رابطة الطالبات

نظم الاتصالات (٢) (١)

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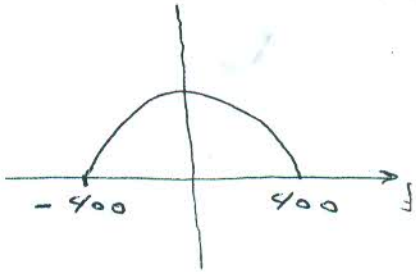
[1] Question one (20 mark)

A- (7 marks)

$$M(f) = \begin{cases} \cos\left(\frac{\pi f}{800}\right) & -400 \le f \le 400 \\ 0 & o\text{-}w \end{cases} \equiv$$

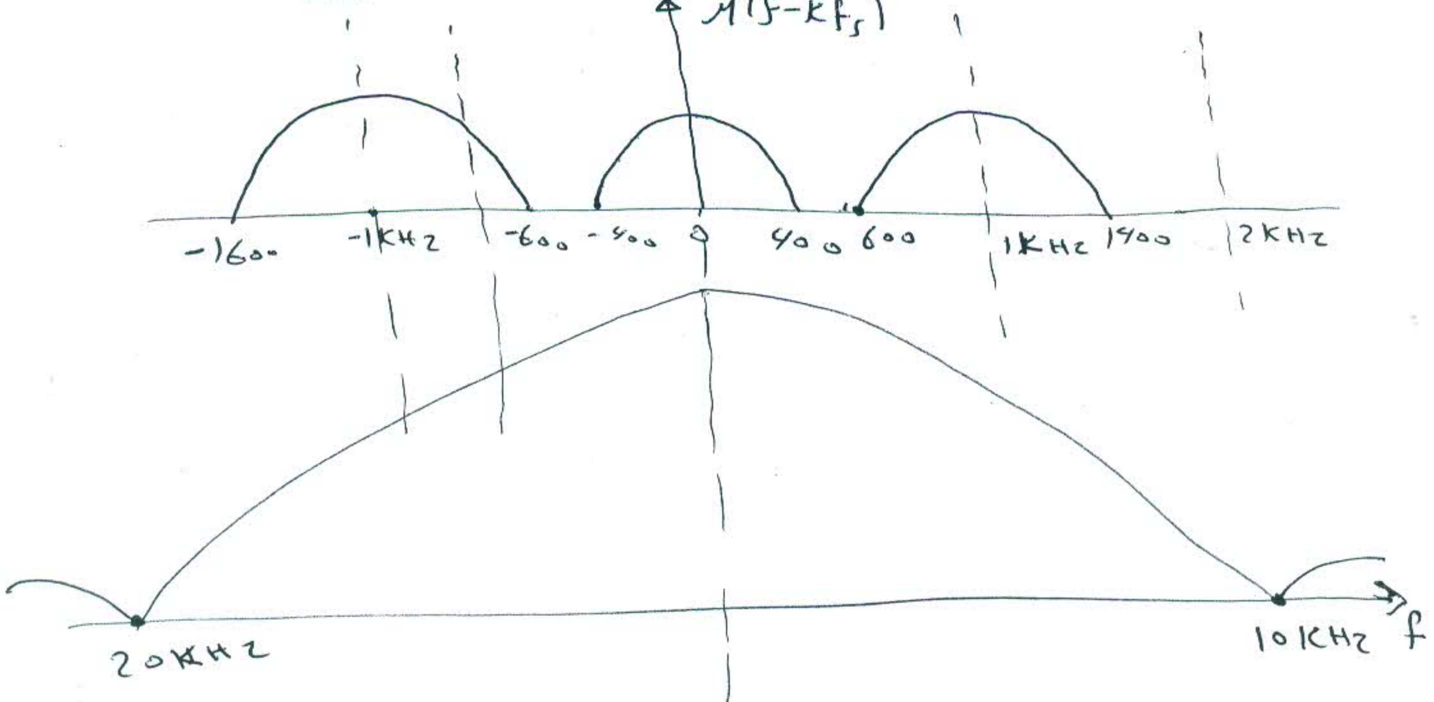


$f_s = 1\ \text{KHz}$, $T_{pulse} = 0.1\ \text{ms}$

$$M(f) = \cos\left(\frac{\pi f}{800}\right) \cdot \text{rect}\ \frac{f}{800}$$

The spectrum of PAM $\quad S(f) = f_s \sum_{k=-\infty}^{\infty} M(f - k f_s) \cdot H(f)$

where $H(f)$ is the transfere function of the pulse



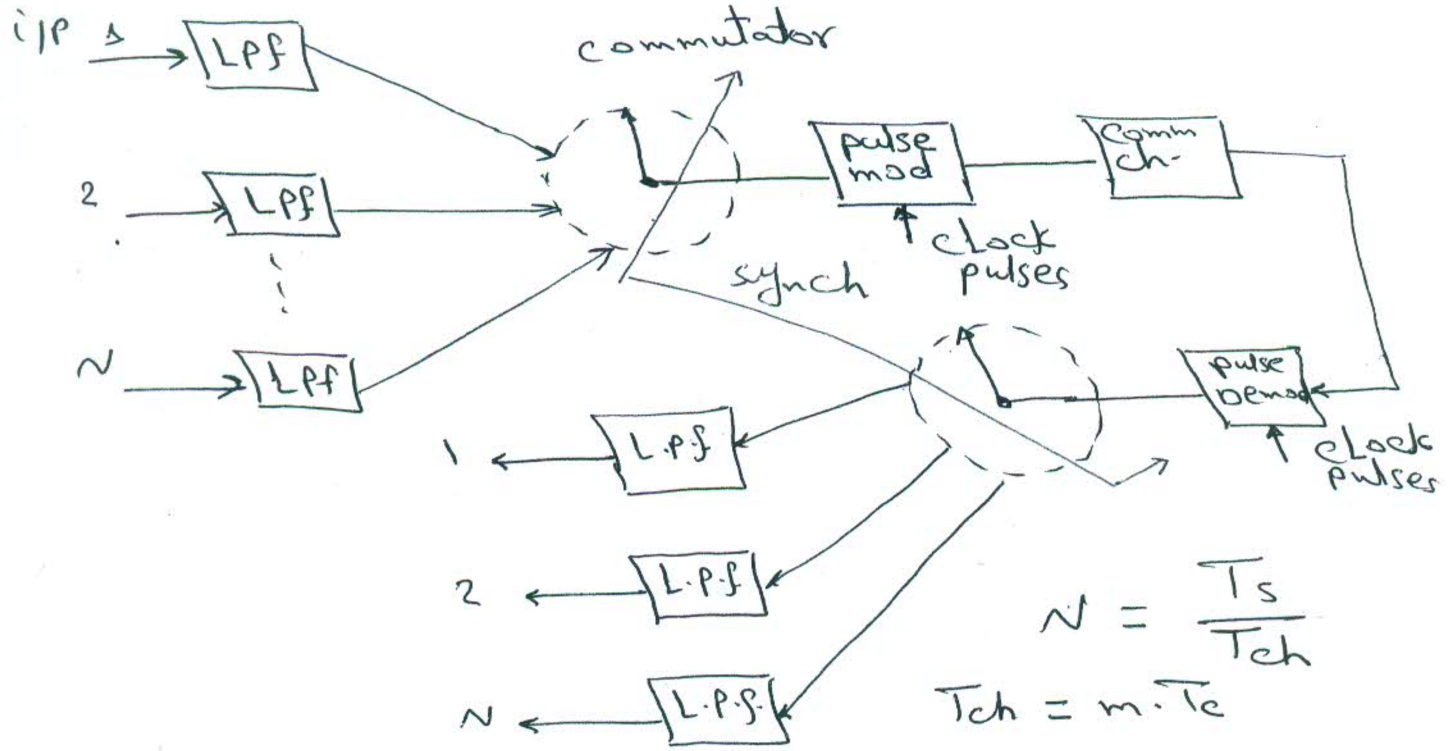
$H(f) = \text{Sinc}(f\tau)$

$\tau = 0.1\ \text{msec} \Rightarrow$ nulls at $\frac{n}{\tau}$ (10 KHz, 20 KHz, - - -

B- (8 marks)

i → The principles of TDM.

- enables the joint utilization of a common ch. by a plurality of independent message sources without mutual interference



$$N = \frac{T_s}{T_{ch}}$$

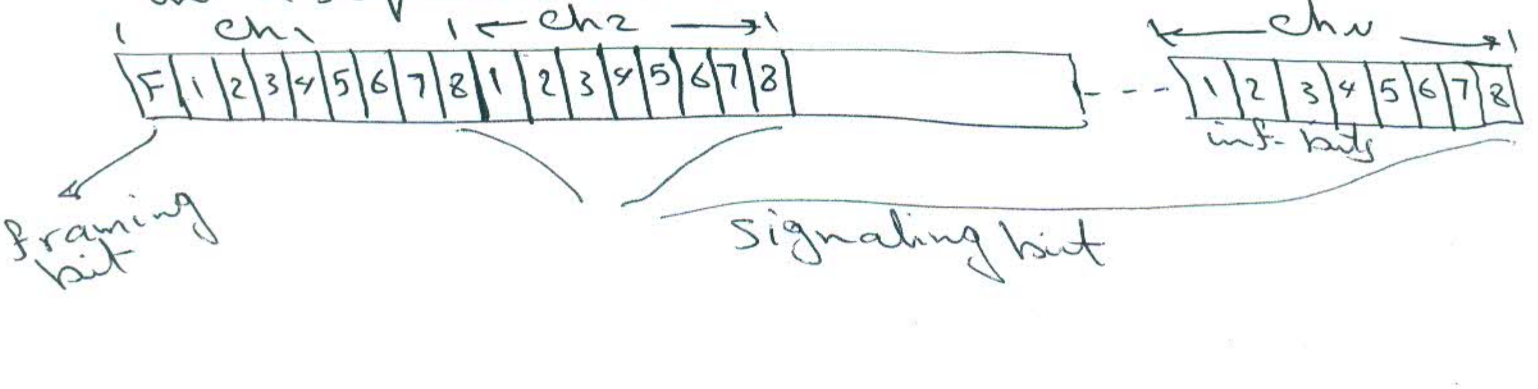
$$T_{ch} = m \cdot T_c$$

→ pulse mod. transform the multiplexed signal into a form suitable for Txion over the comm. ch.

→ TDM introduces a B.W expansion factor N

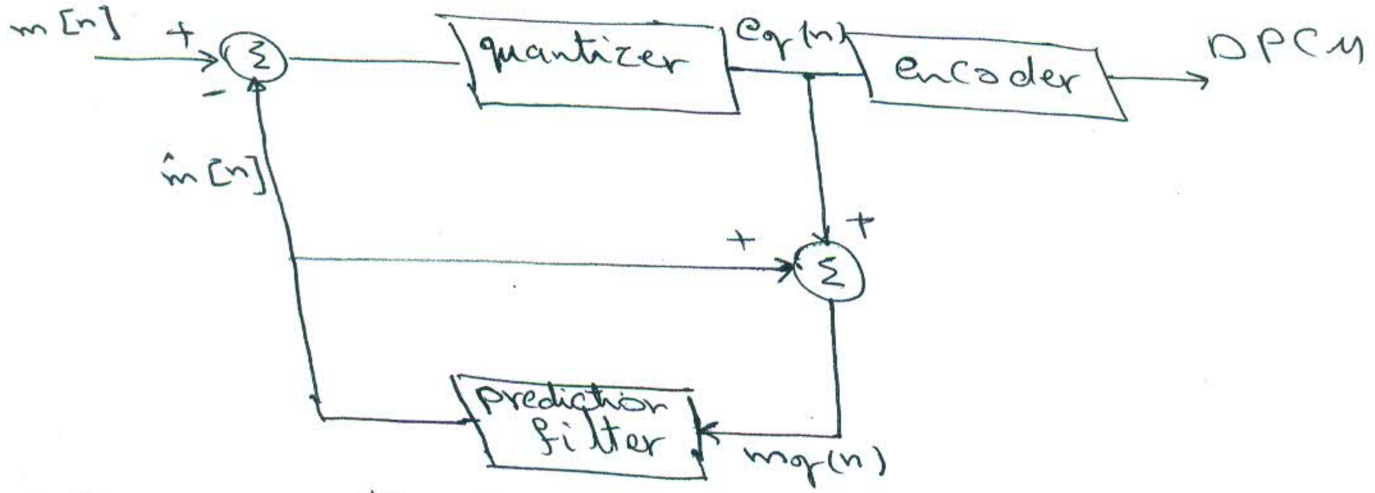
→ In application using PCM, each sample represented by code word

→ Binary code words of each ch. are multiplexed in a sequence as shown



c- show that the performance of DPCM can be controlled by the design of prediction filter.

for DPCM



$m[n]$: unquantized i/p

$e[n] = m[n] - \hat{m}[n]$ → prediction of $m[n]$

prediction error

$$e_q[n] = e[n] + q[n]$$

⇒ using DPCM results in processing gain

$$(SNR)_o = \frac{\sigma_M^2}{\sigma_Q^2}$$

σ_M^2 : variance of the original i/p

σ_Q^2 : variance of $q[n]$

$$(SNR)_o = \left(\frac{\sigma_M^2}{\sigma_E^2}\right) \cdot \left[\frac{\sigma_E^2}{\sigma_Q^2}\right] = G_p \cdot (SNR)_Q$$

σ_E^2 : variance of the prediction error

$(SNR)_Q$: signal to quantization noise ratio

$$G_p = \frac{\sigma_M^2}{\sigma_E^2} > 1$$

for higher G_p = design the prediction filter to minimize σ_E^2

ii –

$N = 24$ TDM

$T_{pulse} = 1 \mu s$ and extra pulse for synch

$f_m = 3.4\ KHz$

Let $f_s = 8\ KHz > 2 f_m$

$\therefore T_s = \frac{1}{f_s} = 125\ \mu sec$

$\therefore$ spacing

each channel assigned time T_{ch}

$$T_{ch} = \frac{T_s}{N + 1\ synch} = \frac{125}{25} = 5\ \mu sec$$

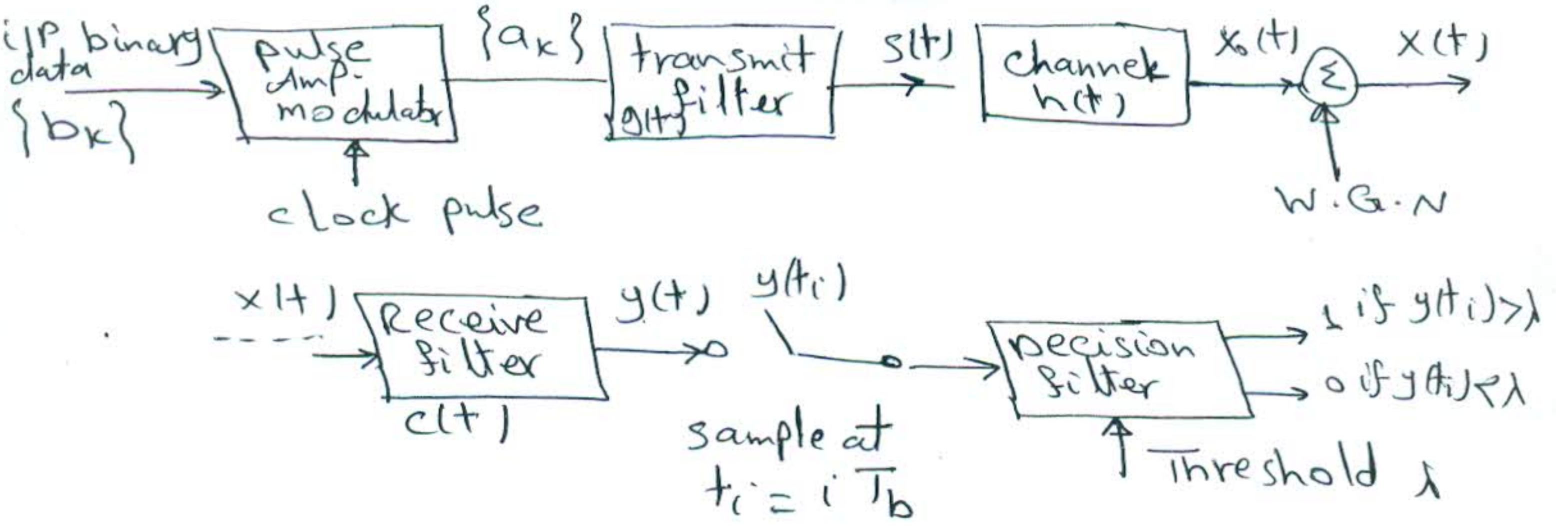
Since the pulse duration $= 1\ \mu sec$

$$\therefore \boxed{S = 4\ \mu sec}$$

[2] Question 2 : (20 Mark) (4)

A_ (7 marks) ISI, arises when the comm ch. is dispersive.

- consider a base band binary PAM system



$$a_k = \begin{cases} +1 & \text{if } b_k \text{ is } 1 \\ -1 & \text{if } b_k \text{ is } 0 \end{cases}$$

$$s(t) = \sum_k a_k \, g(t - kT_b)$$

→ The receiver filter o/p

$$y(t) = \mu \sum_k a_k \, p(t - kT_b) + n(t)$$

μ : scaling factor accounts for amplitude changes.

$$\mu p(t) = g(t) * h(t) * c(t) \quad , p(0) = 1$$

$$\therefore \mu P(f) = G(f) \cdot H(f) \cdot C(f)$$

n(t) : noise produced at the o/p of the receiver filter due to channel noise w(t)

→ the o/p y(t) is sampled at $t_i = i T_b$

$$y(t_i) = \mu \sum_{k=-\infty}^{\infty} a_k \, p[(i-k)T_b] + n(t_i)$$

$$= \mu a_i + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k \, p[(i-k)T_b] + n(t_i)$$

→ μa_i represents the contribution of the ith Txed bit

- 2nd term represents the residual effect of all other txed bits $\equiv$ ISI

Then to eliminate the effect of ISI

$$p(iT_b - kT_b) = \begin{cases} 1 & i = k \\ 0 & i \neq k \end{cases}$$

B-(6)

NRZ , $P_e = 10^{-6}$

- signaling rate is doubled $\Rightarrow T_b' = \frac{1}{2} T_b$

$$P_e = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{E_b}{N_o}}\right) = \frac{1}{2} \operatorname{erfc}(u)$$

$\therefore u = 3.3$

when the signaling rate is doubled

$$\therefore E_b' = A^2 T_b' = A^2 \cdot \frac{T_b}{2}$$

$$\therefore P_e' = \frac{1}{2} \operatorname{erfc}\left(\frac{u}{\sqrt{2}}\right) = \frac{1}{2} \operatorname{erfc}(2.33)$$

$$= 10^{-3}$$

C-(7 marks)

an equalizer

$$C(\omega) = \left[a_1 e^{-j\omega T} + a_2 e^{-j2\omega T}\right] M(\omega)$$

$\therefore$ The transfere function of the channel

$$H_{ch}(\omega) = a_1 e^{-j\omega T} + a_2 e^{-j2\omega T}$$

$\Rightarrow$ The equalizer should be designed so that

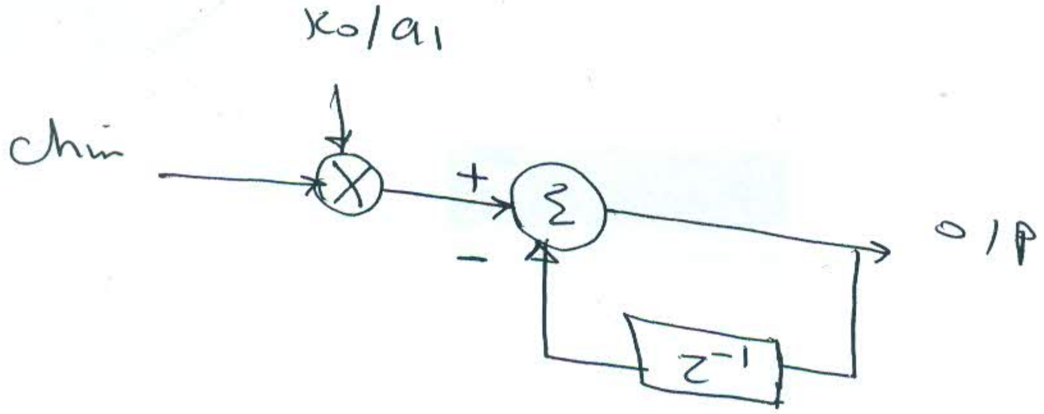
$$H_{ch}(\omega) \cdot H_e(\omega) = k_o \exp(-j\omega T)$$

where T is the transmission delay

$$\therefore H_e(\omega) = \frac{k_o \exp(-j\omega T)}{a_1 e^{-j\omega T} + a_2 e^{-j2\omega T}}$$

$\frac{o/p}{i/p}$

$$= \frac{(k_o/a_1)}{1 + \frac{a_2}{a_1} e^{-j\omega T}}$$



xo/a1
chin
+
Σ
−
z^-1
o/p

[Q3] [6]

A- 6 marks

⇒ Metrics for choice of digital modulation scheme

1- High spectral efficiency $\eta_b = \frac{R_b}{B_T}$

2- " power efficiency $\eta_p = \frac{E_b}{N_o}$

3- " data rates : Bits per second

4- Roubust to multipath effects, fading condition

5- Easy to implement and cost effective

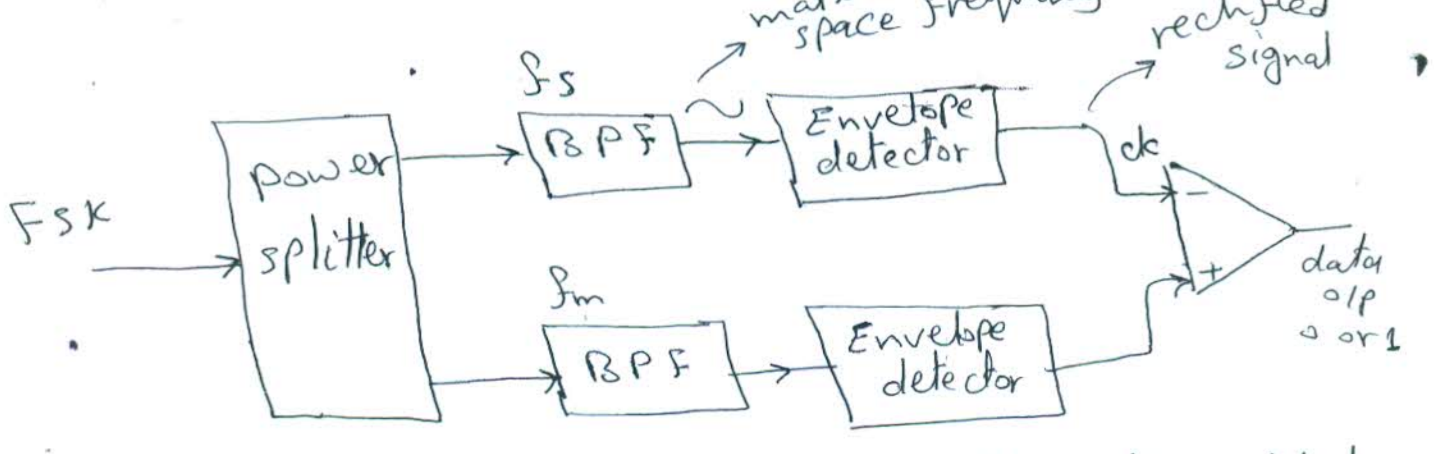
6- Low carrier-to-cochannel signal interference ratio

7- Low out of band radiation

8- Constant or near constant envelope

B- 6 marks

⇒ The operation of the demodulator for non coherent binary FSK.



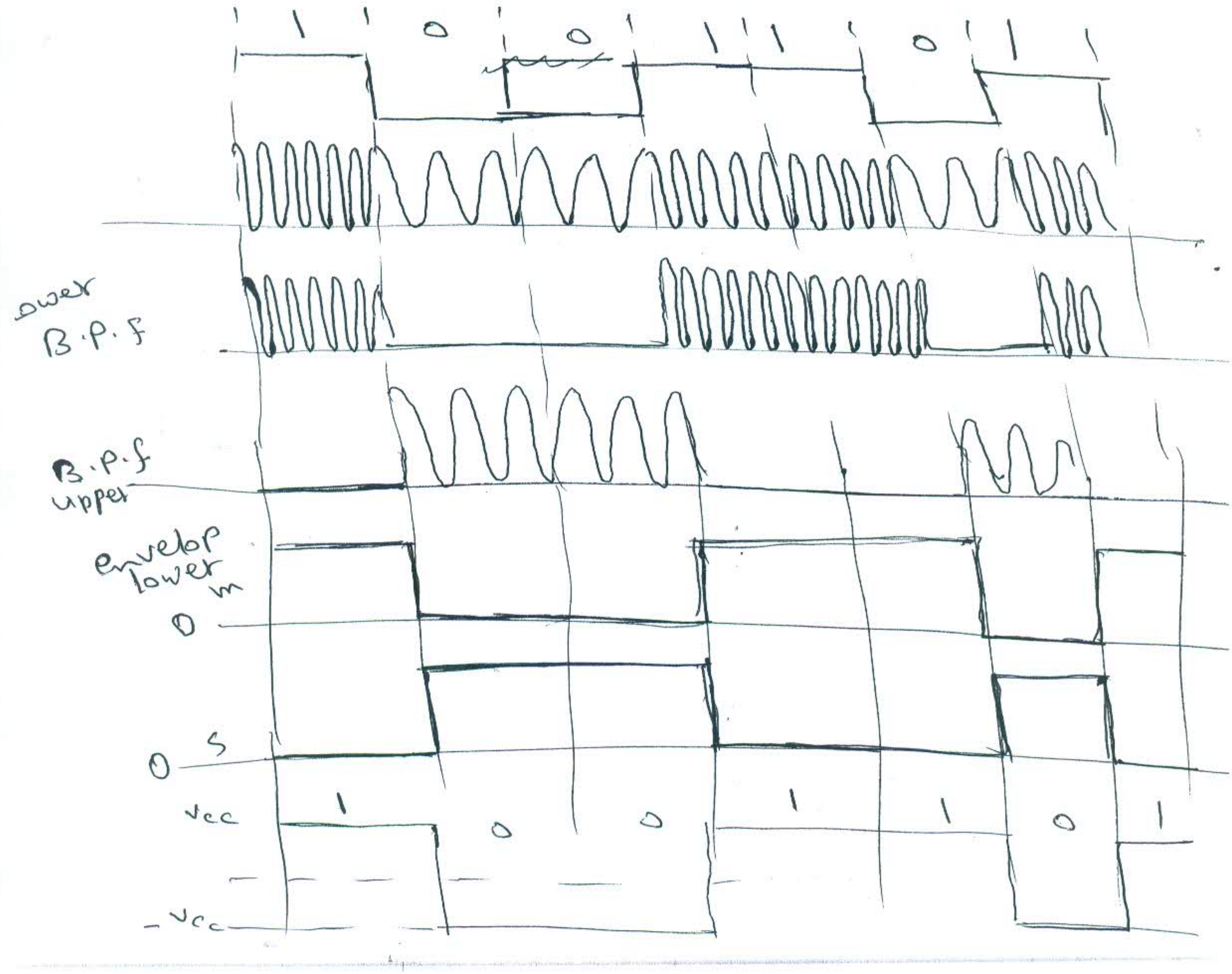
→ No frequency is involved in the demodulator process that is synchronized either in phase, frequency or both with the incomming FSK

→ No carrier recovery circuits

→ The operation the incomming signal is splitted by power splitter

→ The upper B.P.F o/p is related to space freq (0 bits) while the Lower is related to mark freq (1 bit)

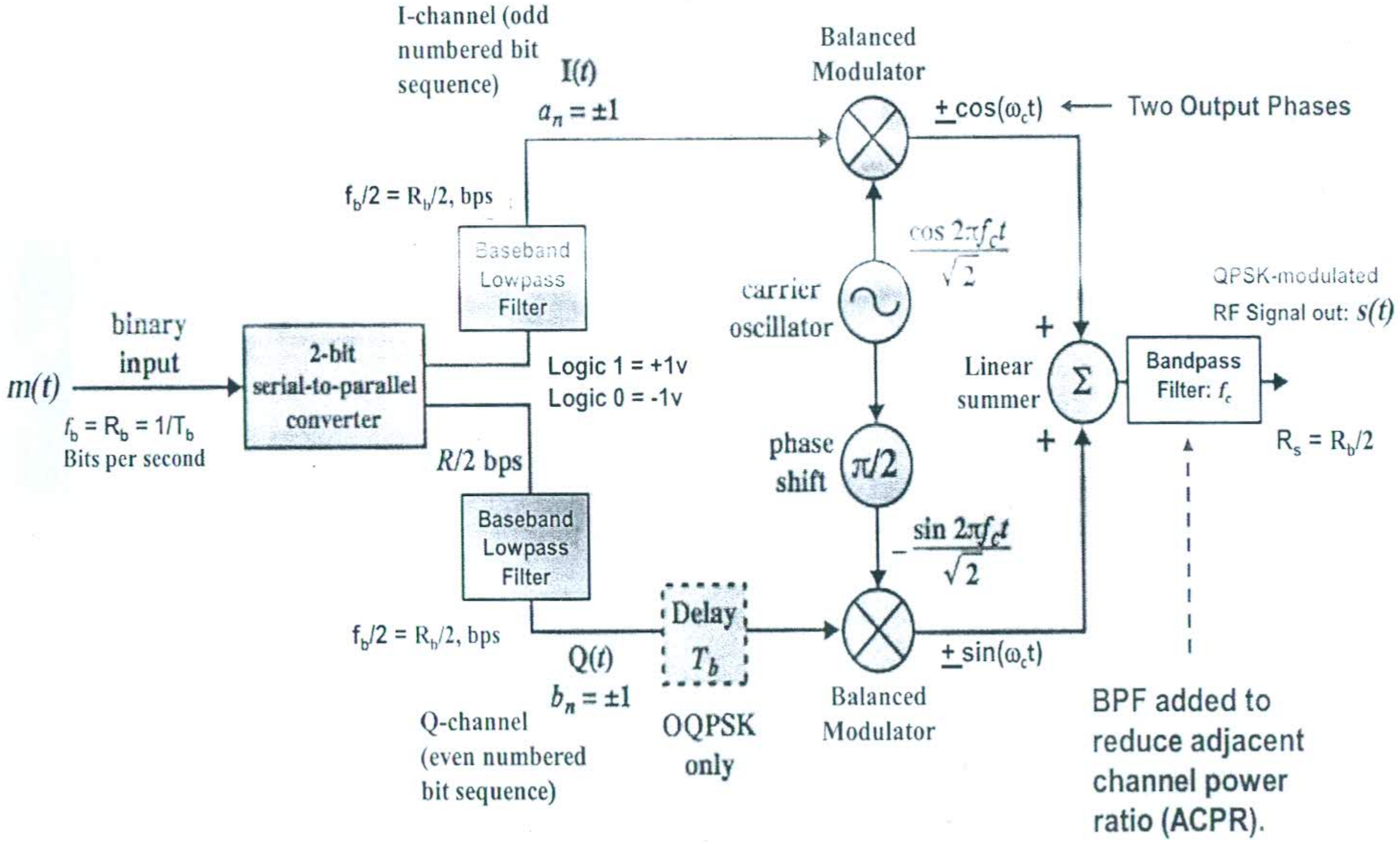
→ The envelop produces dc level corresponds to either mark or space



18)

Transmit Modulator

c) For: Quadrature Phase Shift Keying & Offset-QPSK modulation



ic -

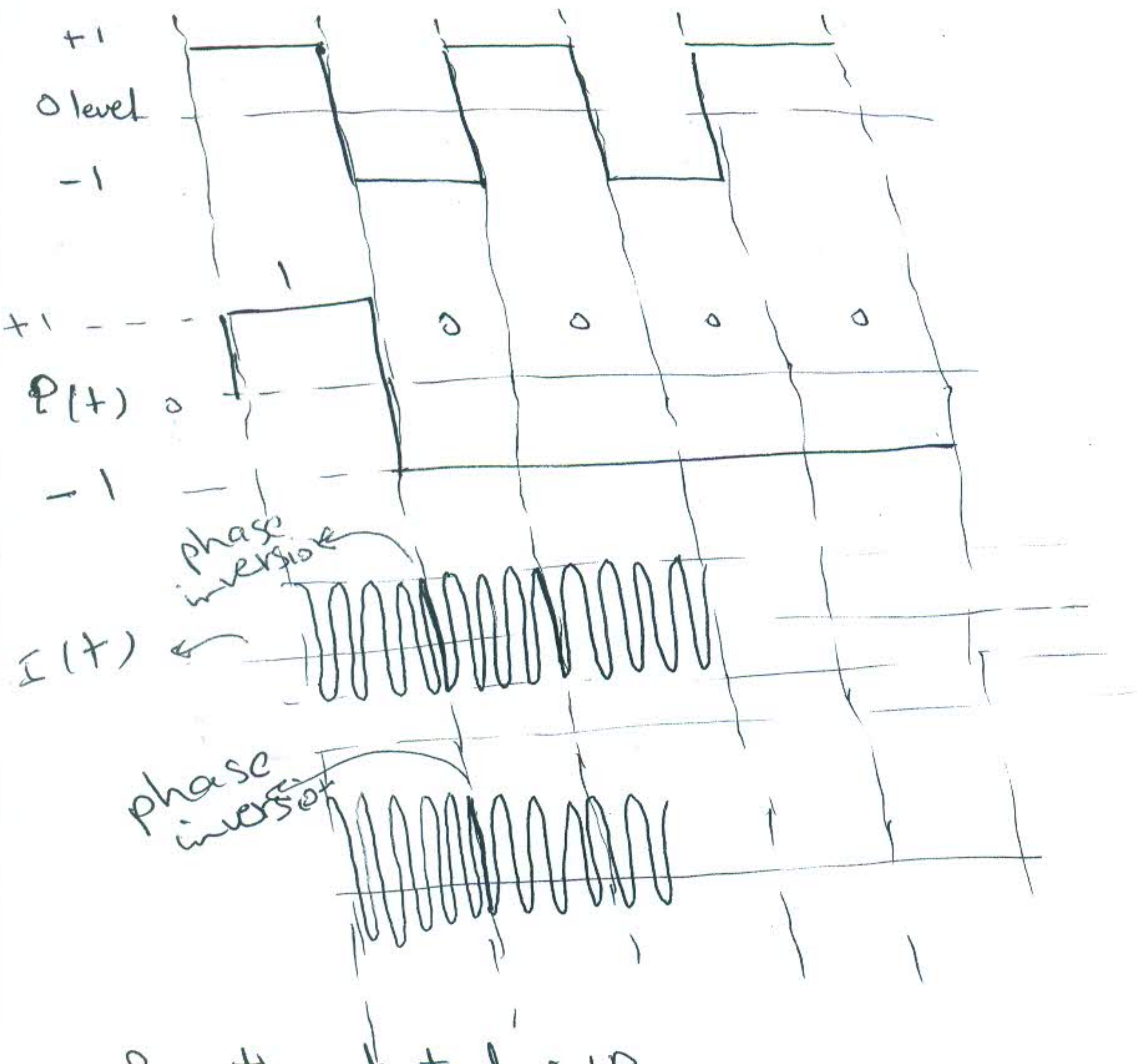
1 1 0 0 1 0 0 0 1 0 1

$I(t) \rightarrow 1\ 0\ 1\ 0\ 1$

$Q(t) \rightarrow 1\ 0\ 0\ 0\ 0$

I(t) wave

1 0 1 0 1



for the total o/p

the first pair $(I, Q) \equiv (1, 1)$

$o/p = \frac{\cos w_c t}{\sqrt{2}} + \frac{\sin w_c t}{\sqrt{2}}$ angle $\underline{45}$

Q3 (c)

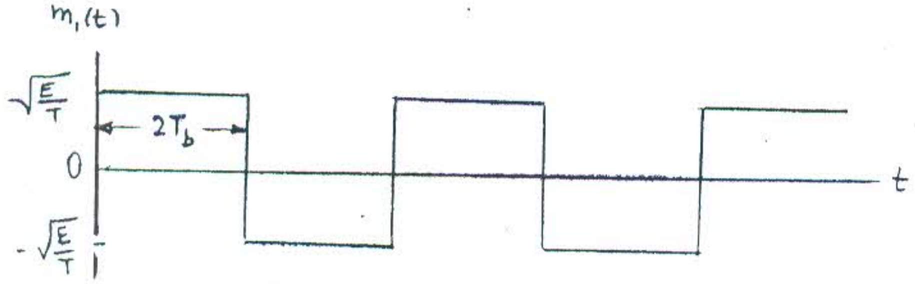
[redacted]

(i) The QPSK wave can be expressed as

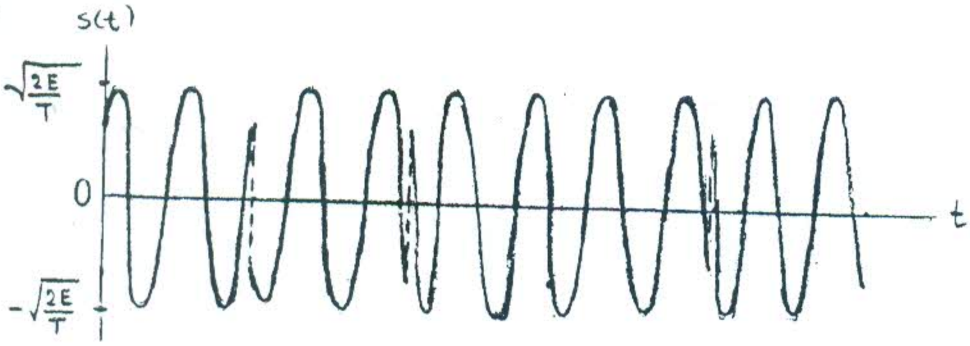
$$s(t) = m_1(t)\cos(2\pi f_c t) + m_2(t)\sin(2\pi f_c t).$$

Dividing the binary wave into dibits and finding $m_1(t)$ and $m_2(t)$ for each dibit:

dibit	11	00	10	00	10
$m_1(t)$	$\sqrt{E/T}$	$-\sqrt{E/T}$	$\sqrt{E/T}$	$-\sqrt{E/T}$	$\sqrt{E/T}$
$m_2(t)$	$\sqrt{E/T}$	$-\sqrt{E/T}$	$-\sqrt{E/T}$	$-\sqrt{E/T}$	$-\sqrt{E/T}$



(b)



[Q 4] 20 Marks

a- State the main differences between error correction block codes and convolutional codes
b- Draw both the encoder shift registers and the encoder state diagram of the convolutional encoder with (n,k,K)=(2,1,3)
c- The previous encoder in part b is used to encode the following sequence 110010. What is the output coded sequence?
d- If an error occurred in the received sequence of part c, explain how the decoder can detect and correct this error.

Solution

A) 6 marks

for block codes.

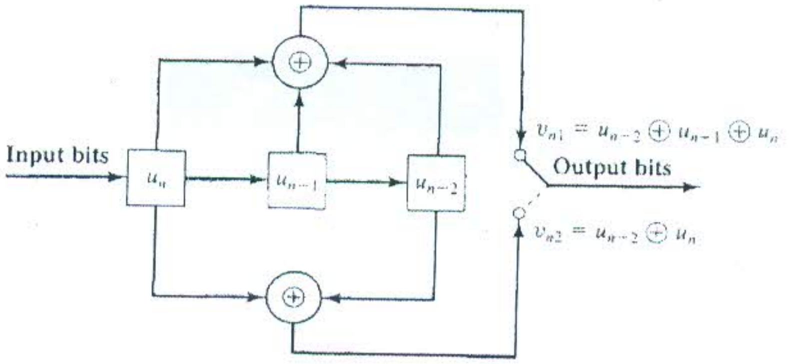
- On the transmission end, each k-bit block of data is mapped into an n-bit block $(n > k)$ called a code word. The first K-bits represents data and the remaining (n-k) represents check bits.
- block code process data in blocks of k bits at a time
- useful when data are transmitted and received in discrete manner.

for convolutional code.

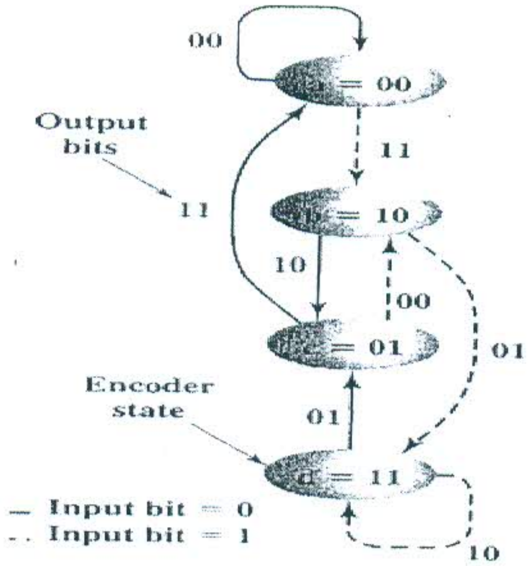
- useful when data are transmitted and received in a more or less continuous stream
- A convolutional code is defined by three parameters: n, k, and K.
- An (n, k, K) code processes input data k bits at a time and produces an output of n bits for each incoming k bits.
- n and k are generally quite small numbers.
- convolutional codes have memory, which is characterized by the *constraint factor* K. In essence, the current *n-bit* output of an (n, k, K) code depends not only on the value of the current block of k input bits but also on the previous K - 1 blocks of k input bits.

B) 7 marks

Encoder shift register



State diagram



C) 7 marks

to encode the following sequence 110010, we use the state diagram

Input	Previous state	Next state	output
0	00	00	00
1	00	10	11
0	10	01	10
0	01	00	11
1	00	10	11
1	10	11	01

Then the output sequence is

011111101100

from right to left

if we started from left

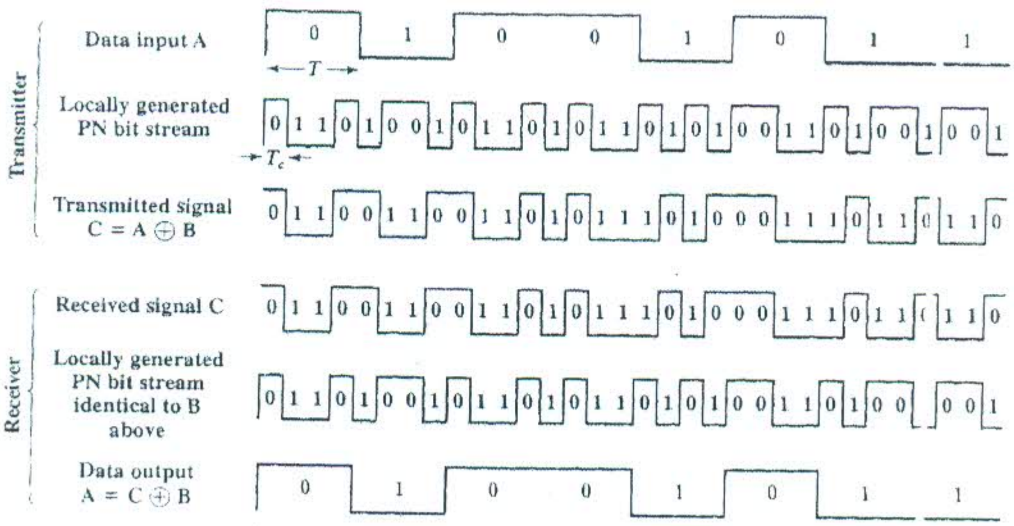
11 0 1 0 1 11 11 10

[Q:5] 10 Marks

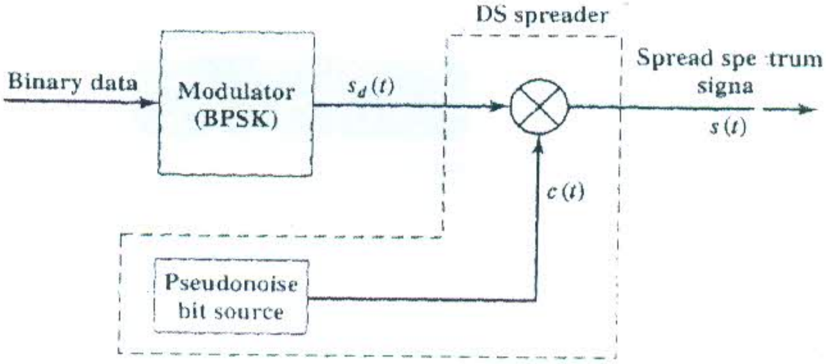
For direct sequence spread spectrum (DSSS), each bit in the original signal is represented by multiple bits in the transmitted signal, using a spreading code. The spreading code spreads the signal across a wider frequency band in direct proportion to the number of bits used. Therefore, a 10-bit spreading code spreads the signal across a frequency band that is 10 times greater than a 1-bit spreading code.

One technique for direct sequence spread spectrum is to combine the digital information stream with the spreading code bit stream using an exclusive-OR (XOR). The XOR obeys the following rules:

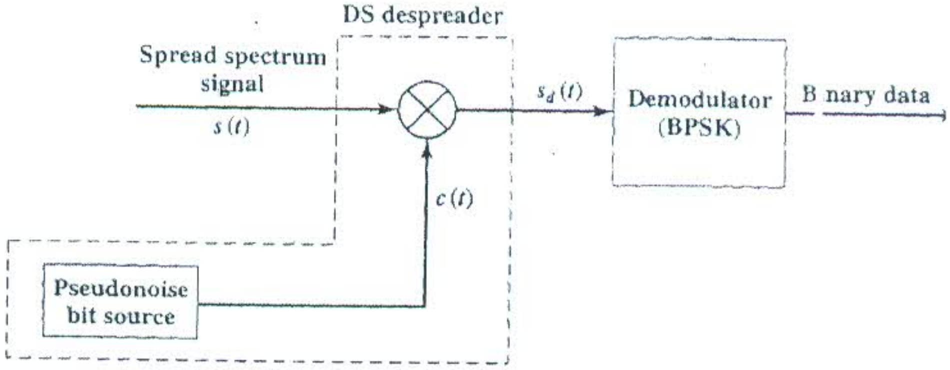
$$0 \oplus 0 = 0 \qquad 0 \oplus 1 = 1 \qquad 1 \oplus 0 = 1 \qquad 1 \oplus \quad = 0$$



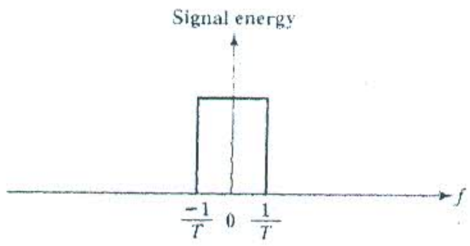
(14)



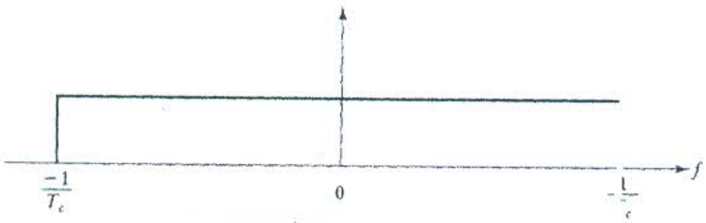
(a) Transmitter



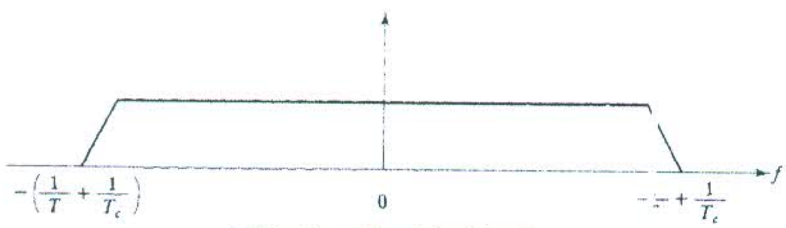
(b) Receiver



(a) Spectrum of data signal



(b) Spectrum of pseudonoise signal



(c) Spectrum of combined signal