Model Answer

Subject Code: 12188 .
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Q.1 Attempt any FIVE of the following: 20 Marks

a) Explain the following terms 4M

i) Channel capacity

The highest rate of information that can be transmitted through a channel is called as channel capacity & is measured in bits/second

ii) Entropy

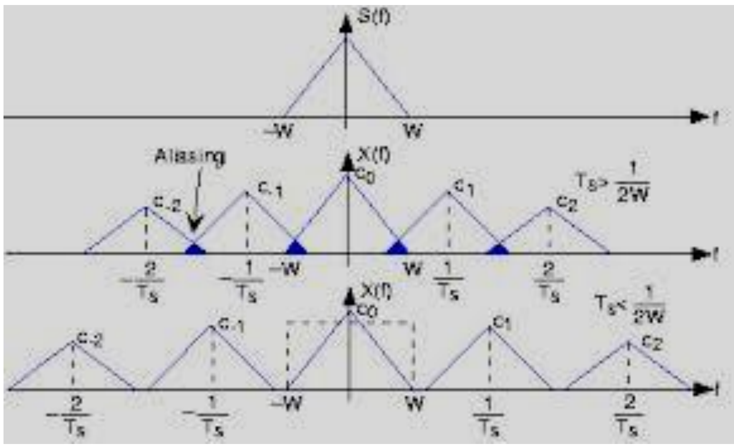
- The entropy is defined as the average information per message. It is denoted by H and its unit is bits/message.
  The entropy must be as high as possible in order to ensure maximum transfer of information.
  Suppose the transmitter is transmitting M different and independent messages m1, m2, m3……….. Let their probabilities of occurrence be p1, p2, p3.
  Then the entropy which is average information per message is denoted by H and

$$H = \sum_{k=1}^{M} p_k \log \frac{1}{p_k}$$

b) What is aliasing? What causes it? How can it be reduced? 4 Marks

- In the sampling theorem i.e. f_s>2W it is assumed that the signal x(t) is strictly bandlimited. But in practice the input signal can contain the wide range of frequencies and cannot be strictly band limited. Therefore the maximum frequencies cannot be predicted; hence the sampling frequency cannot be decided.
  If the sampling frequency becomes less than the twice maximum input frequency, then the adjacent spectrums overlap with each other.
  This problem in which the the higher frequency in the spectrum of the original signal x(t), taking on the identity of lower frequency in the spectrum of sampled signal is called aliasing or foldover error.

Aliasing is caused by poor sampling.



It can be reduced by pre filtering and post filtering

c) Compare pulse modulation with continuous wave modulation by, four points. 4 Marks

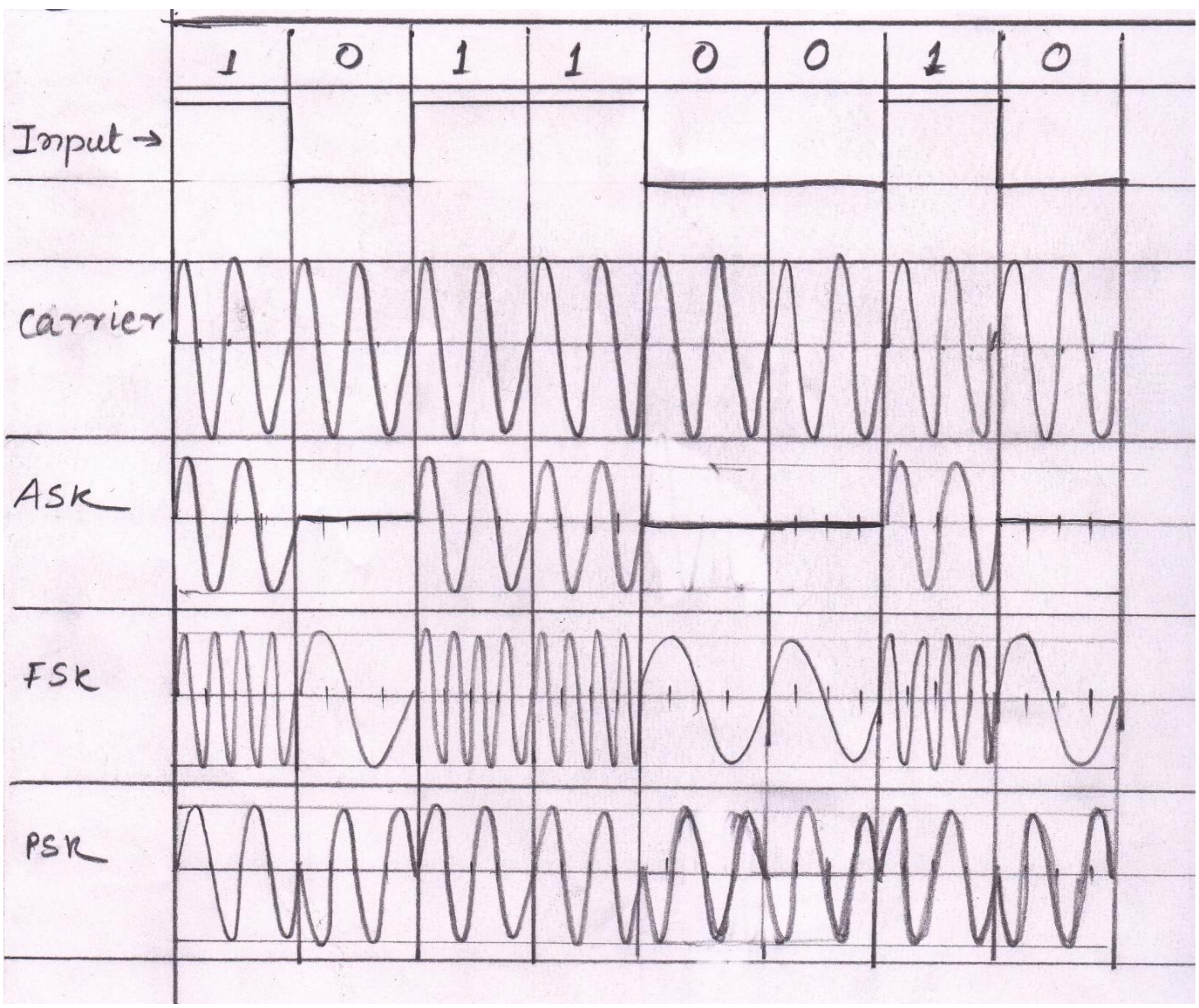
Pulse modulation	Continuous wave modulation
• **The carrier is train of pulses**	• **The carrier is sine wave**
• **The modulation can be done by varying width, position and amplitude of pulse.**	• **The modulation can be done by varying the amplitude, phase and frequency of the sine wave.**
• **The noise immunity is low for AM. While PM and FM work well.**	• **The noise immunity for PAM is least but rest all modulation techniques work well.**
• **Applied in broadcasting radio and TV**	• **Applied in Satellite communication.**
• **Simpler and less costly**	• **Costly and complex.**

d) List different types of digital modulation techniques. Draw waveforms for ASK. FSK and PSK techniques for binary digits

1011001 0 **4 Marks**

➔ Different digital modulation techniques are

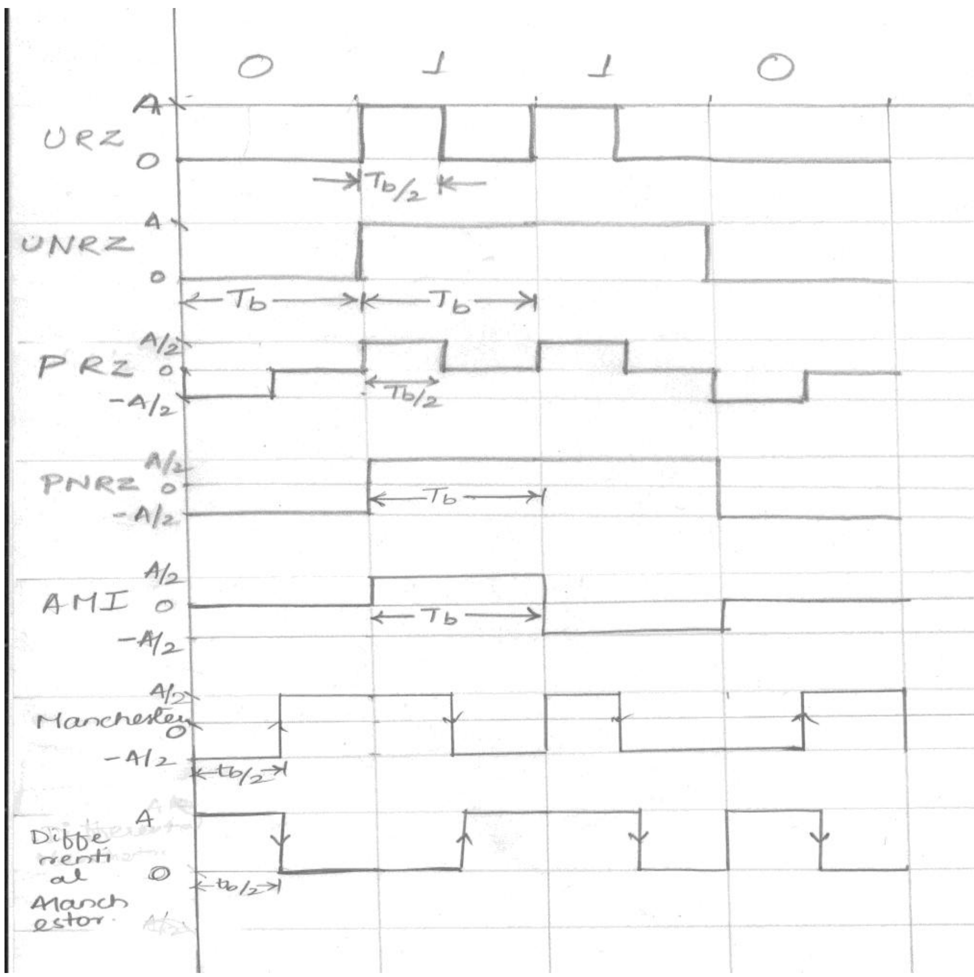
- Pulse code modulation
- Delta modulation
- Adaptive delta modulation



e) What are line codes? List popular line codes along with their waveforms for digital code word 0110. 4 Marks

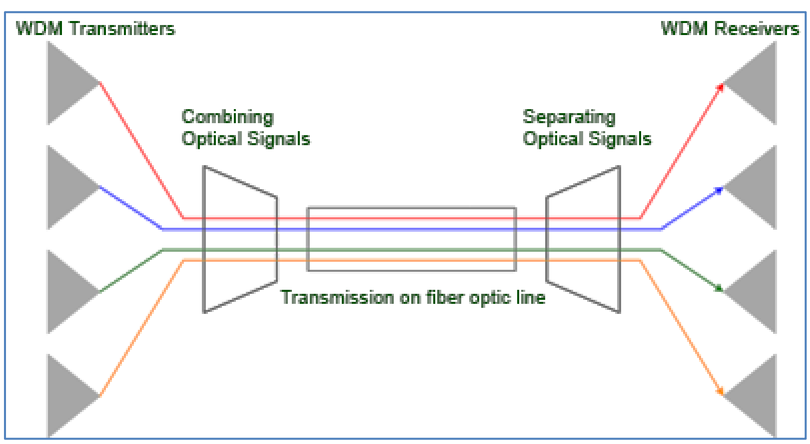
- The analog waveforms are converted in the digital form by PCM, DM ADM etc. techniques. This digital data can be represented by different formats or waveforms. These waveforms are commonly known as data formats format and their graphical representation is called line coding.

  The line coding formats are unipolar RZ, Unipolar NRZ, Polar RZ, Polar NRZ, Bipolar NRZ, Manchester, Differential Manchester and Polar Quaternary.



f) What is WDM? State 2-advantages. **4 Marks**

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



➔ Advantages:

- A wavelength division multiplexed (WDM) system is one where, typically in order to increase system capacity, multiple optical carriers, operating a different wavelengths, share a common optical fiber.

- A WDM system can be simple, operating at two very different wavelengths such as 1310 nm and 1550 nm, or more complex whereby four, eight, or even more optical transmitters operate with very small wavelength separations.
- The amount of data being transmitted can be significantly increased without increasing the number of fibers or repeaters. Hence capacity is more.

g) What is PN sequence? Describe their characteristics. 4 Marks

- PN sequences or Pseudo Noise 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. It is called a pseudo noise sequence as it seems to be a random sequence of zeros and ones for the third person or jammer.
  - For sequence to be a pseudo noise or pseudo random it should follow the following basic rules. The rules mentioned below are simply mentioned in brief:
- The relative frequency of 0′s and 1′s are each ½
- The run lengths of 0′s and 1′s are: ½ of all run lengths are of length 1; ¼ are of length 2; 1/8 are of length 3; and so on.
- If a PN sequence is shifted by any non-zero number of elements, the resulting sequence will have an equal number of agreements and disagreements with respect to the original sequence.
- These properties are known also known as balance property, run property, and correlation property respectively. PN sequence is also known as Maximal Length Sequences**.**

Q.2 Attempt any FOUR of the following: 16 Marks

a) With neat circuit diagram, explain generation of PWM.

➔ The pulse width modulator is as shown in figure below

Model Answer

Subject Code: 12188 .

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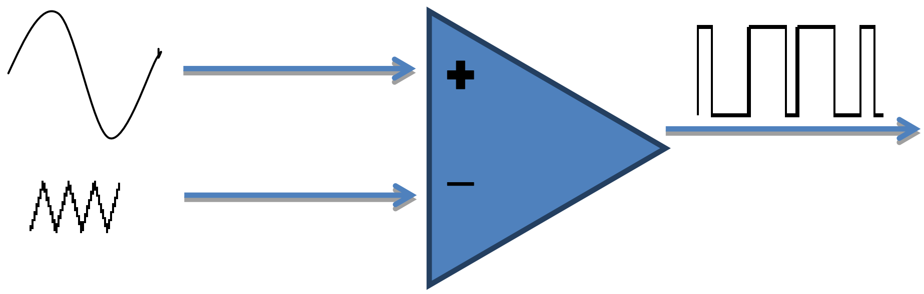
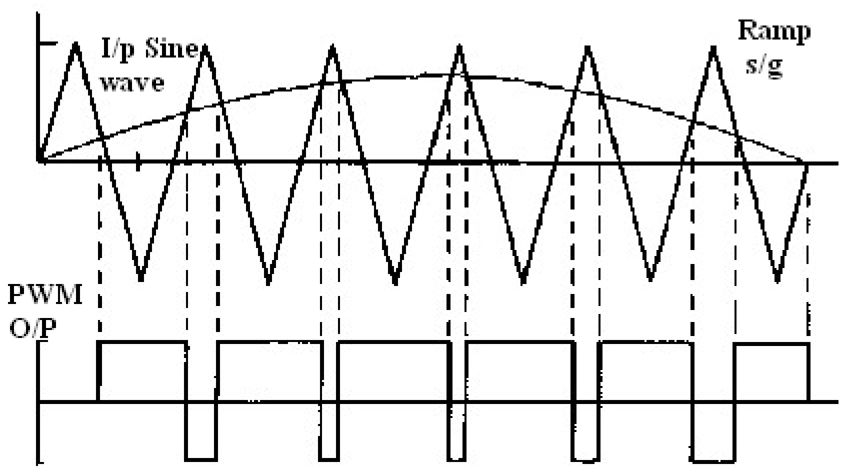
Modulating signal

Ramp signal PWM o/p



1M



1M

2M expl.

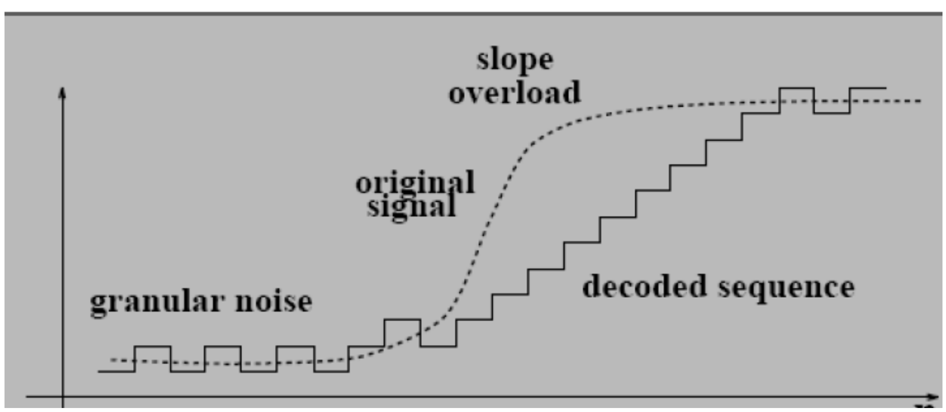
- In this pulse width modulation circuit op amp acts as a comparator.
- It compares both the input voltages that are the saw tooth waveform and the message signal (Sine wave).
- The duration at which the instantaneous value of sine wave is above that of saw tooth, the op amp switches to +Vcc since the sine wave input is connected to the non-inverting input of the op amp.

- Also when the value of sine wave less than the instantaneous value of saw tooth, op amp switches to -Vcc.
- Thus we get a pulse waveform swings between +Vcc and –Vcc.
- That is the pulse width changes according to the message signal (Width of the pulse is modulated)

b) With sketches, explain the terms – 4M

i) slope overload and

ii) granular noise



Slope overload error:

- If slope of analog signal x(t) is much higher than the 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).

- 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.

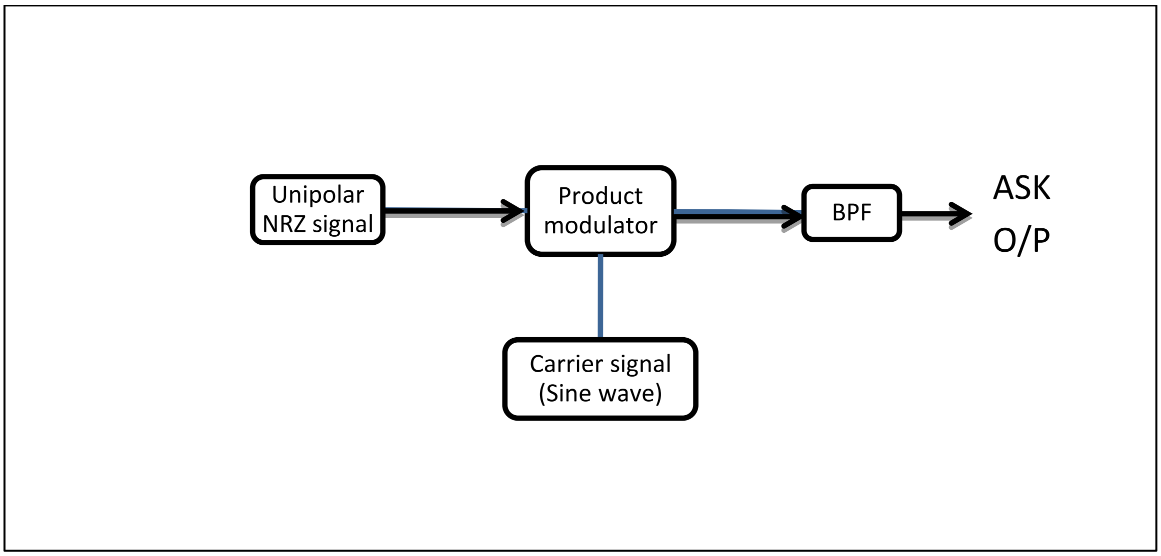
Granular Noise:

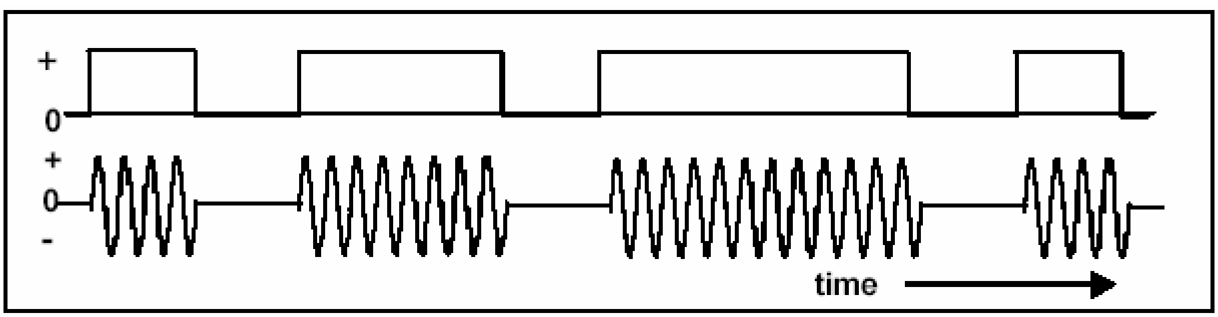
- When the i/p signal is relatively constant in amplitude the approx signal will hunt above and below x(t).
- The granular noise is similar to quantization noise in PCM system.
- It increases with increase in step size. To reduce the granular noise the step size should be small as possible.
- In linear delta modulation step size is not variable.
- If it is made variable then, the slope overload distortion and granular noise can be controlled.

A system with a variable step size is known as adaptive delta modulation ADM.

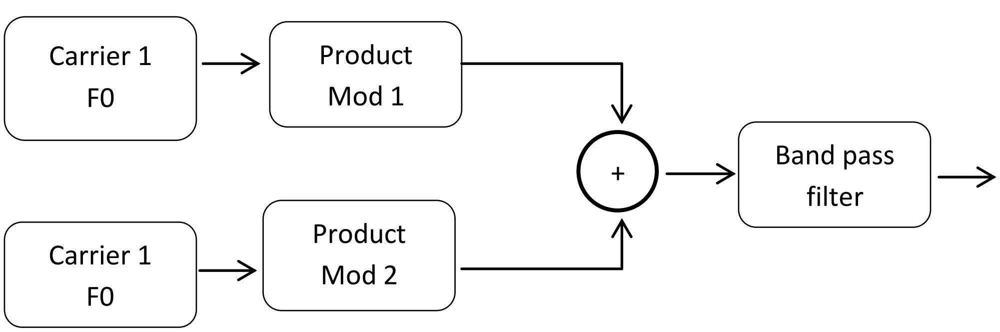
c) Draw block diagram for generation Binary ASK signal and Binary FSK (continuous phase) signal.

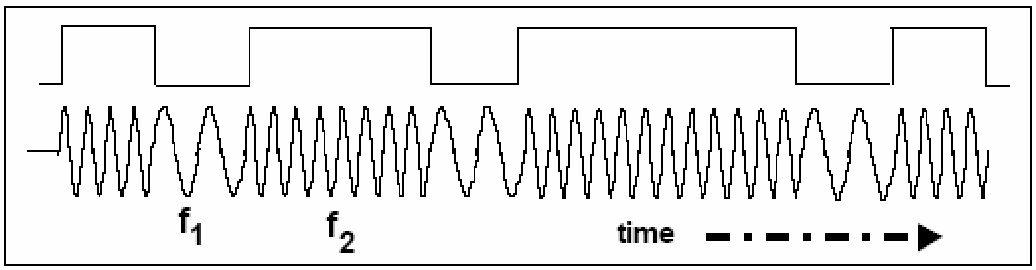
2M EACH





The transmitter block diagram is as shown in figure





d) State principle of DPSK. Draw block diagram of DPSK transmitter.

1M PRINCIPLE

1M EXPL.

2M DIAGRAM.

The differential phase shift keying (DSPK) is a modification of BPSK.

Below Diagram Shows the block diagram of DPSK generator and the relevant waveform are as shown in the below diagram.

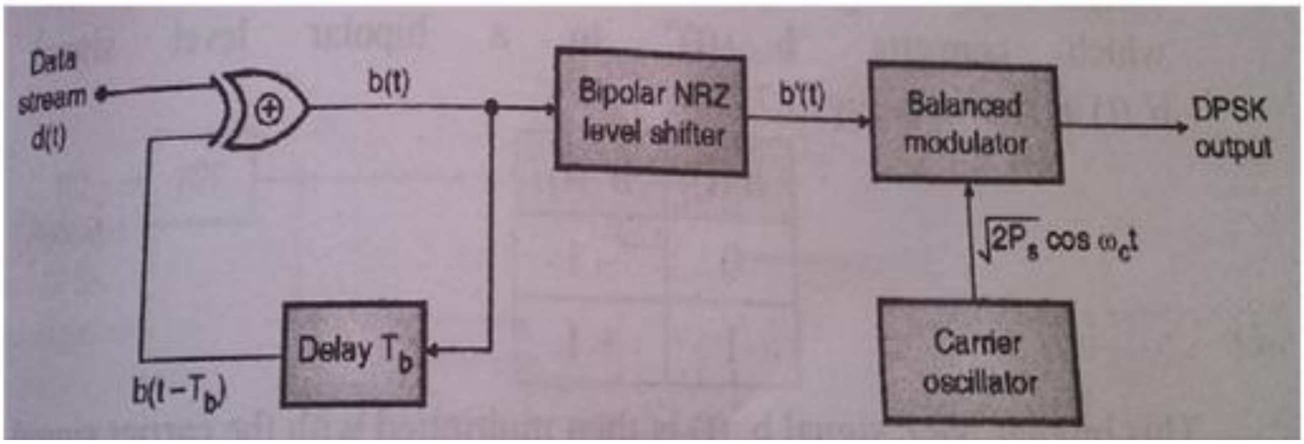
Principle: 1M

It combines, differencial 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 databit, it is squared and thus we can not determine if the received bit is –b(t or b(t.)

Hence DPSK is used to eliminate the ambiguity of the received bit.



2m diag.

Operation of DSPK generator is as follows : 1M

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-ORgate.The delayed represented by "b (t-Tb)".

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

4] Output of EX-OR gate is the applied to a bipolar NRZ level which converts "b(t)" to a bipolar level "b(t)as shown

b(t)	b'(t)
0	-1
1	+1

5] This bipolar NRZ signal b'(t) is then multiplied with the current to produce the DPSK signal. The DPSK output spilt mathematically expressedas:

So when b (t) =1, b'(t) =1 hence

V_{dpsk} (t) $=\sqrt{(2Ps)}\cos wt$

That Means no phase Shift has been introduced

But when b(t) = 0,b' (t) = -1 Hence

V_{dpsk} (t)$=-\sqrt{(2Ps)}\cos wt$

Thus 180

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

e) State the need of source coding? What are the two basic requirements to be met by any source encoder? 4 Marks

A source coder converts the digital signal generated at the source output into other signal in the digital form. Source encoding is used to reduce or eliminate redundancy for efficient representation of of the source output.

Source encoder that satisfies two functional requirements:

1. The code words produced by the encoder are in binary form.
2. The source code is uniquely decodable, so that the original source sequence can be reconstructed perfectly from the encoded binary sequence.

f) List types of error correcting codes. Explain any one in detail.

The error correcting codes are
Linear bock codes
Convolution codes
Hamming codes
Cyclic codes

[List 1m Explanation 1 marks and example 2marks, any relevant data assumed by student should be given 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 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

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

P2	D3	D6	D7
0	1	1	1

= odd parity hence error

So, <u>P2=1</u>

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=

P4	P2	P1
1	1	0

Step 5: decimal equivalent of 110 is 6 hence 6th 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

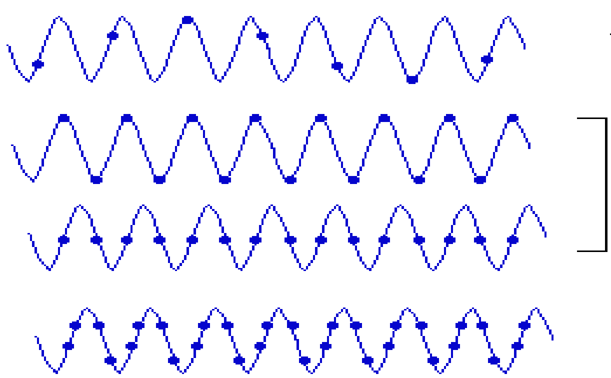
Hence by using this method we can detect as well as correct the error in the transmitted coed word. But it can locate a single bit error and fails in detecting the burst error.

Q.3 Attempt any FOUR of the following: 16

a) With suitable example explain sampling theorem. What is Nyquist Rate?

Example 2M, Nyquist 2M

A continuous time signal can be completely represented as sampled and recovered with least loss if sampling frequency ≥ twice of maximum frequency.



As shown in figure, when the fs< 2f, the points are very distant hence when these samples are joined it doesn't represent the original signal totally.

When the fs= 2f, the sampling points are on proper intervals to obtain minimum required input signal.

When the fs>2f, the input signal can be properly regenerated, when joined the sampling points.

1. A continuous analog signal is shown with finite energy and infinite duration.
2. This signal is strictly band limited that is it does not contain any frequency components above 'w' hz.
3. It is train of pulses, spaced by a period of Ts seconds. This sampling function samples the original sound at rate of fs samples per second.
4. MINIMUM SAMPLING RATE – When the sampling rate becomes exactly equal to twice of maximum input frequency fm , then it is called nyquist criteria.
5. The reciprocal of nyquist rate that is i/2w is called as nyquist interval.
6. Nyquist rate= 2W Hz.

b) State 2-advantages and 2-drawbacks of PPM

Advantages: 1mark each point

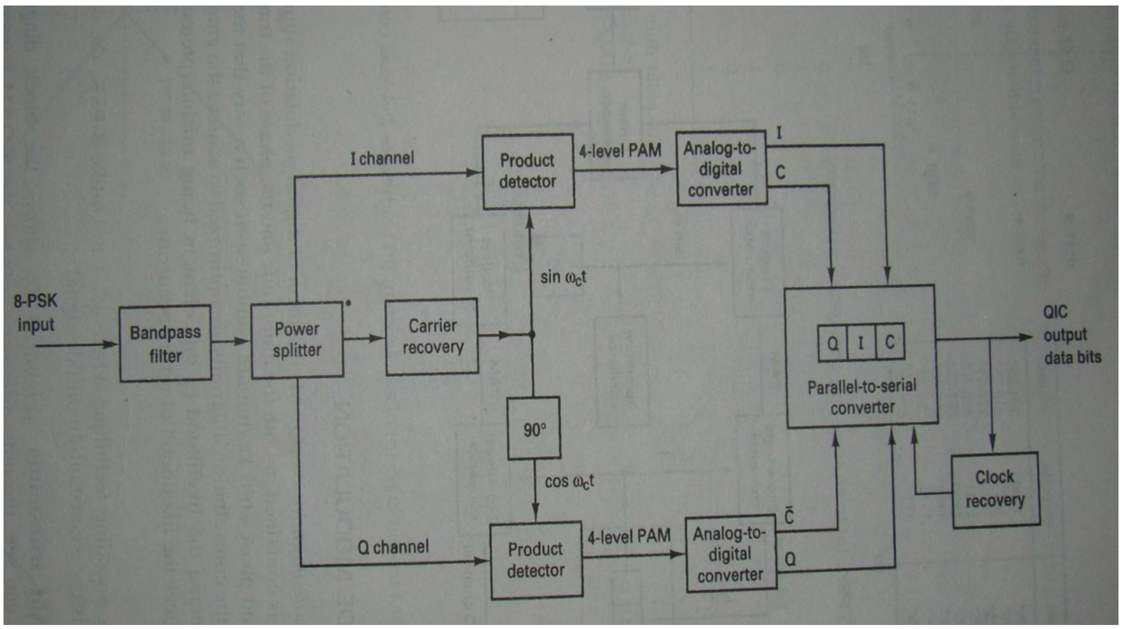
- The noise added in the ppm signal doesn't distort the message.
- The transmitted power is constant.

Disadvantages

- The transmitter has to send synchronizing pulses as the information lies in the position.
- Large bandwidth is required.

c) Explain principle of QAM with block diagram

(2 marks for diagram)



(2 marks for explanation)

- Fig above shows a block diagram of an 8-QAM receiver.
- The power splitter directs the input 8-QAM signal to the I and Q product detectors and the carrier recovery circuit.
- The carrier recovery circuit reproduces the original reference oscillator signal. The incoming 8-QAM signal is mixed with the recovered carrier in the I product detector and with a quadrature carrier in the Q product detector.
- The output of the product detectors are 4-level PAM signals that are fed to the 4 to 2 level analog-to-digital converters (ADCs).

- The outputs from the I channel 4-to-2 level converter are the I and C bits, whereas the outputs form Q channel 4-to-2 level converter are the Q and C bits.
- The parallel to serial logic circuit converts the I/C and Q/C bit pairs to serial I,Q and C output data streams.

d) Explain principle of M-ary encoding

(Explanation 2M; Example 2M)

Ans - In an M-ary signaling scheme, we can send one of the m possible signals such as s1, s2…..sm (t) during each signaling interval of duration of t seconds.

- The number of signals in an M is given as $M=2^m$.
- The M-ary signals are obtained as follows.
- Group of 'N' bits together form a N bit symbols.
- These signals will extend over a period of NTb where tb is duration of one bit.
- Due to grouping of n bit per symbols ,we can have $2^n=M$ possible symbols.
- These M possible signals are represented by sinusoidal signals of duration
- Ts = NTb which differ from one another by a phase of $2\pi/m$ radians .thus M-ary signal is produced at the output.
- Example: QPSK
    - Value of M = 4 also called 4ary psk
    - The number of bits in one symbol are = 2.
    - Hence there are $2^2 = 4$ possible symbols.
    - Symbol1 : 00
    - Symbol2: 01
    - Symbol3: 10
    - Symbol4: 11

e) Compare FSK and DPSK by four points

(1M each)

Parrameter	FSK	DPSK
Variable caharacterstics	Frequency	Phase
Bandwidth	(F1-f0) + fb	Fb
Error probability	Low	Higher
Effect of noise	Lower than dpsk	Higher
Synchronous carrier	Not needed	Not needed
Complexity	Moderately	Very complex

f) State advantages of digital communication over analog communication.

(Any 4,1M each)

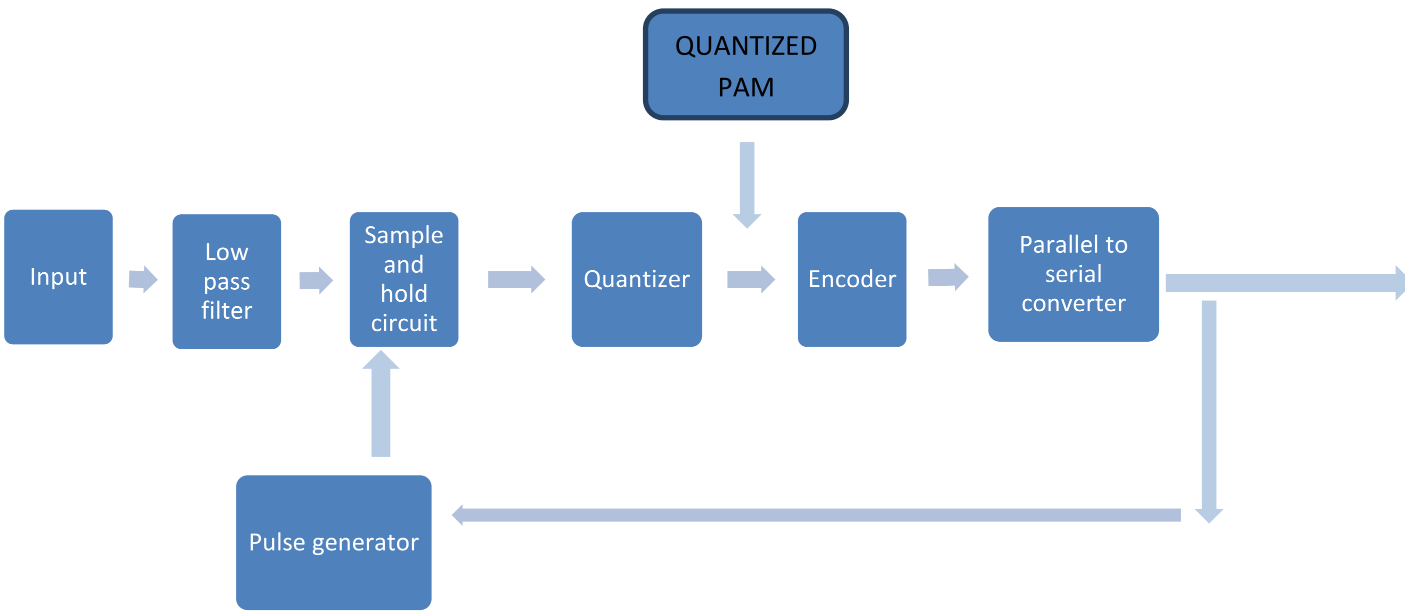
- Noise Immunity
- Digital signals are better suited than analog signals for procession and combining using technique called multiplexing.
- Digital transmission systems are more resistant to analog systems to additive noise because they use signal regeneration rather than signal amplification.
- Digital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

Q.4 Attempt any TWO of the following: 16

a) Draw block diagram of PCM transmitter and receiver system and explain function of each block.

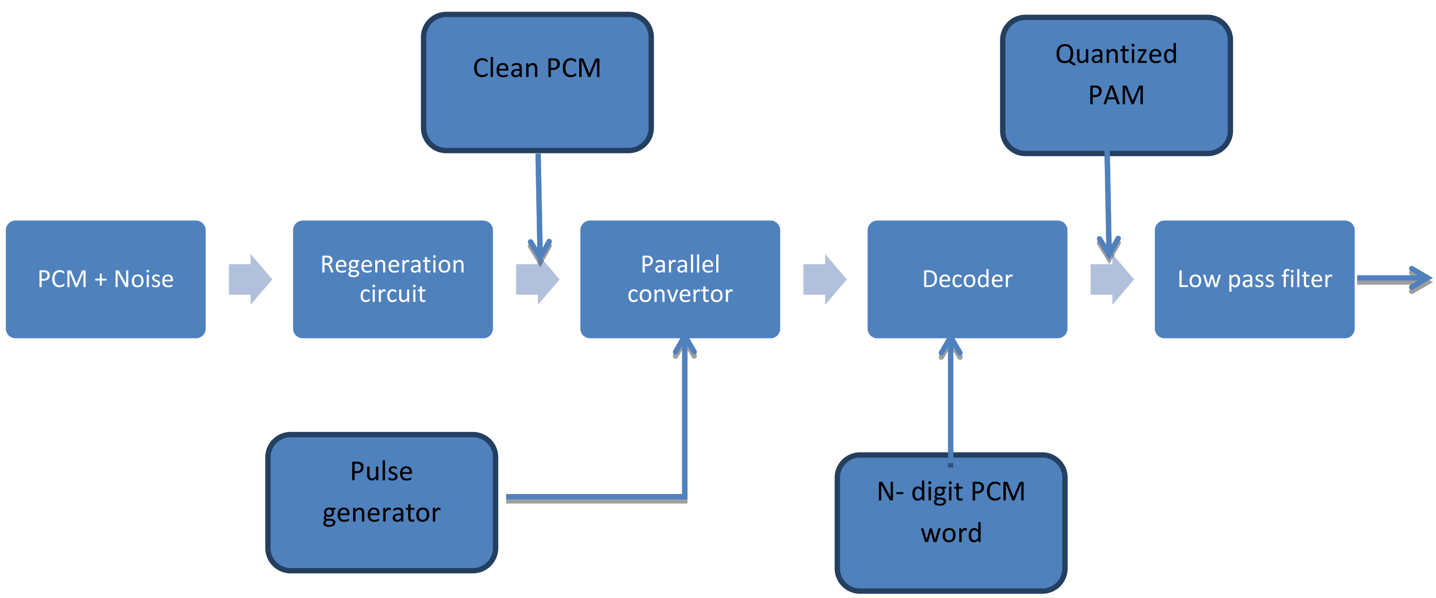
(4M expl for Tx and Diagram, 4M expl Rx and diagram)

Operation of PCM transmitter is as follows:



- The analog signal x (t) is passes through bandlimiting low filter,which has a cut-off frequency fe=W Hz. This will ensure x (t) will not have any frequency component higher than "W". This will eliminate this possibility of aliasing.
- The band limited analog signal is then applied to sample and circuit where adequately high sampling rate .output sample and hold block is a flat topped PAM signal.

- These samples are subjected to operation “quantization” in the “quantizer”.The quantizer is used to reduce effect of noise. The combined effect of sample and quantization produces quantized PAM at the quantizer output.
- 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 value of N can be 8,16,32,64 etc.
- The encoder output is converted into stream of pulses by parallel to serial converted block .thus at the PCM transmitter output train of digital pulses.
- 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.
- Operation of PCM receiver:

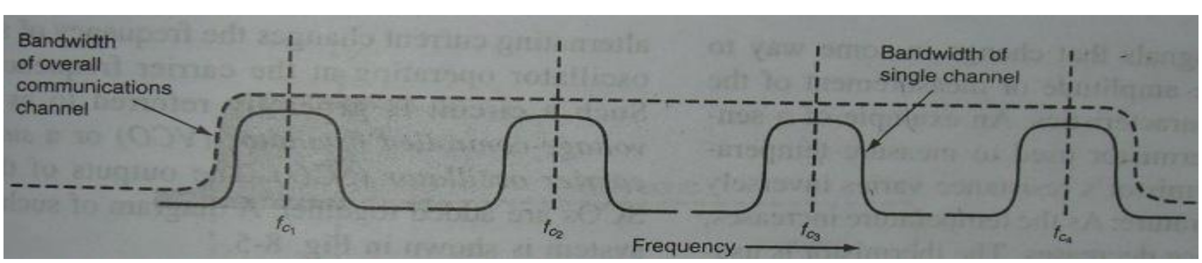


- A PCM signal contaminated with noise is available at the receive input.
- The regeneration circuit at the receiver will separate PCM pulses from noise and will reconstruct original PCM signal.
- The pulse generator has to operate in synchronization with that transmitter .Thus at the regeneration circuit output we get "clear" PCM signal.\
- The reconstruction of PCM signal is possibly due to digital of PCM signal.
- Output of this block is then applied to a decoder.
- the decoder is D to A convertor which performs opposite operation of an encoder
- The decoder output is sequence of quantized PAM signal is thus obtained, at the output of decoder.
- This quantized PAM signal is passed through a low pass filter recovers the analog signal x (t).

b) Draw block diagram of FDM transmitter and FDM receiver. Also, explain its working.

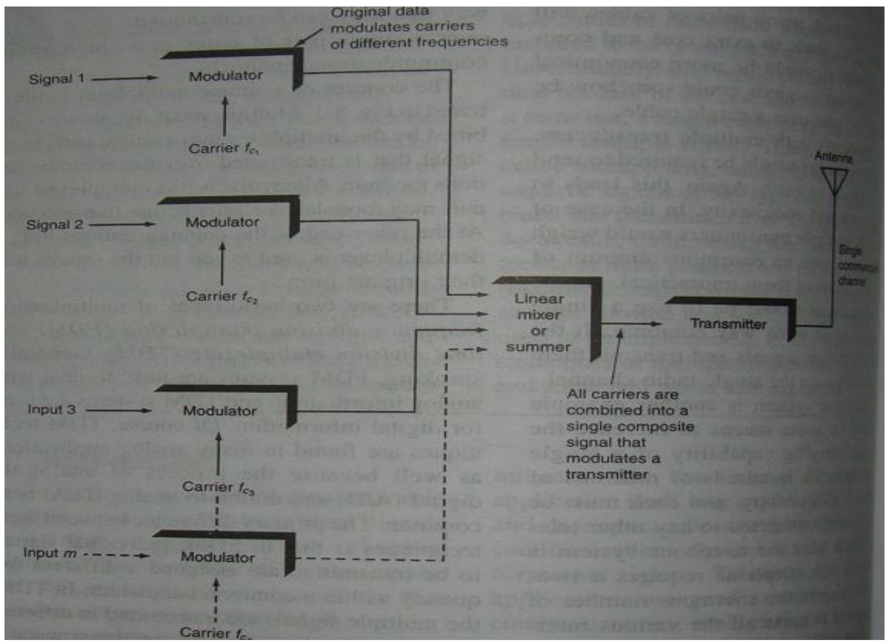
(Concept 2M; Block diagram of transmitter 2M and Receiver 2M ; 2M explanation)

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. The spectrum of the FDM signal is as shown below.



Block diagram of multiplexer of FDM:

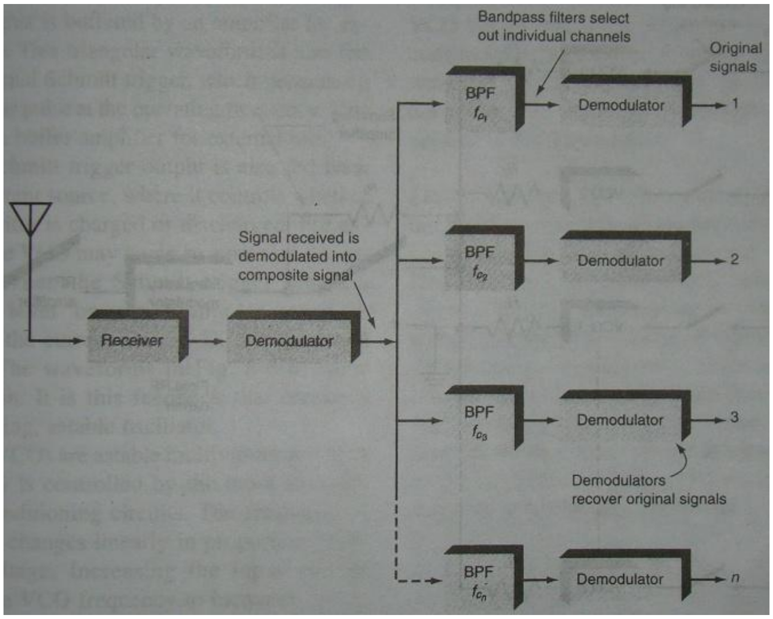
Fig shows a general block diagram of FDM system. Each signal to be transmitted is fed to modulator circuit. The carrier for each modulation fc is on a different frequency. The carrier frequencies are equally spaced from one another over a specific frequency range. Each input signals given portion of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



The modulator output having side band information are 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.

Block diagram of de multiplexer of FDM:

The receiving portion of FDM is as shown below. A receiver picks up the signal and demodulates it. This is sent to a group of band pass filter. Each centered on one of the carrier frequencies. Each filter passes only its channel and rejects all other.



A channel demodulator then recovers each original input signal. The example of FDM system is Telemetry, telephone and FM stereo.

c)

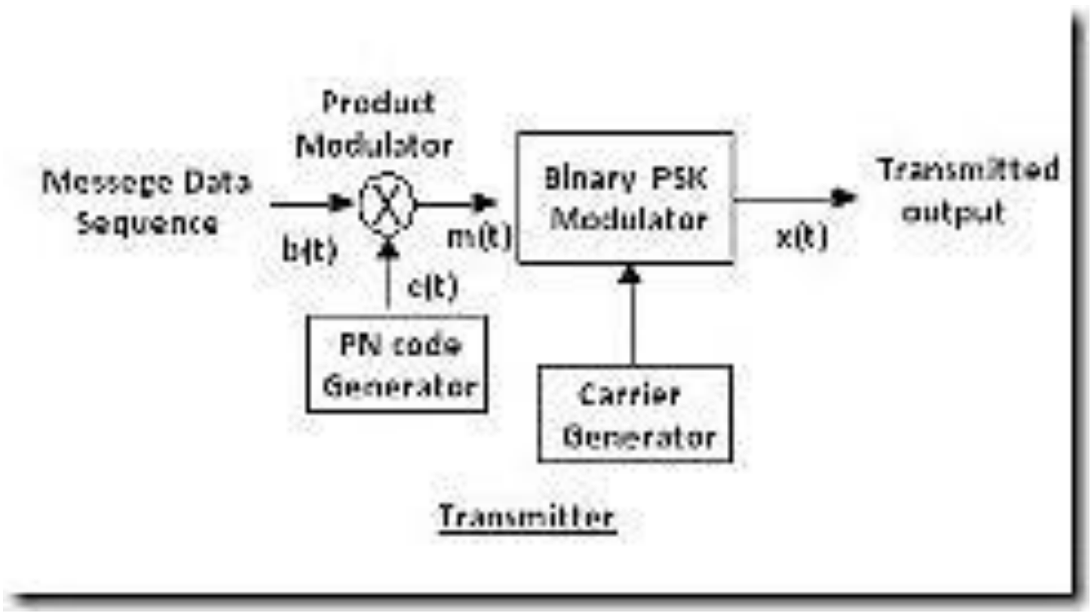
i) With neat diagram explain the DS spread-spectrum signal.

(Diagram 3M; Expl. 3M)

ii) state applications of SS Modulation

(any 2, 1M each)

The figure shows the block diagram of transmitter of direct sequence spread spectrum signal.



1. The binary sequence d(t) is converted into nrz signal b(t) by using an Nrz encoder.
2. The nrz signal b(t) is used to modulate the PN sequence c(t) generated by a pn code generator.
3. The transmitter uses two stages of modulation. The first stage uses a product modulator and a multiplier with b(t) and c(t) as input and the second stage consist of BPSK modulator.
4. The modulated signal at the output of product modulator is m (t) is used to modulate the carrier of BPSK modulation.
5. The transmitted signal x (t) is a direct sequence spread spectrum signal.
    - Application of SS modulation:
1. Rejecting unintentional interference from another user
2. To avoid self-interference from multi path propagation.
3. In detection of Low probability of intercept signal.
4. In obtaining message privacy.

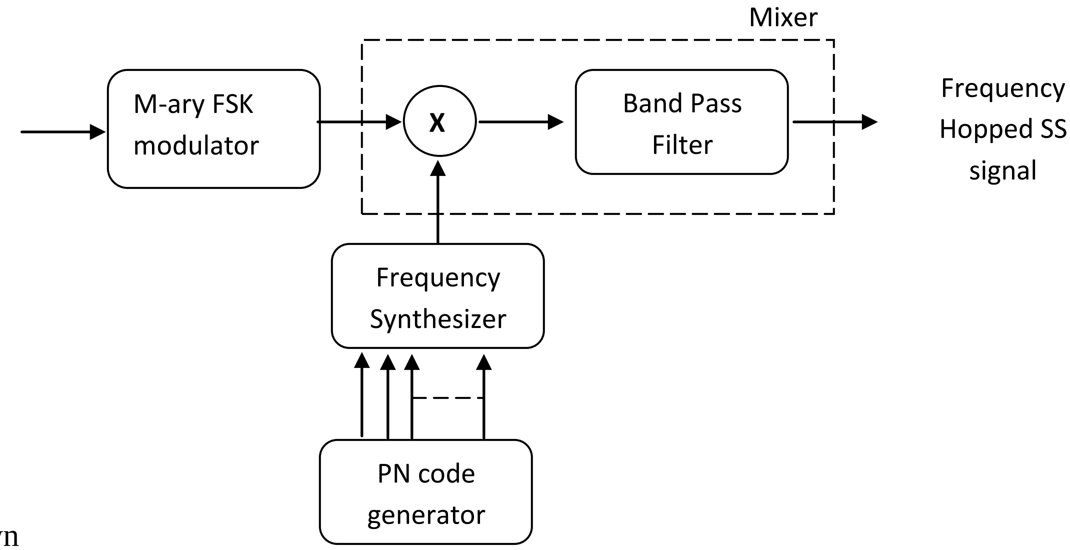
Q.5 Attempt any TWO of the following 16

a) Explain working of FHSS transmitter with block diagram. What is slow hopping FHSS?

Depending on the rate of frequency hopping, there are two types. (3M +3M+2 M)

1) Slow frequency hopping.
2) Fast frequency hopping

The block diagram of frequency hopping technique is as



shown

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

Slow frequency hopping-

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

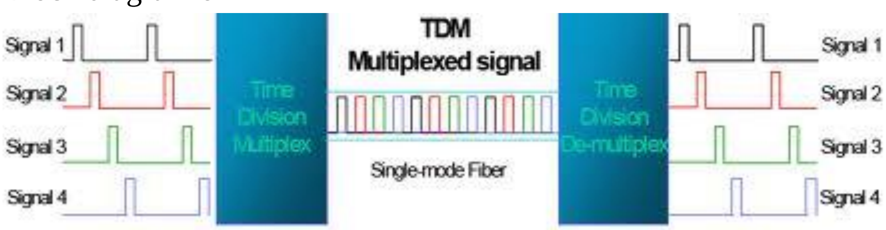
Here frequency hopping takes place slowly.

b) Draw block diagram of TDMA. State advantages of TDMA over FDMA (any 4)

Advantages of TDMA over FDMA

(Diagram 4M; Advantages 4M)

Block diagram of TDMA



- Entire channel bandwidth is allotted to the the single chanmel.
- Frequency selective does not occur in TDMA.
- FDMA cannot work well in digital data.
- It is not necessary to separate out various channels at the receiver.

c)

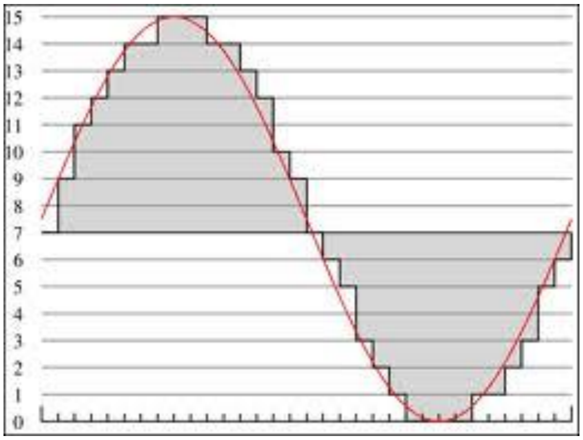
i) Explain the term Quantization and Quantization noise.

(2M diagram, 2M expl)

ii) State advantages and drawbacks of DM over PCM.

(any 2 points each, 2M adv-2-marks Dis adv)

Ans: (i) Quantization:



After the sampling we have a sequence of numbers which can theoretically still take on any value on acontinuous range of values. Because this range is continuous, there are infinitely many possible values for each number.

In order to be able to represent each number fromsuch a continuous range, we would need an infinite number of digits –which is not possible.

Instead,we must represent our numbers with a finite number of digits, that is: after discretizing the time-variable,we now have to discretize the amplitude-variable as well. This discretization of the amplitude values iscalled quantization.

Quantisation noise:

In quantisation process, the infinite values of the samples are rounded off to the nearest quantized values.

When forcing an arbitrary signal value to its closest quantization level, this quantised value induces error.

(ii) advantages

1. Simple system
2. Lass bandwidth
3. Less quantization noise

(ii) Dis advantages:

1. Slope overload occurs when the slope of input analog signal is very high
2. When input analog signal has slope almost equal to zero then granular noise occurs.

Q.6 Attempt any FOUR of the following: 16

a) Draw and explain block diagram of digital Communication System.
(2M Diagram, 2M expl)

Block Explanation-

It consist of following block

<u>1) Information source-</u> it may be in analog forming. Output of microphone gives analog signal. And if source is computer data then it is a digital form.

<u>2) Source encoder-</u> the source encoder converts the signal produced by the information source into DataStream. If i/p signal is analog it can be converted in to digital form using a to d converter. If the i/p to the source encoder is a stream of symbols it can be converted into a stream of 1s and 0s using some coding mechanism.

3) Channel Encoder- if we have to encode the information covertly, even if errors are introduced in the medium. We need to put some additional bits in the source, so that additional information can be used to detect and coolest the errors, this process of adding bits is done by channel encoder. In channel encoding redundancy is introduced so that at the receiving end the redundancy is introduced so that at the receiving end redundant bit can be used for error detection and error correction.

4) Modulator- here the modulation is done for transferring the signal, so that the signal I can be transmitted through the medium easily.

5) Channel-it is the medium through winch the o/p of modulator along with some noise is transmitted and gives to demodulator. This channel is called discrete channel because its input as well as o/p both are in discrete nature.

6) Demodulator- it performs inverse operation than that of modulator.

7) Channel decoder- it checks the received bits and also detect and correct the errors, using additional data introduced by channel encoder.

8) Source decoder-it converts the bit stream in to actual information, here digital to analog conversion is done if the symbols are coded in to is and os at the source decoder the bits are converted in to symbols.

9) Information sink-here the information sink absorbs the original information.

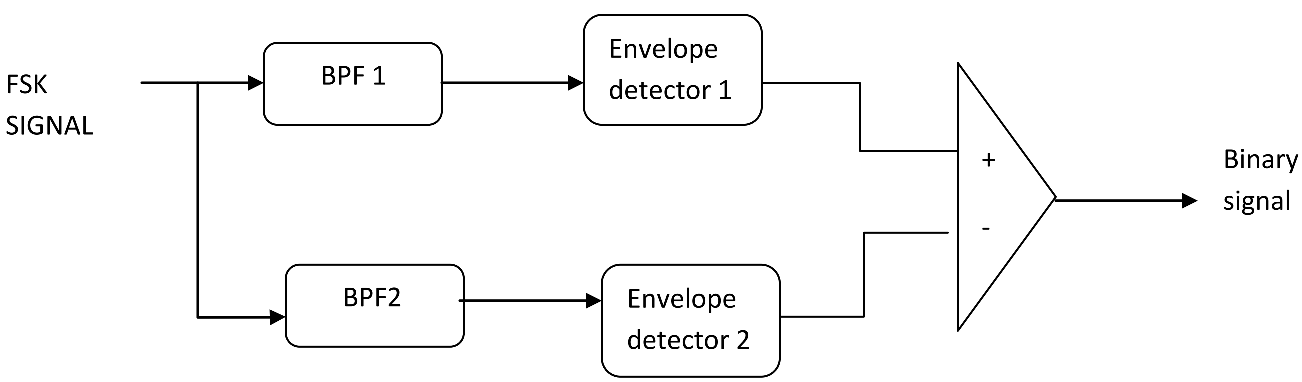
b) Draw and explain ASK and FSK Receiver block diagram with block diagram, 4M (2+2)

Ans:

ASK receiver:

- The received signal is ASK added with channel noise and hence it will not be a copy of the binary sequence TTL waveform. Band limiting will have shaped it.
- Having a very definite envelope, an envelope detector can be used as the first step in recovering the original sequence. Further processing can be employed to regenerate the true binary waveform.
- If the ASK has been bandlimited before or during transmission (or even by the receiver itself) then the recovered message, in the demodulator, will need restoration (‘cleaning up’) to its original bi-polar format.
- Some sort of decision device is then required to regenerate the original binary sequence. This could be done with a regenerator.

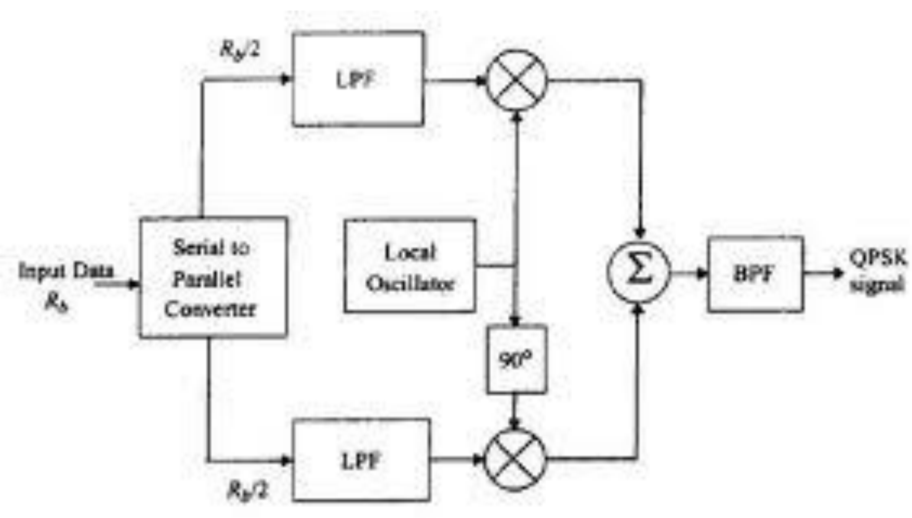
FSK RECEIVER:



- The FSK signal is applied to the input of two band pass filters centered at different frequencies which are carrying the information in the received signal.
- The output from each BPF looks like an amplitude shift keyed (ASK) signal. These can be demodulated asynchronously, using the envelope.
- The decision circuit, to which the outputs of the envelope detectors are presented, selects the output which is the most likely one of the two inputs. It also re-shapes the waveform from a bandlimited to a rectangular form.

- This is, in effect, a two channel receiver. The bandwidth of each is dependent on the message bit rate. There will be a minimum frequency separation required of the two tones.

c) Explain working of QPSK transmitter. 4 M



- First unipolar binary messages are converted into bipolar NRZ sequence.
- Next, the bit stream is split into in-phase (even) and quadrature (odd) bit streams each with a bit rate .
- Two orthogonal carriers separately modulate two binary sequences.
- They are summed to produce a QPSK signal.
- Finally, the bandpass filter at the output prevents the spillover of signal energy into adjacent channels and also removes the out-of-band spurious signals generated during the modulation process. In most wireless and other RF implementations, pulse shaping is done at the baseband to provide proper RF filtering at the output.

d)A sequence of independent symbols A, B, C and D with the probabilities 0.5. 0.25, 0.125 and 0.125 respectively is produced by a source. Determine **2M+2M**

i) **the binary code for output of this source and**

ii) **the source entropy**

$P(B) = 0.25$ - Given.
$P(C) = 0.125$
$P(D) = 0.125$.

Solution →

(i) Binary code for output for the source

Since the symbols are A, B, C & D we have to generate 4 diffirent (unique) codes (symbols).

Symbol with maximum probability should have minimum number of bits. So, binary code for symbol

$A = 1$, $B = 0$, $C = 01$ & $D = 10$

As C & D have same probability they are distinguished from each other.

(ii) $H = P_K \log_2\left(\frac{1}{P_K}\right)$

$$= P_A \log_2\left(\frac{1}{P_A}\right) + P_B \log_2\left(\frac{1}{P_B}\right) + P_C \log_2\left(\frac{1}{P_C}\right) + P_D \log_2\left(\frac{1}{P_D}\right)$$

$$= 0.5 \log_2 \frac{1}{0.5} + 0.25 \log_2\left(\frac{1}{0.25}\right) + 0.125 \log_2\left(\frac{1}{0.125}\right) + 0.125 \log_2\left(\frac{1}{0.125}\right)$$

$$= 0.5 + 0.5 + 0.375 + 0.375$$

$$= \underline{1.75} \text{ bits/message.}$$

c) State two features each for baudot code and EBCDIC code. 2M+2M

→

Baudot Code:

- It consisted originally of groups of five “on” and “off” signals of equal duration composed of short dots and long dashes.
- In Baudot Code, each group of five signals represented a single character; the code therefore provided 32 combinations.
- Modern versions of the Baudot Code usually use groups of seven or eight “on” and “off” signals. Groups of seven permit transmission of 128 characters; with groups of eight, one member may be used for error correction or other function

EBCDIC code:

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

d) Explain terms bit rate and baud rate. Why bit and baud rate are same for transmission of a binary signal, (2M+2M)

Bit rate

It’s the number of bit transmitted or sent one second .its unit is bits per second. If bit duration is ‘tb’ then bit rate will be 1/tb

Baud rate

Baud is the unit of signaling speed or the rate of symbol transmission or it indicates the rate at which a signal level changes over a given period of time.

For a binary signal, the symbols are 0 and 1 which are two individual bits and they require only one bit period each. Hence the bit rate = baud rate.