown with feedback taps [3 ,1**]** **(2 marks)**



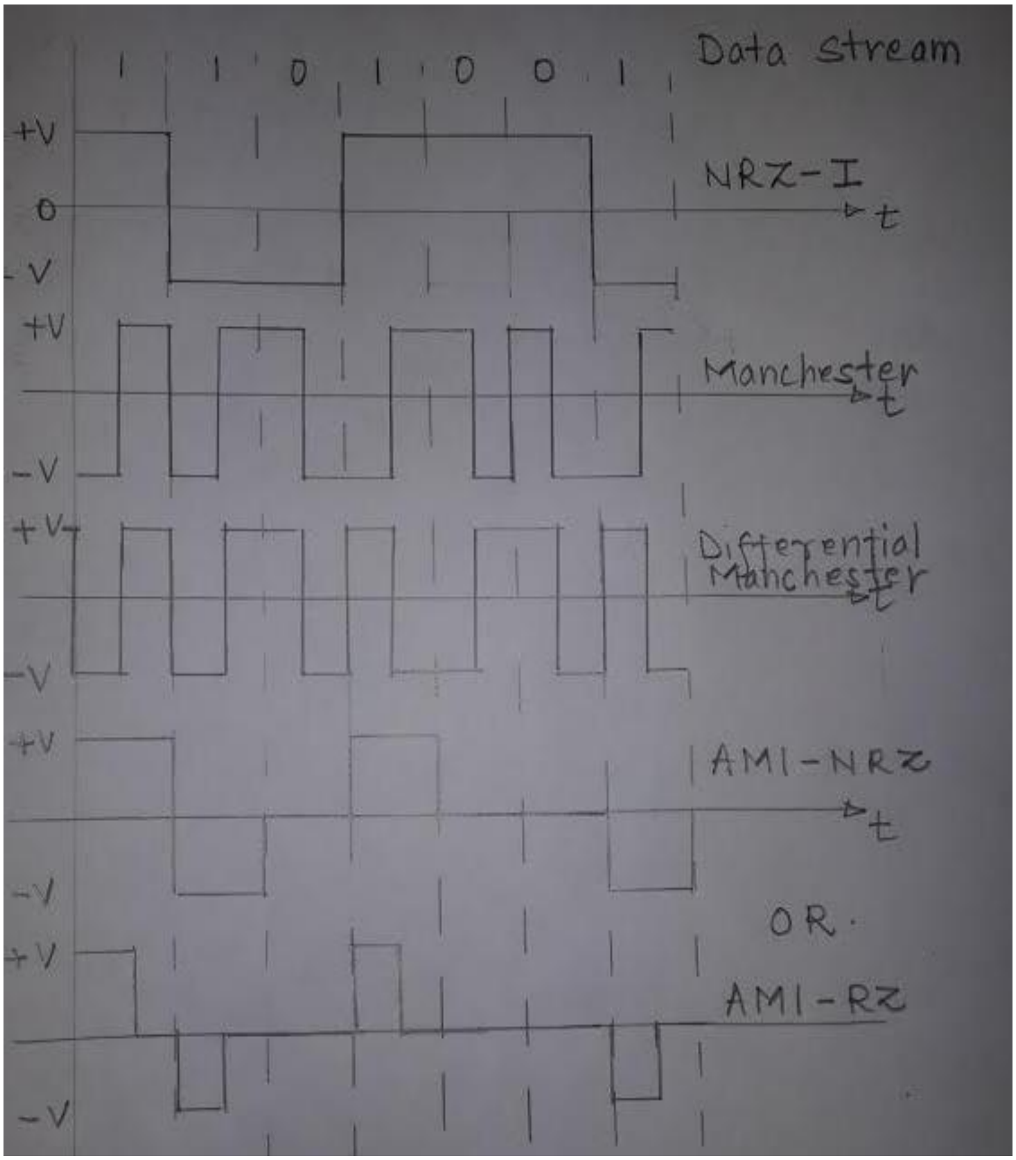
Let initial o/p =100 (any initial value can be taken)
Table is as shown below-

CLK No.	Shift Register Output			Ex-OR output	PN sequence Q3
	Q3	Q2	Q1	Q3 + Q1 = Q1	Q3
0	0	0	1	1	0
1	0	1	1	1	0
2	1	1	1	0	1
3	1	1	0	1	1
4	1	0	1	0	1
5	0	1	0	0	0
6	1	0	0	1	1
7	0	0	1	0	0

PN Sequence of length 7 generated is 0101110

d) Draw NRZ-I, Manchester, differential Manchester and AMI waveforms of line codes for data stream 1101001.

Ans. **(1 Marks each for each line code)**



Q-4 (B) Attempt any one. (6 Marks)

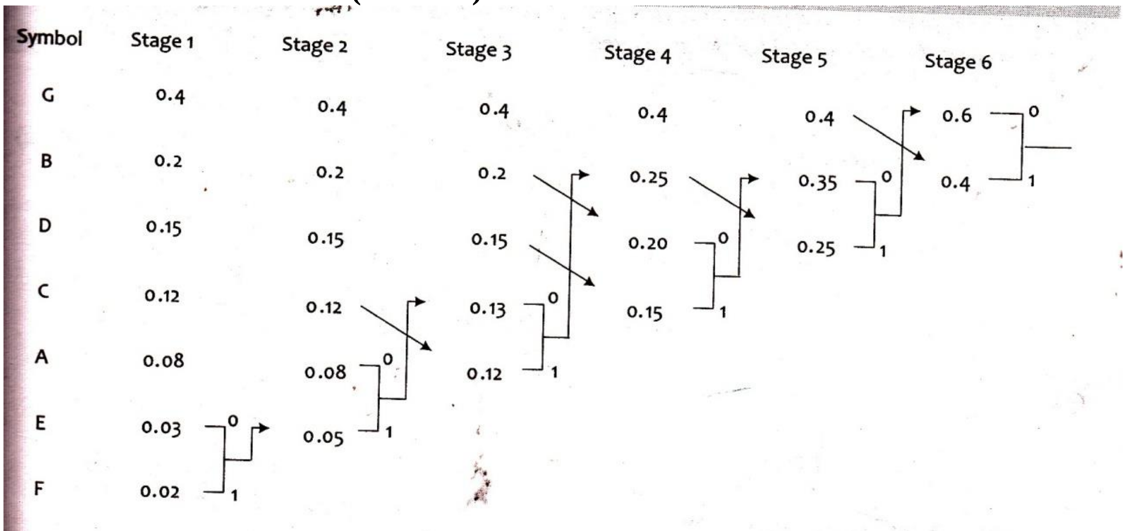
a) A discrete memory less source has the letters A, B, C, D, E, F and G with corresponding probabilities{0.08, 0.2, 0.12, 0.15, 0.03, 0.02, 0.4}

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

ii) Determine the coding efficiency of the Huffman code designed.

Ans:

i) Huffman code- (4 marks)



The code words are listed below:

A — 0100; B — 000; C — 011; D — 001; E — 01010; F — 01011; G — 1

1) The average length of the code word (1 marks)

2) The coding efficiency of the Huffman code (1 marks)

Hence, the average length of a code word is

$$\bar{n} = \sum_{i=0}^{M-1} p(x_i)n_i = 0.08 \times 4 + 0.2 \times 3 + 0.12 \times 3 + 0.15 \times 3 + 0.03 \times 5 + 0.02 \times 5 + 0.4 \times 1$$

$$= 2.38$$

$$\therefore \quad \bar{n} = 2.38 \text{ bits/symbol}$$

(ii) $H(S)$ = Entropy of the source

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

$$= 0.08 \log_2 \frac{1}{0.08} + 0.2 \log_2 \frac{1}{0.2} + 0.12 \log_2 \frac{1}{0.12} + 0.15 \log_2 \frac{1}{0.15}$$

$$+ 0.03 \log_2 \frac{1}{0.03} + 0.02 \log_2 \frac{1}{0.02} + 0.4 \log_2 \frac{1}{0.4}$$

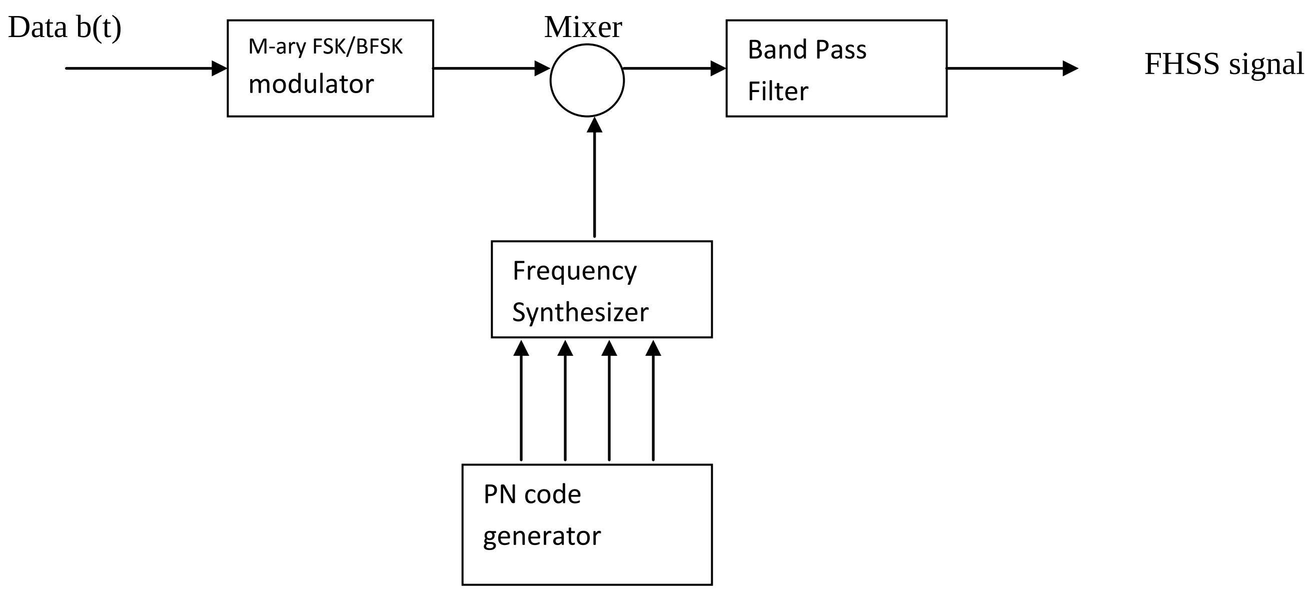
$$= 2.31 \text{ bits/symbol}$$

$$\therefore \text{coding efficiency} = \frac{2.31}{2.38} \times 100 = 97.135\%$$

b) Draw the block diagram of a BFSK/FHSS transmitter and explain its working. State any two advantages.

Ans.

Block diagram of a BFSK/FHSS transmitter. **(2 marks)**



Explanation (2 marks)

The binary data sequence is applied to the M-ary FSK modulator. The output of M-ary FSK is mixed with the frequency synthesizer output. The frequency synthesizer decides the hopping patters of the system.

The output of mixer is the stream of two frequencies. Sum and the difference of both the inputs to it.

The frequency of the mixer input obtained from MFSK modulator is changing continuously.

Other input to the mixer is obtained from the digital frequency synthesizer.

The synthesizer output at a given instant of time is the frequency hop.

Frequency hop at the output of synthesizer are controlled by the successive bit at the output of PN code generator.

The band pass filter is centered at the sum frequency band and rejects all other components. This sum components of the frequency are then retransmitted as FHSS signal.

In slow frequency hopping the symbol rate Rs of the MFSK signal is an integer multiple of the hop rate Rn that means several symbols are transmitted corresponding to each frequency hop.

Each frequency hop: → several symbols

Here frequency hopping takes place slowly.

Advantages **(Any 2 - 2 marks)**

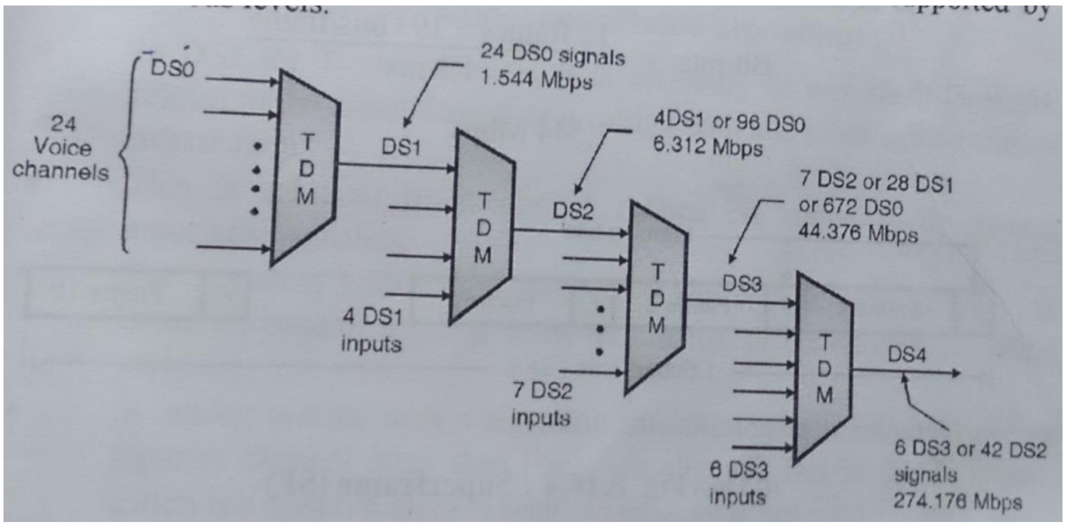
1) The synchronization is not greatly dependent on the distance.
2) The serial search system with FH-SS needs shorter time for acquisition.
3) The processing gain PG is higher than that of DS-SS system.

Q-5 Attempt any two the following (16 Marks)

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

Ans:

Diagram- 4 marks



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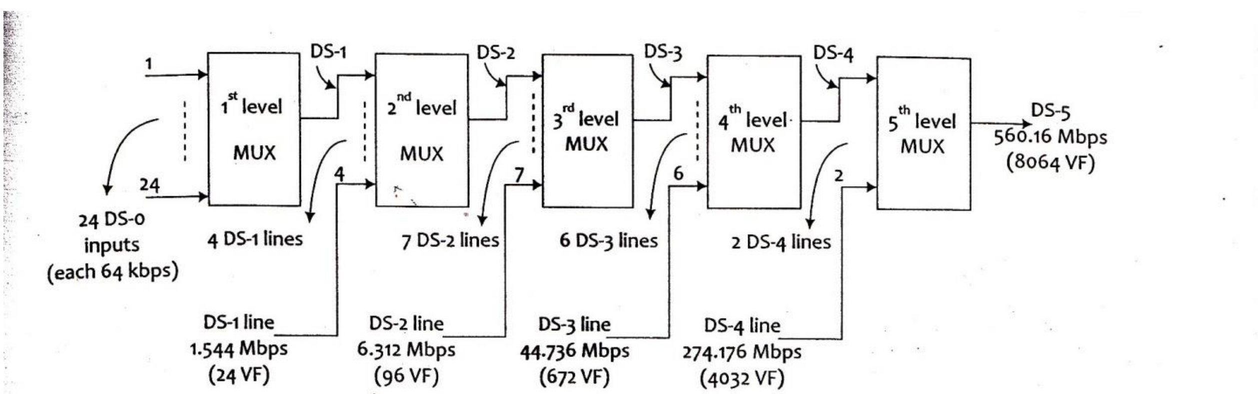
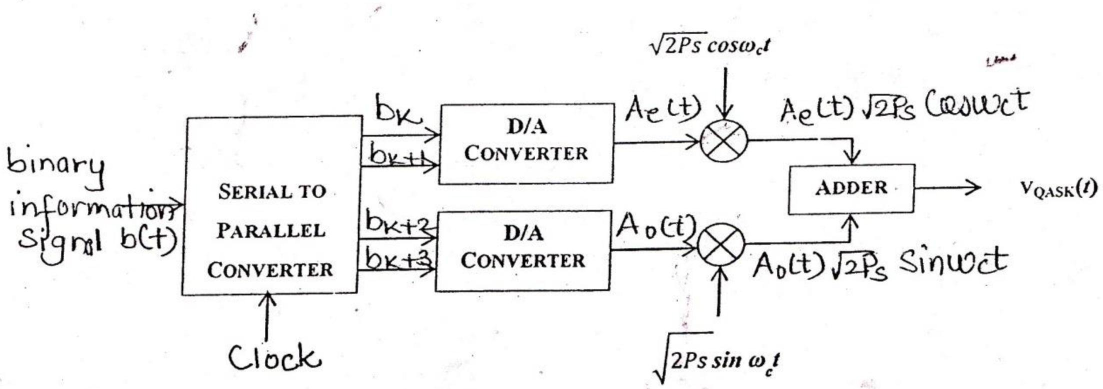


Explanation (4 marks)

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.

b) With the help of block diagram explain the working principle of QAM system.

Ans: **Block diagram (Tx) (2 marks)**



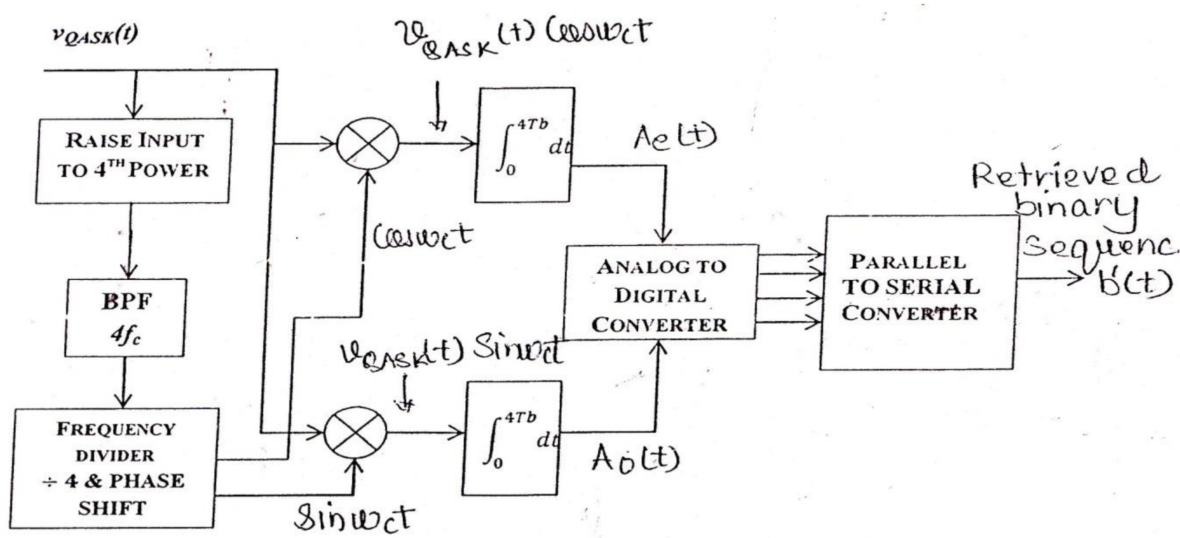
Explanation (2 marks)

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,

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

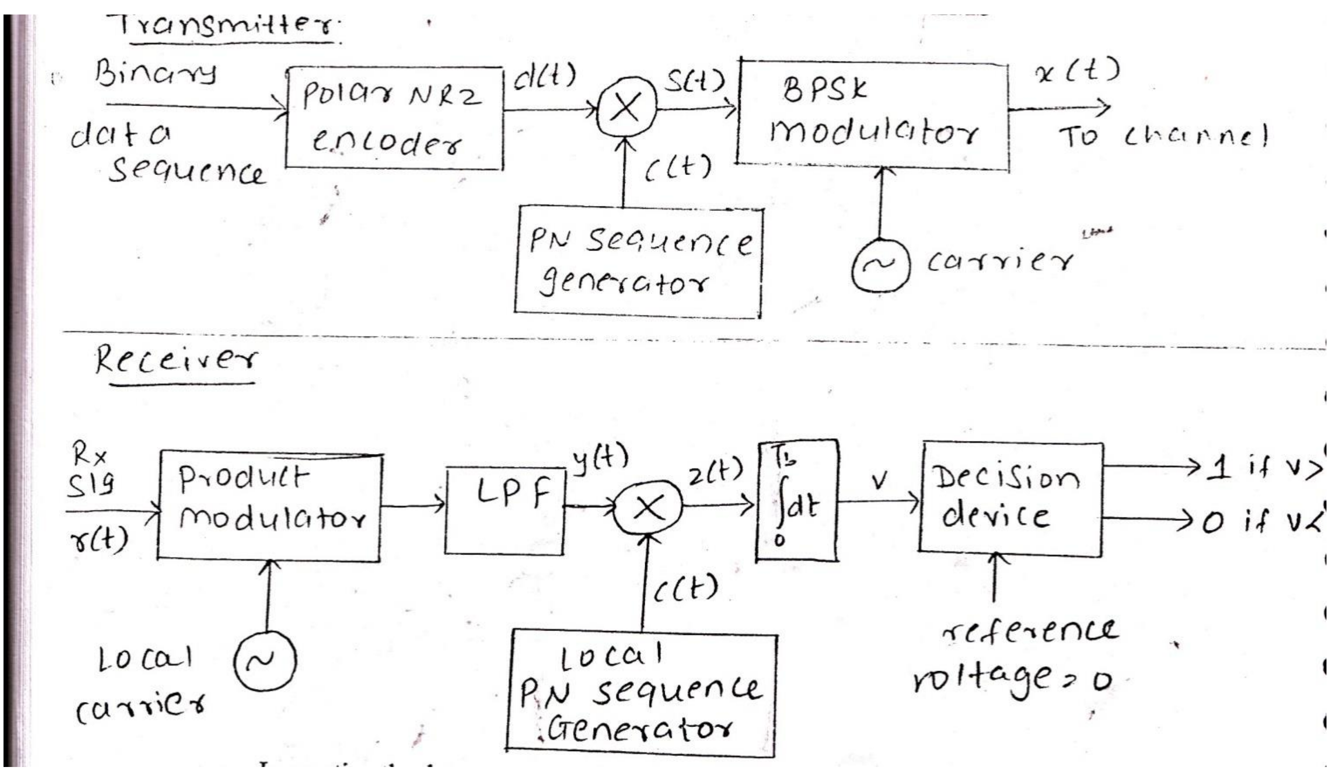
Block diagram (Rx) (2 marks)



Explanation **(2 marks)**

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

c) Draw block diagram of direct sequence spread spectrum and explain its working principle.
Ans: (Diagram 4 marks, explanation 4 marks)



The averaging system reduces the interference by averaging at over a long period. The DSSS system is a averaging system. This technique can be used in practice for transmission of signal over a band pass channel (E.g. satellite channel). For such application the coherent binary phase shift (BPSK) is used in the transmitter and receiver.
The binary sequence b (t) is given to the NRZ encoder. The b (t) is converted NRZ signal d (t). The NRZ signal d (t) is used to modulate the PN sequence c (t) generated by the PN code generator.

The multiplier multiply the signal b (t) * c (t) = s (t). The s (t) signal is given to binary PSK modulator.

The modulated signal at the output of product modulator or multiplier i.e. s (t) is used modulate the 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 $\sqrt{2Ps}\sin 2\pi f_C t$.
The output of BPSK modulator x (t) is transmitted x (t) = s (t) * $\sqrt{2Ps}\ 2\pi f_C t$.
But m (t) = ± 1
Therefore x (t) = $\pm\sqrt{2Ps}\sin 2\pi f_C t$
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,

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

Q-6 Attempt any four (16 Marks)

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

Ans:

Advantages: any 2(2 Marks)

1. PCM has very high noise immunity.
2. Repeaters can be used between the transmitter and the receiver which can further reduce the effect of noise.
3. It is possible to store the PCM signal due to its digital nature.
4. It is possible to use various coding techniques so that only the desired receiver (user) can decode the message.

Disadvantages: any 2(2 Marks)

1. The encoding decoding & quantizing circuitry of PCM is complex.
2. PCM requires a large BW as compared to other systems.

b) State the principle of orthogonality. Explain the concept of single carrier and multi carrier system.

Ans:

Principle (2 Mark)

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

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

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

Single carrier system: (1 mark)

In order to use the available radio spectrum efficiently, in single carrier system, the modulated sub carrier should be placed as close to each other as possible without causing interference. Guard bands are required to be inserted between adjacent spectrum to avoid interference but these increases bandwidth & reduce spectrum efficiency.

Multi carrier system: (1 mark)

Both the problem of multicarrier system can be solved by using a multicarrier system. OFDM divides available spectrum into many sub-channel. Then by making all the sub channel narrowband, it is ensured that all channel experience flat fading. This makes equalizing very easy. In OFDM a DSP based technique is used which allows a overlapping of adjacent spectrum without causing interference.

c) Describe M-ary encoding. State any two advantages and one disadvantages of it.

Ans: **(Explanation- 1mark; 2 Advantages -2mark, 1disadvantages-1 mark)**

- M-ary modulation is a technique of modulation in which N bits are combined together to form M symbols ($2^N = M$) and a signal is transmitted corresponding to each symbol for a duration of $NT_b = T_s$. the signal is generated by changing the amplitude, phase or frequency of a sinusoidal carrier in discrete steps. Thus M-ary modulation / signaling schemes can be categorized into the following types:

1. M-ary ASK
2. M-ary PSK
3. M-ary FSK

- Advantages of M-ary scheme over the binary scheme are as follows:

1. Conservation of channel bandwidth.
2. Utilization of the additional bandwidth to provide increased noise immunity.
3. Increase in system performance.

- Disadvantages of the M-ary scheme are a s follows:

1. Increase in the transmitted power.
2. Increase in error probability

d) With example explain how hamming code is used for single bit error correction.
Ans:**(Explanation with proper example 4 marks)**

Hamming codes are basically linear block codes. It is an error correcting code.The parity bits are inserted in between the data bits as shown below.

D7	D6	D5	P4	D3	P2	P1	
7bit	6bit	5bit	4bit	3bit	2bit	1bit	7-bit hamming code

Where D-data bits and P- parity bits. The hamming coded data is then transmitter. At the receiver it is coded to get the data back.

The bits (1, 3, 5, 7), (2, 3, 6, 7) and (4, 5, 6, 7) are checked for even parity or odd parity, if all the 4-bit groups mentioned above possess the even parity (or odd parity) then the received code word is correct but if the parity is not matching then error exist. Such error can be located by forming a three bit number out of three parity checks. This process can be well explained by following example,

For example: Suppose a 7-bit hamming code is received as 1110101 (for transmitter data 1111) and parity used is assumed to be even .hence we can detect and correct the code as

Step1: Received 7bit hamming code is applied to hamming code format as

D7	D6	D5	P4	D3	P2	P1
1	1	1	0	1	0	1

Step2: Check bits for P4 bit

i.e. P4 D5 D6 D7

0 1 1 1	= odd parity hence error

So, P4=1

Step 3: check bits for P2bit

i.e. P2 D3 D6 D7

0 1 1 1	= odd parity hence error

So, P2=1

Step 4: check bits for P1 bit

i.e. P1 D3 D5 D7

1 1 1 1	= even parity hence no error

So, P1= 0

Hence the error word is E = 1 1 0

Step 5: decimal equivalent of 110 is 6 hence 6^{th}bit is incorrect so invert it and the correct code word will be,

D7	D6	D5	P4	D3	P2	P1
1	0	1	0	1	0	1

Hence by using this method we can detect as well as correct the error in the transmitted coed word. But it can locate a single bit error and fails in detecting the burst error.

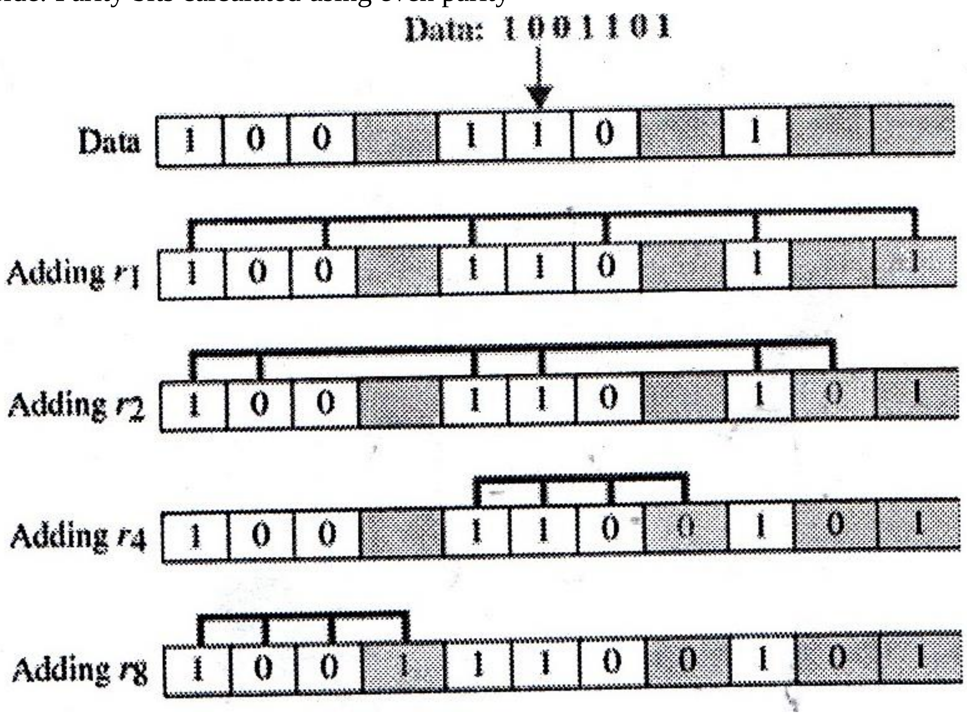
OR

Hamming codes are basically linear block codes. It is an single bit error correcting code. The parity bits are inserted in between the data bits as shown below.

Let the data be 1001101

So the number of parity bits to be added are 4

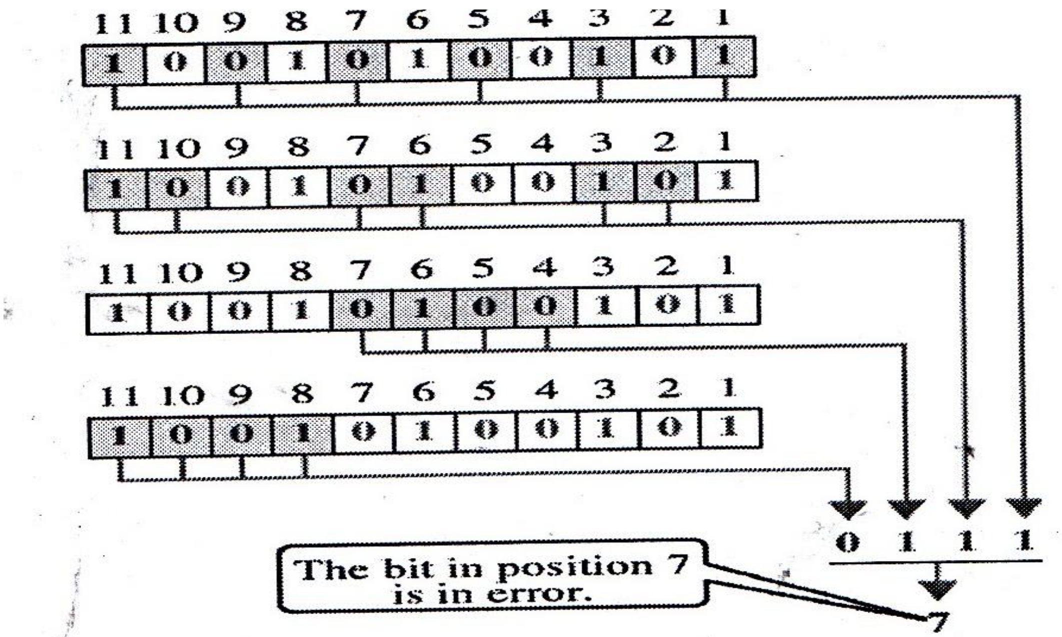
At Transmitter side: Parity bits calculated using even parity



Transmitted data is 10011100101

At Receiver side : Let the received data be 10010100101.Again parity bits are calculated .

The decimal value of parity bit combination gives us the bit position where the error occurred.



Hence the single bit error can be corrected using hamming code.

e) Compare QAM and QPSK (any four points).
Ans:**(4 points- 1 Mark each)**

PARAMETERS	QAM	QPSK
Information is transmitted by change in	Amplitude & phase	Phase
No. of bits per symbol	N=3 or 4 or 5 and so on	N=2
No. of possible symbols $M=2^N$	$M=2^N$	Four
Detection method	Coherent	coherent
Minimum BW	2Fb/N	Fb
Symbol duration	NTb	2Tb