10

PVP 23

Code: 23EC3502

III B.Tech - I Semester - Regular Examinations - NOVEMBER 2025

DIGITAL COMMUNICATIONS
(ELECTRONICS & COMMUNICATION ENGINEERING)

Duration: 3 hours Max. Marks: 70

Note: 1. This question paper contains two Parts A and B.
2. Part-A contains 10 short answer questions. Each Question carries 2 Marks.
3. Part-B contains 5 essay questions with an internal choice from each unit. Each Question carries 10 marks.
4. All parts of Question paper must be answered in one place.

BL – Blooms Level CO – Course Outcome

PART – A

		BL	CO
1.a)	Define sampling and quantization.	L1	CO1
1.b)	Distinguish between PCM and DPCM.	L2	CO1
1.c)	Explain ASK with a neat signal representation.	L2	CO2
1.d)	Write the probability of error expression for FSK.	L1	CO2
1.e)	Define jamming margin in spread spectrum.	L1	CO3
1.f)	Sketch the block diagram of FHSS system.	L3	CO3
1.g)	Define channel capacity.	L1	CO4
1.h)	Explain Shannon-Fano coding briefly.	L2	CO4
1.i)	Write the structure of convolutional code.	L1	CO5
1.j)	Write short notes on Trellis diagram.	L1	CO5

PART – B

			BL	CO	Max. Marks
		UNIT-I			
2	a)	Illustrate the working of DPCM neat block diagram.	L3	CO1	5 M
	b)	Explain about merits and demerits of ADM over DM.	L2	CO1	5 M
		OR			
3	a)	Outline the process of quantization in PCM systems.	L2	CO1	5 M
	b)	Summarize about Manchester line coding scheme.	L2	CO1	5 M
		UNIT-II			
4	a)	Describe about QPSK modulation and demodulation.	L2	CO2	5 M
	b)	Determine the probability of error for PSK.	L3	CO2	5 M
		OR			
5	a)	Explain about M-ary PSK system with a neat diagram.	L2	CO2	5 M
	b)	Distinguish between ASK, FSK, and PSK techniques.	L4	CO2	5 M
		UNIT-III			
6	a)	Summarize about DSSS BPSK system.	L2	CO3	5 M
	b)	Summarize the need of Spread spectrum modulation.	L2	CO3	5 M

		OR			
7	a)	Compare the Slow and Fast Frequency Hopping.	L4	CO3	5 M
	b)	Discuss about the Applications of Spread spectrum Techniques.	L2	CO3	5 M
		UNIT-IV			
8	a)	Summarize about entropy and explain its properties.	L2	CO4	5 M
	b)	Explain about mutual information and its significance.	L2	CO4	5 M
		OR			
9	a)	Illustrate Lempel-Ziv coding with an example.	L3	CO4	5 M
	b)	Summarize about channel coding theorem.	L2	CO4	5 M
		UNIT-V			
10	a)	Explain encoder and decoder for cyclic codes with an example.	L2	CO5	5 M
	b)	Write short notes on error detection and correction using block codes.	L2	CO5	5 M
		OR			
11	a)	Illustrate convolutional codes with an example of encoding sequence.	L3	CO5	5 M
	b)	Explain the Viterbi decoding algorithm.	L2	CO5	5 M

PVP23

III B.Tech - I sem Regular Exam

Sub: Digital Communications.

SubCode: 23EC3502

Scheme of Evaluation

PART-A.

1.(A). Sampling - 1M
Quantization - 1M.

1(B). PCM Points - 1M
DPCM points - 1M.

1(C) ASK wave forms - 2M.

1(d) Error probability expression of FSK - 2M.

1(e) Definition of Hamming Margin - 1M
Expression - 1M

1(f) TX - Block diagram - 1M.
Rx - Block diagram - 1M.

1(g) Definition - 1M, equation - 1M

1(h) Explanation - 2M.

1(i) Structure Concept of convolutional Code - 2M.

1(j) Concept about Trellis diagram - 2M.

PART-B

UNIT-I

2(a) DPCM diagram - 2M. } 5M.
Explanation - 3M

2(b). Merits - 3M. } 5M.
Demerits - 2M

3(a) midtread type - 2M } 5M.
mid rise type - 2M
Explanation - 1M

3(b) Manchester waveform - 2M. } 5M.
Expressions & Explanation - 3M

UNIT-II

4(a). QPSK diagram - 2M. } 5M.
Explanation - 3M

4(b). probability of Error of PSK - 5M.

5(a) M-ary PSK Transmitter - 3M } 5M.
Receiver - 2M

5(b). each point of Comparision - 1M
5x1M = 5M

UNIT-III

6(a). DSSS BPSK - Transmitter - 3M
Receiver - 2M
5M.

6(b). each point - 1M
3×1M = 3M
Explanation = 2M
5M.

7(a) Comparision of each point = 1M
5×1M = 5M.

7(b) Application each point - 1M
5×1M = 5M.

UNIT-IV

8(a). Entropy Explanation - 2M.
Equation - 1M
properties. - 2M
5M.

8(b). Mutual Information
Explanation, equation - 3M.
properties. - 2M
5M.

9(A). Lempel Ziv coding Concept - 1M
Example - 4M. } 5M.

9(B). diagram - 2M.
Explanation & equations, conditions - 3M. } 5M.

UNIT - V

10(a) Cyclic Code Encoder - 3M.
decoder - 2M } 5M.

10(b) Block Code Concept - 1M.
Error detection - 2M.
Error correction - 2M. } 5M.

11(a) Convolutional Encoder Concept - 1M
Example - 4M } 5M.

11(b) Viterbi Algorithm Concept - 3M.
Explanation of important terms - 2X1M = 2M } 5M.

X

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PART-A

1.A)

Sampling:

- Process of converting analog signal into discrete signal.
- Sampling is common in all pulse modulation techniques
- The signal is sampled at regular intervals such that each sample is proportional to amplitude of signal at that instant

Quantization

- The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels.

Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal

1.b) PCM:

(i) PCM provides high noise immunity.
(ii) Due to digital nature of the signal, we can place repeaters between the transmitter and the receivers. Infact, the repeaters regenerate the received PCM signal. This can not be possible in analog systems. Repeaters further reduce the effect of noise.

DPCM

(i) As the difference between $x(nT_s)$ and $\hat{x}(nT_s)$ is being encoded and transmitted by the DPCM technique, a small difference voltage is to be quantized and encoded.
(ii) This will require less number of quantization levels and hence less number of bits to represent them.

1.C)

(i) Definition

Amplitude shift keying (ASK) or ON-OFF keying (OOK) is the simplest digital modulation technique. In this method, there is only one unit energy carrier and it is switched on or off depending upon the input binary sequence.

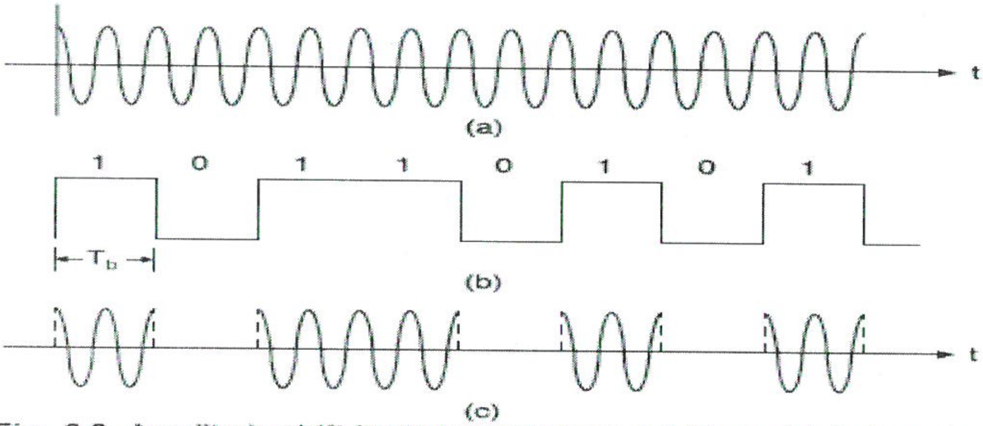


Fig. 6.2. Amplitude-shift keying waveforms: (a) Unmodulated carrier, (b) NRZ Unipolar bit sequence, (c) ASK waveform.

1.D) Probability error of FSK:

$\therefore$ The probability of error $P_e = \frac{1}{2}\text{erfc}\left[0.6\frac{E_s}{\eta}\right]^{1/2}$

When one of two *orthogonal* frequencies are transmitted, $2\Omega T = m\pi$ (m an integer) and

probability of error for FSK $P_e = \frac{1}{2}\text{erfc}\left(\frac{E_s}{2\eta}\right)^{1/2}$

The probability of error of noncoherent FSK is found to be

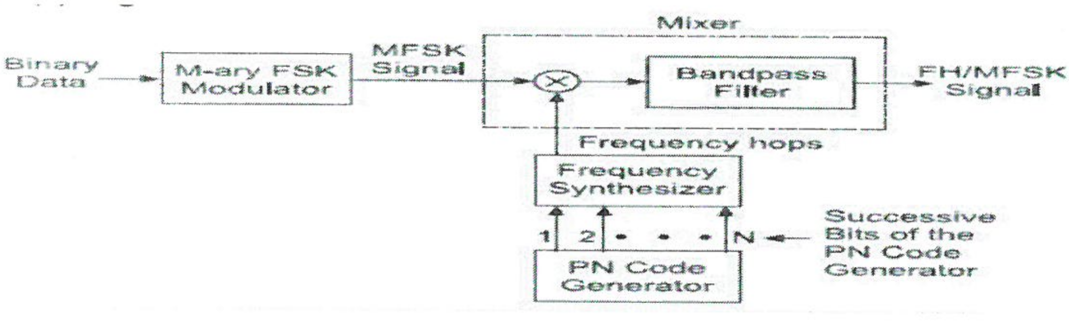
$$P_e = \frac{1}{2}e^{-E_s/2\eta}$$

1.E)

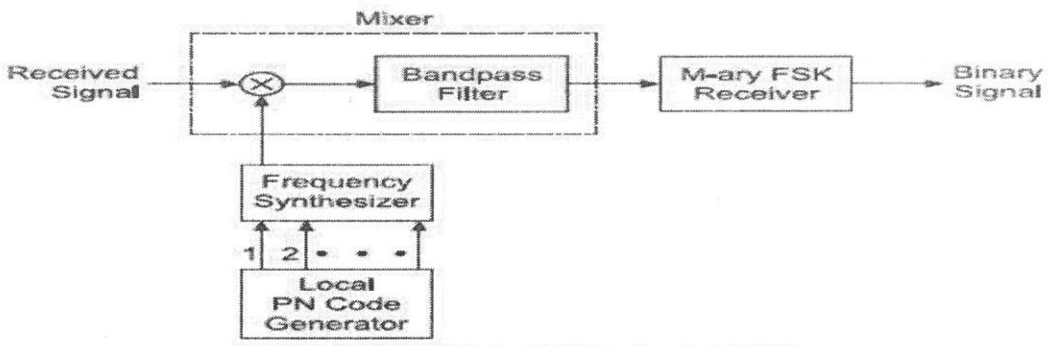
The ratio J/Ps is called as the **jamming margin**. Therefore, the jamming margin may be defined as the ratio of average interference power J and the signal power Ps. If the jamming margin andthe process gain both are expressed in dB, then,

(Jamming margin) dB = (Processing Gain) dB – 10 log10 [E_b/N_0]min

1F) TRANSMITTER



RECEIVER



1.G)Channel capacity:

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

In this expression, $B' =$ channel bandwidth in Hz
$S =$ Signal power
$N =$ Noise power

we can determine the channel capacity. In fact, the channel capacity is the maximum amount of information that can be transmitted per second by a channel. If a channel can transmit a maximum of K pulses per second, then, the channel capacity C is given by

$$C = \frac{K}{2}\log_2\left(1+\frac{S}{N}\right) \text{ bits per second} \qquad ...(8.54)$$

1.h)Shnonfano coding:

An efficient code can be obtained by the following simple procedure, known as Shannon-Fano
Algorithm:

1. List the source symbols in order of decreasing probability.
2. Partition the set into two sets that are as close to equiprobables as possible, and assign 0 to the upper set 1 to the lower set.
3. Continue this process, each time partitioning the sets with as nearly equal probabilities as possible until further partitioning is not possible

1.I)

The code dimension of a convolutional code depends on n, k and L. Here k represents the number of message bits taken at a time by the encoder, n is the number of encoded bits per message bit and L is the encoder's memory. The code dimension is therefore represented by (n, k, L)

1J) **Code Trellis**

A more compactgraphical representation which is popularly known as code trellis. Here, the nodes on the left denote the four possible current states and the nodes on the right are the resulting next rate.

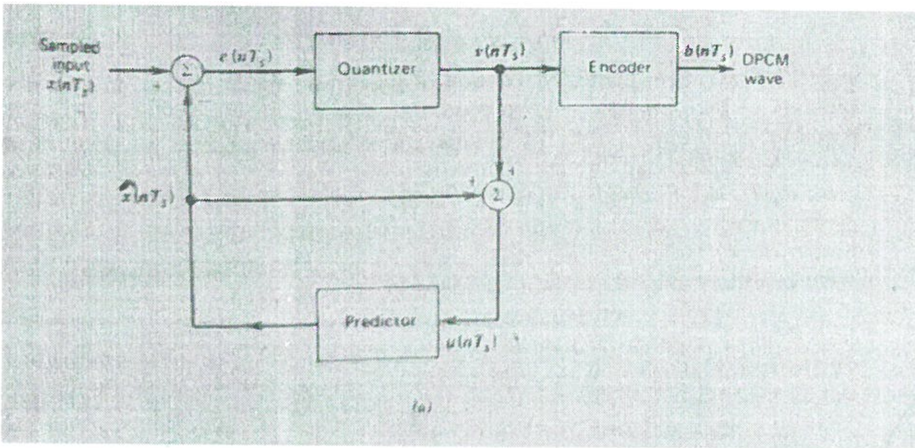
A solid line represents the state transition for m0 = 0 and the dotted line represents the branch for m0 = 1. Each branch s labelled with the resulting output bit

PART-B

UNIT-I

2.A)DPCM:

As a matter of fact, PCM is not a very efficient system because it generates so many bits and requires so much bandwidth to transmit. Many different ideas have been proposed to improve the encoding efficiency of A/D conversion. In general, these ideas exploit the characteristics of the source signals. DPCM is one such scheme.



The sampled signal is denoted by $x(nT_s)$ and the predicted signal is denoted by $\hat{x}(nT_s)$. The comparator finds out the difference between the actual sample value $x(nT_s)$ and predicted sample value $\hat{x}(nT_s)$. This is known as Prediction error and it is denoted by $e(nT_s)$. It can be defined as.

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \qquad ...(3.60)$$

$$\begin{aligned} v(nT_s) &= Q[e(nT_s)] \\ &= e(nT_s) + q(nT_s) \end{aligned} \qquad (5.67)$$

where $q(nT_s)$ is the quantization error. According to Fig. 5.15a, the quantizer output $v(nT_s)$ is added to the predicted value $x(nT_s)$ to produce the predictor input

$$u(nT_s) = \hat{x}(nT_s) + v(nT_s) \qquad (5.68)$$

Substituting the second line of Eq. 5.67 in Eq. 5.68, we get

$$u(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s) \qquad (5.69)$$

However, from Eq. 5.66 we observe that $\hat{x}(nT_s) + e(nT_s)$ is equal to the input signal $x(nT_s)$. Therefore, we may rewrite Eq. 5.69 as follows:

$$u(nT_s) = x(nT_s) + q(nT_s) \qquad (5.70)$$

which represents

The decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal. Thus the signal at the receiver differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

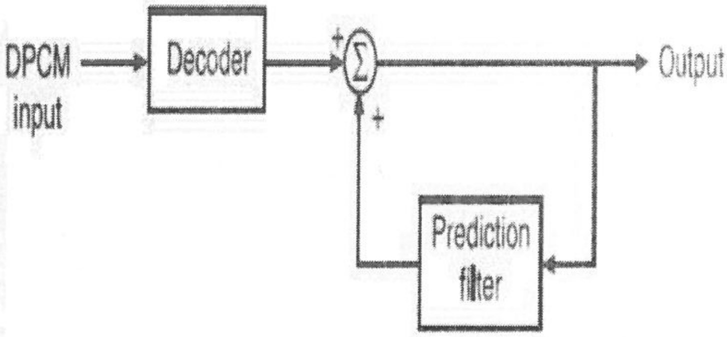


Fig. 3.38. DPCM receiver

2.B)

A PCM system has few drawbacks as under:

(i) The encoding, decoding and quantizing circuitry of PCM is complex.

(ii) PCM requires a large bandwidth as compared to the other systems.

3.32.3. Advantage of DPCM : Salient Features*

(i) As the difference between $x(nT_s)$ and $\hat{x}(nT_s)$ is being encoded and transmitted by the DPCM technique, a small difference voltage is to be quantized and encoded.

(ii) This will require less number of quantization levels and hence less number of bits to represent them.

(iii) Thus signaling rate and bandwidth of a DPCM system will be less than that of PCM.

3.A)QUANTAIZATION in PCM:

The essential operations in the transmitter of a PCM are

1.Sampling
2.Quantaizing
3.Encoding

Quantaizing:

- There are two types of Quantization
    - Uniform Quantization
    - Non-uniform Quantization.
- The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**.
- The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization.**

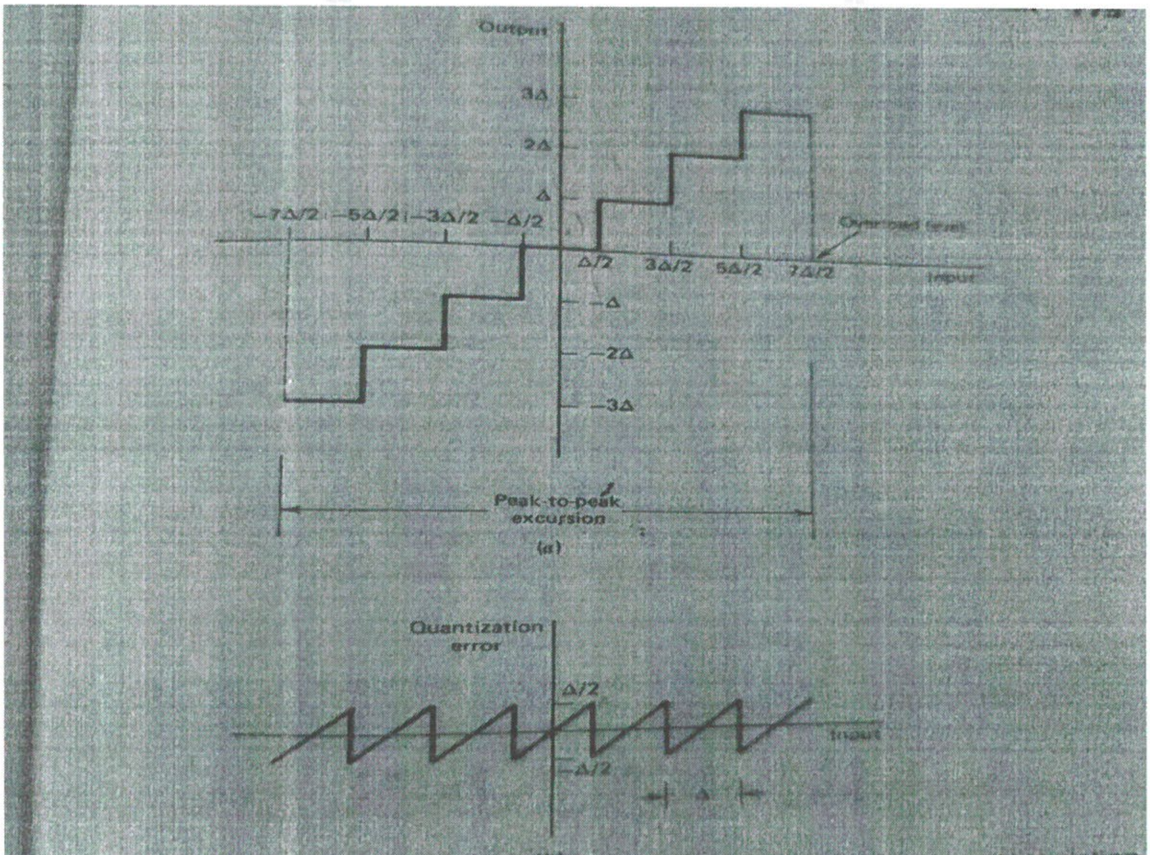


Figure 5.2 (a) Transfer characteristic of quantizer of midtread type. (b) Variation of the quantization error with input.

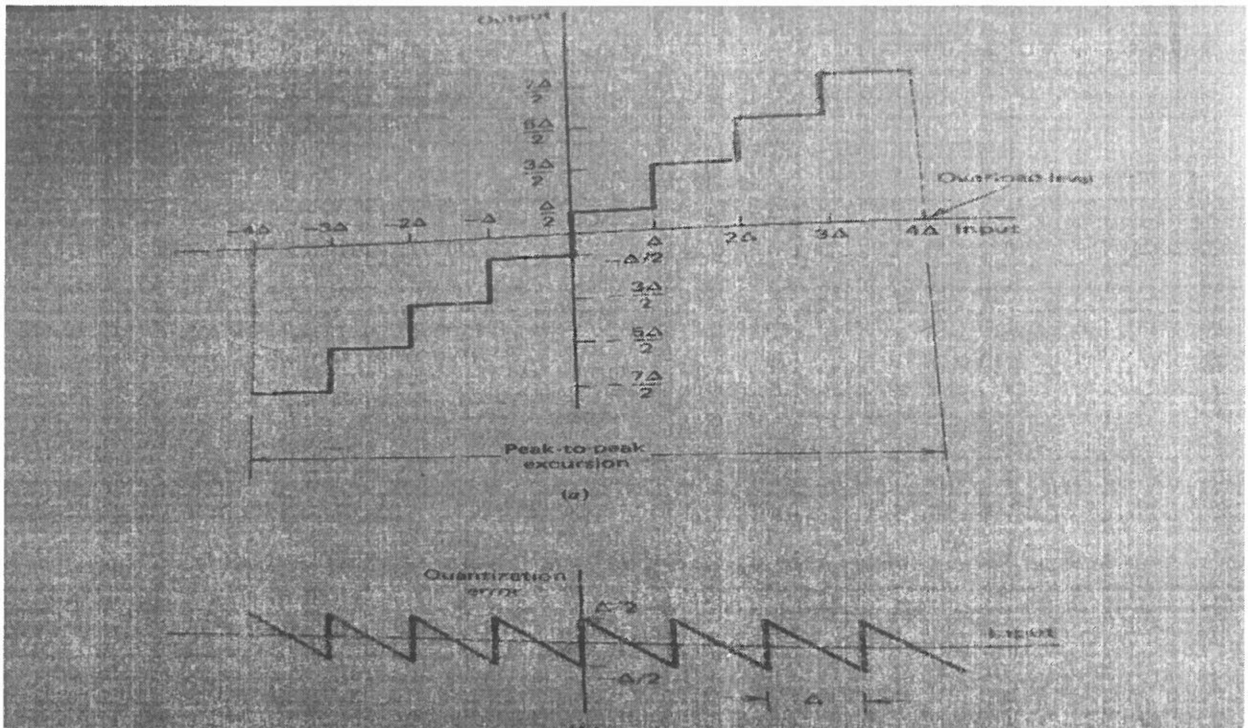


Figure 6.3 (a) Transfer characteristic of quantizer of midriser type. (b) Variation of the quantization error with input.

3.B) **Definition and Waveform**

This type of waveform is shown in figure 5.14. In this case, if symbol '1' is to be transmitted,then a positive half interval pulse is followed by a negative half interval pulse. If symbol '0' is to be transmitted, then a negative half interval pulse is followed by a positive half interval pulse.

Hence, for any symbol the pulse takes positive as well as negative value

2. Mathematical Expressions

If symbol '1' is to be transmitted, then

$$x(t) = \begin{cases} \dfrac{A}{2} & \text{for } 0 \le t < \dfrac{T_b}{2} \\ -\dfrac{A}{2} & \text{for } \dfrac{T_b}{2} \le t < T_b \end{cases} \quad ...(5.15)$$

and if symbol '0' is to be transmitted, then

$$x(t) = \begin{cases} -\dfrac{A}{2} & \text{for } 0 \le t < \dfrac{T_b}{2} \\ \dfrac{A}{2} & \text{for } \dfrac{T_b}{2} \le t < T_b \end{cases} \quad ...(5.16)$$

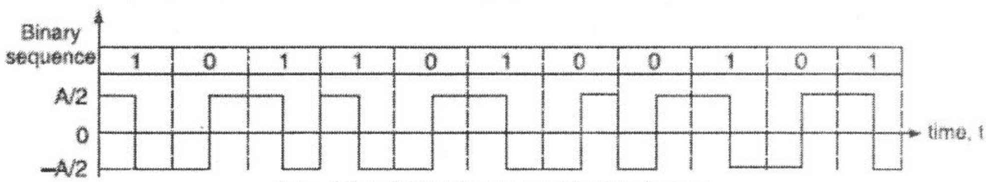


Fig. 5.14. Split phase manchester format

UNIT-II

4.A)QPSK

1. Block Diagram

The block diagram of offset QPSK (OQPSK) is shown in figure 6.30 This shows the mechanism by which a bit stream b(t) generates a QPSK signal for transmission.

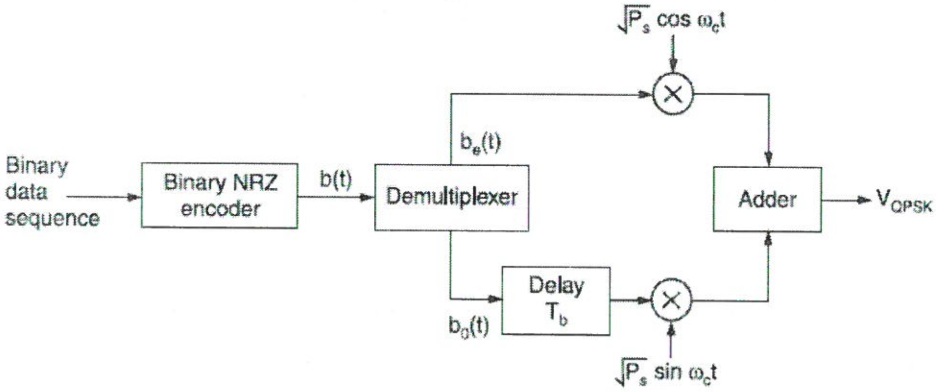


Fig. 6.30. An offset QPSK transmitter

5. Mathematical representation of QPSK

A QPSK signal can be represented mathematically as under:

$$V_{QPSK}(t) = \sqrt{2P_s}\cos\left[\omega_c t + (2m+1)\frac{\pi}{4}\right],\ m = 0, 1, 2, 3$$

By substituting the values of m from 0 to 3, we get the four messages listed in Table 6.9 i.e.,

$$V_{QPSK} = S_1 = \sqrt{2P_s}\cos\left[\omega_c t + \frac{\pi}{4}\right] \quad \text{for } m = 0$$

$$V_{QPSK} = S_2 = \sqrt{2P_s}\cos\left[\omega_c t + \frac{3\pi}{4}\right] \quad \text{for } m = 1$$

Similarly, we can obtain the QPSK output for m = 2 and m = 3. As explained earlier, we can substitute P_s in terms of symbol energy and symbol time duration as under:

$$P_s = \frac{E}{T}$$

The QPSK system of modulation is also called as four state PSK (or simply 4 PSK).

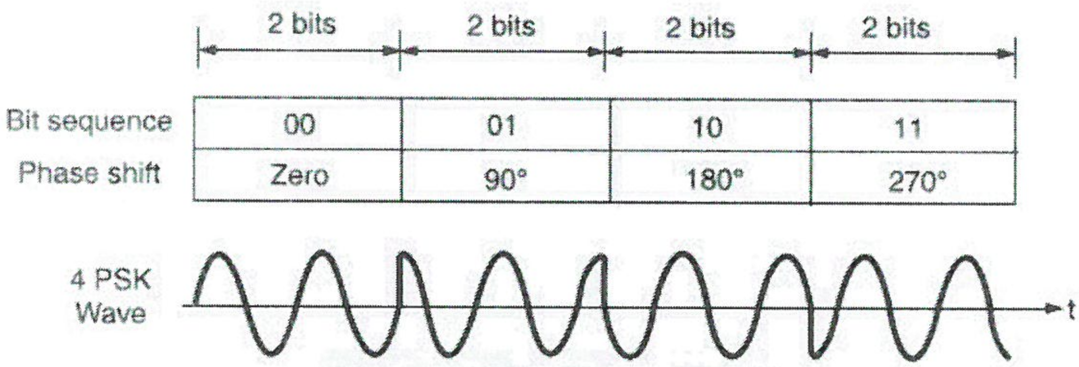


Fig. 6.28. Waveforms of QPSK

6.13.3. The QPSK Receiver

1. Block Diagram

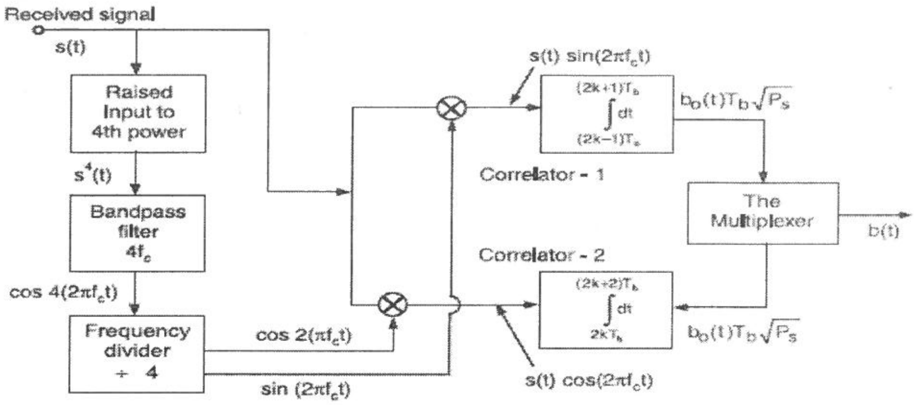


Fig. 6.34. A QPSK receiver

2. Working Operation

Let us represent the received QPSK signal by s(t) instead of V_{QPSK} (t). The received QPSK signal s(t) is raised to fourth power i.e., $s^4(t)$. This signal is then filtered by using a bandpass filter with a center frequency of $4\,\omega_c t$. The output of bandpass filter is $\cos 4\,\omega_c t$. A frequency divider which divides the frequency at the filter output by 4 generates the two carrier signals $\sin \omega_c t$ and $\cos \omega_c t$. The incoming signal s(t) is applied to two synchronous demodulators consisting of a multiplier (balanced modulator) followed by an integrator. Each integrator integrates over a two-bit interval of duration $T_s = 2T_b$. One synchronous demodulator uses $\cos \omega_c t$ as carrier signal and the other one uses $\sin \omega_c t$ as a carrier signal.

4.B)Probability error of PSK:

Probability of Error of PSK

As the Probability of error $P_e = \frac{1}{2} erfc\left[\frac{1}{8}\gamma_{max}{}^2\right]^{\frac{1}{2}}$

$$\gamma_{max}{}^2 = \frac{2}{\eta}\int_0^T [S_1(t) - S_2(t)]^2 dt$$

In PSK system,

$$S_1(t) = A\cos\omega_0 t$$

$$\& \; S_2(t) = -A\cos\omega_0 t$$

$$\therefore P(t)=S_1(t) - S_2(t) = 2A\cos\omega_0 t$$

$$\gamma_{max}{}^2 = \frac{2}{\eta}\int_0^T (2A\cos\omega_0 t)^2\, dt$$

$$= \frac{2}{\eta}.4A^2\int_0^T [\cos^2\omega_0 t]dt$$

$$= \frac{8A^2}{\eta}.\int_0^T \frac{1+\cos 2\omega_0 t}{2}\,dt$$

$$= \frac{4A^2}{\eta}.\int_0^T dt + \int_0^T \cos 2\omega_0 t\,dt$$

$$= \frac{4A^2}{\eta}[t]_0^T + \left[\frac{\sin\omega_0 t}{2\omega_0}\right]_0^T$$

$$\gamma_{max}{}^2 = \frac{4A^2 T}{\eta}$$

Probability of Error of PSK

Probability of error $P_e = \frac{1}{2} erfc \left[\frac{1}{8} r_{max}{}^2\right]^{\frac{1}{2}}$

$$= \frac{1}{2} erfc \left[\frac{1}{8} . \frac{4A^2T}{\eta}\right]^{\frac{1}{2}}$$

$$= \frac{1}{2} erfc \left[\frac{A^2T}{2\eta}\right]^{\frac{1}{2}}$$

$$= \frac{1}{2} erfc \left[\frac{E_S}{\eta}\right]^{\frac{1}{2}} \qquad [\because E_S = \frac{A^2T}{2}]$$

$$\therefore probability\ error\ for\ PSK = \frac{1}{2} erfc \left[\frac{E_S}{\eta}\right]^{\frac{1}{2}}$$

$1/2\ erfc\ [\ v/\sqrt{2}\ ] = Q(v)$

$$\therefore probability\ error\ for\ PSK = Q\left[\frac{2E_s}{\eta}\right]^{\frac{1}{2}}$$

5.A) M-Array PSK

6.15.3. M-ary PSK Transmitter

1. Block Diagram

The block diagram of an M-ary PSK system has been shown in figure 6.42.

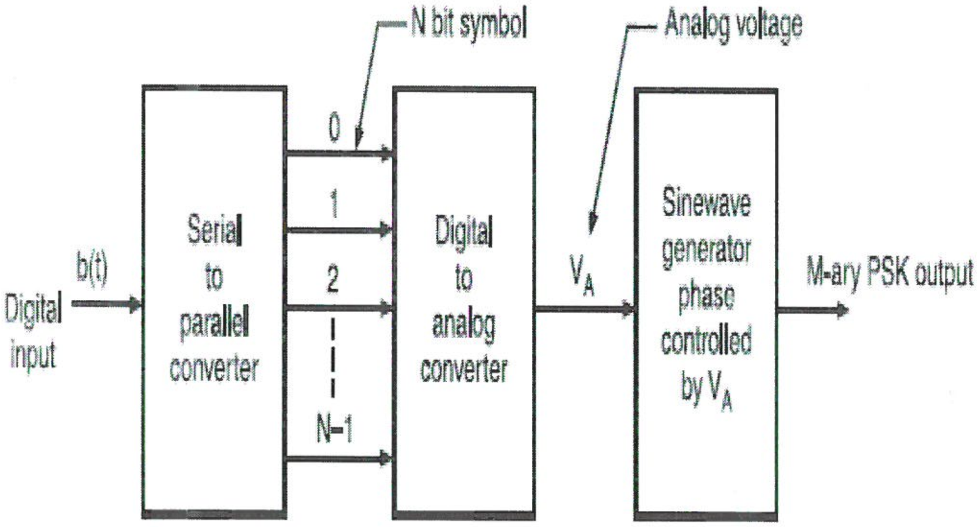


Fig. 6.42. M-ary PSK transmitter

2. Working Operation

The bit stream b(t) is applied to a serial to parallel converter. This block can store the N-bits of a symbol. These N-bits per symbol have been presented serially, in time sequence one after the other. The N-bits per symbol are first assembled by the serial to parallel converter block. Then, all these bits are presented at once on the N output lines of the converter. Thus, the N-bit message appears in the parallel form at the output of the serial to parallel converter. The output of the serial to parallel converter remains unchanged for a duration of NT_b of a symbol which is the time required for the converter to assemble a new group of N-bits. After every NT_b seconds, the converter output is updated. The N-bit output of the converter is then applied to a D/A converter. Depending on the N-bit digital input, it produces an analog output V_A. The N-bit digital input can have $2^N = M$ number of possible combinations. Hence, the D/A converter output V_A will have M number of distinct values, depending on the symbols. Finally this analog voltage is applied to a sinusoidal signal generator, which produces a constant amplitude sinusoidal output voltage, the phase ϕ_m of which is determined by the D/A converter output V_A. Thus, at the output of the transmitter, we get a fixed amplitude sinusoidal waveform, the phase of which has a one to one correspondence to the N-bit symbols. The will change only once per symbol time $T_s = NT_b$. Thus, the M-ary PSK is generated.

6.15.4 M-ary PSK Receiver

1. Block Diagram

The block diagram of M-ary PSK receiver has been shown in figure 6.43. This is same as the non-offset QPSK receiver.

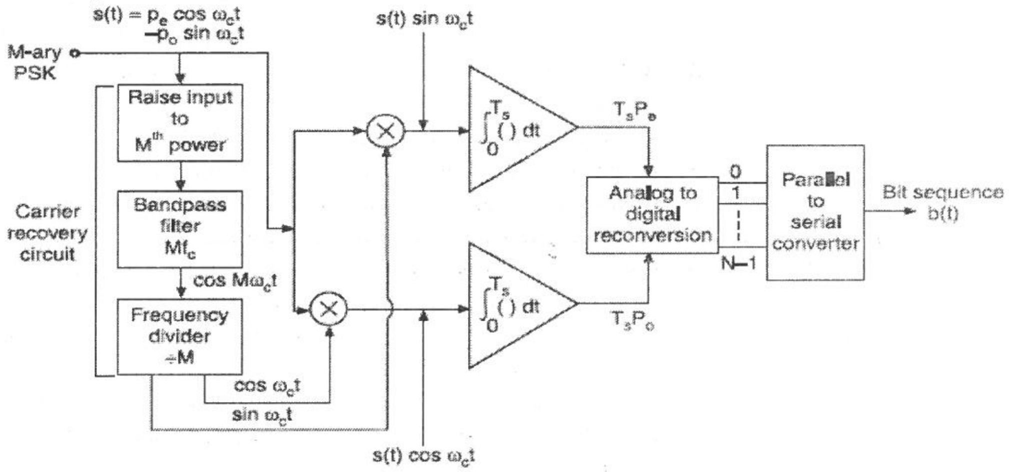


Fig. 6.43. M-ary PSK receiver

2. Working Operation

The M-ary receiver operates on the principle of synchronous demodulation which we had discussed for BPSK and QPSK. The carrier recovery system will require a device which can raise the received signal to M^{th} power. This signal is then applied to a bandpass filter, the center frequency of which has been selected to be Mf_c. At the filter output, we get a sinusoid at frequency Mf_c that is M^{th} harmonic of the carrier frequency f_c. This frequency is then divided by M to obtain the carrier at frequency f_c. The two carriers produced at the filter output are $\cos \omega_c t$ and $\sin \omega_c t$. These recovered carriers are then applied to two multipliers (balanced modulators). The other input of each multiplier block is connected to the received M-ary PSK signal. The outputs of the balanced modulators are applied to the integrators.Since, the M-ary PSK system is a non-offset or non-staggered type of system, the integrators will extend their integration over the same time interval. These integrators will work alongwith a bit synchronizer, which has not been shown in figure 6.43. The outputs of the integrators are proportional to T_sp_e and T_sp_0 respectively and they (outputs) change at the symbol rate. These outputs are then applied to an A to D converter which yields the N-bit transmitted signal. The signal is converted into b(t) by using a parallel to serial converter. Now, the operating systems with N = 4-bits and $M = 2^4 = 16$ are common. The bandwidth of such a system will be given by,

5.B)ASK,FSK,PSK:

S.No.	Parameter of comparison	Binary ASK	Binary FSK	Binary PSK
1.	Variable characteristic	Amplitude	Frequency	Phase
2.	Bandwidth (Hz) (spectral efficiency)	$2 f_b$	$4 f_b$	$2 f_b$
3.	Noise immunity	low	high	high
4.	Probability of error	high	low	low
5.	Performance in presence of noise	poor	Better than ASK	Best of three
6.	System complexity	Simple	Moderately complex	Very complex
7.	Bit rate or data rate	Suitable upto 100 bits/sec.	Suitable upto about 1200 bits/sec.	Suitable for high bit rates
8.	Demodulation method	Envelope detection	Envelope detection	Coherent detection

UNIT-III

6.A)DSSS BPSK System

Direct sequence Spred spectrum:

The spread spectrum technique discussed in the previous section is called as Direct Sequence Spread Spectrum (DS-SS) technique. This technique can be used in practice for transmission of signal over a bandpass channel (e.g., satellite channel).

For such an application, the coherent binary phase shift keying (BPSK) is used in the transmitter and receiver. The transmitter has been shown in figure 10.10

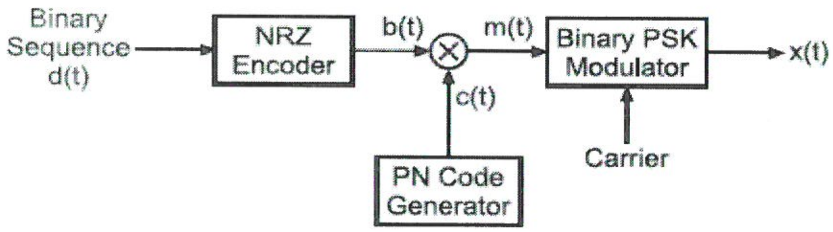


Fig. 10.10. Direct sequence spread spectrum coherent PSK transmitter

Block Diagram The DS-BPSK transmitter has been shown in figure 10.10. 2. Working Operation The binary sequence d(t) is converted into NRZ signal b(t) by using the NRZ encoder. The NRZ signal b(t) is then used to modulate the PN sequence c(t) generated by the PN code generator. The transmitter of figure 10.10 uses two stages of modulation. The first stage uses a product modulator or multiplier with b(t) and c(t) as its inputs and the second stage consists of a BPSK modulator. The modulated signal at the output of the product modulator i.e., m(t) is used to modulate the carrier for BPSK modulation. The transmitted signal x(t) is thus a direct sequence spread BPSK i.e., DS-BPSK signal

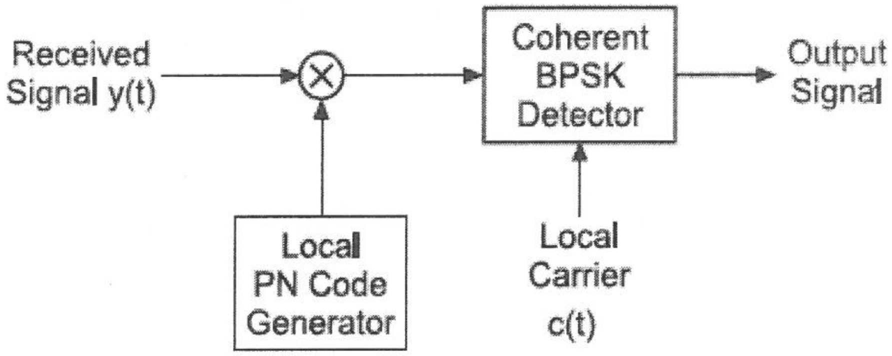


Fig. 10.12. The DS-BPSK receiver

DS-BPSK Receiver :

Block Diagram The DS-BPSK receiver has been shown in figure 10.12. 2. Working Operation The received signal y(t) and the locally generated replica of the PN-sequence are applied to a multiplier. This is the first stage of multiplication. The multiplier performs the de-spreading operation. Output of multiplier is then applied to a coherent BPSK detector with a local carrier applied to it. At the output of the coherent BPSK detector, we get back the original data sequence.

Synchronization For proper operation, the spread spectrum system requires a local PN sequence at the receiver to be synchronized with the PN sequence at transmitter. The synchronization is carried out in following two parts:

6.B)Need of Spread Spectrum Modulation

How to utilize the channel bandwidth efficiently? How to minimize the amount of transmitted power? However, the efficient utilization of bandwidth and minimizing the transmitted power are not the only problems faced by a communication system.
Some other problems encountered by it are as under: Problems encountered by a communication system :

(i) In the areas such as military communication, the information has to be secured. This means that an unauthorized user is not expected to access the information. Also he should not be allowed to interfere the communication by any means.

(ii) ii) Sometimes a hostile transmitter (say used by terrorists) can jam the transmission. To avoid this the channel should be immune to any external interference.

(iii) (iii) Even in the non-military communications an unintentional interference is caused by a user who is transmitting its information through a channel which is already being used. Remedy These problems can be successfully solved by using a technique called **Spread Spectrum Modulation**

7.A) Comparison of Slow and Fast Frequency Hopping

S.No.	Slow frequency hopping	Fast frequency hopping
1.	More than one symbols are transmitted per frequency hop.	More than one frequency hops are required to transmit one symbol
2.	Chip rate is equal to the symbol rate.	Chip rate is higher than symbol rate.
3.	Symbol rate is higher than hop rate.	Hop rate is higher than symbol rate.
4.	Same carrier frequency is used to transmit one or more symbols.	One symbol is transmitted over multiple carriers in different hops.
5.	A jammer can detect this signal if the carrier frequency in one hop is known.	A jammer cannot detect this signal because one symbol is transmitted using more than one carrier frequencies.

7.B) Applications of Spread spectrum Techniques:

(i) For combating the intentional interference (jamming): This can be used in the military applications as well as other commercial applications to avoid the intentional interference.

(ii) For reducing the unintentional interference.

(iii) For suppressing the interference due to multipath reception.

(iv) In the low probability of intercept (LPI) application.

(v) Due to large bandwidth of a spread spectrum signal, the fading (specially, the frequency selective fading) does not affect the entire spectrum. Infact, only a small portion of the complete spectrum is affected in the fading process. Therefore, the spread-spectrum signals are used in the mobile communication.

(vi) Due to the use of pseudo-noise code sequence, the spread spctrum signal can be recognised only by the authorised receiver. All other receivers consider this signal as noise. Thus, the SS communication is a secured communication.

(vii) The spread spectrum signals are used in the RADAR and other navigation systems for ranging or distance measurement. We know that wideband signals are time-limited. Hence, they can be used for the measurement of time delays very precisely.

VIII).The most important application of the spread spectrum technique is code division multiple access (CDMA). This is a multiuser communication system in which many users can access the available channel bandwidth simultaneously

UNIT-IV

8.A) ENTROPY (i.e., AVERAGE INFORMATION)

In a practical communication system, we usually transmit long sequences of symbols from an information source. Thus, we are more interested in the average information that a source produces than the information content of a single symbol.

Thus, we require to talk about the average information content of the symbols in a long message. Thus, the entropy is defined as the average information per message

It is a measure of average information content per source symbol.

- $H(s) = \Sigma (Pk) \, Log2(1/Pk)$; $k=0,1,2,\ldots (K-1)$

Properties:

1. H(S)=0 IF AND ONLY IF P0=1,
   P1=P2=--Pk-1=0

2. H(S)= Log2 k IF AND ONLY IF Pi=1/k FOR ALL I
   i=1,2,3,... (k-1)

The source entropy H(X) satisfies the following relation:

$$0 \le H(X) \le \log_2 m \qquad \text{...(8.9)}$$

where, m is the size (number of symbols) of the alphabet of source X. The lower bound corresponds to no uncertainty, which occurs when one symbol has probability $P(x_i) = 1$ while, $P(x_j) = 0$ for $j \neq i$, so X emits the same symbol x_i all the time. The upper bound corresponds to the maximum uncertainty which occurs when $P(x_i) = 1/m$ for all i, that is, when all symbols are equal likely to be emitted by X.

8.B) Mutual information:

The mutual information denoted by $I(X:Y)$ of a channel is defined by

$$I(X:Y) = H(X) - H(X|Y) \text{ b/symbol} \qquad \text{...(8.30)}$$

Since H(X) represents the uncertainty about the channel input before the channel output is observed and H(X|Y) represents the uncertainty about the channel input after the channel output is observed, the mutual information I(X : Y) represents the uncertainty about the channel input that is resolved by observing the channel output.

$I(x;y) = H(x)-H(x|y)$
x – Channel input
y – Channel output
H(x)– Entropy
H(x|y)– Conditional Entropy

The amount of uncertainty remaining about the channel input after observing the channel output, is called as Conditional Entropy

Properties of Mutual Information I(X; Y)

(i) $I(X:Y) = I(Y:X)$
(ii) $I(X:Y) \geq 0$
(iii) $I(X:Y) = H(Y) - H(Y|X)$
(iv) $I(X:Y) = H(X) + H(Y) - H(X,Y)$

9.A) **LEMPEL-ZIV CODING :**
The Lempel-Ziv algorithm is a variable-to-fixed length code.
Basically, there are two versions of the algorithm presented in the literature: the theoretical version and the practical version. Theoretically, both versions perform essentially the same. However, the proof of the asymptotic optimality of the theoretical version is easier. In ractice, the practical version is easier to implement and is slightly more efficient.
Here, the **basic idea is to parse the input sequence into non- overlapping blocks of different lengths while constructing a dictionary of blocks seen thus far**

Binary Sequence:

0 1 | 1 0 | 0 0 | 1 0 0 | 1 1 | 1 1 1 | 0 1 0

Use the Lempel-Ziv algorithm to encode this sequence assume that the binary symbol 0 and 1 are already in the code book.

Numerical Position	1	2	3	4	5	6	7	8	9
Subsequence	0	1	01	10	00	100	11	111	010
Numerical representation			12	21	11	41	22	72	31
Binary encoding of numerical representation			0011	0100	0010	1000	0101	1111	0110

9.B) CHANNEL CODING THEOREM

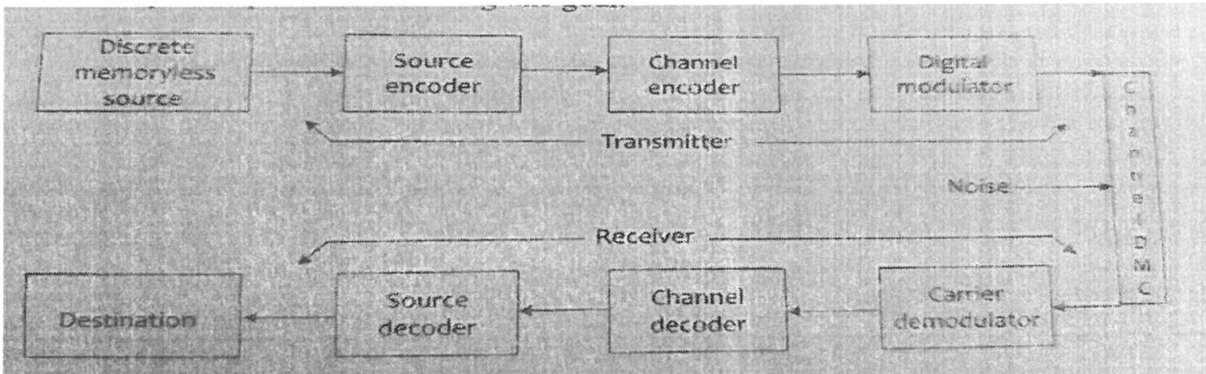


Fig. 6.18 *Block diagram of a digital communication system*

Noise causes errors

- probability of error < 10 is required,to achieve this channel coding isneeded
- encoder: in channel coding incomingdata sequence is mapped into channelinput sequence
- block codes: message sequence is divided into blocks of 'k' bits long eack 'k' bit block is mapped into 'n' bit block (n > k)

No of redundant bits = (n-k) are parity check bits

code rate (r)= k/n

- channel capacity= c bits per use of channel
- channel capacity per unit time=c/tc
- at the source encoder output each bit ocuppies tsseconds,and the output of the Channel encoder each bit ocupies only tc seconds.
- ts=(k/n)tc

Given a discrete memoryless source with an entropy of $H(S)$ bits per symbol emitting symbols at the rate of $(1/T_s)$ symbols per second, and given a discrete memoryless channel with a capacity of C_s bits per symbol and through which the symbols are transmitted at the rate of $(1/T_c)$ symbols per second, it is possible to construct a channel code which would make it possible to transmit the source symbols through the channel and be reconstructed with arbitrarily small probability of error if and only if

$$\frac{H(S)}{T_s} \leq \frac{C_s}{T_c} \qquad ...(6.63)$$

UNIT-V

10 A)Cyclic codes:

Cyclic codes are also linear block codes.Infact many important linear block codes are cyclic codes.

The flip-flops (F/F) of figure 9.34 are used to construct a shift register. Operation of all these flip-flops is governed by an external clock, which is not shown in figure 9.34. The flip-flop contents will get shifted in the direction of the arrow corresponding to each clock pulse. The feedback switch is closed and the output switch is connected to the message input. All the flip-flops are initialized to zero state. First k message bits are shifted to the transmitter and also shifted into the shift register. After shifting the k message bits the shift register will contain the (n – k) parity (or check) bits. Therefore, after shifting the k message bits, the feedback switch is open circuited and the output switch is thrown to parity bit position. Now with every shift, the parity bits are transmitted over the channel. Thus, this encoder generates the code words in the format shown in figure 9.33.

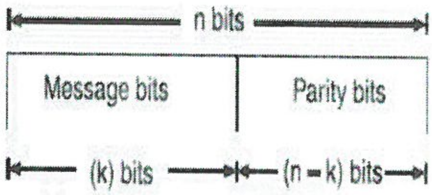


Fig. 9.33. Format of the code word

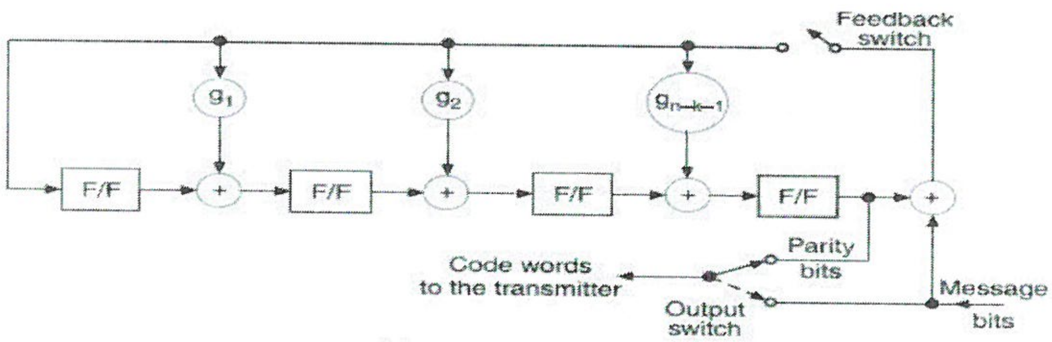


Fig. 9.34. Encoder for an (n, k) cyclic code

9.14.9. Syndrome Decoding for the Cyclic Codes

When a code word X is transmitted over a noisy channel, errors are likely to get introduced into it. Thus, the received code word Y is different from X. For a linear block code, the first step in decoding is to calculate the syndrome for the received code word. If the syndrome is zero, then there are no transmission errors in the received code word. But, if the syndrome is non-zero, then received code word contains transmission errors which requires correction. In case of a cyclic code, in the systematic form, the syndrome can be calculated easily. Let the received code word be a polynomial of degree (n – 1) or less. Let it be given by,

$$y(p) = y_0 + y_1\,(p) + \ldots + y_{n-1}\,p^{n-1} \qquad ..(9.45)$$

Now, we divide y(p) by the generator polynomial G(p). Let Q(p) represent the quotient polynomial and R(p) be the remainder polynomial.

Therefore, $$\frac{y(p)}{G(p)} = Q(p) + \frac{R(p)}{G(p)} \quad ...(9.46)$$

$$y(p) = Q(p) \cdot G(p) + R(p) \quad ...(9.47)$$

The remainder R(p) is a polynomial with degree (n − k − 1) or less. It is called as the syndrome polynomial. The coefficients of the syndrome polynomial will make up the (n − k) by 1 syndrome S.

Thus, syndrome polynomial S(p) = Remainder of $\frac{y(p)}{G(p)}$.

When the syndrome polynomial S(p) is non-zero, the presence of errors in the received code word is detected.

10.B)block codes:

Block Codes :

The generation of block codesTo generate an (n, k) block code, the channel encoder accepts the information in successive k-bit blocks. At the end of each such lock (of k message bits), it adds (n – k) parity bits .

for example d min value= 3 then as follow how to detect and correct error in block codes

The code vector represented by equation (9.4) can be mathematically represented as:

$$X = [M : C]$$

where M = k-message vectors

and C = (n − k). parity vectors

We know that the minimum distance d_{min} is equal to the minimum weight of any non-zero code vector. Looking at Table 9.5, we obtain

$$d_{min} = 3. \text{ Ans.}$$

Number of errors that can be detected is given by

$$d_{min} \geq s + 1$$

$$3 \geq s + 1$$

or $$s \leq 2$$

Hence, at the most two errors can be detected.

Number of errors that can be corrected is given by

$$d_{min} \geq 2t + 1$$

or $$3 \geq 2t + 1$$

or $$t \leq 1$$

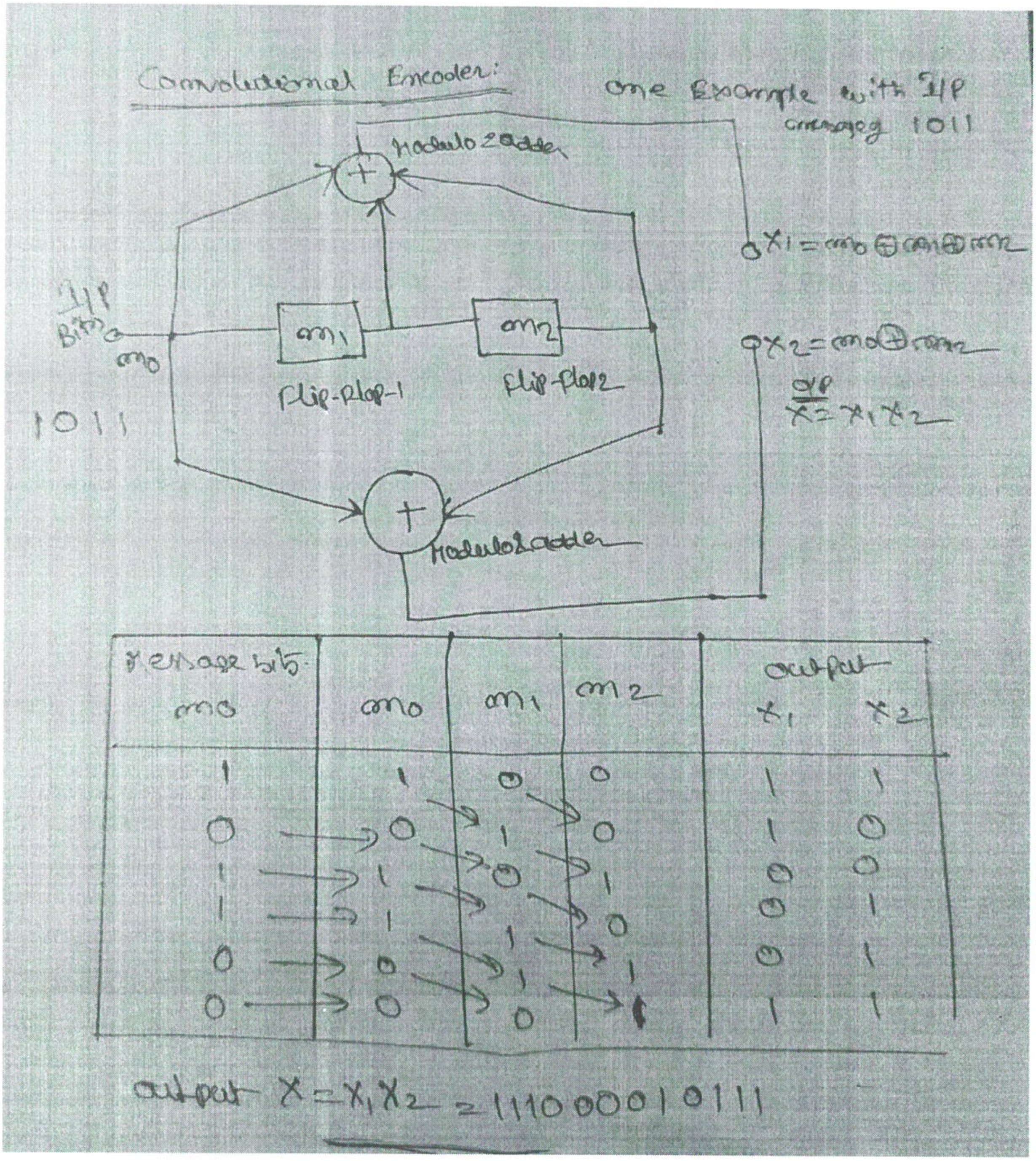
This means that at the most one error can be corrected.

Thus, for the (7, 4) linear block code, at the most two errors can be detected and at the most only one error can be corrected.

11.A) Convolutional Encoder:

Convolutional Code In the convolutional codes, the block of n bits generated by the encoder in a time slot depends not only on the k message bits within that time slot, but also on the preceeding 'L' blocks of the message bits (L > 1). Generally, the values of k and n will be small The fundamental hardware unit for the convolutional encoding is a taped shift register with (L + 1) stages,. The message bits enter one by one into the tapped shift register, which are then **combined by mod-2 addition** to form the encoded bit x.

Consider one example:



11.B) Viterbi Algorithm:

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 there are 2N 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. Before we explain the Viterbi algorithm for the decoding of convolutional codes, it is necessary to define certain important terms

Metric
It is defined as the Hamming distance of each branch of each surviving path from the corresponding branch of Y (received signal). The metric is defined by assuming that 0's and 1's have the same transmission error probability

Surviving Path
The surviving path is defined as the path of the decoded signal with minimum metric.
(i) Let the received signal be represented by Y. The Viterbi decoder assigns to each branch of each surviving path of a metric.
(ii) By summing the branch matrices we get the path metric.