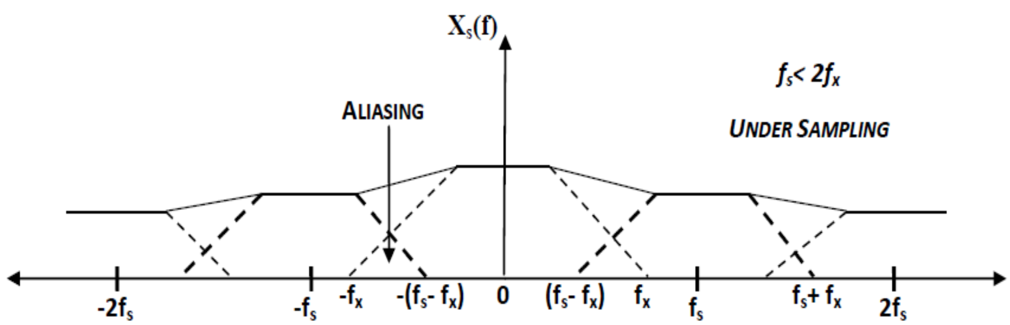
WINTER– 18 EXAMINATION

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

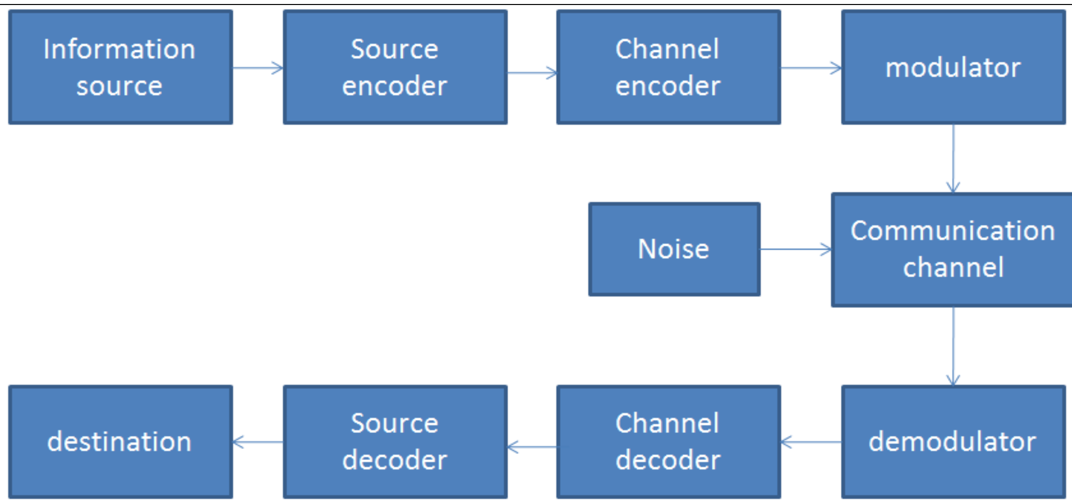
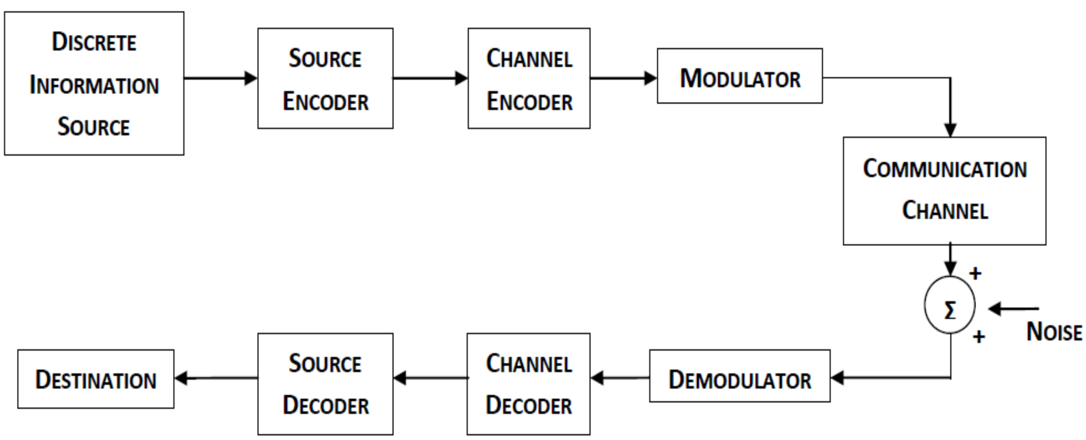
Important Instructions to examiners:

1) The answers should be examined by key words and not as word-to-word as given in themodel answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may tryto assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given moreImportance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in thefigure. The figures drawn by candidate and model answer may vary. The examiner may give credit for anyequivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constantvalues 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.

Q. No.	Sub Q. N.	Answers	Marking Scheme
1	(A)	**Attempt any THREE:**	**12- Total Marks**
	(a)	**Define bit rate and baud rate. Write relationship between them.**	**4M**
	Ans:	**Bit rate:** Bit rate is defined as the number of bits transmitted per second that is bits/sec. **Bit rate** = $\frac{1}{Bit\ interval}$ **Baud rate**: Baud rate is defined as the number of signal units per second or it is defined as rate of symbol transmission that is baud/sec. **Relation:** • Baud rate is less than or equal to the bit rate. • When binary bits are transmitted as electrical signal with two levels '0' and '1' then the bit rate and baud rate are same. • When a signal unit is composed of 2 or more bits then baud rate is less than bit rate. • **Baud rate** = $\frac{Bit\ rate}{Number\ of\ bits\ per\ signal\ unit}$	**2M for two definitions,** **2M for relation**

	(b)	**State aliasing effect. Draw neat diagram showing aliasing effect and hoe it can be overcome.**	**4M**
	Ans :	**Aliasing Effect :** • If signal x(t) is not strictly bandlimited or if the sampling frequency fs is less than 2fm, then an error occurs called as aliasing. • If the sampling rate *fs< 2fx (Under Sampling)*, then the sidebands of the signal overlap and high frequencies of x(t) are reflected into low frequencies in x(f) and information signal *x(t)* cannot be recovered without distortion from sampled signal, *Xs(f)*. • This phenomenon of a high frequency in the spectrum of the original signal x(t) taking an identity of lower frequency in the spectrum of the sampled signal Xs(f) is called aliasing or fold over error. • Here the sideband frequency from one harmonic will fold-over or overlap with the sideband frequency of another harmonic as shown in fig.  **Elimination of Aliasing:** 1) Using antialiasing filter: Use a bandlimiting low pass filter and pass the signal x(t) through it before sampling. This filter has cut off frequency fc which is same as maximum modulating signal frequency which will bandlimit the signal. 2) Keeping sampling frequency fs > 2fm: Increase the sampling frequency fs to great extend fs >> 2fm. Due to this even if signal is not bandlimited still spectrum will not overlap due to presence of guard band between the adjacent spectrum.	**State-1M** **Diagram-1M** **Overcome -2M** **(1M for each overcome technique)**
	(c)	**What is multiplexing and state its need.**	**4M**
	Ans :	**Multiplexing:-** • Multiplexing is the process of simultaneously transmitting multiple signals over a single communication channel.	**Definition-multiplexing-02M**

		OR • Multiplexing divide the physical line or a medium into logical segment called channel. **Need of multiplexing:** • As the data and telecommunications usage increases, so does the traffic. We can accommodate this increase by installing higher capacity links and use each to carry multiple signals. • 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 which is practically impossible. • 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. • If the bandwidth of the link is greater than the transmission needs of the devices connected to it, the excess capacity is wasted; this bandwidth utilization is possible by using multiplexing. • Bandwidth is one of the most precious resources in data communications, multiplexing maximizes the utilization of this resource.	**Any two Needs of multiplexing:2 M**
	(d)	**Define PN sequence. Comment on maximum length sequence.**	**4M**
	Ans:	**PN sequence**: A PN sequence is defined as a pseudorandom coded sequence of 1's and 0's with certain auto correlation properties. Or **PN sequence** is a periodic binary code which is random in nature generated by the use of shift registers, but generated with taking into considerations some generator polynomials. **Maximum length sequence:** • Maximum length of PN Sequence 'L' is the no. of bits in a PN sequence • Maximum length of PN Sequence 'L' it depends upon the number of flip-flops 'n' used for the PN Sequence generator and given $L=2^n -1$. • For example if PN sequence generator uses 3 flip flops then Maximum length sequence is • $L= 2^n -1 = 2^3- 1 = 7$. • PN sequence repeats after every maximum length sequence. • PN sequence is also known as Maximal Length Sequences. • PN sequence code is orthogonal in nature. • PN sequences are used in spread spectrum systems, like CDMA, WCDMA, and Radar etc. • PN sequences follow the correlation property.	**Definition-2M** **Any two comments of Maximum length sequence-2M**
	(B)	**Attempt any ONE :**	**06- Total**

			Marks
	(a)	**Draw the block diagram of digital communication system and illustrate its working.**	**6M**
	Ans :	 **OR**  **INFORMATION SOURCE:** • An *Information source* generates a message, examples of which include human voice, television picture, teletype data, atmospheric temperature and pressure. • The message signal can be of an *analog* or *digital* type. An analog signal can be converted into digital form through the process of *sampling*, *quantizing* and *encoding*. • In a digital signal, on the other hand, both amplitude and time take on *discrete values*. Computer data and telegraph signals are examples of digital signals.	**Block diagram-3M** **Brief Working or function of each block-3M**

SOURCE ENCODER:

- The input to the source encoder (also referred to as the source coder) is a string of symbols occurring at a rate *symbols/sec*.
- The source encoder converts the symbol sequence into a binary sequence of 0's and 1's by assigning code words to the symbols in the input sequence by using either assigning fixed-length binary code word to each symbol or assigns variable-length code words to these blocks.
- Second function it performs is data compression. It reduces the redundancy by performing a one-to-one mapping of its input bit stream into another bit stream at its output but with fewer digits.

CHANNEL ENCODER:

- The channel coder provides some amount of error controlled capability to the data to be transmitted.
- It adds some extra bits to the output of the source coder. While these extra bits themselves convey no information, they make it possible for the receiver to detect and/or correct some of the errors in the information bearing bits.
- This is needed because the data gets corrupted by the additive noise on the channel and this gives rise to the possibility of the channel decoder committing mistakes in decoding the data received from the channel.

MODULATOR:

- The modulator accepts a bit stream as its input and converts it to an electrical waveform suitable for transmission over the communication channel as they are basically analog in nature.
- Modulation can be effectively used to minimize the effects of channel noise, to match the frequency spectrum of the transmitted signal with channel characteristics, to provide the capability to multiplex many signals and to overcome some equipment limitations.

COMMUNICATION CHANNEL:

- The communication channel provides the electrical connection between the source and the destination.
- The channel may be a pair of wire or a telephone link or free space over which the information bearing signal is radiated.
- Due to physical limitations, communication channels have only finite bandwidth (*B* Hz) and the information bearing signal often suffers amplitude and phase distortion as it travels over the channel.
- While some of the degrading effects of the channel can be removed or compensated for, the effects of noise cannot be completely removed.

DEMODULATOR:

- Modulation is a reversible process and the extraction of the message from the information bearing waveform produced by the modulator is accomplished by

		the demodulator. • There are a variety of techniques available for demodulating a given modulated waveform; the actual procedure used determines the equipment complexity needed and the accuracy of demodulation. **CHANNEL DECODER:** • The channel decoder recovers the information bearing bits from the coded binary stream. Error detection and possible correction is also performed by the channel decoder. • The decoder operates either in a block mode or in a continuous sequential mode depending on the type of coding used in the system. **SOURCE DECODER:** • At the receiver, the source decoder converts the binary output of the channel decoder into a symbol sequence. • The decoder for a system using fixed-length coding is quite simple, but the decoder for a system using variable-length coding will be very complex. • Decoders for such systems must be able to cope with a number of problems such as growing memory requirements and loss of synchronization due to bit errors.	
	(b)	**Generate Hamming code for binary data 110110 at the transmitter and decode the coded sequence at the receiver.**	**6M**
	Ans :	**(Note: Student can solve by assuming odd parity. So consider the logic and understanding of the student. Also while writing code students may write from left to right or from right to left, consider both.)** **By Assuming Even Parity:** In the Hamming code, *k* parity bits are added to *n*-bit data word, forming a new word of *n* + *k* bits. The bit positions are numbered in sequence from 1 to *n* + *k*. Those positions numbered with powers of two are reserved for the parity bits. The remaining bits are the data bits. The given 6-bitdata word is 110110. We include four parity bits with this word and arrange the 10bits as follows:	**Generation-3M** **Decoding-3M** **(in generation stepwise marks to be given in decoding stepwise marks to be given)**

P1	P2	D1	P4	D2	D3	D4	P8	D5	D6
		1		1	0	1		1	0
1	2	3	4	5	6	7	8	9	10

The 4 parity bits *P*1 through *P*8 are in positions 1, 2, 4, and 8, respectively. The 6 bits of the data word are in the remaining positions. Each parity bit is calculated as follows:

$P1$=XOR of bits (3, 5, 7, 9) = $1 \oplus 1 \oplus 1 \oplus 1 = 0$

$P2$= XOR of bits (3, 6, 7, 10) = $1 \oplus 0 \oplus 1 \oplus 0 = 0$

$P4$= XOR of bits (5, 6, 7) = $1 \oplus 0 \oplus 1 = 0$

$P8$= XOR of bits (9, 10) = $1 \oplus 0 = 1$

The 6-bit data word is written into the memory together with the 4 parity bits as a 10-bit composite word. Substituting the 4 parity bits in their proper positions, we obtain the 10-bit composite word as a generated final Hamming code.

P1	P2	D1	P4	D2	D3	D4	P8	D5	D6
0	0	1	0	1	0	1	1	1	0
1	2	3	4	5	6	7	8	9	10

Final Hamming code at Transmitter is: 0 0 1 0 1 0 1 1 1 0

Decoder:

When the 10 bits are read from memory, they are checked again for errors. The parity of the word is checked over the same groups of bits, including their parity bits. The four check bits are evaluated as follows:

$C1$ = XOR of bits (1, 3, 5, 7, 9) = $0 \oplus 1 \oplus 1 \oplus 1 \oplus 1 = 0$

$C2$ =XOR of bits (2, 3, 6, 7, 10) = $0 \oplus 1 \oplus 0 \oplus 1 \oplus 0 = 0$

$C4$ = XOR of bits (4, 5, 6, 7) = $0 \oplus 1 \oplus 0 \oplus 1 = 0$

$C8$ = XOR of bits (8, 9, 10) = $1 \oplus 1 \oplus 0 = 0$

In this case, there is no error in the 10-bit word, as C = C8 C4 C2 C1 = 0000=0.
A 0 check bit designates an even parity over the checked bits, and a 1 designates an odd parity. Since the bits were written with even parity, the result, $C = C8C4C2C1$ =0000, indicates that no error has occurred. However, if $C \neq 0$, the 4-bit binary number formed by the check bits gives the position of the erroneous bit if only a single bit is in error.
If there is an error in bit position number 5 because it changed from 1 to 0.

P1	P2	D1	P4	D2	D3	D4	P8	D5	D6
0	0	1	0	0	0	1	1	1	0
1	2	3	4	5	6	7	8	9	10

Then,

C_1 = XOR of bits (1, 3, 5, 7, 9) = $0 \oplus 1 \oplus 0 \oplus 1 \oplus 1 = 1$

C_2 =XOR of bits (2, 3, 6, 7, 10) = $0 \oplus 1 \oplus 0 \oplus 1 \oplus 0 = 0$

C_4 = XOR of bits (4, 5, 6, 7) = $0 \oplus 0 \oplus 0 \oplus 1 = 1$

C_8 = XOR of bits (8, 9, 10) = $1 \oplus 1 \oplus 0 = 0$

***The result C =C8C4C2C*1 = 0101, indicates that error has occurred.**

Hence, when $C \neq 0$, the decimal value of *C* gives the position of the bit in error. Here **C=0101=5. This indicates 5th bit is in error.** The error can then be corrected by complementing the corresponding bit. An error can occur in the data or in one of the parity bits.

OR

The given data is 6-bit, 110110. Hence the Hamming code will have 6-data bit and 4 parity bits. The format is as follows:

D5	D4	P8	D3	D2	D1	P4	D0	P2	P1
1	1		0	1	1		0		
10	9	8	7	6	5	4	3	2	1

Each parity bit is calculated as follows:

$P1$=XOR of bits (1, 3, 5, 7, 9) = $P1 \oplus 0 \oplus 1 \oplus 0 \oplus 1 = 0$

$P2$= XOR of bits (2, 3, 6, 7, 10) = $P2 \oplus 0 \oplus 1 \oplus 0 \oplus 1 = 0$

$P4$= XOR of bits (4, 5, 6, 7) = $P4 \oplus 1 \oplus 1 \oplus 0 = 0$

$P8$= XOR of bits (8, 9, 10) = $P8 \oplus 1 \oplus 1 = 0$

The final hamming code is:

D5	D4	P8	D3	D2	D1	P4	D0	P2	P1
1	1	0	0	1	1	0	0	0	0
10	9	8	7	6	5	4	3	2	1

At the decoder check bits are calculated by considering the bits as follows including parity bits.

When the 10 bits are read from memory, they are checked again for errors. The parity of the word is checked over the same groups of bits, including their parity bits. The four check bits are evaluated as follows:

$C1$ = XOR of bits (1, 3, 5, 7, 9) = $0 \oplus 0 \oplus 1 \oplus 0 \oplus 1 = 0$

$C2$ = XOR of bits (2, 3, 6, 7, 10) = $0 \oplus 0 \oplus 1 \oplus 0 \oplus 1 = 0$

$C4$ = XOR of bits (4, 5, 6, 7) = $0 \oplus 1 \oplus 1 \oplus 0 = 0$

$C8$ = XOR of bits (8, 9, 10) = $0 \oplus 1 \oplus 1 = 0$

This indicates no error in the received code as C = C8 C4 C2 C1 =0 0 0 0 = 0.

A 0 check bit designates an even parity over the checked bits, and a 1 designates an odd parity. Since the bits were written with even parity, the result, $C = C8C4C2C1 = 0000$, indicates that no error has occurred. However, if $C \neq 0$, the 4-bit binary number formed by the check bits gives the position of the erroneous bit if only a single bit is in error.

For error detection and correction :

Suppose bit 7 is in error , bit 7 changes from 0 to 1 and the received code word is:

1 1 0 1 1 1 0 0 0 0

D5	D4	P8	D3	D2	D1	P4	D0	P2	P1
1	1	0	1	1	1	0	0	0	0
10	9	8	7	6	5	4	3	2	1

As it is even parity considered at the transmitter so same should be at receiver i. e. even parity.

The four check bits are evaluated as follows:

$C1$ = XOR of bits (1, 3, 5, 7, 9) = $0 \oplus 0 \oplus 1 \oplus 1 \oplus 1 = 1$

$C2$ = XOR of bits (2, 3, 6, 7, 10) = $0 \oplus 0 \oplus 1 \oplus 1 \oplus 1 = 1$

$C4$ = XOR of bits (4, 5, 6, 7) = $0 \oplus 1 \oplus 1 \oplus 1 = 1$

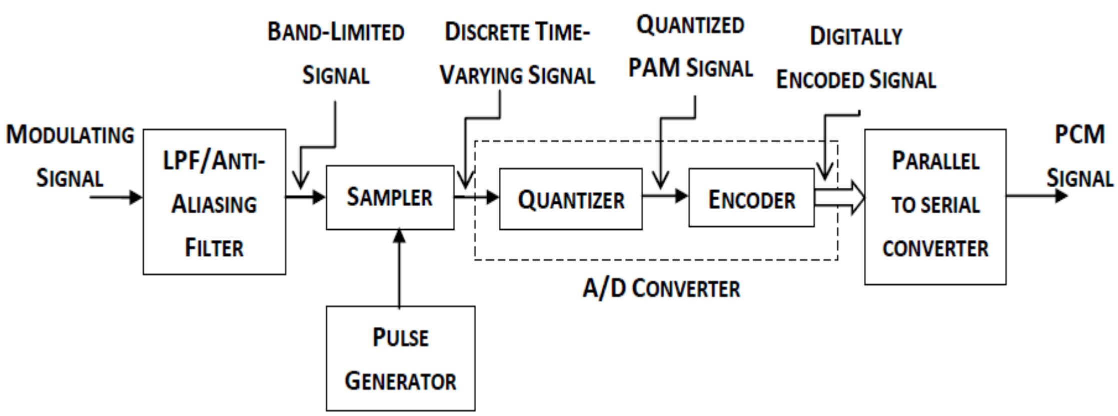
$C8$ = XOR of bits (8, 9, 10) = $0 \oplus 1 \oplus 1 = 0$

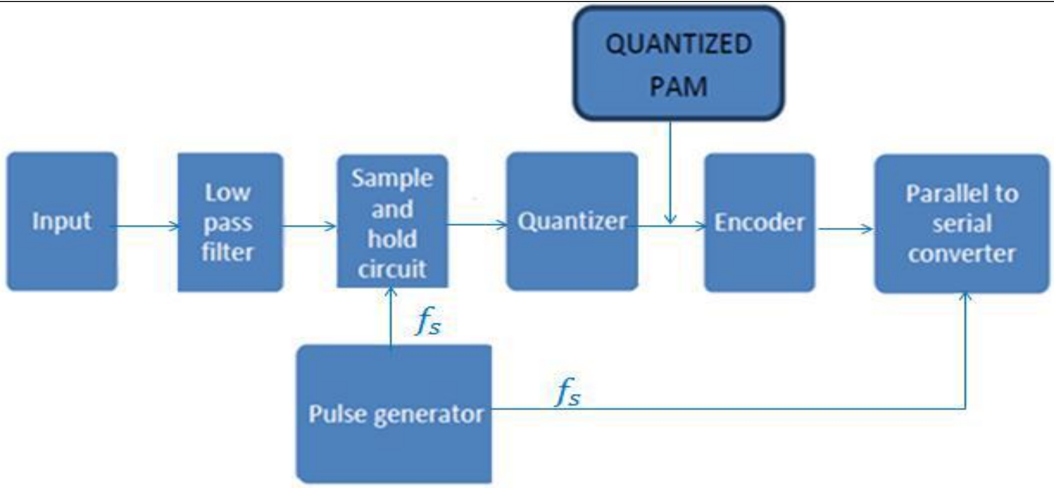
This indicates error in the received code as C = C8 C4 C2 C1 =0 1 1 1 = 7.

This shows 7^{th} bit is in error, by inverting the bit at 7^{th} position from 1 to 0 error can be corrected.

WINTER– 18 EXAMINATION

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

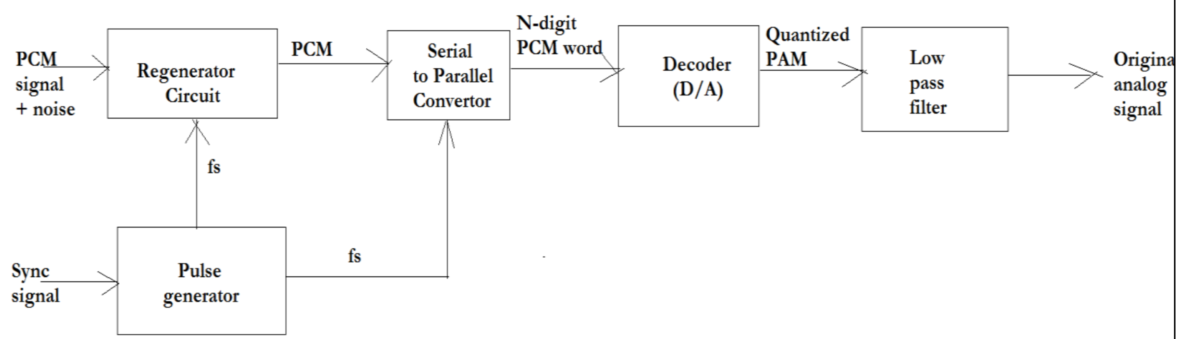
Q. No.	Sub Q. N.	Answers	Marking Scheme
2		**Attempt any TWO:**	**16- Total Marks**
	a)	**Describe the working of PCM transmitter and receiver. State advantage of PCM.**	**8M**
	Ans:	**Block diagram of PCM transmitter:**  **OR**	**PCM Transmitter-3M** **(diagram - 1.5M and explain-1.5M)** **PCM Receiver-3M** **(diagram 1.5M explain 1.5M)** **Any 2 Advantages-2M**



PCM Transmitter Explanation: 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 such that $Q = 2^N$ where Q is the total number of quantization levels.
- 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 serial converter which provides synchronization.

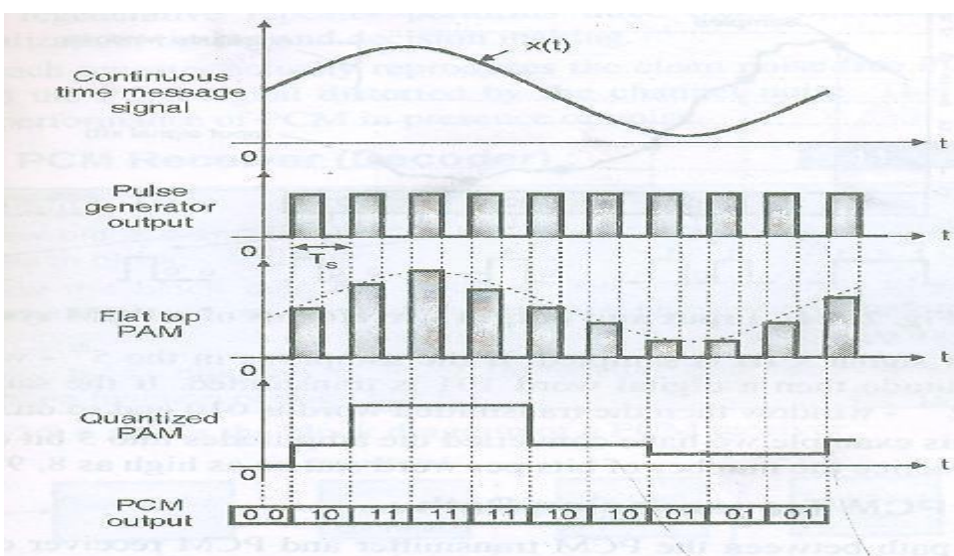
Block diagram of PCM Receiver:



PCM Receiver Explanation:

- 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.
- The pulse generator has to operate in synchronization with that at transmitter.
- 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).

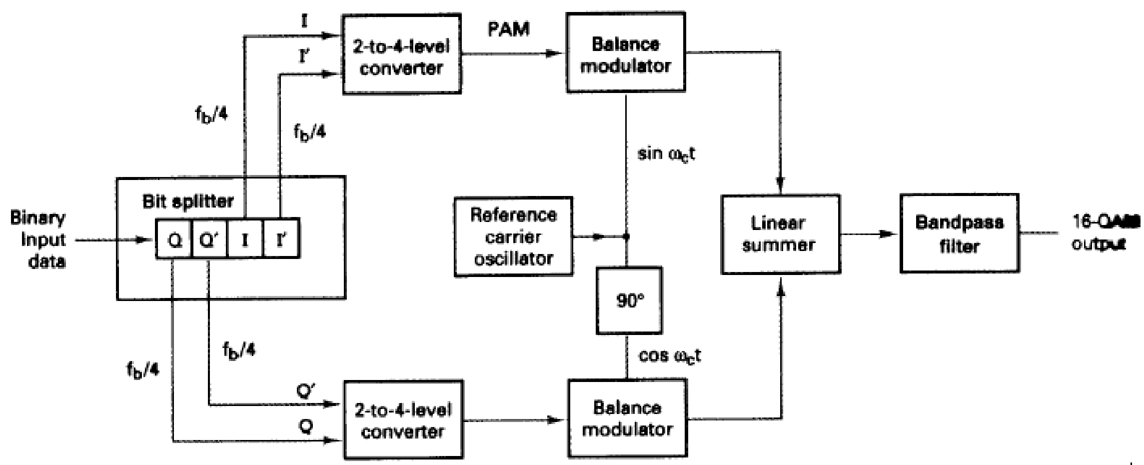
Waveforms:(Optional)



Advantages: (any 2)-2Marks

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 which provides security and privacy.	
	b)	**Draw the block diagram of 16 QAM system. Explain its working. Draw its constellation diagram.**	**8M**
		Block diagram of 16 QAM System:	**Diagram-3M** **Explain-3M** **Constellation diagram-2M**



Explaination:

- The input binary data are divided into four channels: I, I', Q, and Q'. The bit rate in each channel is equal to one-fourth of the input bit rate (fb/4).
- The I and Q bits determine the polarity at the output of the 2-to-4-level converters (a logic 1 = positive and a logic 0 =negative).
- The I' and Q' bits determine the magnitude (a logic 1 = 0.821 V and a logic 0 = 0.22 V).
- For the I product modulator they are +0.821 sinωct, -0.821 sinωct, +0.22 sinωct, and -0.22 sinωct.
- For the Q product modulator, they are +0.821 cosωct, +0.22 cosωct, -0.821 cosωct, and -0.22 cosωct.
- The linear summer combines the outputs from the I and Q channel product modulators and produces the 16 output conditions necessary for 16-QAM.

The truth table for the I and Q channel 2-to-4-level converters is as follows:

I	I'	Output
0	0	−0.22 V
0	1	−0.821 V
1	0	+0.22 V
1	1	+0.821 V

(a)

Q	Q'	Output
0	0	−0.22 V
0	1	−0.821 V
1	0	+0.22 V
1	1	+0.821 V

(b)

- For a quadbit input of I= 0, I' = 0, Q = 0, and Q' = 0 (0000),The inputs to the I channel 2-to-4-level converter are I = 0 and I' = 0. From truth table, the output is -0.22 V.
- The inputs to the Q channel 2-to-4-level converter are Q= 0 and Q' = 0.Again from truth table, the output is -0.22 V.
- Thus, the two inputs to the I channel product modulator are - 0,22 V and sin ωct. The output is I = (-0.22)(sin ωct) = -0.22 sin ωct
- The two inputs to the Q channel product modulator are -0.22 V and cos ωct. The output is Q = (-0.22)(cos ωct) = -0.22 cos ωct
- The outputs from the I and Q channel product modulators are combined in the linear summer and produce a modulated output of
- summer output = -0.22 sin ωct - 0.22 cos ωct= 0.311 sin(ωct - 135°)
- For the remaining quadbit codes, the procedure is the same. The results are

Binary input				16-QAM output	
Q	Q'	I	I'		
0	0	0	0	0.311 V	−135°
0	0	0	1	0.850 V	−165°
0	0	1	0	0.311 V	−45°
0	0	1	1	0.850 V	−15°
0	1	0	0	0.850 V	−105°
0	1	0	1	1.161 V	−135°
0	1	1	0	0.850 V	−75°
0	1	1	1	1.161 V	−45°
1	0	0	0	0.311 V	135°
1	0	0	1	0.850 V	165°
1	0	1	0	0.311 V	45°
1	0	1	1	0.850 V	15°
1	1	0	0	0.850 V	105°
1	1	0	1	1.161 V	135°
1	1	1	0	0.850 V	75°
1	1	1	1	1.161 V	45°

With a 16-QAM, the bit rate in the I, I', Q, or Q' channel is equal to one-fourth of the binary input data rate (fb/4).

Constellation diagram:

WINTER– 18 EXAMINATION

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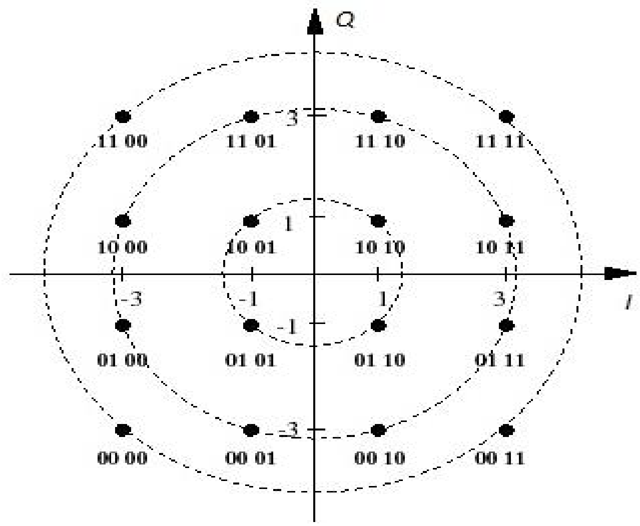
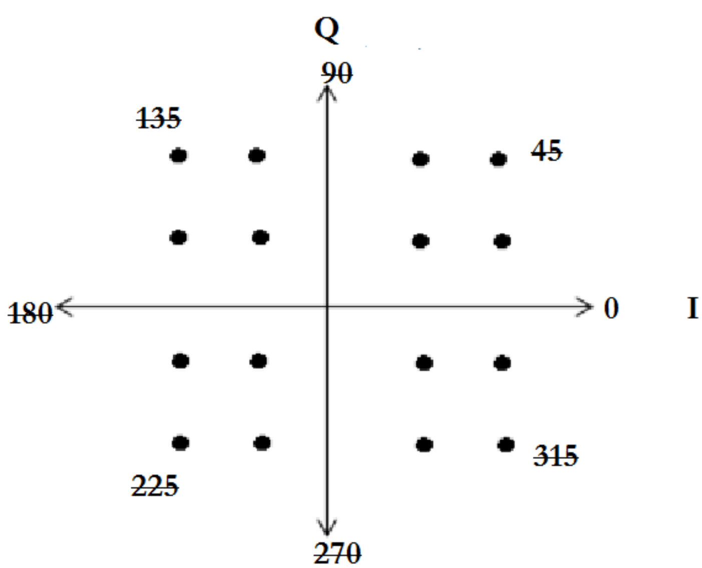
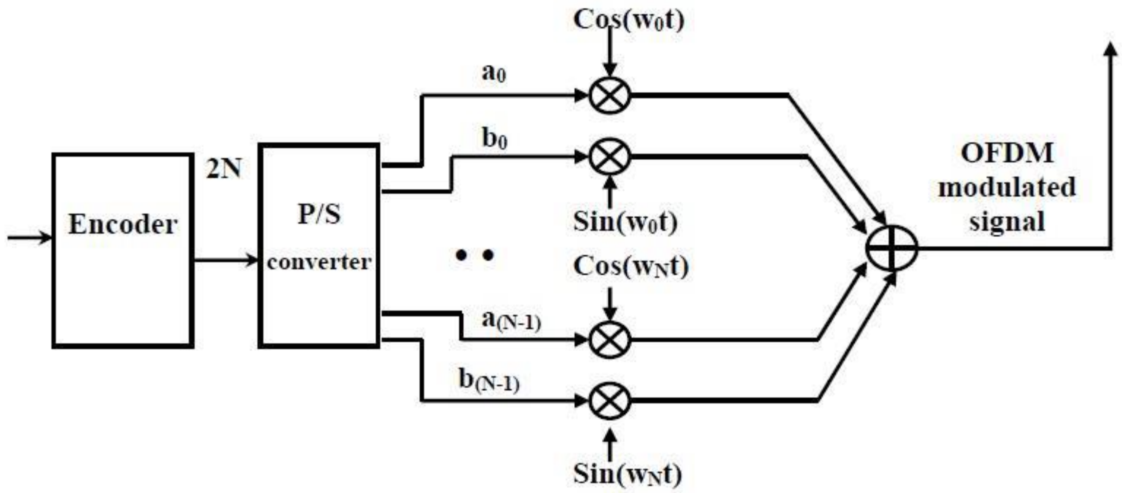
		 **OR** 	
	c)	**Draw the block diagram and explain the working of OFDM multicarrier system.**	**8M**
	Ans :	**Block diagram of OFDM:**	**Block diagram-4M** **Explain-4M**



OR

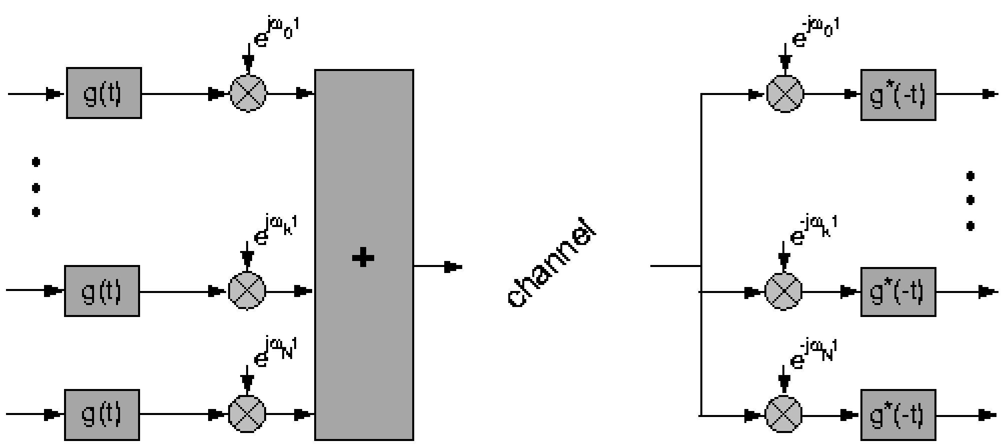
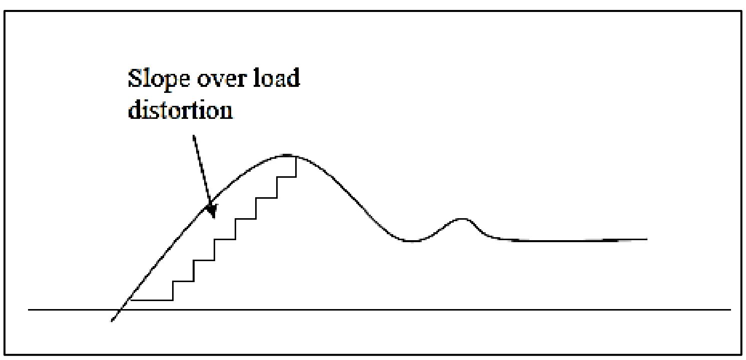


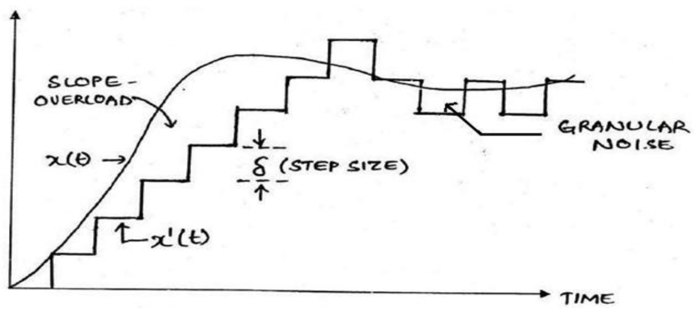
Figure :Basic structure of a OFDM multicarrier system

Explanation:

- OFDM stands for Orthogonal Frequency Division Modulation. It is based on the principle of orthogonality. Two signals are said to be orthogonal if they are independent of each other in specified time interval & do not interact with each other. It is possible to transmit multiple signals over a common channel without interference & get detected on the receiving end without interference.
- OFDM is single carrier and multicarrier system.
- In single carrier system: The complexity involved in removing ISI interference is tremendous so multi carrier approach has become so popular.
- In multi carrier approach: The original data stream of rate R is multiplexed into N parallel data streams Each of the data streams is modulated with a different frequency and the resulting signals are transmitted together in the same

		band. • Correspondingly the receiver consists of *N* parallel receiver paths. Due to the prolonged distance in between transmitted symbols the ISI for each sub system reduced. • Such little ISI can often be tolerated and no extra counter measure such as an equalizer is needed.	
Q. No.	**Sub Q. N.**	**Answers**	**Marking Scheme**
3		**Attempt any FOUR :**	**16- Total Marks**
	a)	**What is slope overload distortion. Explain how it can be avoided using ADM with neat waveform.**	**4M**
	Ans :	**Slope over load distortion:** If the input signal amplitude changes fast, the step by step accumulation process may not catch up with the rate of change as shown in Fig  **Fig : slope-overload problem** An intuitive remedy for this problem is to increase the step-size δ but that approach has	**2M for slope overload distortion** **&** **2M for avoidance with waveform**

another serious problem given below.



The delta modulation has major drawbacks as under:

(i) Slope overload distortion

- This distortion is arises because of large dynamic range of the input signal.
- As can be observed from figure the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it, the steep size „Δ‟ becomes too small for staircase signal x‟(t)to follow the step segment of x(t).
- Hence, there is a large error between the staircase approximated signal and the original input signal x (t).
- This error or noise is known as slope overload distortion.
- **Adaptive delta modulation technique**
- 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.
- To reduce slope overload error, the step size must be increased when slope of signal x (t) is high.
- Thus due to variable step size the in ADM, approximated signal follows the original signal and reduces the slope overload effect

Waveform of adaptive delta modulation transmitter

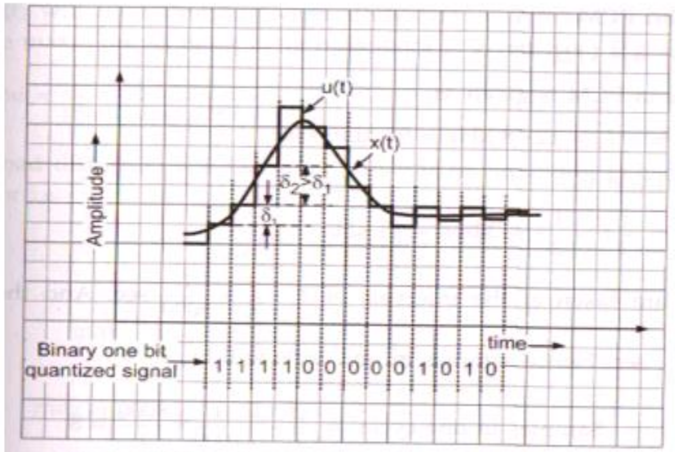


Fig. 5.4.2 Waveforms of adaptive delta modulation

b) **Compare PCM with DM w.r t.** — **4M**

(i) No. of bits required to encode one sample
(ii) Band width requirement
(iii) Complexity of circuit

Ans:

Parameter	PCM	DM
No. of bits required to encode one sample	It can use 4,8 or 16 bit per sample	One bit is used per sample
Band width requirement	Highest bandwidth is required since number of bits/sample is high	Lowest channel bandwidth is required
Complexity of circuit	System complex	Simple

2M for correct comparison of any 2given points as marks allocated is 4 and given points are 3

c) **Compare TDMA, FDMA and CDMA technique for following points:** — **4M**

(i) Definition
(ii) Bandwidth available
(iii) Synchronization
(iv) Application

Ans:

PARAMETER	TDMA	FDMA	CDMA
Definition	Time Division Multiple Access, here entire	Frequency Division Multiple Access, here	Code Division Multiple Access, here entire

1M each for correct comparison point

			bandwidth is shared among different subscribers at fixed predetermined or dynamically assigned time intervals/slots.	entire band of frequencies is divided into multiple RF channels/carriers. Each carrier is allocated to different users.	bandwidth is shared among different users by assigning unique codes.	
		Bandwidth available	Entire Bandwidth is available to each user for predefined same time period.	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)	**Outline working principle of DPSK to convert digital into analog signal.**				**4M**
	Ans :	• In DPSK transmitter, each symbol is modulated relative to the previous symbol and modulating signal, for instance in BPSK 0 = no change, 1 = $+180^0$ • In the receiver, the current symbol is demodulated using the previous symbol as a reference. The previous symbol serves as an estimate of the channel. • A no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous symbol.				**2M for Transmitter & Receiver block diagram of DPSK & 2M for**

explanation

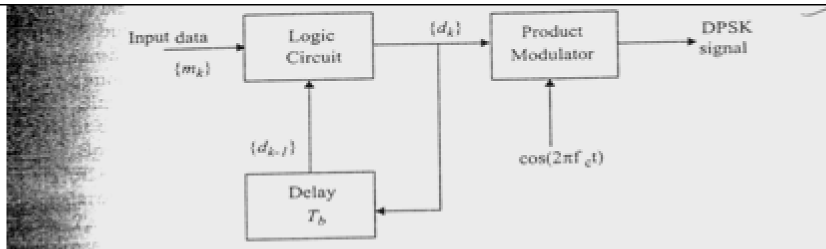


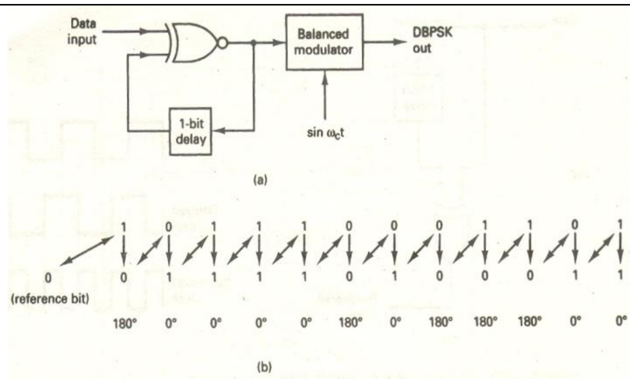
Figure 5.24
Block diagram of a DPSK transmitter.

Table 5.1 Illustration of the Differential Encoding Process

$\{m_k\}$		1	0	0	1	0	1	1	0
$\{d_{k-1}\}$		1	1	0	1	1	0	0	0
$\{d_k\}$	1	1	0	1	1	0	0	0	1

- Let $\{d_k\}$ denote the differentially encoded sequence with this added reference bit. We now introduce the following definitions in the generation of this sequence:
- If the incoming binary symbol b_k is 1, leave the symbol d_k unchanged with respect to the previous bit.
- If the incoming binary symbol b_k is 0, change the symbold_k with respect to the previous bit.
- To send symbol 0, we advance the phase of the current signal waveform by 180 degrees, To send symbol 1, we leave the phase of the current signal waveform unchanged.

OR



DBPSK modulator: (a) block diagram; (b) timing diagram

Working of DPSK transmitter:

- An incoming information bit is XNORed with the preceding bit prior to entering the BPSK modulator(balanced modulator).
- For the first data bit ,there is no preceding bit with which to compare it. Therefore, an initial reference bit is assumed,
- If the initial logic bit is assumed to be "1"the o/p of the XNOR circuitry is simply the complement as shown.
- The first data bit is XNORed with the reference bit. IF they are same the XNOR o/p is a logic1, if they are different ,the XNOR o/p is logic "0".
- The balanced modulator operates the same as a conventional BPSK modulator, logic 1 produces +sin (ωct) at the o/p and logic 0 produces-sin (ωct)at the o/p.

	e)	**Compare BASK, BFSK, QPSK and 16- PSK w. r. t. bandwidth requirement.**	**4M**
	Ans :	**Bandwidth:** **BASK=2f_b** **BFSK=4f_b** **QPSK=f_b** **16-PSK=$\frac{2fb}{N}$ (for M-ary PSK)** Wheref_b is bit rate of the input binary data.	**1M each for correct comparison of bandwidth requirement**

		OR Type / Minimum BANDWIDTH: BASK — fb BFSK — fb QPSK — fb/2 16PSK — fb/4	

Type	**Minimum BANDWIDTH**
BASK	**fb**
BFSK	**fb**
QPSK	**fb/2**
16PSK	**fb/4**

Q. No.	Sub Q. N.	Answers	Marking Scheme
4	**a)**	**Attempt any THREE :**	**12- Total Marks**
	(a)	**What is channel modeling? Explain any one with neat sketch.**	**4M**
	Ans :	**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 **Additive Gaussian noise channel**	**channel modeling -1M** **any one type diagram- 1M** **explain-2M**

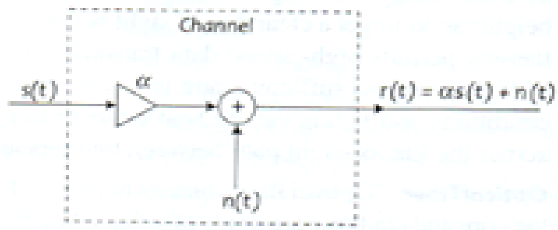


Fig. 1.2 *Additive Gaussian noise channel*

- 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 < α < 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.

or

Bandwidth limited linear channel

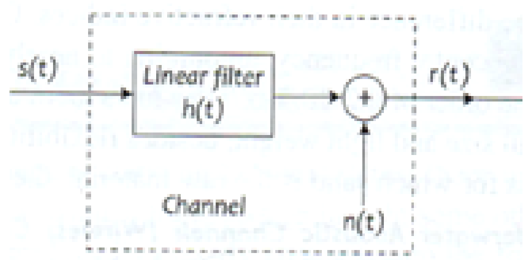
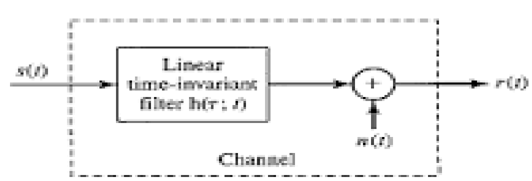


Fig. 1.3 *Bandwidth-limited linear channel*

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

Or

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
- 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)

(b) **With the help of neat sketch illustrate uniform quantization.** **4M**

Ans :

2M for diagram &

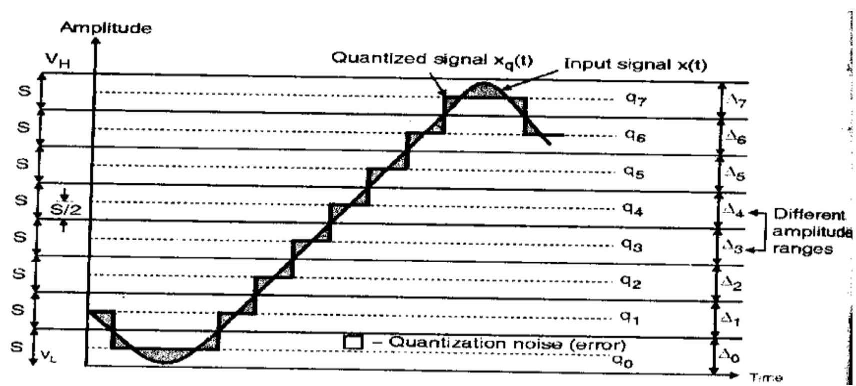
2M for explanation

Quantization :

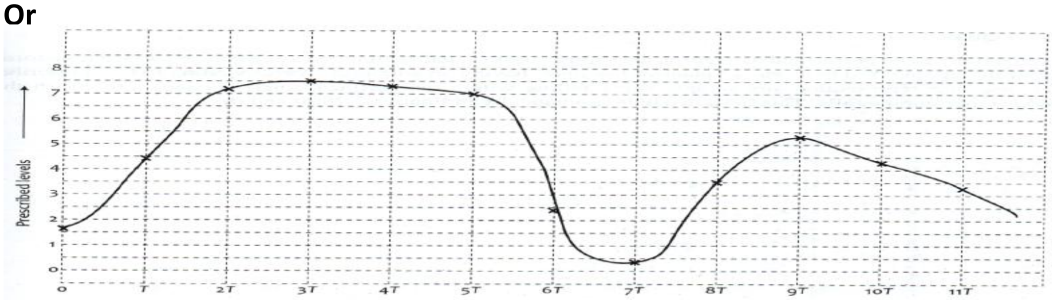
- **It** is the process of approximation or rounding off the sampled signal.
- The quantizer converts sampled signal into approximated rounded values consisting of only finite no. of pre decided voltage levels called as quantization levels.
- In the process of A to D conversion , after sampling, quantization is the next step.
- The input signal x(t) is assumed to have a peak swing of V_L to V_H volts.
- This entire voltage range has been divided into Q equal intervals each of size "s". s is called as step size and its value is given as

$$S= V_H-V_L / Q$$

Diagram of the Process quantization is as shown below-



Or

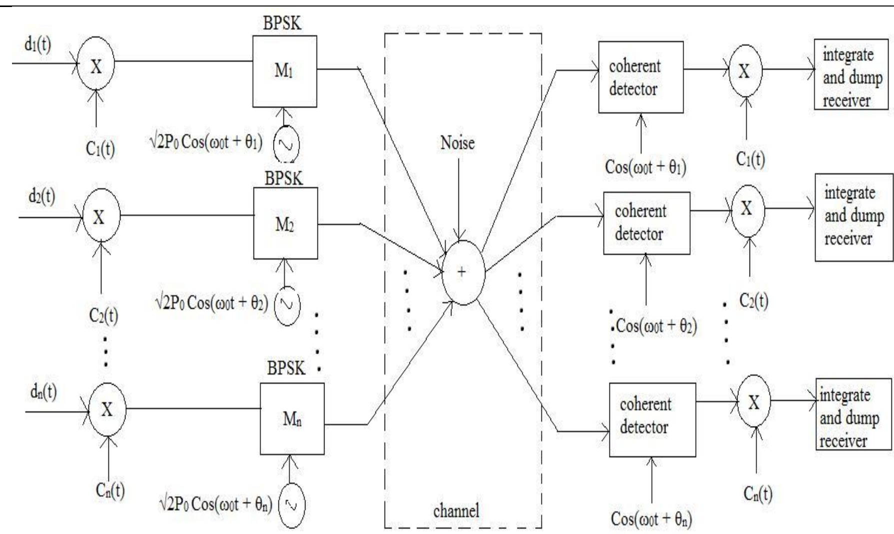


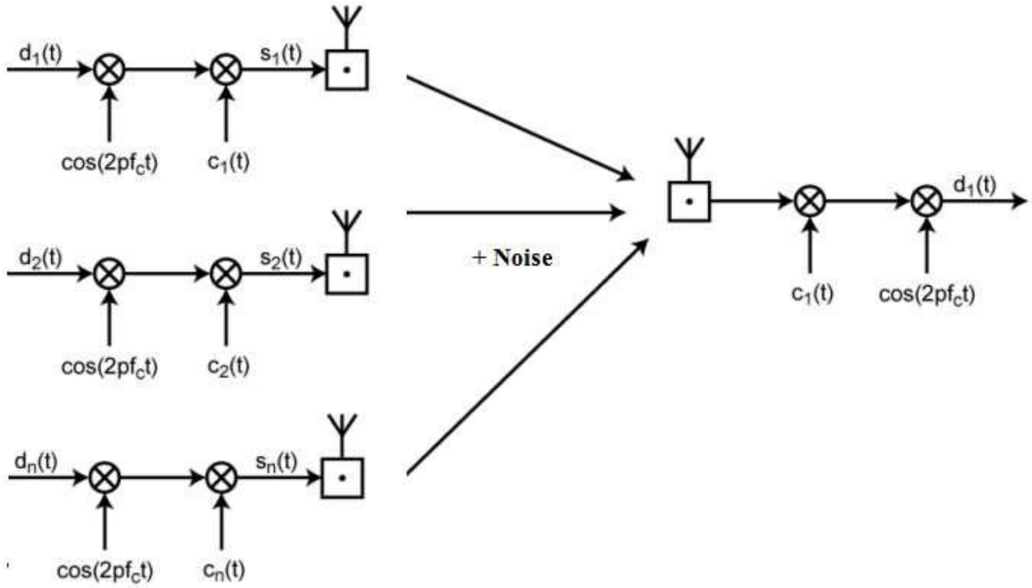
Sample Number	S_0	S_1	S_2	S_3	S_4	S_5	S_6	S_7	S_8	S_9	S_{10}	S_{11}
Actual values of the samples	1.7	4.4	7.2	7.4	7.3	7.0	2.4	0.4	3.4	5.3	4.3	3.2
Rounded off values	2	4	7	7	7	7	2	0	3	5	4	3
Binary code for rounded value	010	100	111	111	111	111	010	000	011	101	100	011

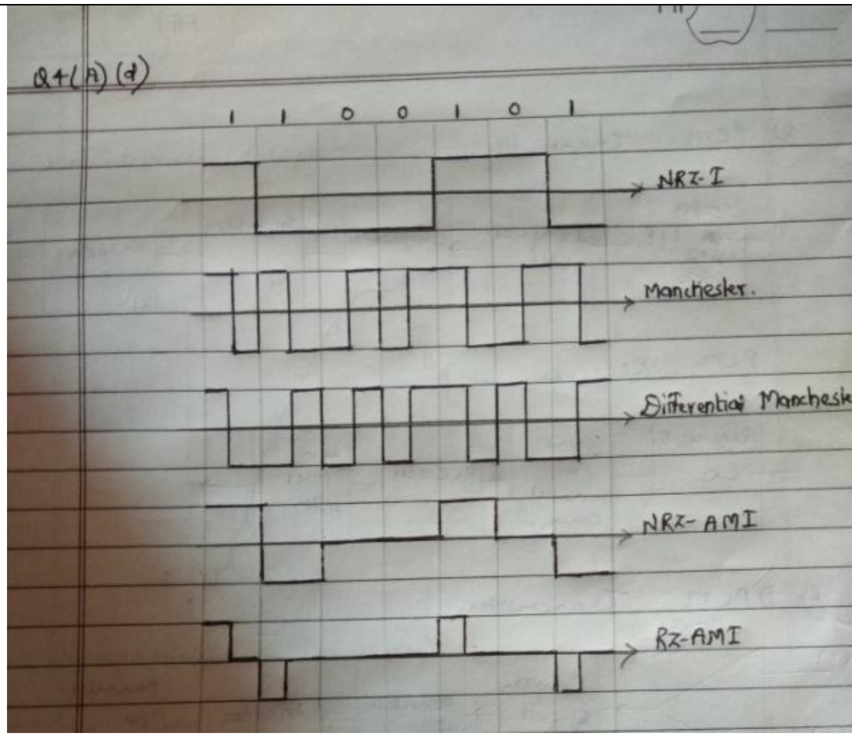
Fig. 3.27 *Illustrating the quantization process*

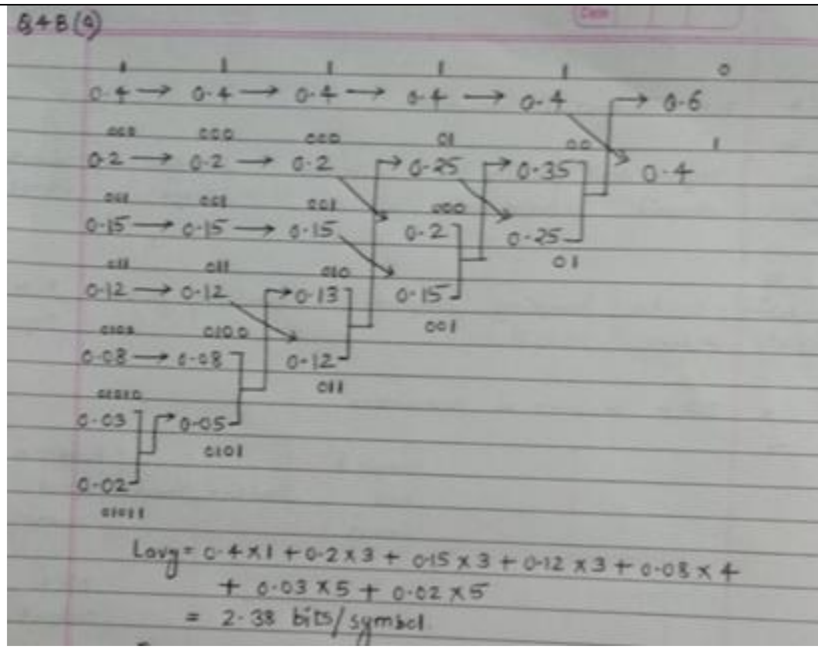
Uniform Quantization:

- In Uniform Quantization a step size "s" is constant and so variance of quantization noise power Nq=s^2/12 as shown in fig.
- As a result signal to noise ratio SNRq decreases with decrease in the input signal power level which is unacceptable. So Non Uniform Quantization, is used companding.

(c)	**Draw block diagram of DSSS based CDMA system.**	**4M**
Ans:	 **DSSS based CDMA system**	**Diagram-4M**

		OR  	
	(d)	**Encode binary sequence 1100101 using line encoding techniques:** **(i) NRZ-I** **(ii) Manchester** **(iii) Differential** **(iv) AMI**	**4M**

	Ans :		1M each for correct waveforms
	b)	**Attempt any ONE :**	**06- Total Marks**
	(a)	**A discrete memoryless 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 and 0.4}. Derive Huffman code for above source and determine the average length of the code words.**	**6M**

	Ans :	 **The code words are:** **A – 0100** **B – 000** **C – 011** **D – 001** **E – 01010** **F – 01011** **G - 1**	**6M for correct answer of Huffman coding Numerical** **(Method-2M,** **Code-2M & Average length-2M)**
	(b)	**Draw the block diagram of FHSS system and illustrate its working.**	**6M**

WINTER– 18 EXAMINATION

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

Ans :

FREQUNCY HOPPING SPREAD SPECTRUM

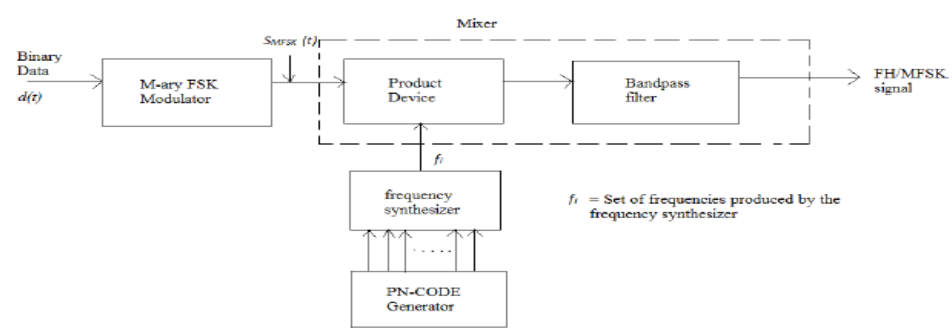


Fig : FHSS/M-ary FSK Transmitter

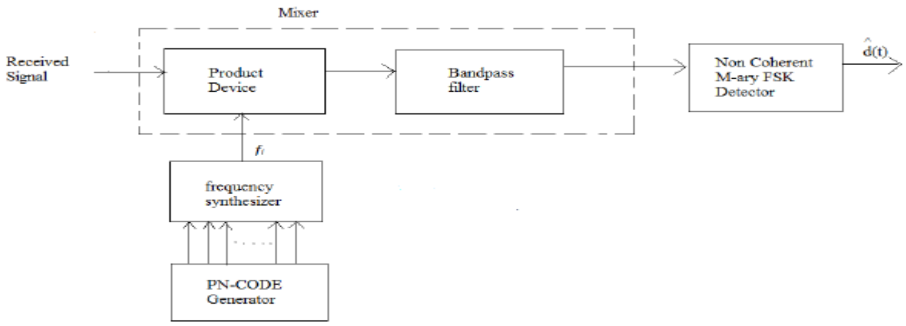
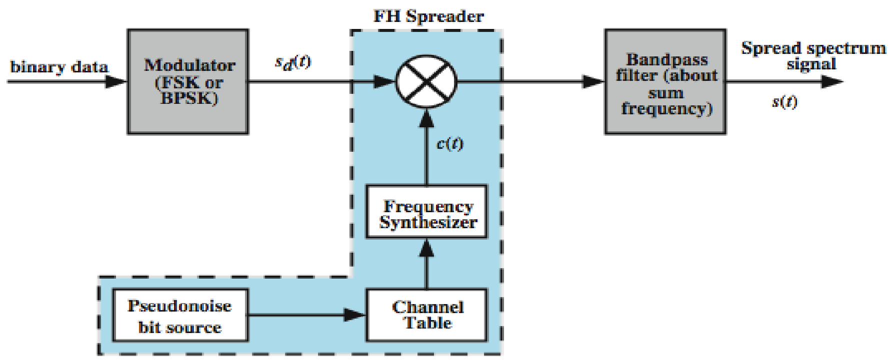


Fig : FHSS/M-ary FSK Receiver

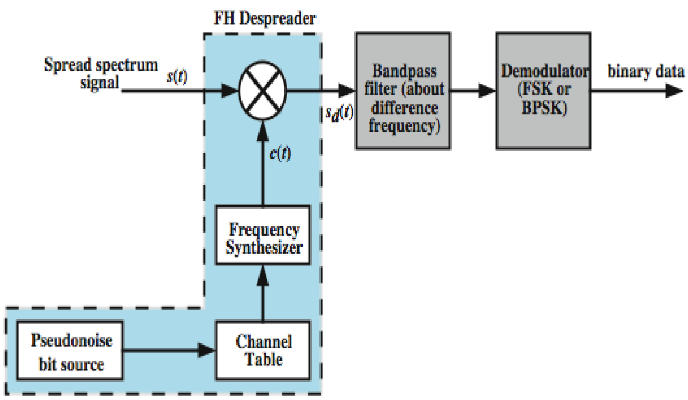
- The binary digital data modulates a carrier using a traditional modulation scheme like M-ary FSK.
- This M-ary FSK modulated signal is then modulated a second time by another carrier frequency, but this carrier frequency changes its value, or rather hops, at regular intervals of T_c, the chip period, from one value to another form among a given set of values, accordingly to a pre-determined, pseudo-random pattern.
- This carrier frequency hopping is controlled at the transmitter by a pseudo-random code generator, as shown in fig.
- The binary data is first used to produce an M-ary FSK modulated signal. This is again modulated by a carrier produced by a frequency synthesizer that is controlled by a PN code generator.

OR

Diagram-3M expalination-3M



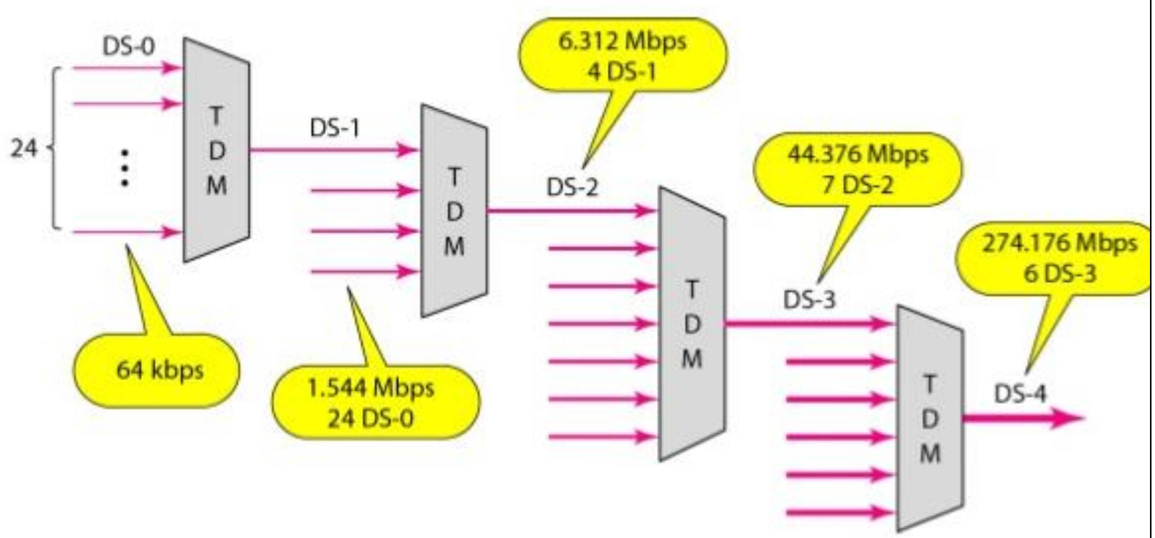
(a) Transmitter



(b) Receiver

Fig shows a typical block diagram for a frequency-hopping system.

- For transmission, binary data are fed into a modulator using some digital-to-analog encoding scheme, such as frequency shift keying (FSK) or binary phase shift keying (BPSK).
- The resulting signal $s_d(t)$ is centered on some base frequency.
- A pseudonoise (PN), or pseudorandom number, source serves as an index into a table of frequencies; this is the spreading code referred to previously.
- Each k bits of the PN source specifies one of the 2^k carrier frequencies.
- At each successive interval (each k PN bits), a new carrier frequency is selected.
- The frequency synthesizer generates a constant-frequency tone whose frequency hops among a set of 2^k frequencies, with the hopping pattern determined by k bits from the PN sequence. This is known as the spreading or **chipping signal** $c(t)$.
- This is then modulated by the signal produced from the initial modulator to

		produce a new signal with the same shape but now centered on the selected carrier frequency. • A bandpass filter is used to block the difference frequency and pass the sum frequency, yielding the final FHSS signal *s*(*t*). • On reception, the spread spectrum signal is demodulated using the same sequence of PN-derived frequencies and then demodulated to produce the output data. • At the receiver, a signal of the form *s*(*t*) defined on the previous slide, will be received. • This is multiplied by a replica of the spreading signal to yield a product signal. • A bandpass filter is used to block the sum frequency and pass the difference frequency, which is then demodulated to recover the binary data	
Q. No.	**Sub Q. N.**	**Answers**	**Marking Scheme**
5.		**Attempt any TWO:**	**16- Total Marks**
	a)	**Illustrate the North American digital multiplexing hierarchy with neat diagram.**	**8M**
	Ans :	 **Explanation:-** **T1 Carrier System** T1 carrier systems were designed to combine PCM and TDM Techniques for the transmission of 24 64Kbps channels with each channel Capable of Carrying Digitally	**Diagram 4M &** **Explanation 4M**

		encoded voice band telephone signals or data. The transmission bit rate (line speed) for a T1 carrier is 1.544 Mbps. All 24 DS-0 channels combined has a data rate of 1.544Mbps, this digital signal level is called DS-1. Therefore T1 lines are referred as DS-1 lines.	

Service	Line	Rate (Mbps)	Voice Channels
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

DS and T Line rates

T2 Carrier System

T2 carriers time division multiplex 96 64-Kbps voice or data channels into a single 6.312 Mbps data signal for transmission over twisted pair copper wire upto 500 miles over a special metallic cable.

T3 Carrier system

T3 carriers Time division multiplex 672 64-kbps voice or data channels for transmission over a single coaxial cable. The transmission rate is 44.736 Mbps.

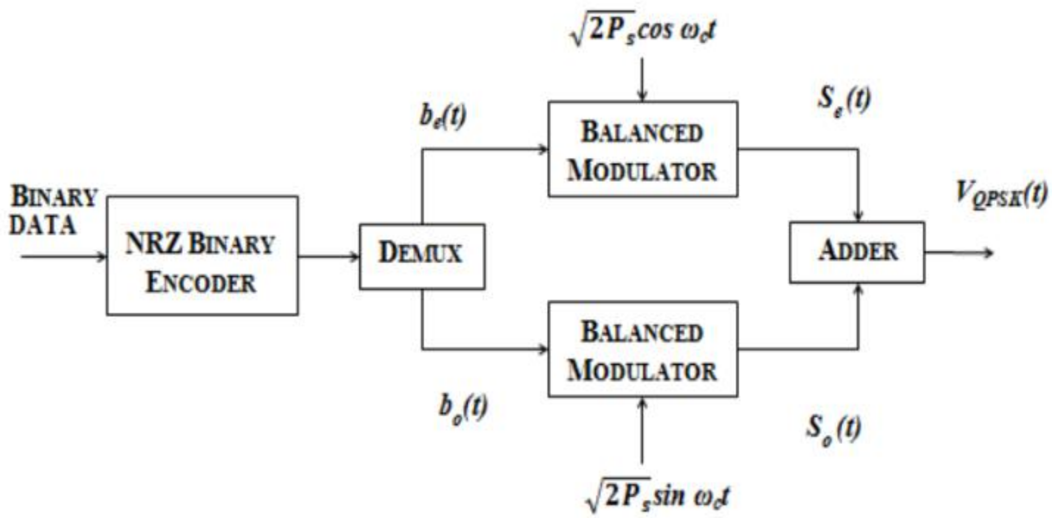
T4 Carrier System

T4 carriers time division multiplex 4032 64-kbps voice or data channels for transmitting over a single T4 coaxial cable upto 500 mile. The transmission rate is very high i.e. 274.16Kbps.

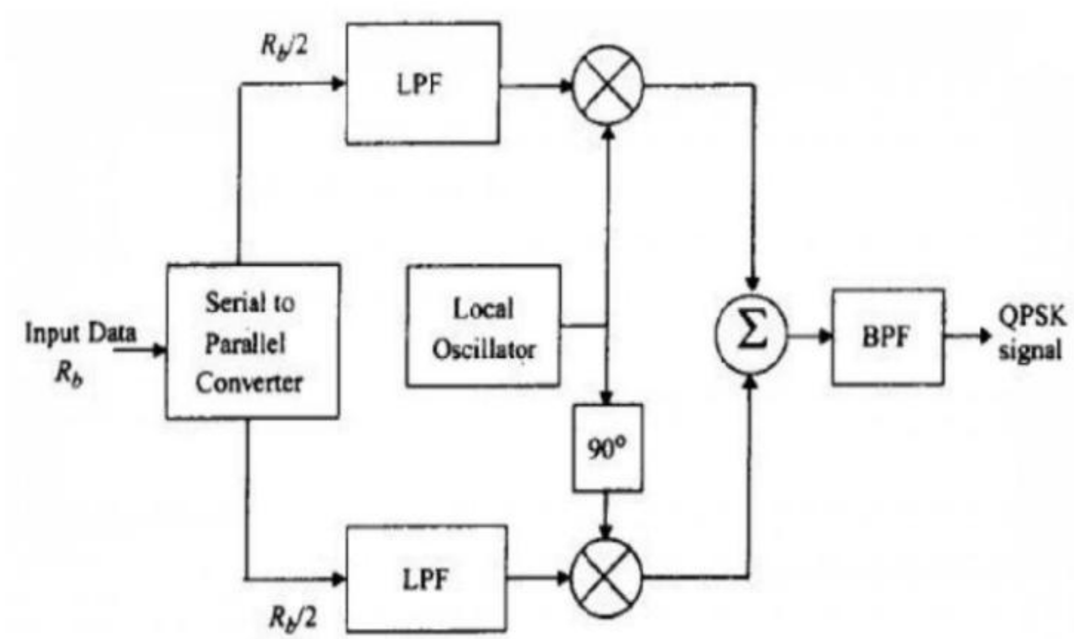
T5 Carrier System

T5 carriers time division multiplex 8064 64Kbps voice or data channels and transmit them at 560.16Mbps over a single coaxial cable.

	b)	**With the help of block diagram explain QPSK system.**	**8M**
	Ans:	**Note: Credit should be given for any pair of transmitter and receiver.** **QPSK transmitter Diagram:-**	**(QPSK transmitter & Explanation: 4M)**



OR



QPSK Transmitter Explanation:

- 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 De-multiplexer (DEMUX) will divide *b(t)* into two separate bit streams $b_o(t)$ and $b_e(t)$. The bit stream $b_e(t)$ consists of only the even numbered bits 2, 4, 6, 8, ... whereas $b_o(t)$ bit stream consists of only the odd numbered bits i.e., 1, 3, 5, ... as shown in Figure
- Each bit in the even and odd stream will be held for a period of $2T_b$. This duration is

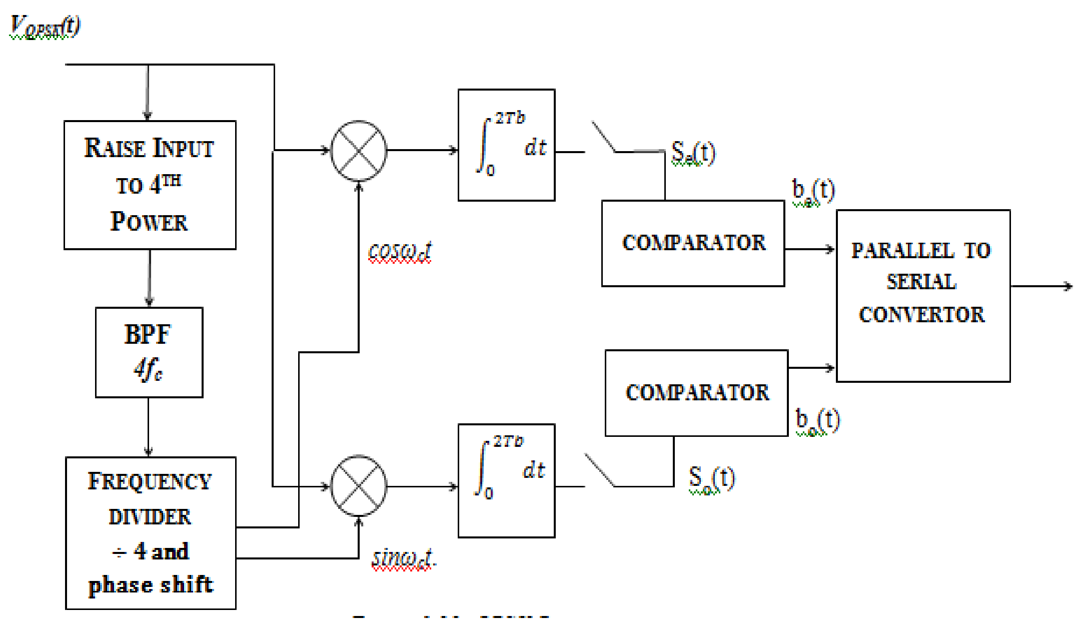
called as symbol duration T_s. Thus, every symbol contains two bits.

- The bit stream $b_e(t)$ is superimposed on a carrier $\sqrt{2P_s}\cos\omega_c t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sqrt{2P_s}\sin\omega_c t$ by using two balanced modulators (or multipliers) to generate $s_e(t)$ and $s_o(t)$. These two signals are basically BPSK signals.
- These signals are then added to generate the QPSK output signal $V_{QPSK}(t)$ given by,

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

QPSK Receiver Diagram:-

In QPSK receiver, we use coherent detection technique. Hence, it is necessary to locally generate the carriers $\sin\omega_c t$ and $\cos\omega_c t$. The block diagram of QPSK receiver is shown in Figure



- Let the received QPSK signal be $v_{QPSK}(t)$. The received QPSK signal $v_{QPSK}(t)$ is raised to the fourth power i.e., $v^4_{QPSK}(t)$.
- This signal is then filtered by using a BPF with a center frequency of $4\omega_c$. The output of the BPF is $-\cos 4\omega_c t$.
- A frequency divider divides the frequency at the filter output by 4 and generates the two carrier signals $\sin\omega_c t$ and $\cos\omega_c t$.
- The incoming signal $v_{QPSK}(t)$ is applied to two synchronous demodulators consisting of multipliers followed by an integrator. Each integrator integrates over a two-bit interval
- $T_s = 2T_b$.
- One synchronous demodulator uses $\cos\omega_c t$ as the carrier signal and the other

synchronous demodulator uses $sin\omega_c t$ as the carrier signal. The input to the lower integrator is given by,

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

- The lower integrator output is given by,

$$= b_o(t)\sqrt{2P_s}\int_0^{2Tb}\sin^2 \quad \omega ct + b_e(t)\sqrt{2P_s}\int_0^{2Tb} sin\omega ct\ cos\omega ct$$

We know that, $sin^2\omega_c t=\frac{1}{2}[1 - cos2\omega_c t]$

$sin\omega_c\, tcos\omega_c t=\frac{1}{2}sin2\omega_c t$

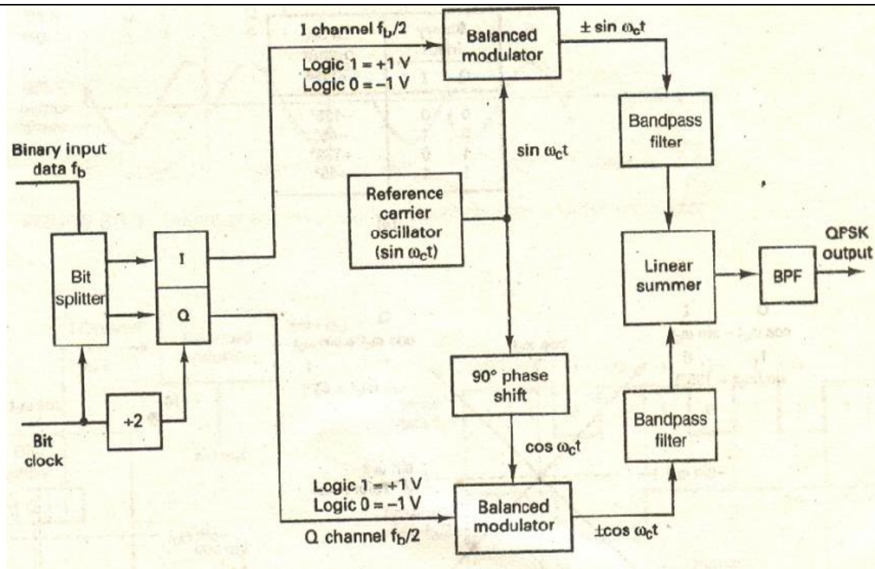
$\int_0^{2Tb}\frac{1}{2}sin2\omega ct=0$

Integrator output = $\frac{1}{2}b_o(t)\sqrt{2P_s}\int_0^{2Tb} dt$

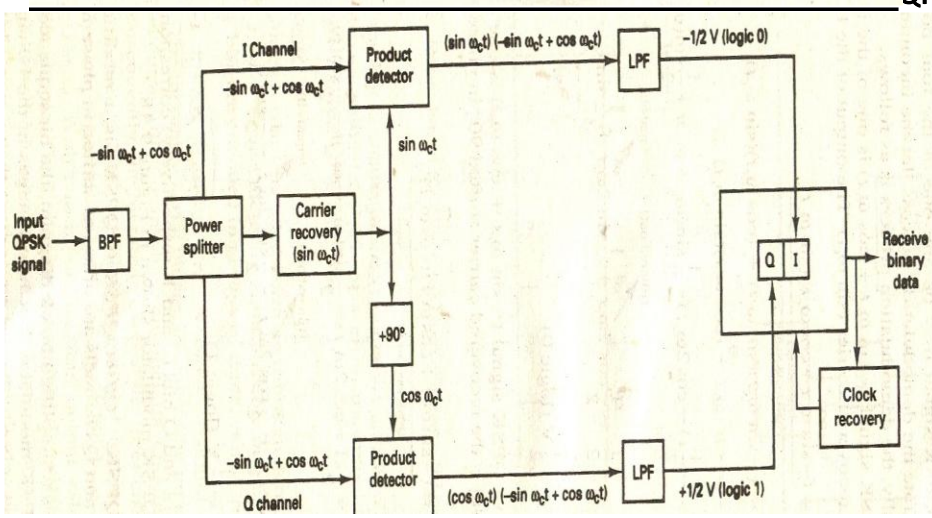
$= b_o(t)\sqrt{2P_s}T_b$

- Similarly, the output of the lower integrator is given by $b_e(t)\sqrt{2P_s}T_b$
- Thus, at the output of the two integrators we obtain the bit streams $b_e(t)$ and $b_o(t)$.
- Bit synchronizer is used to establish the beginning and end of the bit intervals of each bit stream. It is also used to operate the sampling switch.
- The integrator output is sampled at the end of each integration time for each integrator. The samples are taken alternately from the two integrator outputs at the end of each bit time T_b and these samples are then held in the latch for the bit time T_b. Each integrator output is thus sampled at intervals $2T_b$. At the output of the latch we get the signal $b(t)$.

Or



QPSK transmitter



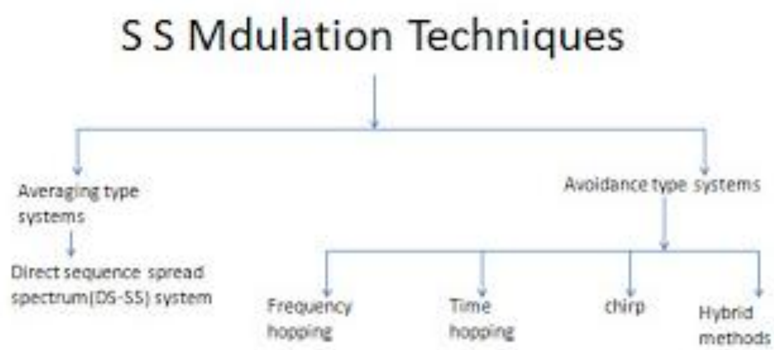
QPSK recevier

Working of QPSK :

Digital signal acts as modulating or information signal which is fed to bit splitter which splits the data with the help of bit clock in I and Q bits having I channel fb/2 and Q bit fb/2.

- Logic 1 represents=+1v
  Logic 0 represents=-1v

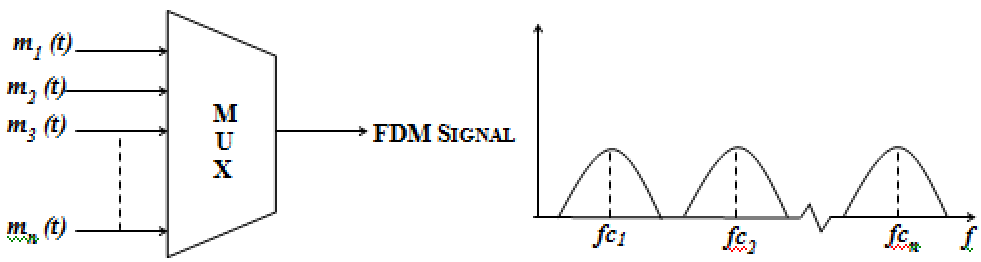
- I i/p is fed to balanced modulator1 which have the other input analog carrier sin(ωct)
- Q i/p is fed to balanced modulator2 which have the other input analog carrier cos(ωct)
- O/P of balanced modulator1is +-sin(ωct) and O/P of balanced modulator1is +-cos(ωct)
  - O/P of balance modulator is fed to BPF to allow only wanted frequency to pass through it.

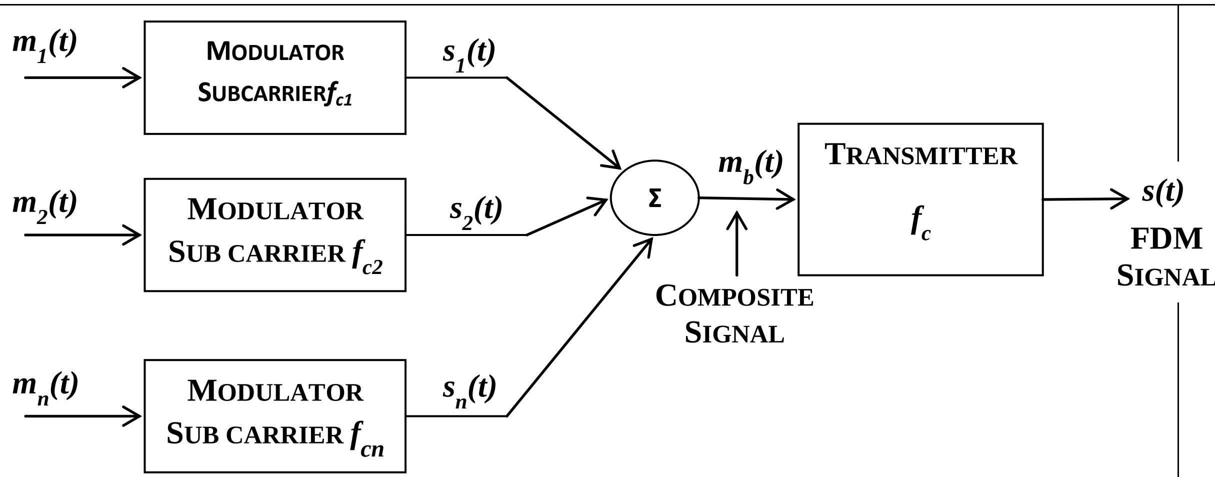
		• O/P from BPF is fed to linear summer and BPF to give QPSK o/p • Output of QPSK transmitter: 1) +sin(ωct) +cos(ωct) 2) +sin(ωct) -cos(ωct) 3) -sin(ωct) +cos(ωct) 4) -sin(ωct) -cos(ωct At the receiver given any one of these input will be recover the original binary data at the receiver	
	c)	**State the types of SS modulation and list its application.**	**8M**
	Ans :	**Types of SS Based** on the kind of spreading modulation, spread spectrum systems are broadly classified as- (i) Direct sequence spread spectrum (DS-SS) systems (ii) Frequency hopping spread spectrum (FH-SS) systems (iii) Time hopping spread spectrum (TH-SS) systems. (iv) Hybrid systems **Or**  **Applications of SS modulation:-** 1. In combating the intentional interference or jamming 2. In rejecting the unintentional interference from some other user 3. To avoid the self interference due to multipath propagation 4. In low probability of intercept signals 5. In obtaining the message privacy 6. The spread spectrum Communications is widely used today for Military, 7. Industrial, Avionics, Scientific, and Civil uses. 8. Bluetooth Technology. 9. CDMA radios: It is useful in multiple access communications wherein many users communicate over a shared channel. Here the assignment of a unique spread spectrum sequence to each user allows him to simultaneously transmit over a	**Types:4M** **&** **listing any 4 application 4M.**

		common channel with minimal mutual interference. Such access technique often simplifies the network control requirements considerably. 10. High Resolution Ranging: spread spectrum communications is often used in high resolution ranging. It is possible to locate an object with good accuracy using spread spectrum techniques. One example where it could be used is Global Positioning System (GPS). Here an object can use signals from several satellites transmitting spread spectrum signals according to a predefined format to determine its own position accurately on the globe. 11. WLAN: Wireless LAN (Local Area Networks) widely use spread spectrum communications. 12. Cordless Phones: Several manufacturers implement Spread Spectrum in Cordless phones. The advantages of using spread spectrum in cordless phone include the following: Security: Inherently, ass communication is coded. Immunity to Noise: SS modulation is immune to noise when compared with other modulation schemes such as AM and FM. Longer Range: Due to noise immunity, it is possible to achieve a longer range of communications, for a very small transmitted power. 13. Long-range wireless phones for home and industry 14. Cellular base stations interconnection.	
Q. No.	Sub Q. N.	Answers	Marking Scheme
6.		**Attempt any FOUR :**	**16- Total Marks**
	a)	**State two advantages and two disadvantages of DPCM system.**	**4M**
	Ans :	**Advantages:-** 1. As the difference between sample signal and predicted value is being encoded and transmitted by the PCM technique, a small difference voltage is to be quantized and encoded. 2. It needs less number of quantization levels and hence less number of bits to represent it, 3. The signaling rate and bandwidth of a DPCM system will be less than that of PCM. **Disadvantages:-** 1. High bit rate 2. Needs the predictor circuit to be used which is very complex.	**Any two Advantage-2M** **Any two Disadvantage-2M**

WINTER– 18 EXAMINATION

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

	b)	**How different signals can be multiplexed using FDM ? Explain with block diagram.**	**4M**
	Ans :	• 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. Figure explains the concept of FDM.  **Concept of FDM** • FDM is based on the concept of a common communication channel. The modulating signals $m_1(t), m_2(t), \ldots, m_n(t)$ are transmitted simultaneously on separate carriers $fc_1, fc_2, \ldots, fc_n$. • These bandwidth ranges are the channels through which the various signals travel. Channels must be separated by strips of unused bandwidths (***guard bands***) to prevent the signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies. **OR** **FDM TRANSMITTER:** • 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	**Diagram 2M & Explanation 2M**

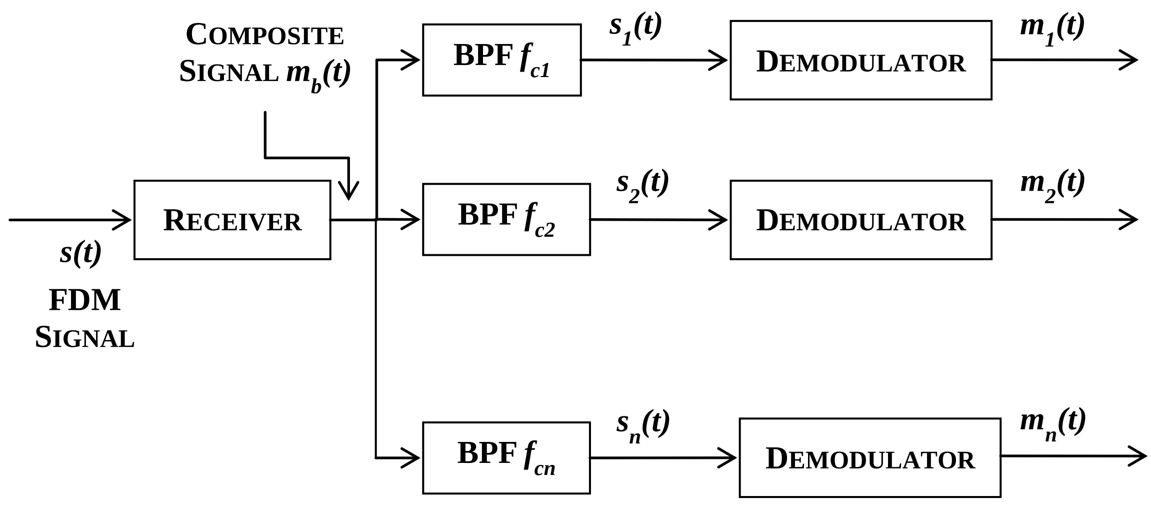


FDM Transmitter

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

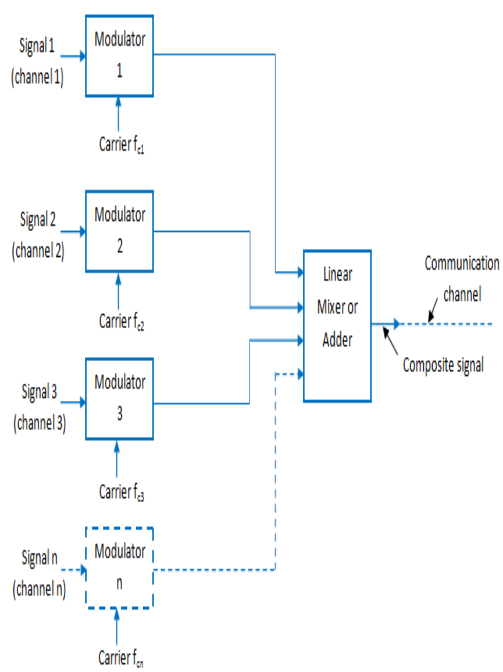
FDM RECEIVER:

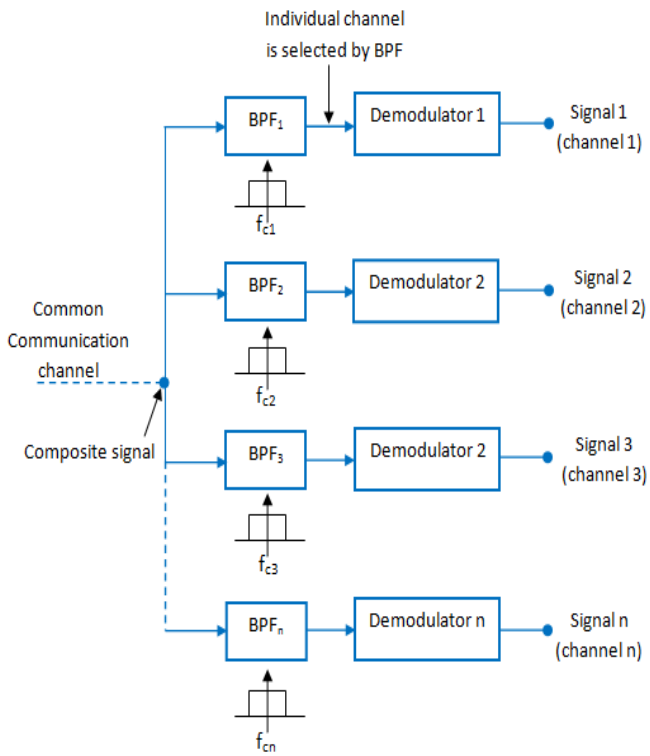
- The FDM receiver is shown in Figure The FDM signal is received by the receiver and demodulated to retrieve the composite signal $m_b(t)$ which is further amplified.
- This composite signal is passed through n band pass filters, each filter having a center frequency equal to the subcarrier frequency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

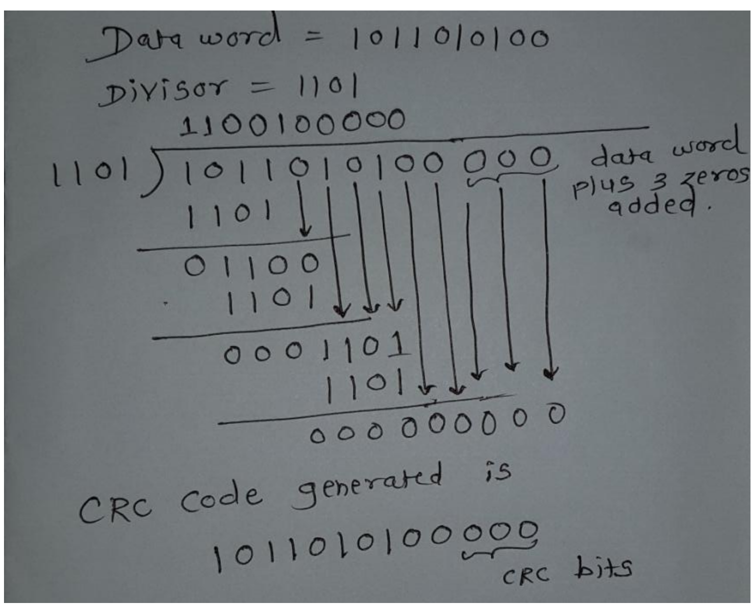


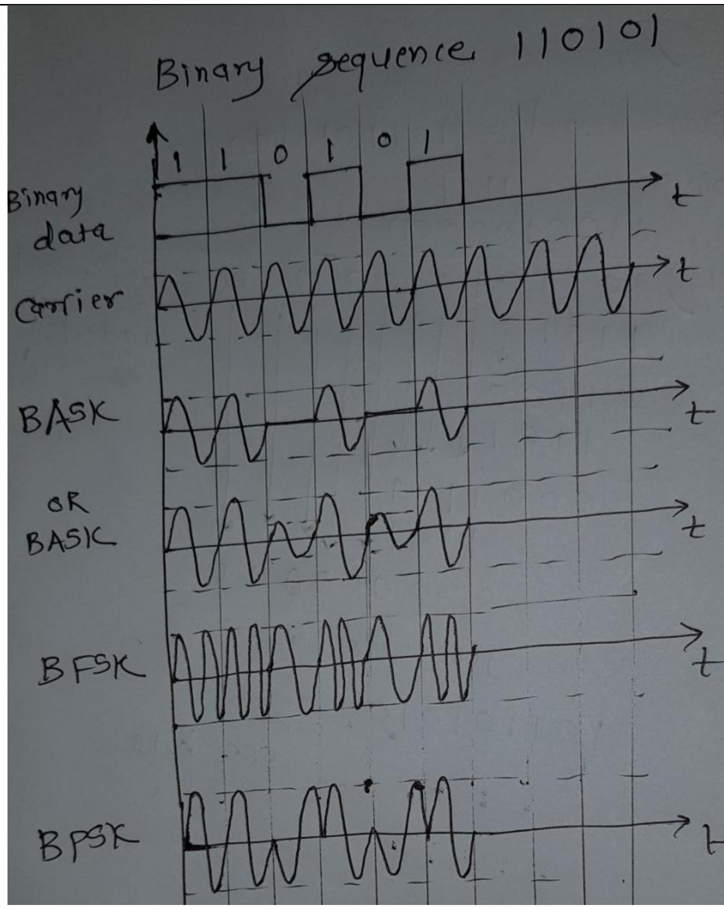
FDM Receiver

Or



		 **Block diagram of FDM transmitter and receiver:** The multiple signal to be transmitted are modulated in the modulator with the carrier(f1,f2 ..fn) which are equally spaced from each other and are than added in the linear adder to form a composite FDM signal . This FDM signal is feed to the different BPF of specific range and then the filtered signal is feed to demodulator to retrieve the original signal.	
	c)	**What is M-ary encoding? State any two advantages and one disadvantage.**	**4M**
	Ans :	• 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:	**M-ary encoding -1M**

		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. Disadvantages of the M-ary scheme are a s follows: 1. Increase in the transmitted power. 2. Increase in error probability.	**Any 2Advantages:1 M each** **Any one Disadvantage 1M**
	d)	**Generate cyclic redundant bits for binary sequence 1011010100 using divisor 1101.**	**4M**
	Ans :		**Adding no of zero to data-1M** **Division-2M** **Appending CRC code-1M**
	e)	**Draw the waveforms of BASK, BFSK, BPSK for binary sequence 110101.**	**4M**

	Ans :		data -1M, BASK,BFSK,BPSK 1M each