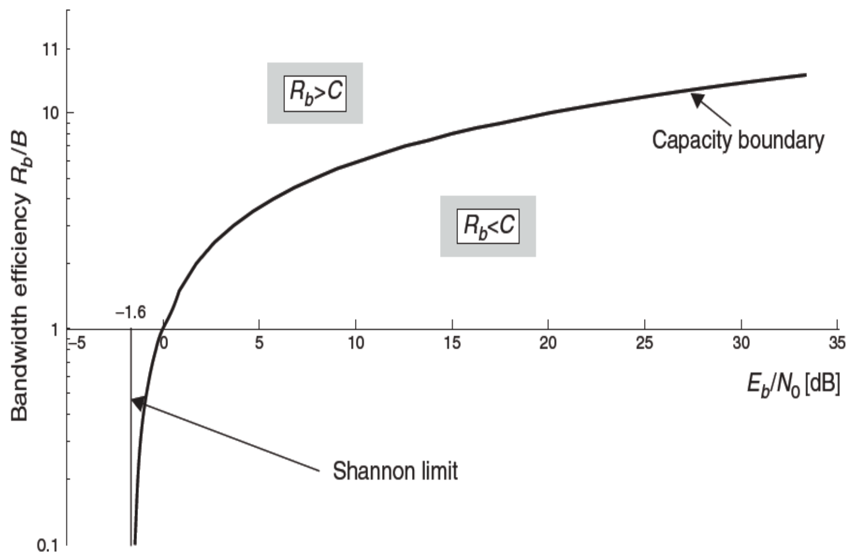
Q.2 a. Draw the bandwidth efficiency curve w.r.t E_b/N_o. Compute the value of E_b/N_o required to achieve the data rate equal to the channel capacity if the channel bandwidth tends to infinity

Answer:

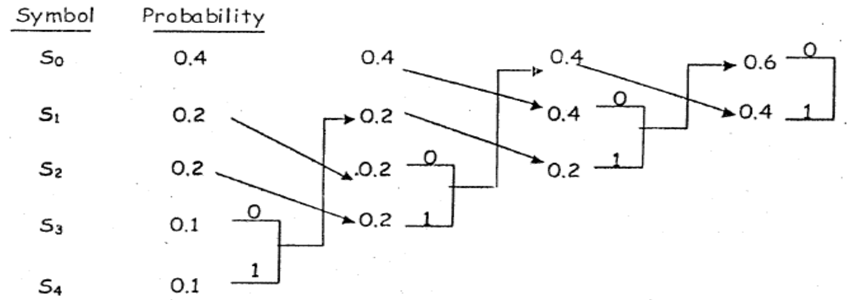


$$\lim_{B\to\infty}\frac{E_b}{N_0} = \lim_{B\to\infty}\frac{2^{\frac{C}{B}}-1}{\frac{C}{B}} = \lim_{B\to\infty}\frac{\ln 2\cdot 2^{\frac{C}{B}}\cdot\left(-\frac{C}{B^2}\right)}{\left(-\frac{C}{B^2}\right)} = \ln 2$$

b. A discrete memory less source has an alphabet of five symbols with probabilities 0.4, 0.2, 0.2, 0.1, 0.1 respectively. Compute the minimum variance and maximum variance Huffman code for this source and the average code word length of each code.

Answer:

i. **Minimum Variance Code**

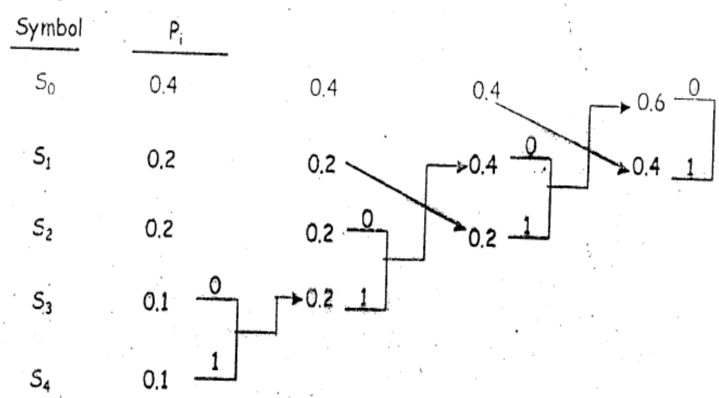


S_i	P_i	Code	n_i
S_0	0.4	00	2
S_1	0.2	10	2
S_2	0.2	11	2
S_3	0.1	010	3
S_4	0.1	011	3

Variance, $\sigma^2 = \sum_{i=0}^{4} (n_i - 2.2)^2 P_i = 0.16$

Average code word length, $\bar{L} = 2.2$. bits / symbol

ii. Maximum variance code



S_i	P_i	Code	n_i
S_0	0.4	1	1
S_1	0.2	01	2
S_2	0.2	000	3
S_3	0.1	0010	4
S_4	0.1	0011	4

Average code word length, $\bar{L} = 2.2$.

variance, $\sigma^2 = \sum_{i=0}^{4} (n_i - 2.2)^2 P_i = 1.36$

Q.3 a. State and explain the sampling theorem for the band – pass signal. Consider a signal g(t) having the upper cut-off frequency f_u = 120 kHz and lower cut-off frequency f_l = 70 kHz.

Answer:

The sampling theorem for bandpass signal says that in order to reconstruct the exact replica of transmitted signal, the sampling frequency must be either equal to or greater than 2(fu/m) , where m=[fu/(fu-fl)]

Fs=2(120K/m) where m=[120K/50K]=[2.4]=2

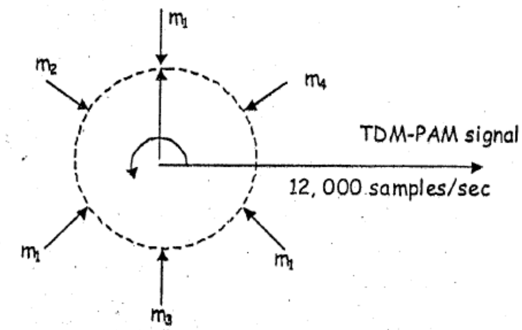
Thus, fs=2*120K/2=120KHz.

b. Explain TDM in brief with the help of block diagram. A signal $m_1(t)$ is bandlimited to 3 kHz and three other signals $m_2(t)$, $m_3(t)$ and $m_4(t)$ are bandlimited to 1 kHz each. (8)
(i) Set up a system for multiplexing the above signals with each signal sampled of its Nyquist rate.
(ii) What is the output rate of commutator?
(iii) What is the speed of commutator in rotations per second?
(iv) Sketch one frame of TDM-PAM signal
(v) If the output of the commutator converted to binary signal with M=256 what is the output bit rate
(vi) Determine the minimum channel bandwidth required for TDM-PCM signal

Answer:

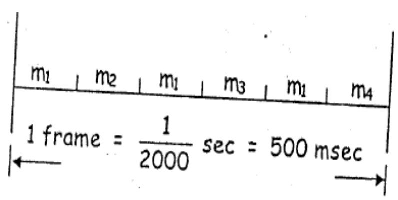
Signals	f_m	Nyquist rate
$M_1(t)$	3000 Hz	6000 samples/ second
M_2 (t)	1000 Hz	2000 samples/second
M_3 (t)	1000 Hz	2000 samples / second
M_4 (t)	1000 Hz	2000 Samples/second

a.



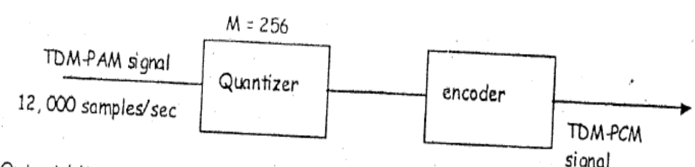
b. Output rate of commutator = 12,000 samples/sec

c. Speed = 2000 rotations/sec



One frame consists of 6 channels, 3 channels for m_1 (t) and one channel for m_2 (t), m_3 (t) and m_4 (t) each.

e.



Output bit rate = 12,000 samples / sec x $\log_2$ 256 bits / samples = 12,000 x 8 = 96 kbps

f. minimum BW = $\frac{\text{Bit rate}}{2}$ Hz = $\frac{96000}{2}$ = 48 KHz

Q.4 a. Obtain the expression of SNR of PCM system for a sinusoidal input signal. Hence, determine the bit rate and SNR of PCM of one telephone voice channel.

Answer:

a)

<u>Maximum signal to quantization noise ratio in PCM system</u>

Let the signal is m(t) and is ranging from -v to +v voltage with a uniform probability density function.

Signal is from -v to +v ∴ mean = 0

PDF, $f(x) = \frac{1}{V-(-V)} = \frac{1}{2V}$

mean square input signal power, $S_i = \int_{-V}^{+V} f(x)\, x^2\, dx = \frac{1}{2V}\int_{-V}^{+V} x^2\, dx = \frac{V^2}{3}$

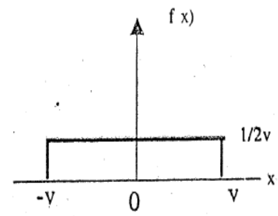
mean square quantization error, $N_Q = \frac{S^2}{12}$

∴ Input signal to quantization noise ratio, $\frac{S_i}{N_Q} = \frac{4V^2}{S^2}$

$\therefore \frac{S_i}{N_Q} = Q^2 = 2^{2N}$

Step Size $S = \frac{V_H - V_L}{Q} = \frac{2V}{Q} = \frac{2V}{2^N}, \quad \therefore V = \frac{SQ}{2}$

$\therefore \frac{S_i}{N_Q} = Q^2 = 2^{2N}$



For a good communication system $\frac{S_i}{N_Q} \cong \frac{S_0}{N_Q}$

$\therefore \frac{S_0}{N_Q} \cong 2^{2N}$, where S_0 = Output signal power

$$\left(\frac{S_0}{N_Q}\right)_{dB} = 10\log_{10}\left(\frac{S_0}{N_Q}\right) = 6N dB$$

This is the maximum signal to quantization noise ratio possible and 6N dB is not for the entire information signal. Because.

$\frac{S_0}{N_Q} = \frac{4V^2}{S^2}$

V = maximum voltage, but for the entire time signal amplitude is not equal to V

∴ $S_o / N_Q \downarrow$ as $V \downarrow$ because S is constant.

Step size $s = \frac{2V}{2^N}$ Where $V_{max} = V$ and $V_{min} = -V$

N = Number of bits / sample and $2^N = Q$ = number of quantization levels

∴ Mean square quantization error $= \frac{s^2}{12} = \frac{4V^2}{12.2^{2N}} = \frac{V^2}{3.2^{2N}}$

For voice signal, most of the time the signal amplitude is very small. ∴ SNR is also small

<u>For PCM if modulating signal is A $\cos\omega_0 t$</u>

$(SNR)_0 = \frac{3P2^{2N}}{V^2}$

The modulating signal is $A\cos\omega_0 t$ ∵ $v_{max} = A$ and $V_{min} = -A$ and $P = \frac{A^2}{2}$

$\therefore (SNR)_0 = \frac{3\frac{A^2}{2}2^{2N}}{A^2} = 1.5.2^{2N}$

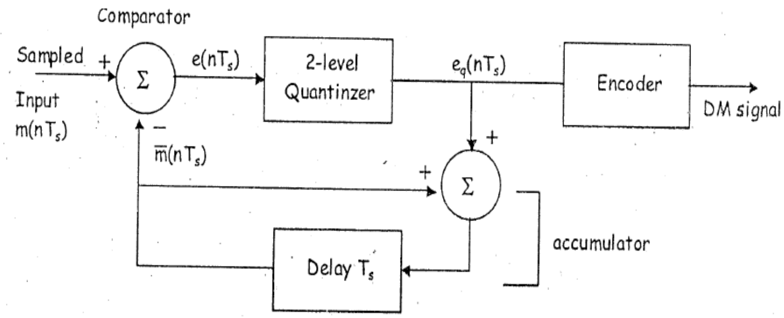
in dB; $10\log_{10}(SNR)_0 = 1.8 + 6N$ dB

Sampling frequency=8000Hz, number of quantization levels=256 , bit rate=8x8000Hz, SNR=1.8+6x8=49.8dB.

b. Draw and explain the block diagram of delta modulation. Discuss its

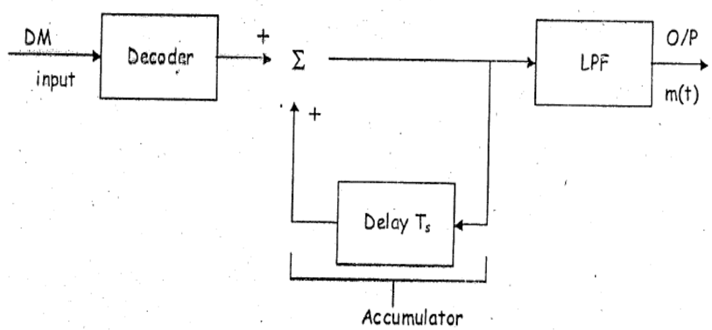
merits and limitations. **(8)**

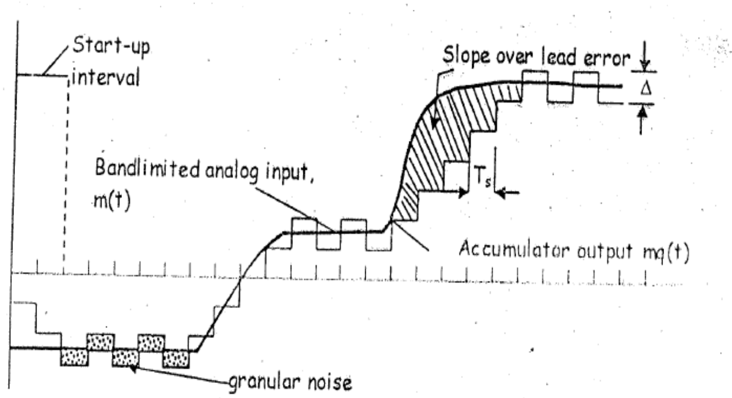
Answer:



Delta modulation is a one bit transmission technique. If $m(t) > m_q(t)$ then one step (Δ) increment and DM wave will be logic 1 and $m(t) < m_q(t)$ then one step decrement and DM wave will be logic 0. For DM the f_s is very high compare to the f_s of PCM.

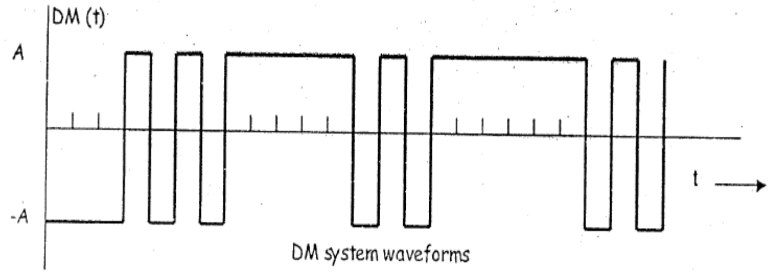
In its basic form (Linear delta modulation, LDM), DM provides a staircase approximation to the over sampled version ($f_s \gg 2 f_m$) of the message signal. The difference between the input and the approximation is quantized in to only two levels $\pm \Delta$. So the quatizer is a two level quantizer or hard limiter.



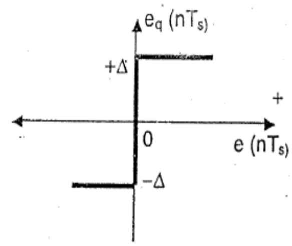


DM involves

1. Comparator, 2. Quantizer, 3. Accumulator



DM system waveforms



Input-Output relatin of 2 level quantizer

Limitations of DM System

1. Slope overload error (distortion)

If the slope of the approximated signal is smaller than the slope of the information signal, the approximate signal will not follow the input signal properly. This error is called slope overload error. ∴ For avoiding slope overload

$\Delta f_s = \frac{\Delta}{T_s} >$ maximum slope of input signal i.e. for a given signal, slope overload can be decreased by

increasing the step size Δ and or increasing the sampling frequency f_s

2. Granular noise or Hunting error.

Granular noise occurs when the step size Δ is too large relative to the local slope characteristics of the input wave form m(t), there by causing the stair case approximation to hunt around a relatively flat segment of the input waveform. This can be reduced by decreasing step size and/or decreasing f_s.

3. Start-up interval error

If the initial amplitude of the input signal is not small, the staircase signal will take some time to reach the information signal. This time interval is called start-up interval. If this time is large, reconstructed signal and original signal will not be same. Start-up interval error can be decreased by increasing the step size and/or increasing f_s.

4. Quantization Error

Quantization error is the difference between e (nT_s) and $e_q(nT_s)$. Quantization error can be reduced by reducing in the step size and the maximum quantization error = $\pm\Delta$Volt.

Q.5 a. Explain Inter symbol interference. (8)

Answer: Refer Pages 243 to 245 of Text Book-I

b. Explain Duo binary signalling. (8)

Answer: Refer Pages 252 to 255 of Text Book-I

Q.6 a. Explain the difference between DPSK and PSK. How should PSK senders and receivers be adjusted, if they are to be used for DPSK? Name one advantage of

DPSK over PSK and name one advantage of PSK over DPSK.

Answer:

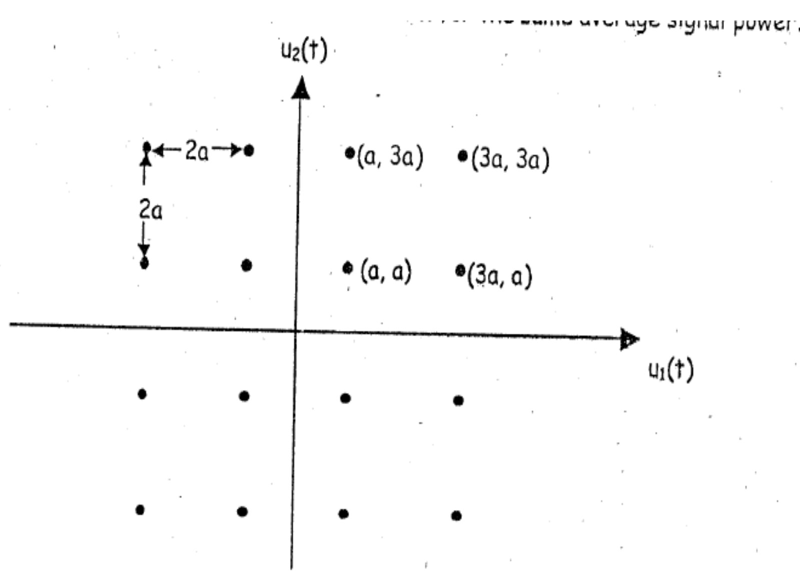
In PSK, the information is represented by the absolute phase of the signal. DPSK is a variant of PSK where the difference between the phase in two consecutive symbol intervals represents the information.

A PSK sender maps information on the phase of a carrier. The needed adjustment of a PSK sender to be used as a DPSK sender is a preprocessor that updates the phase by adding the intended phase of one signal interval to the one from the previous signal interval.A PSK receiver estimates the phase of the received signal and maps that on the closest phase in the constellation. Assuming that the absolute phase error changes slowly compared to the duration of a symbol, we can do the following. The corresponding adjustment of the receiver is to insert a block that remembers the estimated phase of the previous symbol interval, and subtracts that from the estimated phase of the present symbol interval. That difference is then what is compared to the phases of the signal constellation.

An advantage of PSK over DPSK: If we have coherent detection, i.e. the frequency references of the sender and the receiver are fully syncronized, then the resulting error probability is smaller for PSK than for DPSK, since an error on the channel in one symbol interval results in one symbol error in PSK, while we typically get errors in two consecutive symbol intervals after detection in DPSK. This costs sowewhere between two and three dB of signal power.
An advantage of DPSK over PSK: PSK demands full synchronization. If we have non-coherent detection, i.e. the receiver does not know the absolute phase of the frequency reference of the sender, then ordinary PSK is completely useless. However, DPSK can be used in this situation as long as any change in the the absolute phase error is slow compared to the symbol interval.

b. Show that probability of error in 16-PSK is higher than 16-QAM.

Answer:



The points are placed symmetrically about the origin of the signal space to simplify the hardware design of the system while keeping the energy per signal near a minimum.

Let as assume that all 16 signals are equally likely. Because of the symmetry displayed in figure the average normalized energy of a signal is

$$E_s = \frac{1}{4}\left[\left(a^2 + a^2\right) + \left(9a^2 + a^2\right) + \left(a^2 + 9a^2\right) + \left(9a^2 + 9a^2\right)\right] = 10a^2$$

$\therefore a = \sqrt{0.1E_s}$ and $d = 2a = 2\sqrt{0.1E_s}$. Here $E_s = 4E_b$ $\therefore a = \sqrt{0.4E_b}$

$d = 2a = 2\sqrt{0.4\,E_b}$

But for 16.PSK, $d = \sqrt{16E_b \sin^2\left(\frac{\pi}{16}\right)} = 2\sqrt{0.15E_b}$

$d_{16.PSK} < D_{16QAM}$ for the same P_s $\therefore P_{e\,16\,QAM} < P_{e\,16\,PSK}$ for the same P_s

Thus, 16 QAM will be shown to have a lower error rate then 16 PSK for the same average signal power.

Q.7 a. Obtain the output of matched filter if input g(t) as shown in Fig.1 is applied

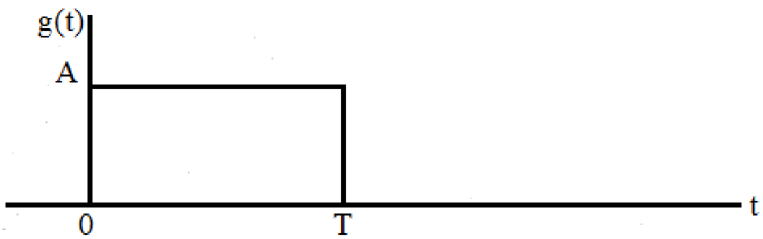
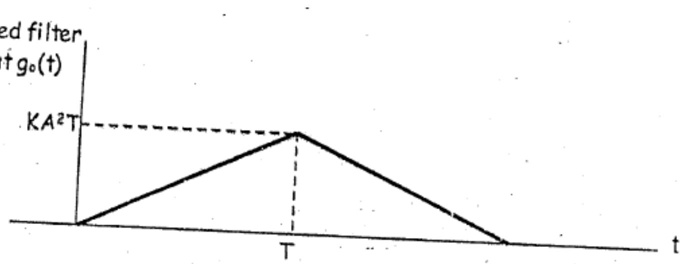


Fig.1

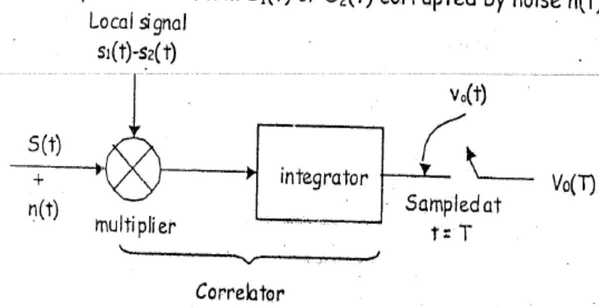
Answer:



b. Explain the correlator receiver. Obtain its signal output and noise output.

Answer:

Correlator is an alternate type of receiving system, which is identical in performance with the matched filter receiver. The input to the correlator is a binary data waveform $S_1(t)$ or $S_2(t)$ corrupted by noise $n(t)$. The bit length is T.



A coherent system of signal reception

There is a correlation between received signal and $S_1(t) - S_2(t)$. ∴ Correlator receiver. $S_1(t) - S_2(t)$ is generating from the received signal.

Signal & Noise Output Of Correlator

$$V_0(T) = \frac{1}{\tau}\int_0^T V_i(t)[S_1(t) - S_2(t)]\,dt,$$

Where $\tau = RC$ = time constant of integrator

$$V_0(T) = S_0(T) + n_0(T)$$

$$S_0(T) = S_{01}(T) \text{ or } S_{02}(T)$$

$$V_i(t) = S_i(t) + n(t)$$

$$S_i(t) = S_1(t) \text{ or } S_2(t)$$

$$\therefore S_0(T) = \frac{1}{\tau}\int_0^T S_i(t)\,[S_1(t) - S_2(t)]\,dt \qquad \text{..... (C)}$$

$$\therefore n_0(T) = \frac{1}{\tau}\,n(t)\,[S_1(t) - S_2(t)]\,dt \qquad \text{.....(D)}$$

Q.8 a. Explain the DS/BPSK spread spectrum with the help of suitable block diagram.

Answer:

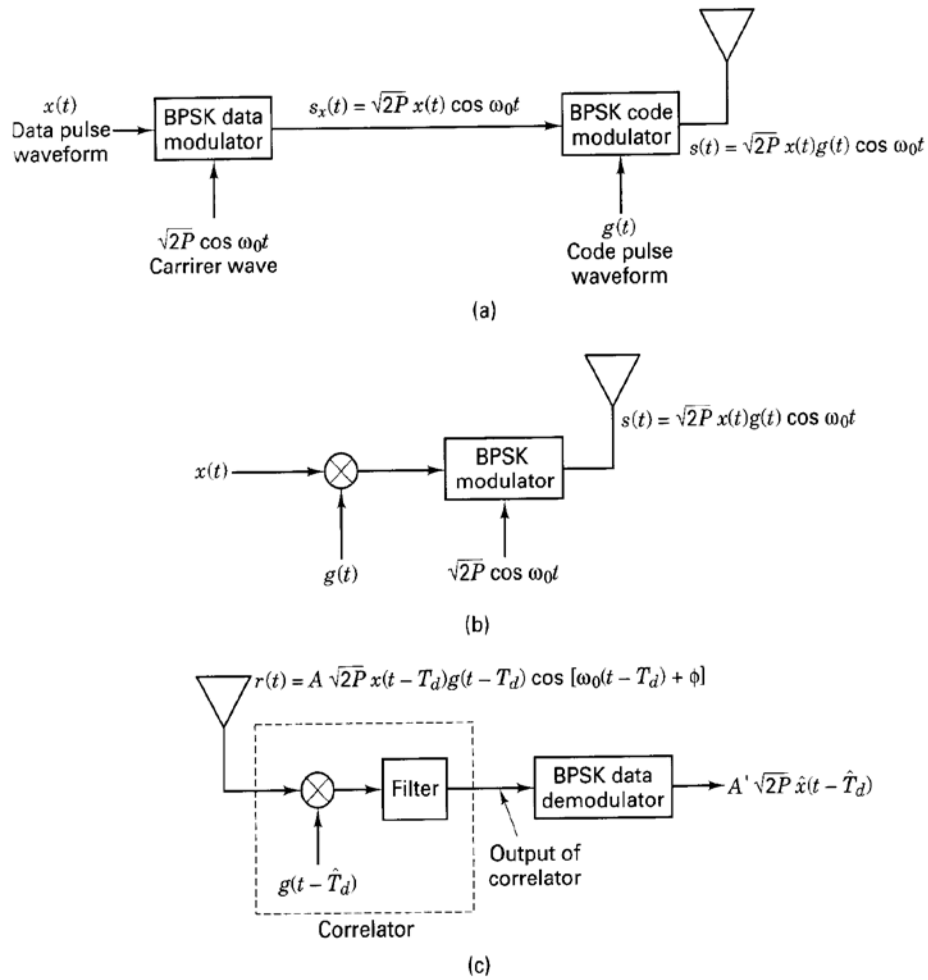


Figure Direct-sequence spread-spectrum system. (a) BPSK direct-sequence transmitter. (b) Simplified BPSK direct-sequence transmitter. (c) BPSK direct-sequence receiver.

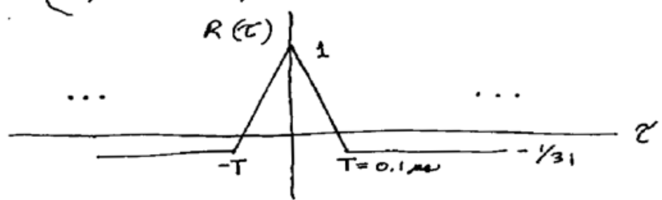
b. **A feedback shift register PN generator produces a 31-bit PN sequence at a clock rate of 10MHz. What are the equation and graphical form of the autocorrelation function of the sequence? Assume that the pulses have values of ±1. (8)**

Answer:

The 31-bit sequence has an autocorrelation function with a maximum value at $\tau = 0$ decreasing linearly to $-1/31$ at $\tau = |T|$ for T equal to a chip interval (0.1 μs in this case).

$$R(\tau) = \frac{1}{T_0}\int_{-T_0/2}^{T_0/2} x(t)\,x(t+\tau)\,d\tau$$

$$= \frac{1}{p}\left\{\begin{array}{l}\text{\# agreements} - \text{\# disagreements in one} \\ \text{full period of the sequence with a } \tau \\ \text{position cyclic shift}\end{array}\right\}$$



R(τ) repeats for offsets modulo-31 chip times (T_0 = 3.1 μs).

Q.9 Write short note on any <u>TWO</u> of the following: (2×8)

(i) Digital Radio

(ii) CDMA

(iii) Digital Multiplexer

Answer: (i) Refer pages 350 to 353 of Text Book-I

(ii) Refer pages 468 to 469 of Text Book-I

(iii) Refer pages 218 to 220 of Text Book-I

<u>TEXT BOOK</u>

I. Digital Communications, Wiley Student Edition, Simon Haykin