PAPER SOLUTION

Class: S.E. Computer SEM. IV (Rev.) Summer 2010

Subject: Analog and Digital Communication

PAPER CODE: AN-3442

Q.1 (a)

$V(t) = 12\ (1 + \sin 12.566 \times 10^3 t) \sin 18.85 \times 10^6 t$ Volts.

Comparing with standard AM equation:

$E_{AM} = E_C[1 + m \sin\omega_M t] \sin\omega_C t$

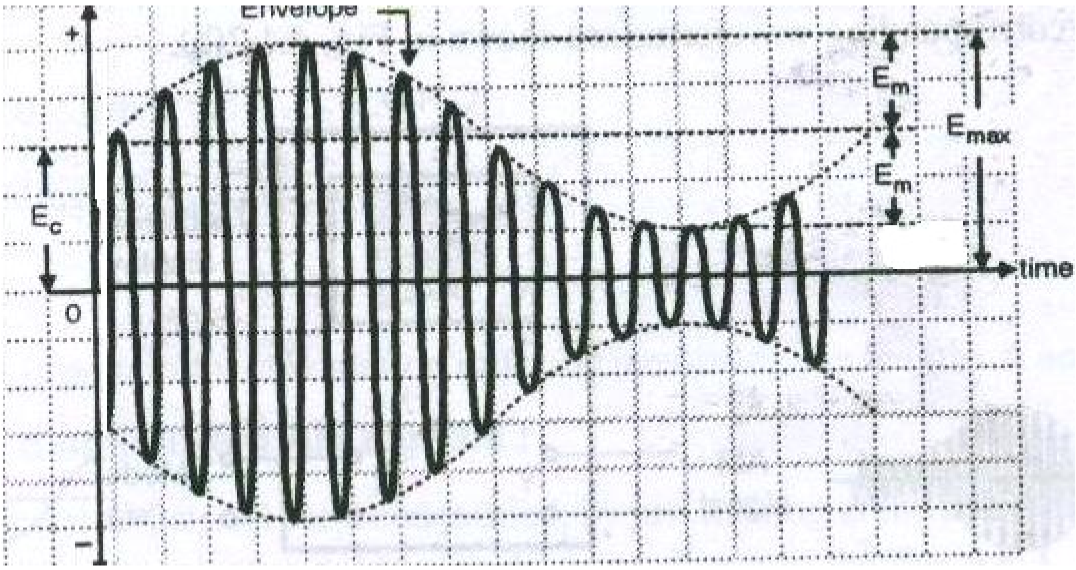
$E_C = 12V \quad m=1 \quad$ So $E_M = E_C = 12V$

$$f_C = \frac{18.5 \times 10^6}{2\pi}$$

$= 3$ MHz

$$f_M = \frac{12.566 \times 10^3}{2\pi}$$

$= 2$ KHz



M=1 --------------------Modulation Index

$f_{USB} = f_C + f_M = 3002$ KHz

$f_{LSB} = f_C - f_M = 2998$ KHz

$P_T = P_C[\ 1 + m*m/2]$

$= 1.5P_C$

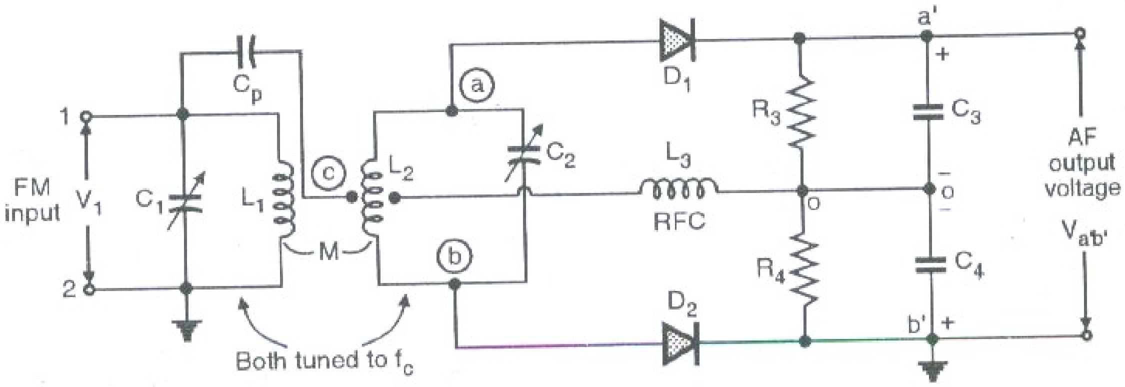
$$= \frac{1.5 \times E_C \times E_C}{2 \times 50}$$

$= 2.16W$

B.W. $= 2 * f_M$

$= 4$ KHz

(b)



Operation:

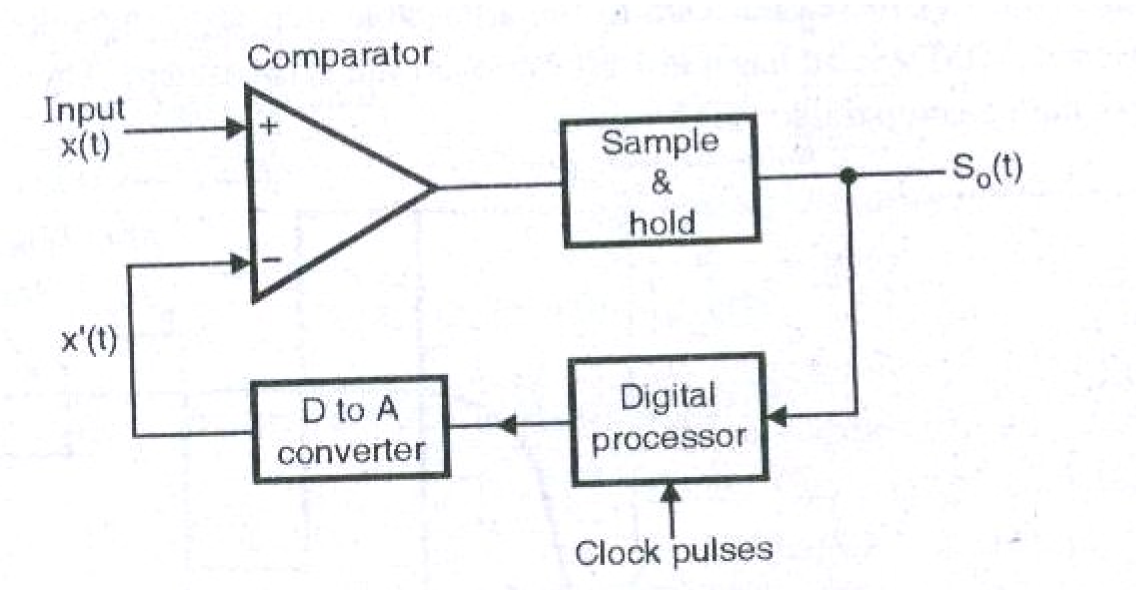
O/p voltage is equal to half of difference between O/P voltages from individual diodes. Same like phase discriminator

Amplitude limiting is incorporated due to large capacitance C5, due to this amplitude limiter is not required to ratio detector.

Advantages:

1. Easy to align.
2. Very good linearity due to linear phase relationship between primary and secondary.
3. Amplitude limiter is provided internally so additional limiter is not required.

Q.2 (a)



$S_0(t)$= + 1 If x(t)>x'(t) just before k^{th} clock edge.
= -1 If x(t)<x'(t) just before k^{th} clock edge.

- In response to the kth clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. (k- 1))" clock edge.
- If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "8". If the directions are opposite then the processor will decrease the magnitude of the present step by "8",
- So(t)in the Fig. 10.11.1, i.e. the output of the ADM system is given as,

So (t) = + 1 if x (t) > x' (t) just before kth clock edge, and
So (t) = - 1 if x (t) < x' (t) just before the kth clock edge.

- Then the step size at the sampling instant k is given by,

$\delta (k) = [\delta (k-1)] So(k) + \delta So(k-1)$

let us take an example,
refer fig. below. Assume k=6

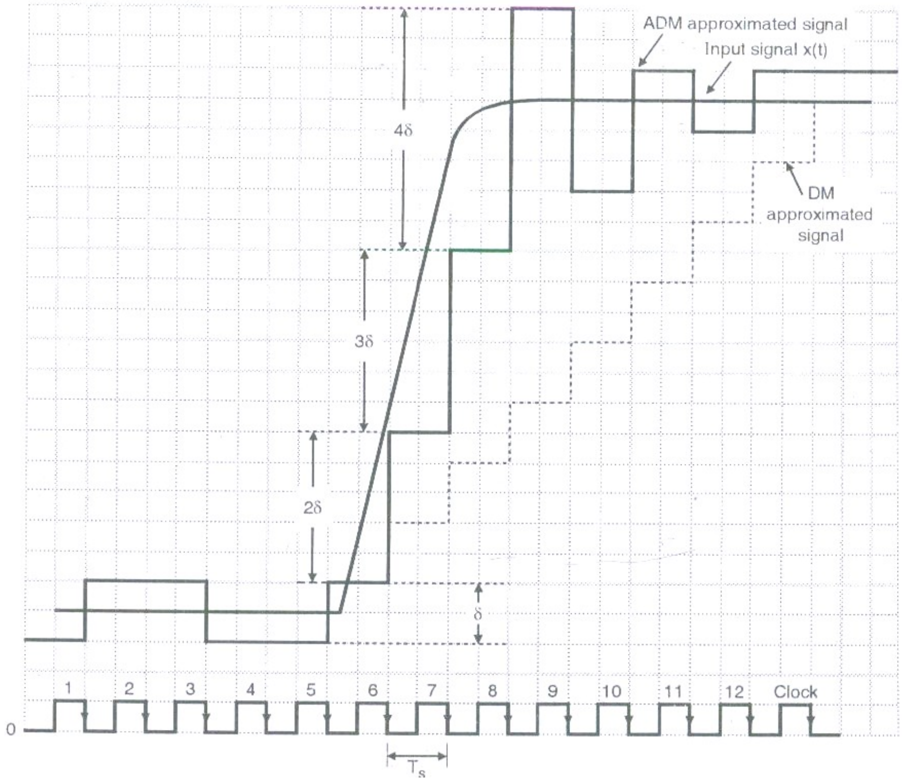
k-1=5,

$\delta(k-1) = \delta(5) = \delta$
So(k)=So(6)=+1
So(k-1)=So(5)=+1

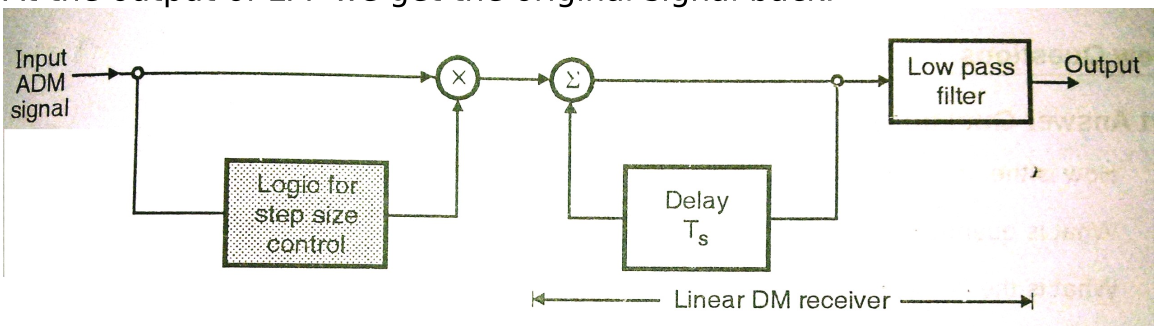
"Substitute in Equation
to get, $\delta(6) = \delta + \delta = 2\delta$

Look at the Fig. , the step size at the 6th clock edge is 2 δ.

As shown in Fig. 10.11.2, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rale than the PCM system. Therefore the BW required is also less than a comparable PCM system.



ADM signal is first converted into a D.M. signal with the help of a step size control logic and then applied to D.M. receiver.
At the output of LPF we get the original signal back.



Q.2 (b)
Shannon Hartley capacity theorem:
The information capacity of a continuous channel of bandwidth B Hz perturbed by AWGN of PSD of $N_0/2$ is given by,

$$C = B \log_2[1 + P/(N_0 B)] \text{ bits per second}$$

Where P is average transmitted power.

It relates channel bandwidth, avg. transmitted power amd noise PSD at channel output.
Channel capacity C depends linearly on bandwidth B but dependence of C on ratio $P/(N_0B)$ is of logarithmic in nature.

So it is to increase information capacity of channel by increasing bandwidth rather than by increasing signal power P for constant noise variance.
It is not possible to transmit at a rate higher than C bps by any encoding system without definite probability of error .
If communication channel is noiseless then N=0, hence $P/N \to \infty$ and so $C \to \infty$.
Thus noiseless channel will have infinite capacity.

Shannon's Limit:
Consider that same AWGN is present hence P/N is not infinite.
As BW approaches ∞ the channel capacity does not become infinite since $N = N_0 B$ will also increase with bandwidth B.
This will reduce value of P/N with increase in B, assuming signal power P to be constant.
Thus we conclude that an ideal system with infinite bandwidth has a finite channel capacity.

$$C_\infty = 1.44\, P/N_0$$

Q.3 (a)

Intersymbol Interference : (Pulse Transmission over Bandlimited Channel) ________________________________

- In a communication system when the data is being transmitted in the form of pulses (bits), the output produced at the receiver due to the other bits or symbols interferes with the output produced by the desired bit.
- This is called as intersymbol interference (ISI). The intersymbol interference will introduce errors in the detected signal.
- The ISI arises due to the imperfections in the overall frequency response of the system.

When a short pulse of duration Tb seconds is transmitted through a bandlimited system, then the frequency components contained in the input pulse are differentially attenuated and more importantly differentially delayed by the system.

Due to this the pulse appearing at the output of the system will be "dispersed" over an interval which is longer than " Tb" seconds.

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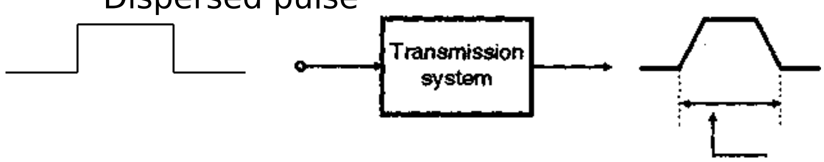
Due to this dispersion, the symbols each of duration " Tb" will interfere with each other when
transmitted over the communication channel. This will result in the intersymbol interference (1SI).

The transmitted pulse of duration Tb seconds and the dispersed pulse of duration more than Tb seconds are shown in Fig

Input

Dispersed pulse



I/O Duration longer than Tb

Cause of ISI

Effect of ISI

- In the absence of ISI and noise, the transmitted bit can be decoded correctly at the receiver.The presence of ISI will introduce errors in the decision device at the receiver output.
- Thus the receiver can make an error in deciding whether it has received a logic 1 or a logic 0.
- Another effect of ISI is the cross talk whjich may take place due to the overlapping of the spreading of adjucent pulses.
- It is necessary to use the special filters called equalizers in order to reduces ISI and its effects.

Equalization :

Whenever a signal is passed through a communication channel, distortion will be introduced. To compensate for the linear distortion, we can use a network called "equalizer" connected in cascade with the channel or system as shown in Fig.

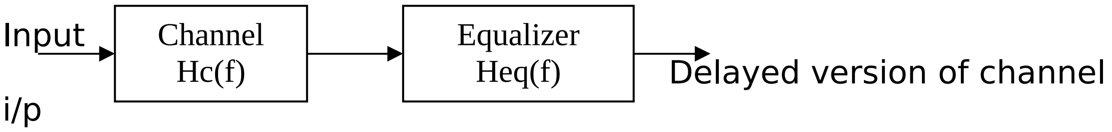


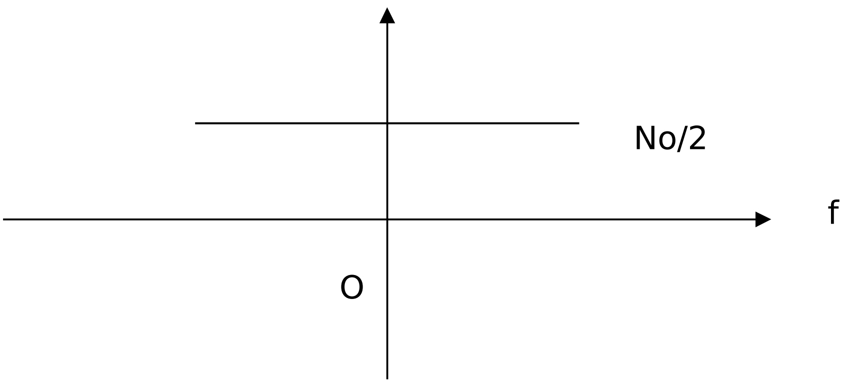
Fig. Block diagram of equalization Principle of Equalizer :
The equalizer is designed in such a way that inside the frequency band of interest, tlic overall amplitude and phase responses of the cascade system shown in Fig. are approximately equal to amplitude and phase responses for the distortionless transmission.
Consider a communication channel with a transfer function Hc (f).
Let the transfer function of an equalizer be Heq(f) . Then the overall transfer function of the cascade connection is given by Hc(f)Heq(f).
For the overall transmission through the cascaded connection of Fig. 8.25.1 to be distortionless, the overall transfer function should satisfy the following expression :

$$H_c(f)\,H_{eq}(f) = K e^{-j2\pi nft}$$

where K =   Scaling factor and t = Constant time delay
Therefore the transfer function of the equalizer is given by,

$$H_{eq}(f) = \frac{K e^{-j2\pi nft}}{H_c(f)}$$

Q.3(a)
ii)
White Gaussian noise is a noise whose PSD is uniform over entire range of interest.



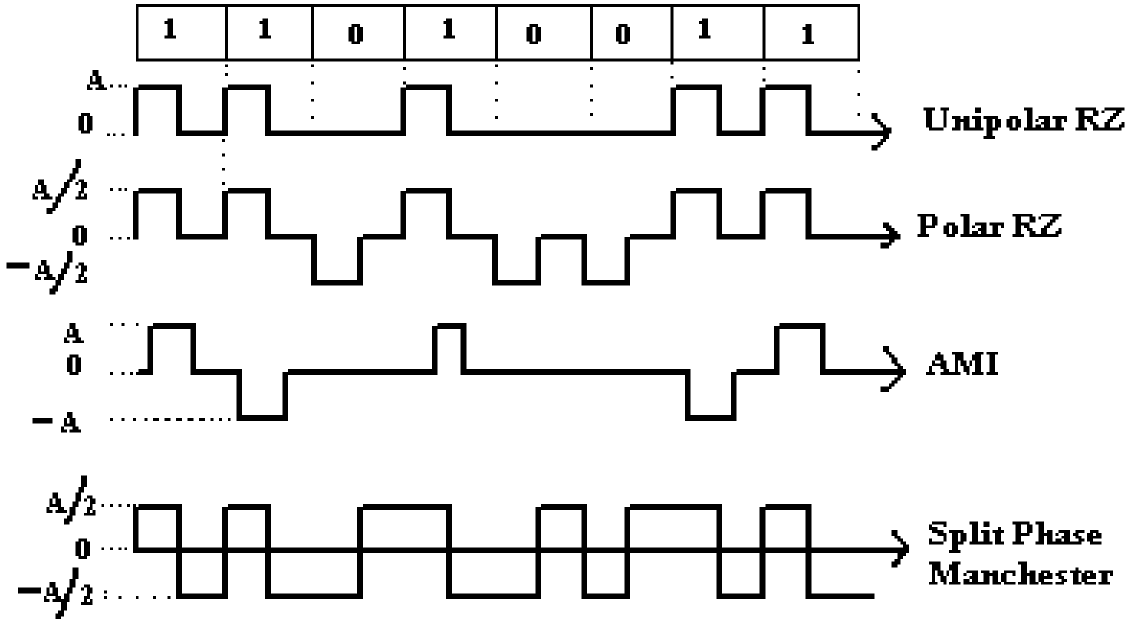
It contains all frequency components in equal proportion. White noise has a Gaussian distribution, i.e. it has Gaussian PDF. Hence it is called as Gaussian noise.

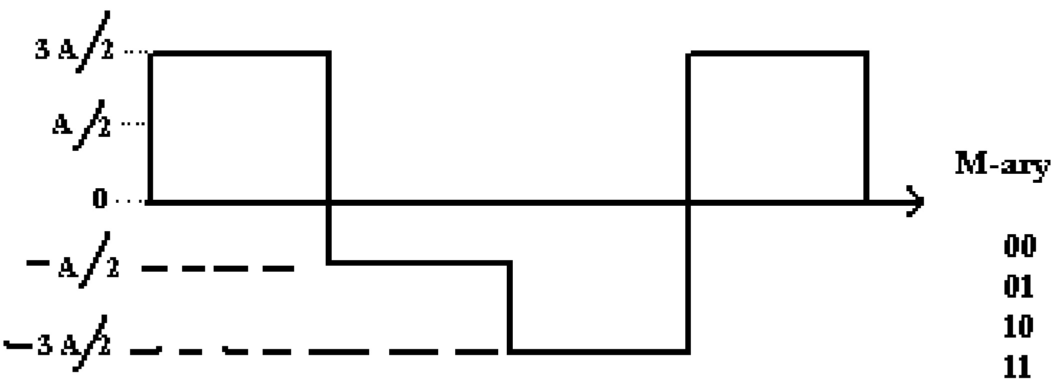
Power Spectral Density   Sn(f)= No/2   Where   No = KTe
K= Boltzmann’s constant

Te=Equivalent noise temp. of system.

Q.3 (b)

Line code represents a particular waveform used for conveying the bit sequence.





00	-3A/2
01	-A/2
10	A/2
11	3A/2

Q.4 (a)

i)Viterbi Algorithm:

- The ML decoding is practically achieved via a minimum length decoder which is usually referred to as Viterbi algorithm.
- The VA is therefore an optimal decoding technique for the memory less channel.
- The viterbi algorithm operates on the principle of maximum likelihood decoding and achieves
  Optimum performance.
- The maximum likelihood decoder has to examine the entire received sequence Y and find a valid path which has the smallest hamming distance from Y.
- But here are 2^N possible paths for a message sequence of N bits. These are a large number
  of paths.
- The viterbi algorithm applies the maximum likelihood principle to limit the comparison of so many surviving paths, to make the maximum likelihood decoding possible.

ii) Cyclic Codes:

A code is cyclic if

i) It is linear code and

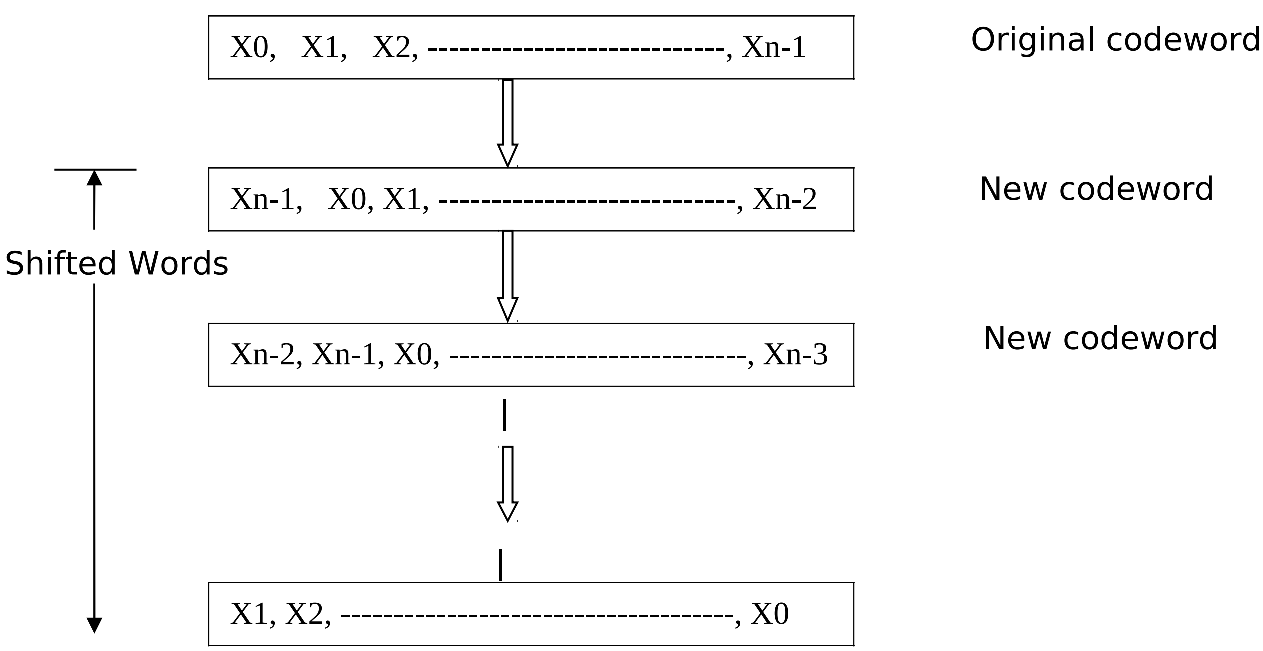
ii) If it satisfies the cycjic property. So a cyclic code exhibits the two properties namely linearity and cyclic properties.

Linearity Property:

A code is said to be linear if sum of any two codeword is another codeword. This property states that
cyclic codes are linear block codes.

Cyclic Property:

Code is said to be cyclic if any cyclic shift of a codeword results in the formation of another codeword.



New codeword

Q.4 (b)

$$G = \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 \\ 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 & 1 & 0 \end{bmatrix}$$

$$G = [\ I_k : P]$$

$$P = \begin{bmatrix} 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \end{bmatrix}$$

Code vectors:

Msg. Words	Parity Bits	Codeword
000	000	000000
001	110	001110
010	101	010101
011	011	011011
100	011	100011
101	101	101101
110	110	110110
111	000	111000

$$H = [\ P^T : I]$$

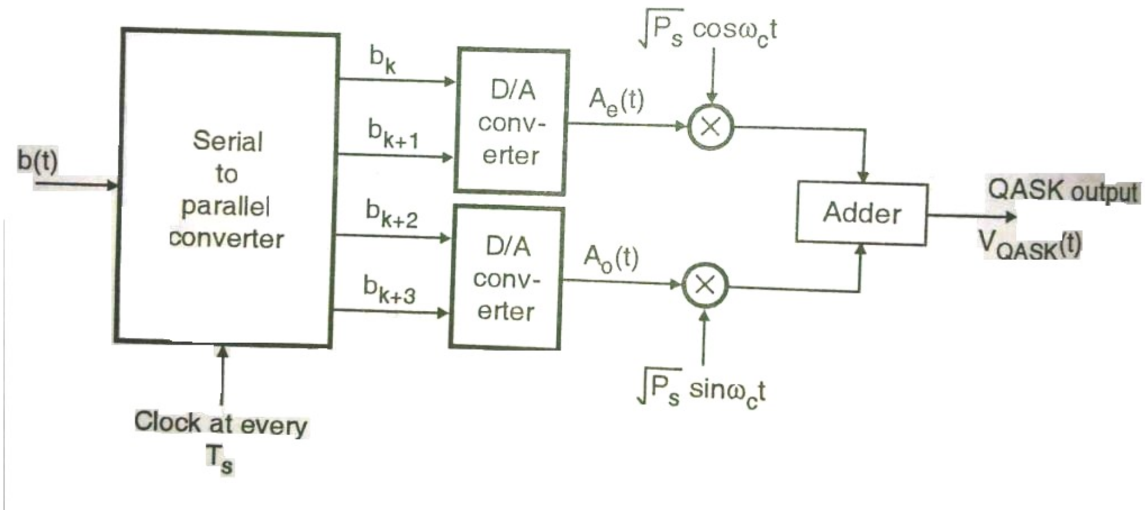
$$H = \begin{bmatrix} 011100 \\ 101010 \\ 110001 \end{bmatrix}$$

Error Syndrome

$$S = E \quad H^T$$

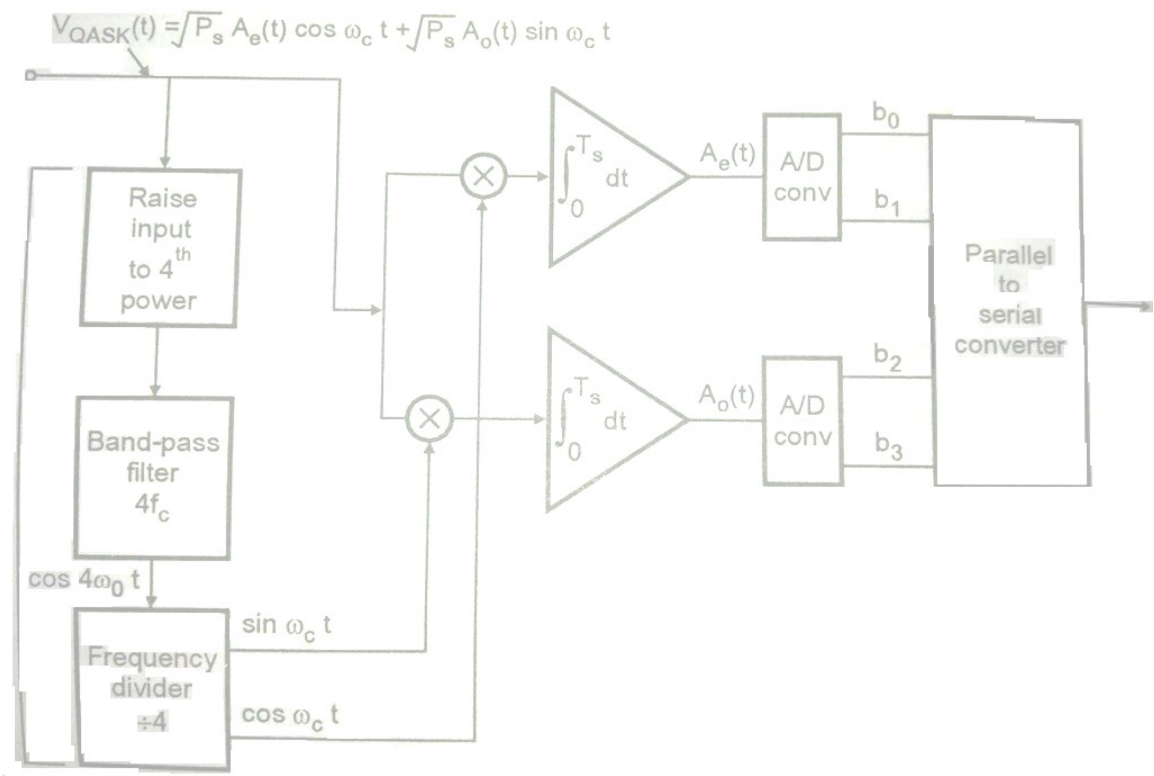
Error Vector	Syndrome Vector
000000	000
100000	011
010000	101
001000	110
000100	100
000010	010
000001	001

Q. 5(a)



Explanation :

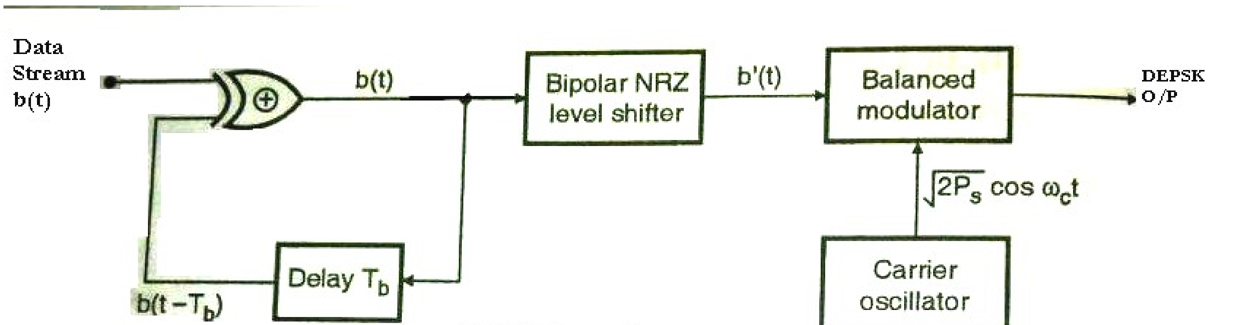
Working of each block.
Equation for QAM signal.



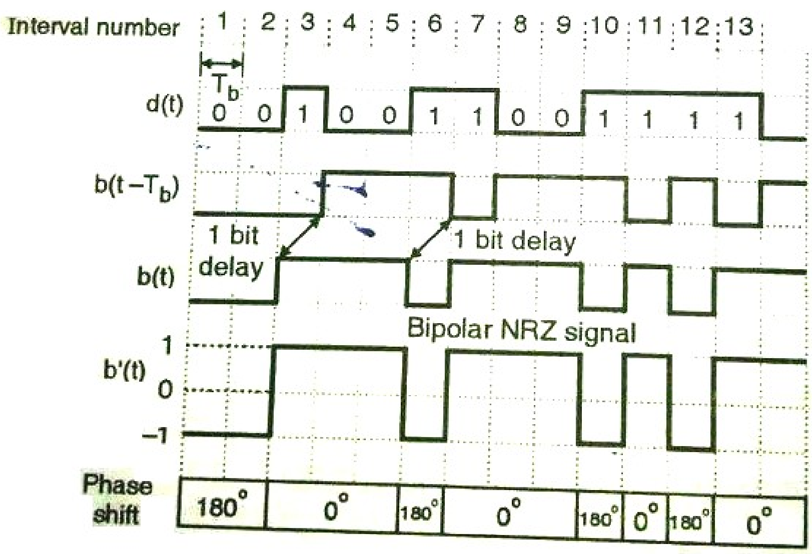
Explanation:

Working of each block.
Recovery of original signal from received QAM signal.

Q. 5(b)



DEPSK Transmitter



Operation :

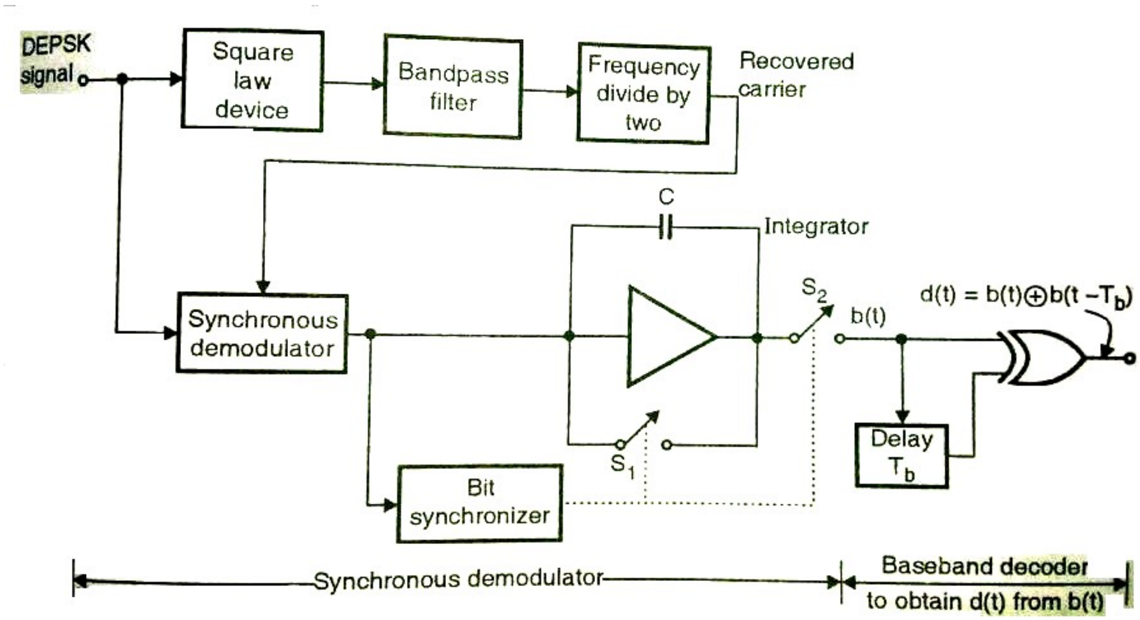
Operation of the DEPSK generator is as follows :

• d(t) represents the data stream which is to be transmitted. It is applied to one input of an EXOR logic gate

• The EX-OR gate output "b (t)" is delayed by one bit period Tb and applied to the other input of the EX-OR gate. The delayed output is represented by “b (t - Tb)".

• Depending on the values of "d(t)" and “b (t-Tb)", the EX-OR gate produces the output seguence"b(t)". The waveforms jbr DPSK generator are as shown in Fig. These waveforms have been drawn by arbitrarily assuming that in the first interval b(O) = 0.

• Output of EXOR gate is then applied to bipolar NRZ level shifter, which converts “b (t)" to a bipolar level signal b' (t) as shown in Fig.



• The block diagram of DEPSK receiver is shown in Fig. It shows that the signal b (t) is recovered from the received signal, using the synchronous demodulation technique.
This is same as the BPSK detection. Once the signal b (t) is recovered, it is applied to one input of an EX-OR gate.
The signal b (t) is also applied to a time delay circuit and the delayed signal b (t - Tb) is applied to the other input of the EX-OR gate as shown in Fig. 8.10.1.
If b (t) = b (t - T.) then output of the EX-OR gate will be 0.
d(t) = 0 ... if b(t) =b(t-Tb).
And if b (t) b (t - Tb) then output of the EX-OR gate will be 1.

b(k): 0 1 1 0 1 1 0 0
b(k-1): 0 1 1 0 1 1 0 0
d(t) 1 0 1 1 0 1 0 ----DEPSK output

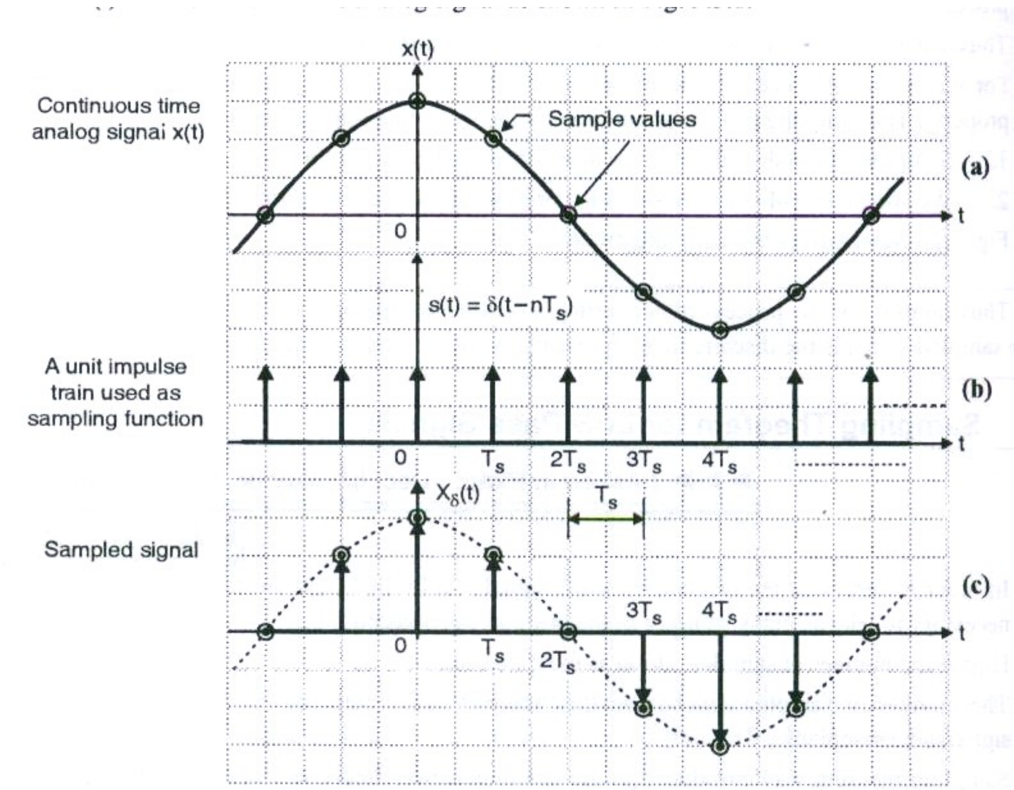
b(k): 0 1 1 1 1 1 0 0
b(k-1): 0 1 1 0 1 1 0 0
d(t) 1 0 0 0 0 1 0 ----DEPSK output

2 errors

Q.6(a)

Statement:

1. If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.
2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (I /2W) seconds apart



$$S(t) = \partial(t+2Ts)+ \partial(t+Ts)+ \partial(t)+ \partial(t-Ts)+ \partial(t-Ts)+\ldots$$

$$S(t)= \sum_{N=-\infty}^{\infty} \partial(t-nTs)$$

$$x\partial(t)= \sum_{N=-\infty}^{\infty} x(nTs) . \partial(t-nTs)$$

Fourier Transform:

$$X(f) = f_o \sum_{N=-\infty}^{\infty} \partial(f-nf0)$$

$$S(f) = f_s \sum_{N=-\infty}^{\infty} \partial(f-nf_s)$$

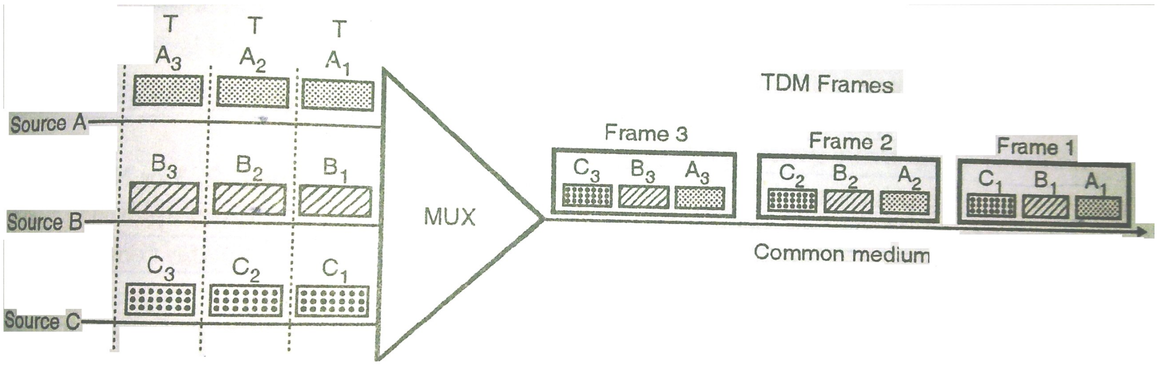
$$X\partial(t)=x(t)S(t)$$

$$X\partial(f)=x(f)s(f)$$

$$S(f) = x(f) \left[f_s \sum_{N=-\infty}^{\infty} \partial(f-nf_s)\right]$$

$$X\partial(f) = f_s \sum_{N=-\infty}^{\infty} \partial(f-nf_s)$$

Q.6 (b)



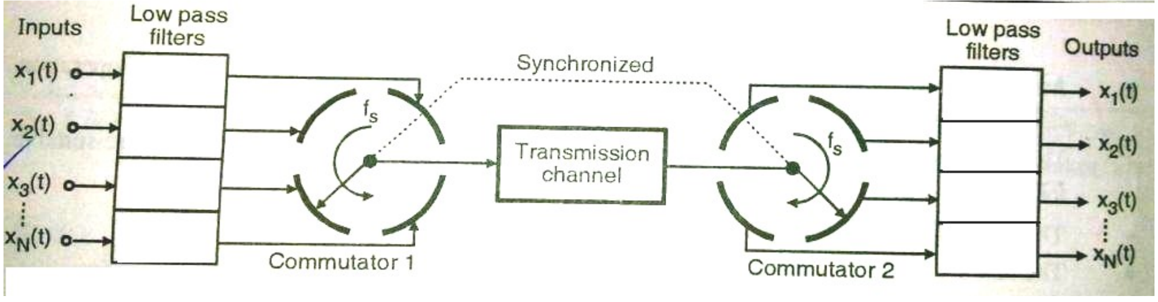
In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one.

.Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel. The TDM principle is illustrated in Fig

Fig shows the frames of TDM signal For 3 inputs being multiplexed, a frame of TDM will consist of 3 units i.e. one unit from each source.

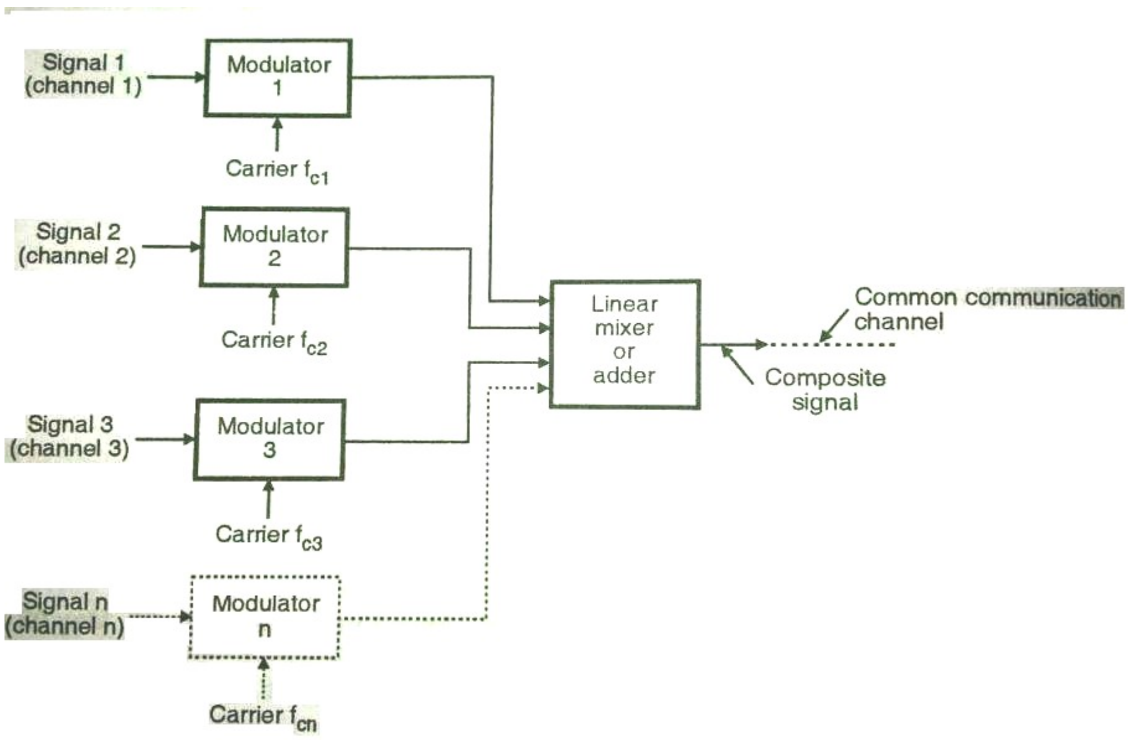
Similarly for n number of inputs, each TDM frame will consist of n units.

The TDM system can be used to multiplex analog or digital signals, however it is more suitable For the digital signals.



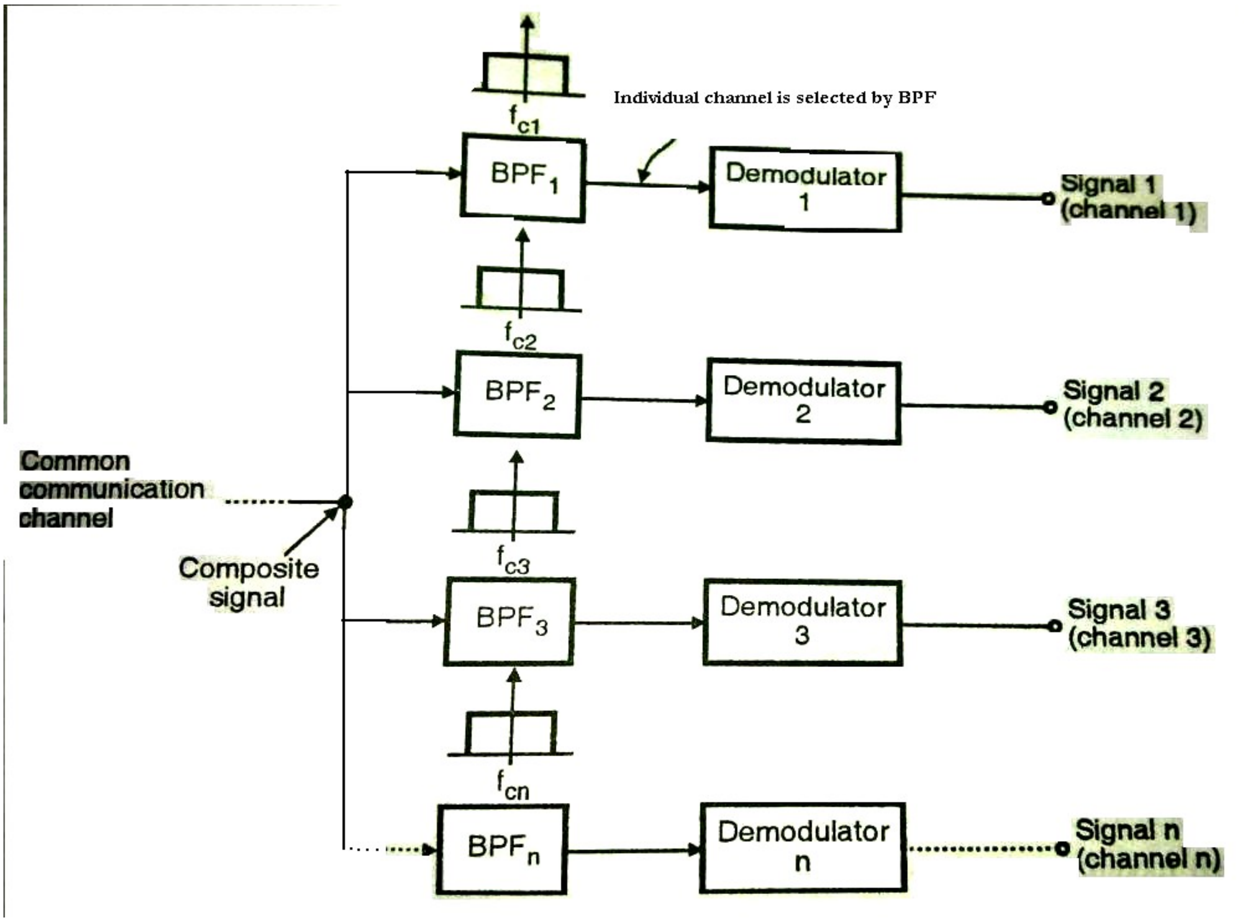
FDM transmitter:

- Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the ' corresponding signals.
- The modulator outputs are added together in a linear mixer or adder. The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place.
- Different signals are thus added together in the time domain but they have a separate identity in the frequency domain. This is as shown in the Fig.
- The composite signal at the output of mixer is transmitted over the single communication channel as shown in Fig.. This signal can be used to modulate a radio transmitter if the FDM signal is to be transmitted through air



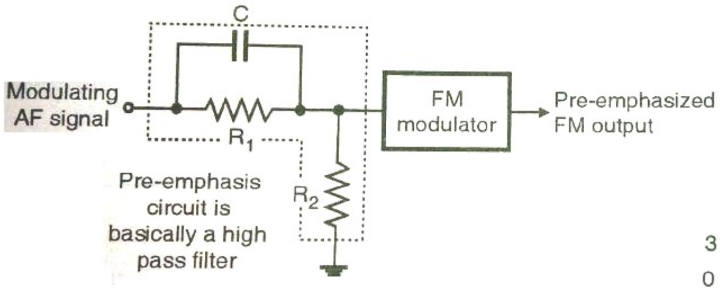
FDM Receiver:

- The block diagram of an FDM receiver is as shown in Fig. The composite signal is applied to a group of band pass filters (BPF).
- Each BPF has a center frequency corresponding to one of the carriers.
- The BPFs have an adequate bandwidth to pass all the channel information without any distortion

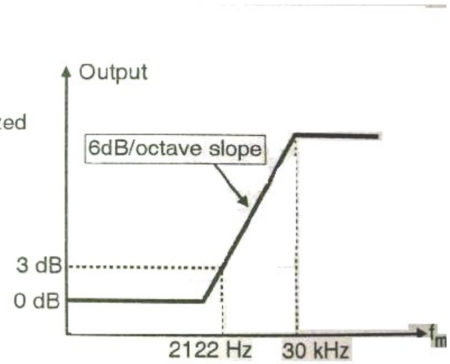


Q.7 (a)

Pre-Emphasis:



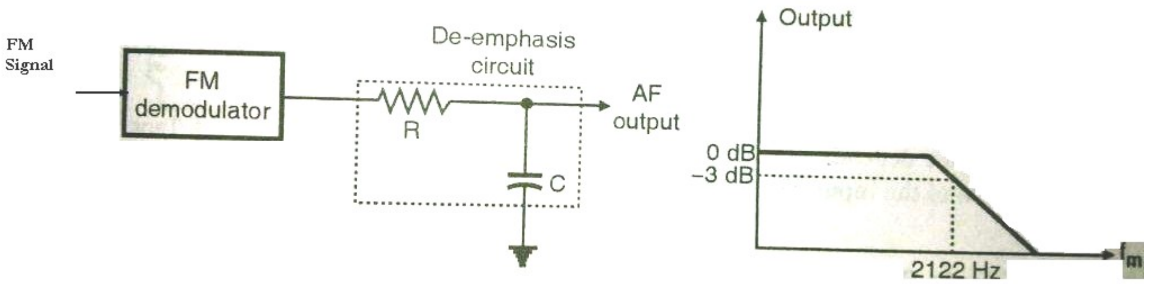
Pre Emphasis Circuit



Characteristics

It has been proved, that in FM. the noise has a greater effect on the higher modulating frequencies.
This effect can be reduced by increasing the value of modulation index (mf) forjiigher
modulating frequencies (fm).
This can be done by increasing the deviation "8" and 5 can be increased byjncreasing the amplitude of modulating signal at higher modulating frequencies.
Thus if we "boost" the amplitude of higher frequency modulating signals artificially then it will be possible to improve the noise immunity at higher modulating frequencies. "The artificial boosting of higher modulating frequencies is called as pre emphasis. Boosting of higher .frequency modulating signal is achieved by using the pre-emphasis circuit.
The modulating AF signal is passed through a high pass RC filter, before applying it to the FM modulator.
As fm increases, reactance of C decreases and modulating voltage applied to FM modulator goes on increasing.
The frequency response characteristic of the RC high pass network is shown in fig.
The amount of pre-emphasis in US FM transmission and sound transmission in TV has been standardized at 75 microsecond.

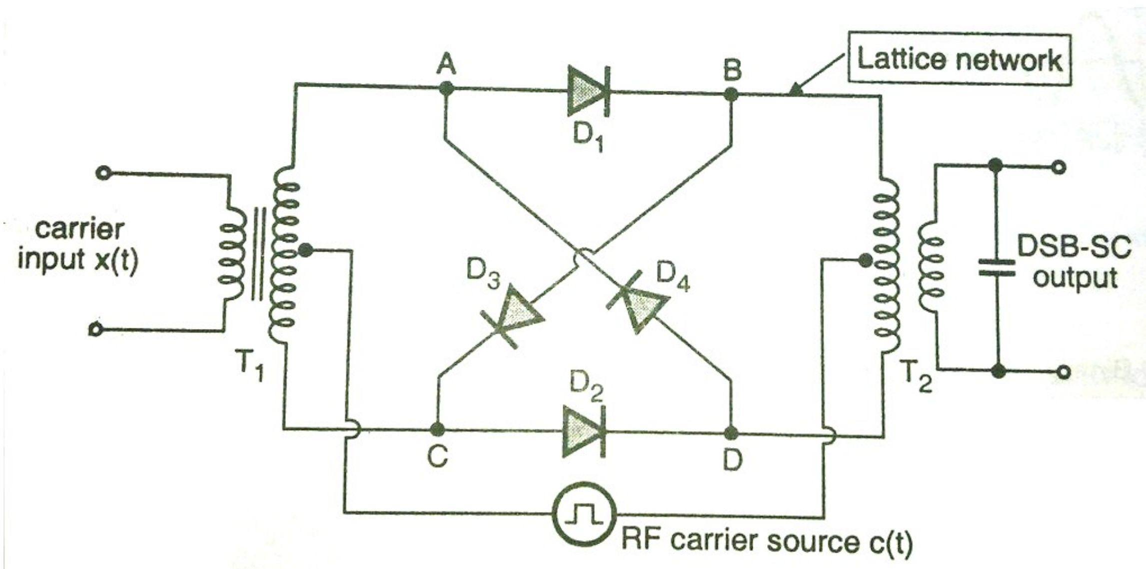
De-Emphasis:



The artificial boosting given to the high modulating frequencies in the process of pre-ernphasis is nullified or compensated at the receiver by process called "De-emphasis". The artificially boosted high frequency signals are brought to their original amplitude using the de-emphasis circuit.
The 75 microsecond de-emphasis circuit is standard and it is as shown in Fig. shows that it is a low Pass filter .75 microsecond de-emphasis corresponds to a frequency response curve that is 3 dB down at a frequency whose RC time constant is 7.5
The demodulated FM is applied to the De-emphasis circuit with increase in fm the reactance of C goes on decreasing and the output of de-emphasis circuit will also reduce

Q.7(b)

- In the BPSK generator we have a product modulator which is actually a balanced ring modulator with two inputs. One input is the carrier frequency while other input is the digital data.



Operation of the circuit:

- The operation is explained with the assumptions that the diodes act as perfect switches and that they are switched on and off by the digital data signal.
- The operation can be divided into different modes.

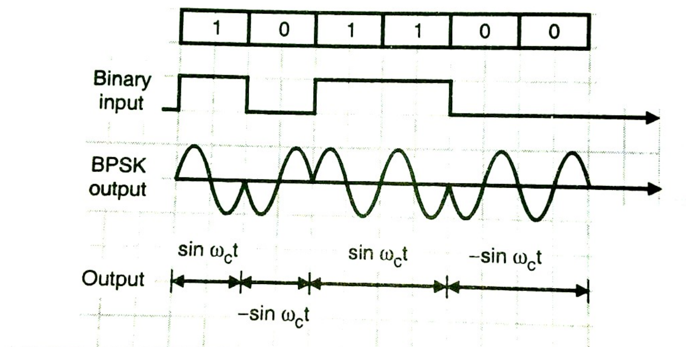
Operation with binary input = Logic 1 :

- If the binary input is equal to logic 1 then the equivalent circuit is as shown in Fig. 9.8.3(a). The diodes Dj and D2 are on (forward biased) while D3 and D4 are reverse biased and off.
- With the polarity shown, the carrier voltage is developed across the transformer T2 in phase with the carrier voltage across Tj.
- Hence the output signal is in phase with the carrier input signal.

Operation with binary input = logic 0:

- If the binary input is equal to 0 (actually -V) then D1 D2 are reverse biased and remain off whereas D3 and D4 are forward biased and hence conduct.

The output voltage across T2 is 180° out of phase with respect to the carrier input. Hence the output voltage is
- sin ω_c t.

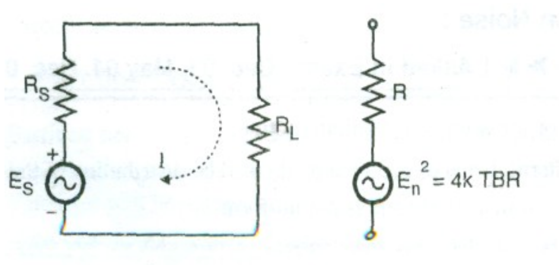


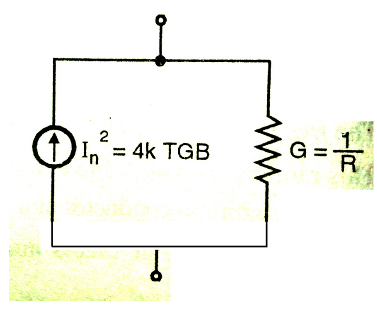
Phase	0°	180°	0°	180°
Conducting diodes	D_1D_2	D_3D_4	D_1D_2	D_3D_4

Q.7(c)

Thermal Noise:

The free electrons within a conductor are always in random motion.
This random motion is due to the thermal energy received by them. The distribution of these free
electrons within a conductor at a given instant of time is not uniform.
It is possible that an excess number of electrons may appear at one end or the other of the
conductor.
The average voltage resulting from this non-uniform distribution is zero but the average power is
not zero.
As this power results from the thermal energy, it is called as the "thermal noise power".





The average thermal noise power is given by,

$$Pn = kTB \text{ watts}$$

Where, k = Boltzmann's constant = 1.38 x 10" Joules/Kelvin. B = Bandwidth of the noise spectrum (Hz). T = Temperature of the conductor, "Kelvin

- Equation indicates that a conductor operated at a finite temperature can work as generator of electrical energy.
- The thermal noise power Pn is proportional to the noise BW and conductor temperature.

Q.7 (d)

Companding:

Companding is non-uniform quantization. It is required to be implemented to improve the signal
to quantization noise ratio of weak signals.
The quantization noise is given by,

$$N_q = s^2/12$$

This shows that in the uniform quantization once the step size is fixed, the quantization noise power remains constant.
But the signal power is not constant. It is proportional to the square of signal amplitude. Hence signal power will be small for weak signals, but quantization noise power is constant. Therefore the signal to quantization noise ratio for the weak signals is very poor. This will affect the quality of signal. The remedy is to use companding.
Companding is a term derived from two words, compression and expansion.

Companding = Compressing + Expanding

Practically it is difficult to implement the non-uniform quantization because it is not known in advance about the changes in the signal level.
Therefore a trick is used. The weak signals are amplified and strong signals are attenuated before applying them to a uniform quantizer.
This process is called as "compression" and the block that provides it is called as a "compressor".
At the receiver exactly opposite process is followed which is called expansion. The circuit used
for providing expansion is called as an "expander".
The compression of signal at the transmitter and expansion at the receiver is combined to be as "companding".

Types of Companding :
There are two possible types of companding :

1. Analog companding 2. Digital Companding