UNIT III DIGITAL MODULATION AND TRANSMISSION

Memoryless modulation techniques

Modulation is defined as the process by which some characteristics of a carrier is varied in accordance with a modulating wave. In digital communications, the modulating wave consists of binary data or an M-ary encoded version of it and the carrier is sinusoidal wave.

Different Shift keying methods that are used in digital modulation techniques are

- **Amplitude shift keying [ASK]**
- **Frequency shift keying [FSK]**
- **Phase shift keying [ASK]**

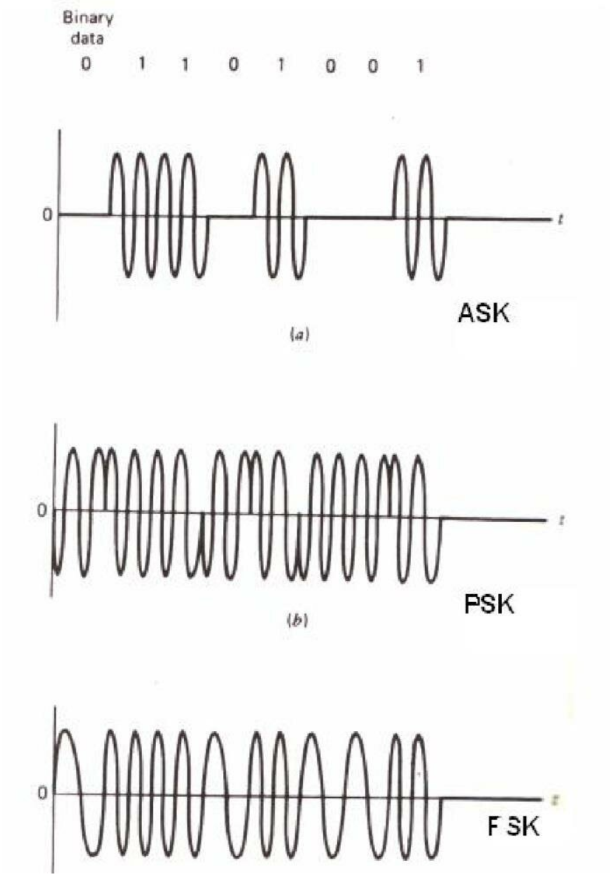

Fig 3.1 Digital Modulation Schemes

Amplitude Shift Keying (ASK) Modulation:

Amplitude shift keying (ASK) is a simple and elementary form of digital modulation in which the amplitude of a carrier sinusoid is modified in a discrete manner depending on the value of a modulating symbol. Let a group of 'm' bits make one symbol. Hence one can design $M = 2^m$ different baseband signals, $d_m(t)$, $0 \le m \le M$ and $0 \le t \le T$. When one of these symbols modulates the carrier, say, $c(t) = \cos\omega_c t$, the modulated waveform is:

$$s_m(t) = d_m(t).\cos\omega_c t \qquad 5.23.1$$

This is a narrowband modulation scheme and we assume that a large number of carrier cycles are sent within a symbol interval, i.e. $\frac{T}{\left(2\pi/\varpi_c\right)}$ is a large integer. It is obvious that the information is embedded only in the peak amplitude of the modulated signal. So, this is a kind of digital amplitude modulation technique. From another angle, one can describe this scheme of modulation as a one-dimensional modulation scheme where one basis function $\varphi_1(t) = \sqrt{\frac{2}{T}}.\cos\varpi_c t$, defined over $0 \le t \le T$ and having unit energy is used and all the baseband signals are linearly dependent.

Implementation of binary ASK:

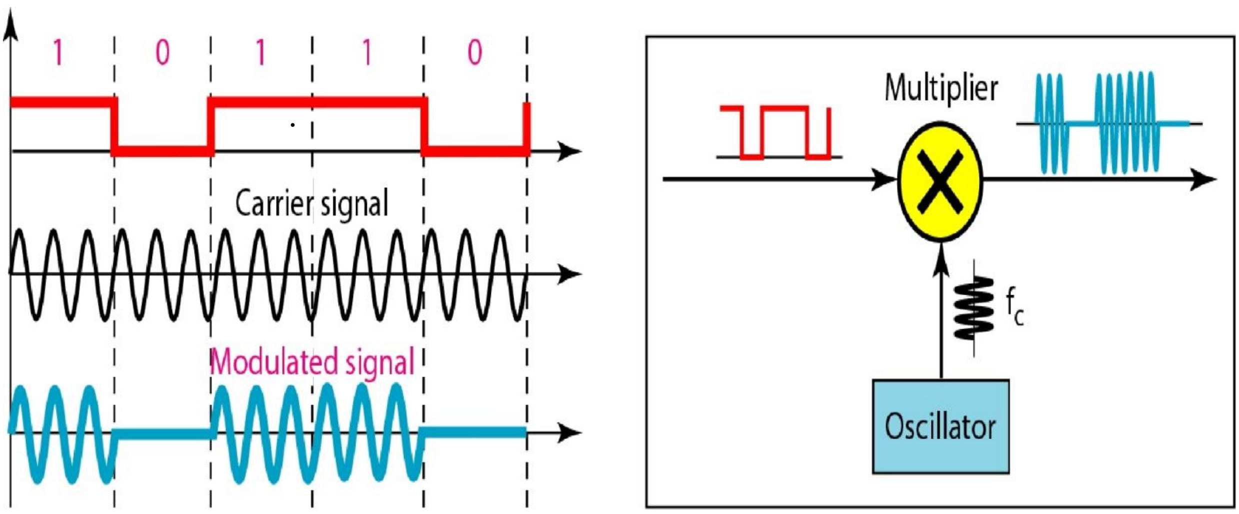

Fig 3.2 - Implementation of BASK

BINARY AMPLITUDE SHIFT KEYING, BANDWIDTH:

- $d \ge 0$-related to the condition of the line

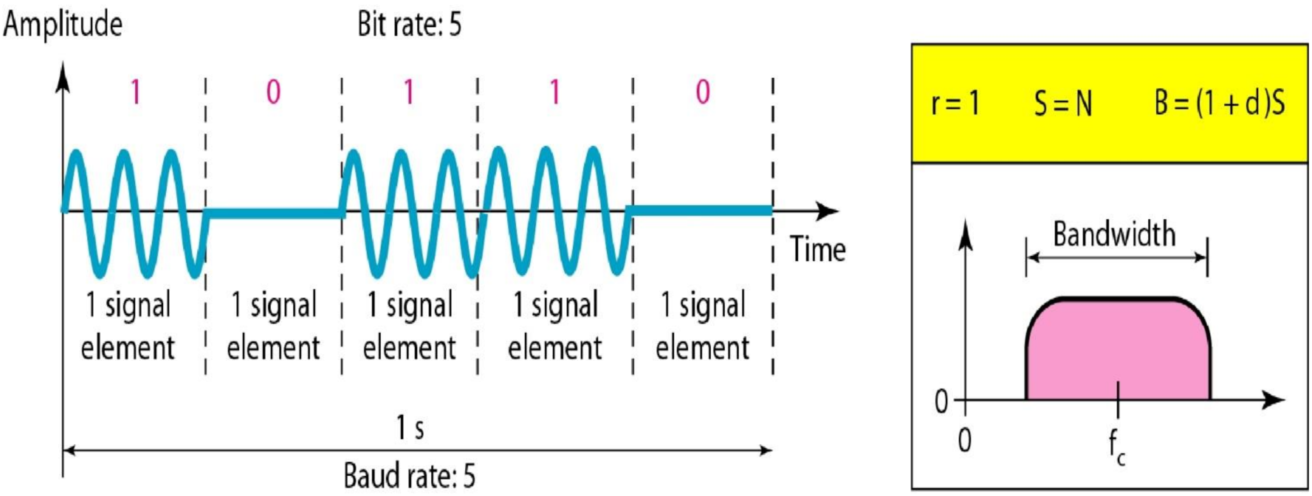

Fig 3.3. ASK waveform and Bandwidth

B = (1+d) x S = (1+d) x N x 1/r

The BASK system has one dimensional signal space with two messages (N=1, M=2)

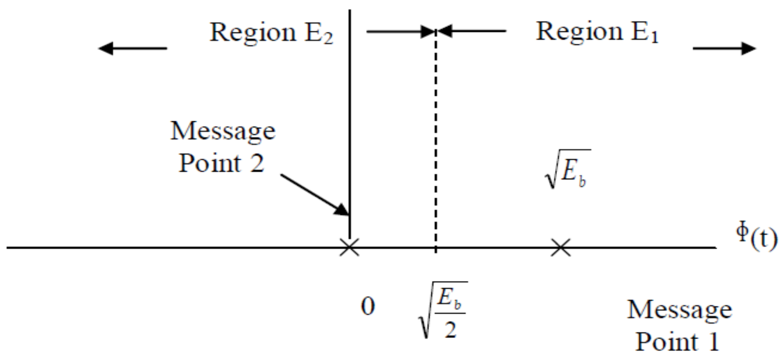

Fig 3.4 Signal Space representation of BASK signal

In transmitter the binary data sequence is given to an on-off encoder. Which gives an output $\sqrt{E_b}$ volts for symbol 1 and 0 volt for symbol 0. The resulting binary wave [in unipolar form] and sinusoidal carrier are applied to a product modulator. The desired BASK wave is obtained at the modulator output.

In demodulator, the received noisy BASK signal x(t) is apply to correlator with coherent reference signal. The correlator output x is pared with threshold λ.

- If x > λ the receiver decides in favour of symbol 1.
- If x < λ the receiver decides in favour of symbol 0.

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Frequency Shift Keying Modulation

Frequency Shift Keying (FSK) modulation is a popular form of digital modulation used in low-cost applications for transmitting data at moderate or low rate over wired as well as wireless channels. In general, an M-ary FSK modulation scheme is a power efficient modulation scheme and several forms of M-ary FSK modulation are becoming popular for spread spectrum communications and other wireless applications. In this lesson, our discussion will be limited to binary frequency shift keying (BFSK).

Two carrier frequencies are used for binary frequency shift keying modulation. One frequency is called the 'mark' frequency (f_2) and the other as the space frequency (f_1). By convention, the 'mark' frequency indicates the higher of the two carriers used. If T_b indicates the duration of one information bit, the two time-limited signals can be expressed as :

$$s_i(t) = \begin{cases} \sqrt{\dfrac{2E_b}{T_b}} \cos 2\pi f_i t, & 0 \le t \le T_b,\ i = 1,2 \\ 0, & \text{elsewhere.} \end{cases} \qquad 5.23.2$$

The binary scheme uses two carriers and for special relationship between the two frequencies one can also define two orthonormal basis functions as shown below.

$$\varphi_l(t) = \sqrt{\frac{2}{T_b}} \cos 2\pi f_i t \quad ; \quad 0 \le t \le T_b \ \text{ and } \ j = 1,2 \qquad 5.23.3$$

<u>Generation and</u> <u>Detection:-</u>

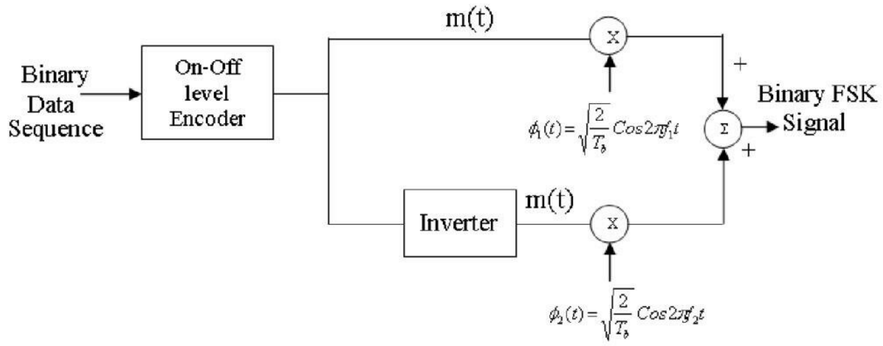

Fig 3.5 Generation of FSK

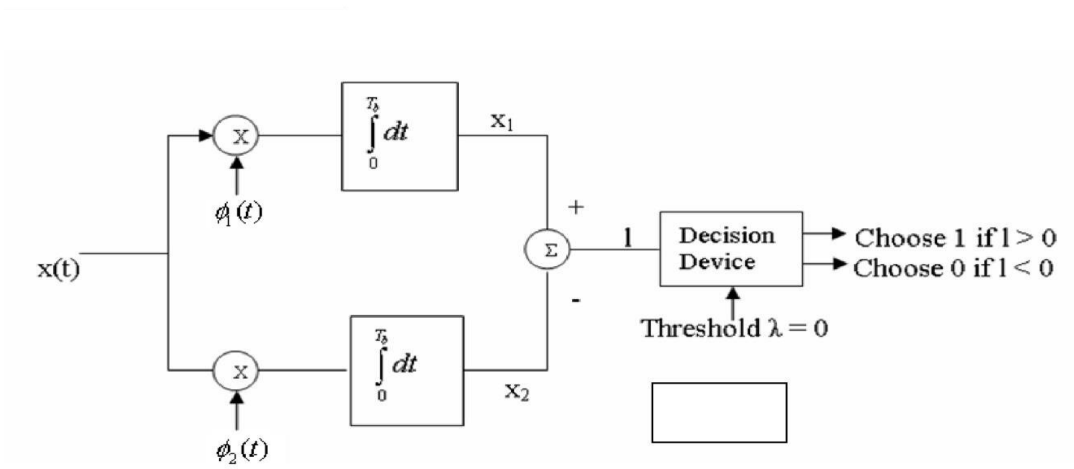

Fig 3.6 Detection of FSK

A binary FSK Transmitter is as shown, the incoming binary data sequence is applied to on-off level encoder. The output of encoder is $\sqrt{E_b}$ volts for symbol 1 and 0 volts for symbol „0‟. When we have symbol 1 the upper channel is switched on with oscillator frequency f1, for symbol „0‟, because of inverter the lower channel is switched on with oscillator frequency f2. These two frequencies are combined using an adder circuit and then transmitted. The transmitted signal is nothing but required BFSK signal.

The detector consists of two correlators. The incoming noisy BFSK signal x(t) is common to both correlator. The Coherent reference signal $\phi 1(t)$ & $\phi 2(t)$ are supplied to upper and lower correlators respectively.

The correlator outputs are then subtracted one from the other and resulting a random vector „l‟ (l=x1 - x2). The output "l" is compared with threshold of zero volts.

If l > 0, the receiver decides in favour of symbol 1.
IF l < 0, the receiver decides in favour of symbol 0.

FSK Bandwidth:

- Limiting factor: Physical capabilities of the carrier
- Not susceptible to noise as much as ASK
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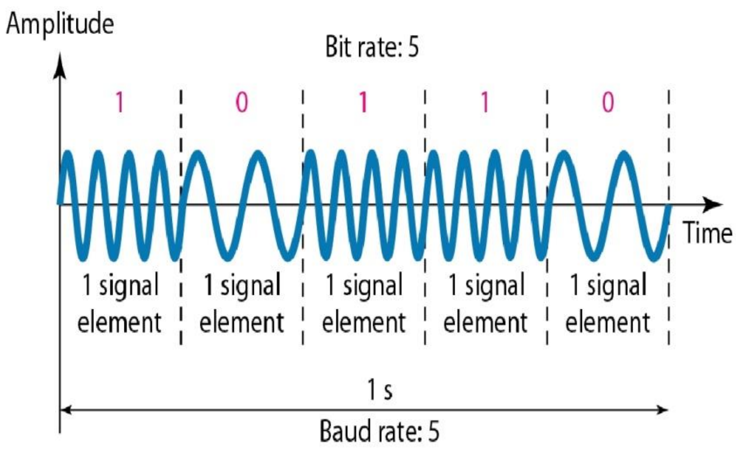


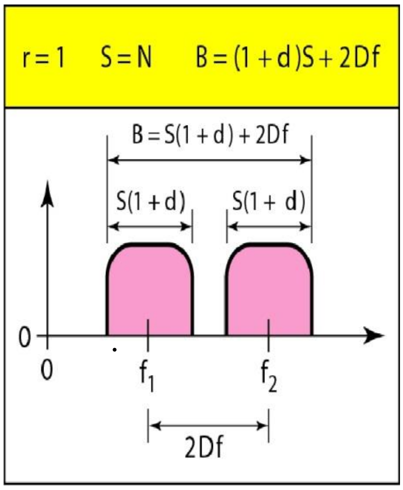


Fig 3.7 FSK Waveform and bandwidth

- Applications
  - On voice-grade lines, used up to 1200bps
  - Used for high-frequency (3 to 30 MHz) radio transmission
  - used at higher frequencies on LANs that use coaxial cable.

Therefore Binary FSK system has 2 dimensional signal space with two messages $S_1(t)$ and $S_2(t)$, [N=2 , m=2] they are represented,

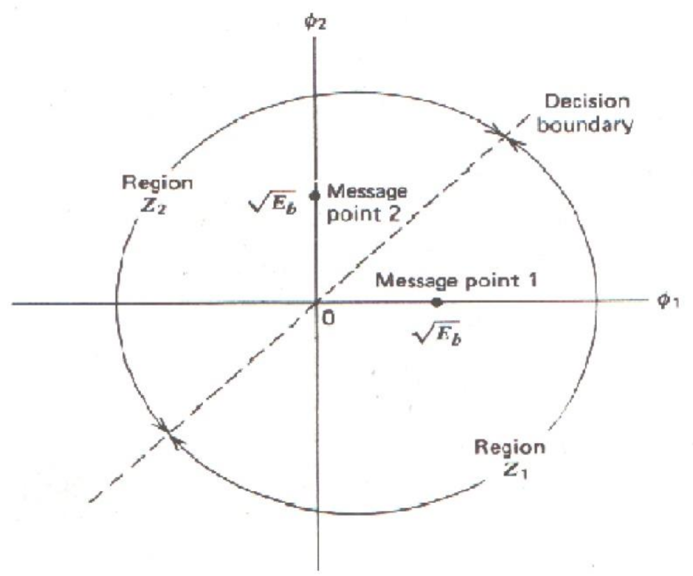
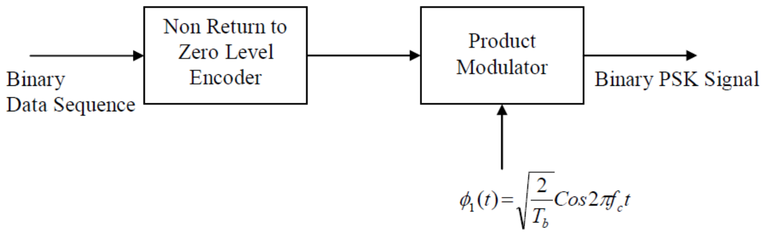


Fig.3.8 Signal Space diagram of Coherent binary FSK system.

PHASE SHIFT KEYING(PSK):



Fig(a) Block diagram of BPSK transmitter

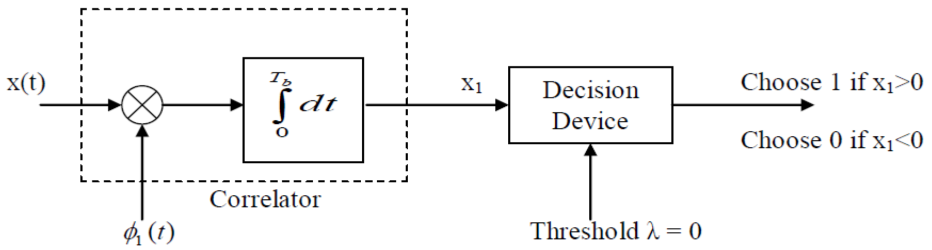


Fig (b) Coherent binary PSK receiver

Fig 3.9 (a) & (b) BPSK Transmitter and Receiver

In a Coherent binary PSK system the pair of signals S1(t) and S2(t) are used to represent binary symbol „1" and „0" respectively.

To generate a binary PSK signal we have to represent the input binary sequence in polar form with symbol „1" and „0" represented by constant amplitude levels.

To detect the original binary sequence of 1"s and 0"s we apply the noisy PSK signal x(t) to a Correlator, which is also supplied with a locally generated coherent reference signal.

The correlator output x1 is compared with a threshold of zero volt.

If x1 > 0, the receiver decides in favour of symbol 1.

If x1 < 0, the receiver decides in favour of symbol 0.

The signal space representation is as shown in fig (N=1 & M=2)

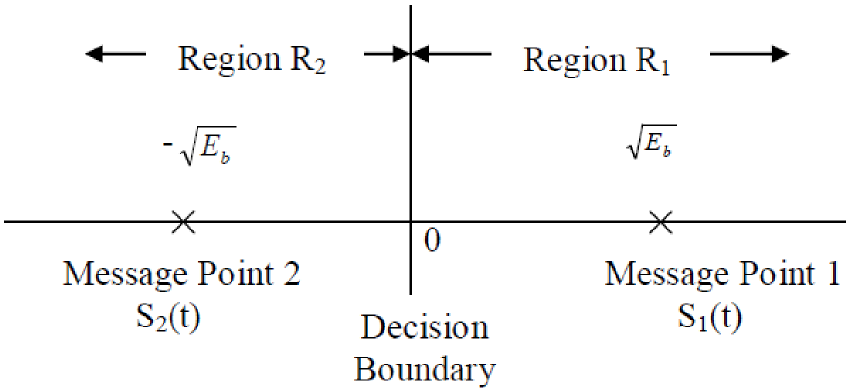


Fig. Signal Space Representation of BPSK

Fig 3.10 Signal space Diagram of BPSK

QUADRATURE PHASE – SHIFT KEYING(QPSK)

In a sense, QPSK is an expanded version from binary PSK where in a symbol consists of two bits and two orthonormal basis functions are used. A group of two bits is often called a 'dibit'. So, four dibits are possible. Each symbol carries same energy.

Let, E: Energy per Symbol and T: Symbol Duration = 2.* Tb, where Tb: duration of 1 bit.

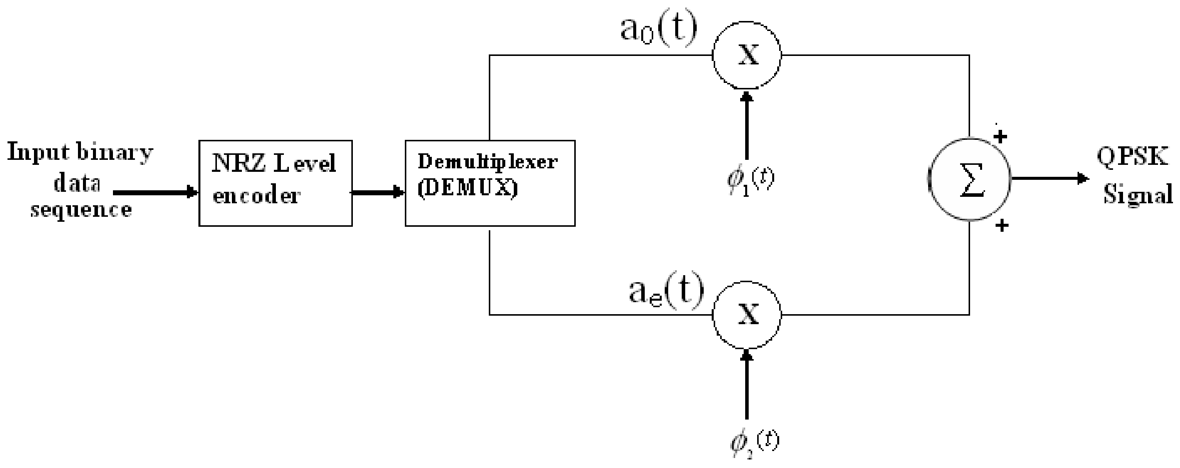


Fig. (a) QPSK Transmitter

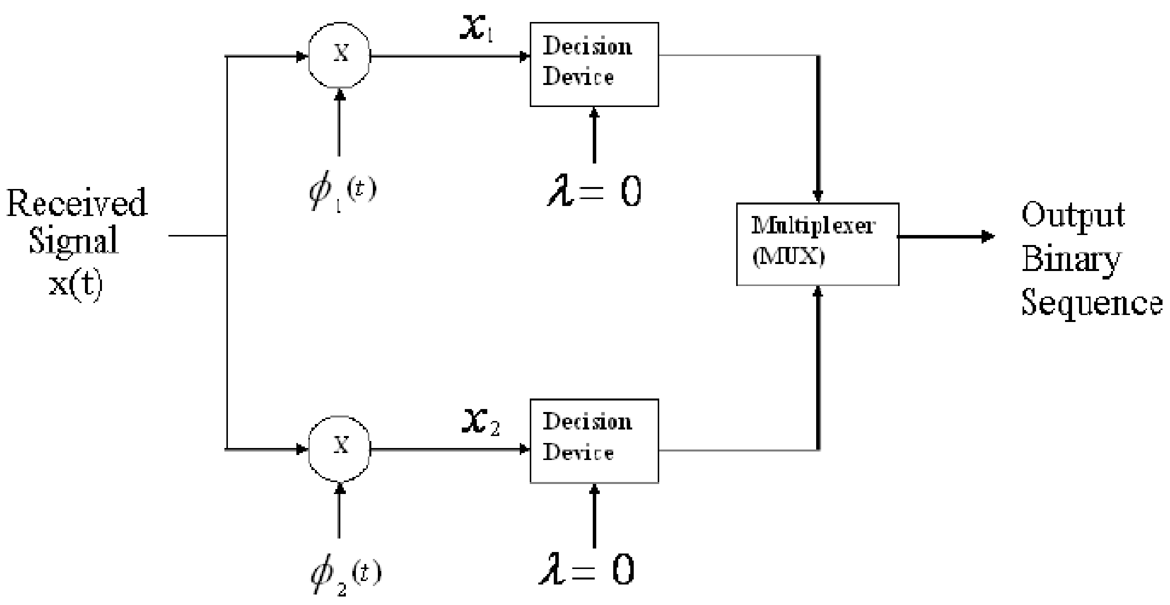


Fig. (b) QPSK Receiver

Fig 3.11 QPSK Transmitter and Receiver

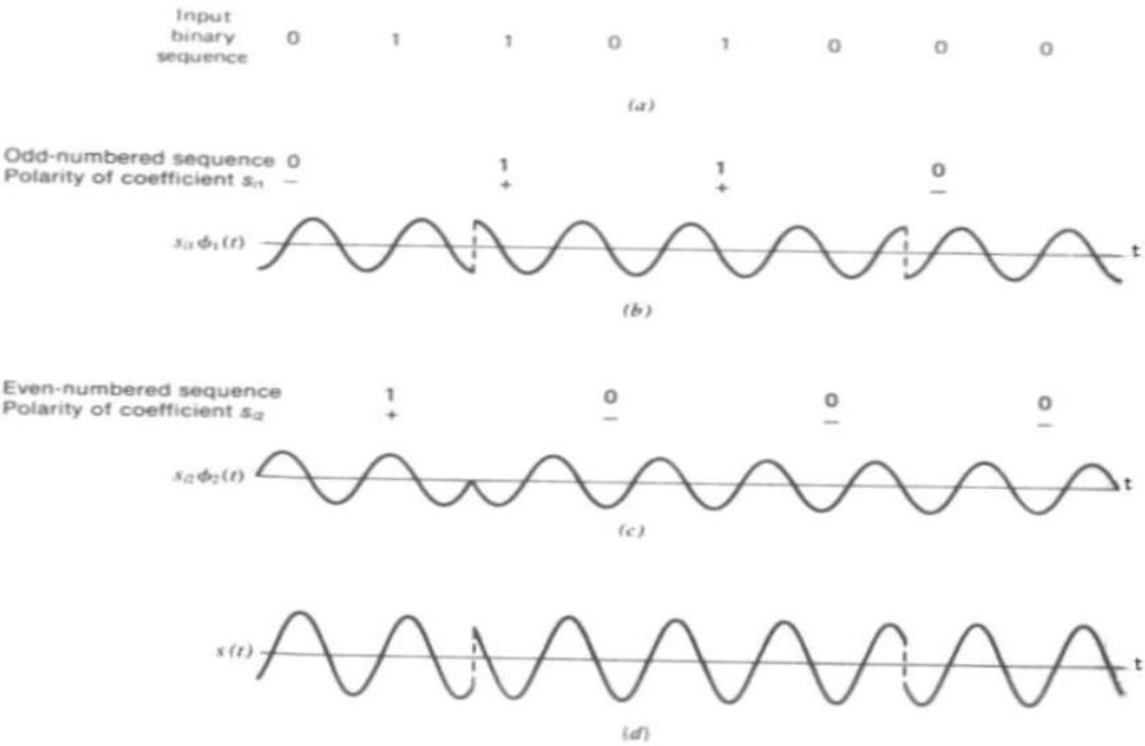


Fig. QPSK Waveform

Fig 3.12 QPSK Waveform

In QPSK system the information carried by the transmitted signal is contained in the phase.

QPSK Receiver:-

The QPSK receiver consists of a pair of correlators with a common input and supplied with a locally generated pair of coherent reference signals $\phi 1(t)$ & $\phi 2(t)$as shown in fig(b).The correlator outputs x1 and x2 produced in response to the received signal x(t) are each compared with a threshold value of zero.

The in-phase channel output :

If x1 > 0 a decision is made in favour of symbol 1

x1 < 0 a decision is made in favour of symbol 0

Similarly quadrature channel output:

If x2 >0 a decision is made in favour of symbol 1

And, x2 <0 a decision is made in favour of symbol 0

Finally these two binary sequences at the in phase and quadrature channel outputs are combined in a multiplexer (Parallel to Serial) to reproduce the original binary sequence.

Input	Dibit		Phase of QPSK	Coordinates of signal points		
	(b_0)	(b_e)		s_{i1}	s_{i2}	i
$\overline{s_1}$	1	0	$\pi/4$	$+\sqrt{E/2}$	$-\sqrt{E/2}$	1
$\overline{s_2}$	0	0	$3\pi/4$	$-\sqrt{E/2}$	$-\sqrt{E/2}$	2
$\overline{s_3}$	0	1	$5\pi/4$	$-\sqrt{E/2}$	$+\sqrt{E/2}$	3
$\overline{s_4}$	1	1	$7\pi/4$	$+\sqrt{E/2}$	$+\sqrt{E/2}$	4

Probability of error:-

A QPSK system is in fact equivalent to two coherent binary PSK systems working in parallel and using carriers that are in-phase and quadrature.

The in-phase channel output x1 and the Q-channel output x2 may be viewed as the individual outputs of the two coherent binary PSK systems. Thus the two binary PSK systems may be characterized as follows.

The signal energy per bit $\sqrt{E/2}$

The noise spectral density is $N_0/2$

The bit errors in the I-channel and Q-channel of the QPSK system are statistically independent. The I-channel makes a decision on one of the two bits constituting a symbol (di bit) of the QPSK signal and the Q-channel takes care of the other bit.

QAM(Quadrature Amplitude Modulation):

QAM is a combination of ASK and PSK. Two different signals sent simultaneously on the same carrier frequency
ie,M=4, 16, 32, 64, 128, 256

As an example of QAM, 12 different phases are combined with two different amplitudes. Since only 4 phase angles have 2 different amplitudes, there are a total of 16 combinations.

With 16 signal combinations, each baud equals 4 bits of information (2 ^ 4 = 16). Combine ASK and PSK such that each signal corresponds to multiple bits. More phases than amplitudes. Minimum bandwidth requirement same as ASK

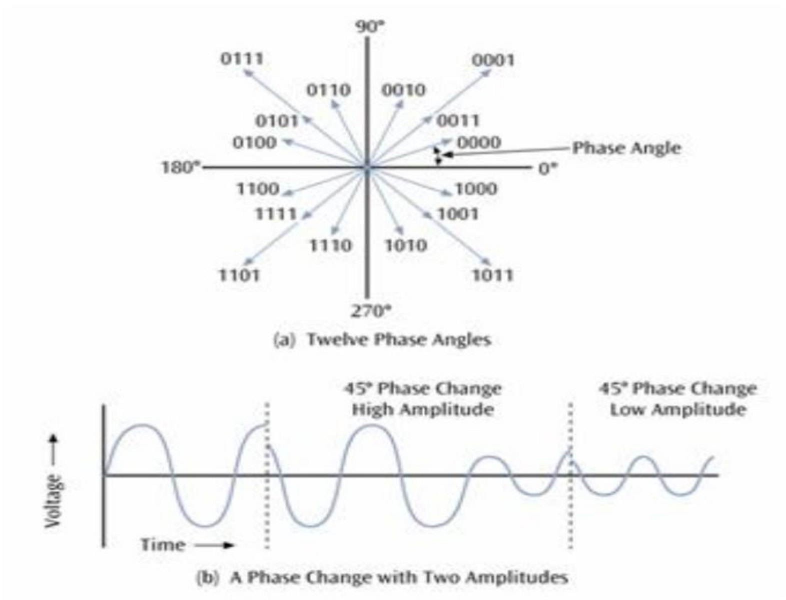


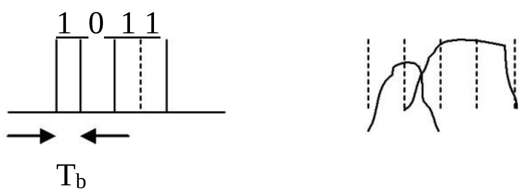
Fig 3.13 QAM Signal space and Waveform

Inter symbol Interference

Generally, digital data is represented by electrical pulse; communication channel is always band limited. Such a channel disperses or spreads a pulse carrying digitized samples passing through it. When the channel bandwidth is greater than bandwidth of pulse, spreading of pulse is very less. But when channel bandwidth is close to signal bandwidth, i.e. if we transmit digital data which demands more bandwidth which exceeds channel bandwidth, spreading will occur and cause signal pulses to overlap. This overlapping is called **Inter Symbol Interference**.

In short it is called ISI. Similar to interference caused by other sources, ISI causes degradations of signal if left uncontrolled. This problem of ISI exists strongly in Telephone channels like coaxial cables and optical fibers.

In this chapter main objective is to study the effect of ISI, when digital data is transmitted through band limited channel and solution is to overcome the degradation of waveform by properly shaping pulse.



Transmitted Waveform Pulse Dispersion

Fig 3.14. Transmitted and Received Pulse wave

The effect of sequence of pulses transmitted through channel is shown in fig. The

Spreading of pulse is greater than symbol duration; as a result adjacent pulses interfere. i.e. pulses get completely smeared, tail of smeared pulse enter into adjacent symbol intervals making it difficult to decide actual transmitted pulse.

First let us have look at different formats of transmitting digital data. In base band transmission best way is to map digits or symbols into pulse waveform. This waveform is generally termed as **Line codes**.

EYE PATTERN

The quality of digital transmission systems are evaluated using the bit error rate. Degradation of quality occurs in each process modulation, transmission, and detection. The eye pattern is experimental method that contains all the information concerning the degradation of quality. Therefore, careful analysis of the eye pattern is important in analyzing the degradation mechanism.

- Eye patterns can be observed using an oscilloscope. The received wave is applied to the vertical deflection plates of an oscilloscope and the sawtooth wave at a rate equal to transmitted symbol rate is applied to the horizontal deflection plates, resulting display is eye pattern as it resembles human eye.
- The interior region of eye pattern is called eye opening

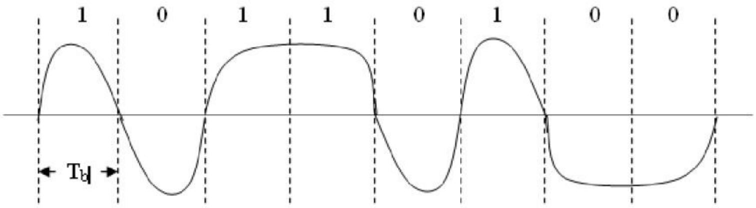


Fig 3.15 Received Waveform

We get superposition of successive symbol intervals to produce eye pattern as shown below.

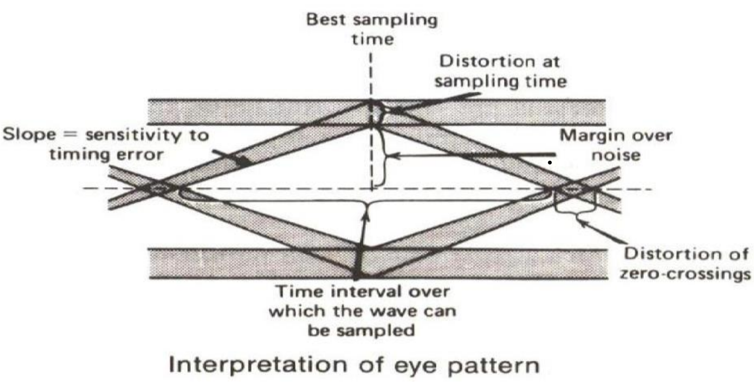


Fig 3.16 Eye Pattern

- The width of the eye opening defines the time interval over which the received wave can be sampled without error from ISI
- The optimum sampling time corresponds to the maximum eye opening
- The height of the eye opening at a specified sampling time is a measure of the margin over channel noise.
- The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.

Any non linear transmission distortion would reveal itself in an asymmetric or squinted eye. When the effected of ISI is excessive, traces from the upper portion of the eye pattern cross traces from lower portion with the result that the eye is completely closed.

Example of eye pattern:

Binary-PAM Perfect channel (no noise and no ISI)

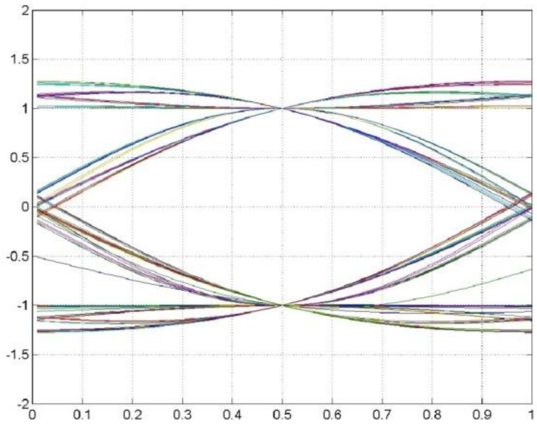


Fig 3.17 Eye Pattern Example

An equalizer is a filter that compensates for the dispersion effects of a channel.
Adaptive equalizer can adjust its coefficients continuously during the transmission of data.

Pre channel equalization:

- requires feed back channel
- causes burden on transmission.

Post channel equalization

Achieved prior to data transmission by training the filter with the guidance of a training sequence transmitted through the channel so as to adjust the filter parameters to optimum values.

Adaptive equalization – It consists of tapped delay line filter with set of delay elements, set of adjustable multipliers connected to the delay line taps and a summer for adding multiplier outputs.

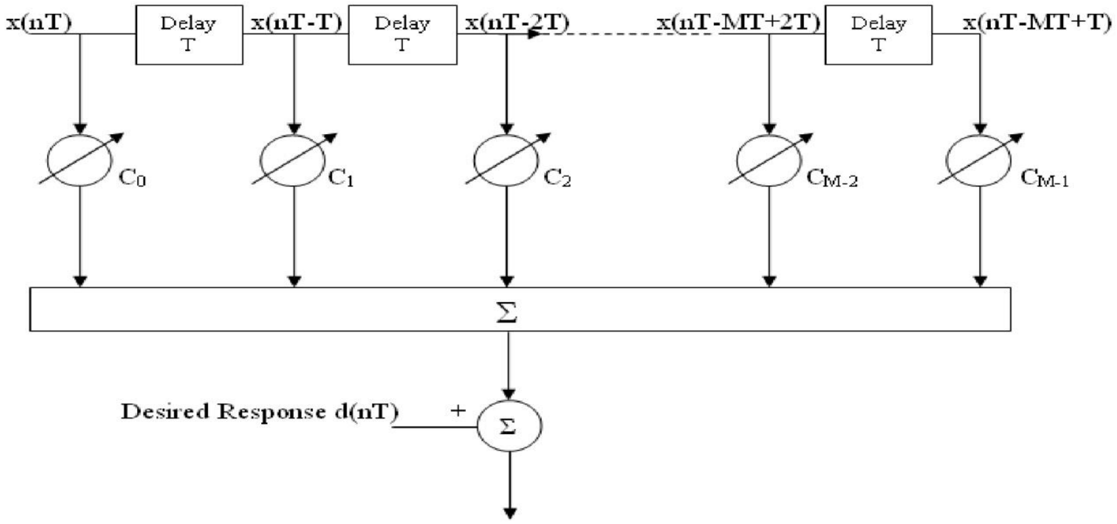


Fig 3.18 Adaptive Equalizer

The output of the Adaptive equalizer is given by

$$Y(nt)=\sum C_i \; x(nT-iT)$$

C_i is weight of the i^{th} tap.
Total numbers of taps are M.
Tap spacing is equal to symbol duration T of transmitted signal

In a conventional FIR filter the tap weights are constant and particular designed response is obtained. In the adaptive equaliser the **C** 's are variable and are adjusted by an algorithm

Two modes of operation

1. Training mode
2. Decision directed mode

Mechanism of adaptation

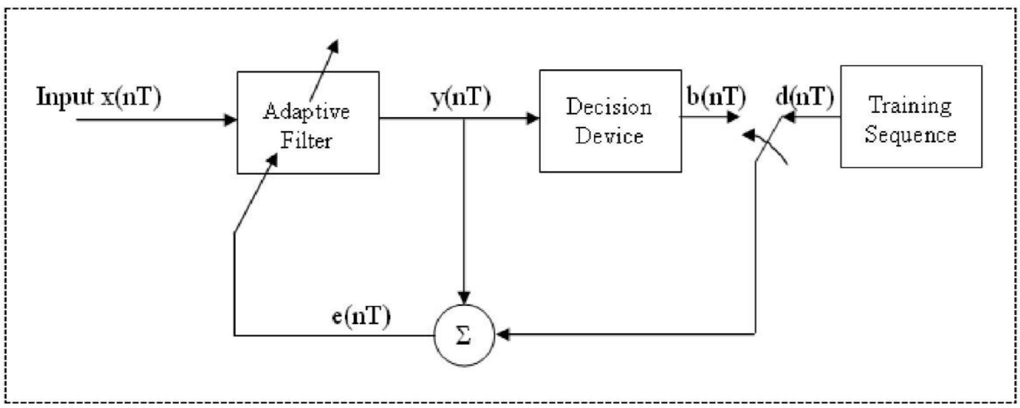


Fig 3.19 Mechanism of adaptation

Training mode

A known sequence d(nT) is transmitted and synchronized version of it is generated in the receiver applied to adaptive equalizer. This training sequence has maximal length PN Sequence, because it has large average power and large SNR, resulting response sequence (Impulse) is observed by measuring the filter outputs at the sampling instants.

The difference between resulting response y(nT) and desired response d(nT)is error signal which is used to estimate the direction in which the coefficients of filter are to be optimized using algorithms.

RECEIVING FILTER:

Correlative receiver

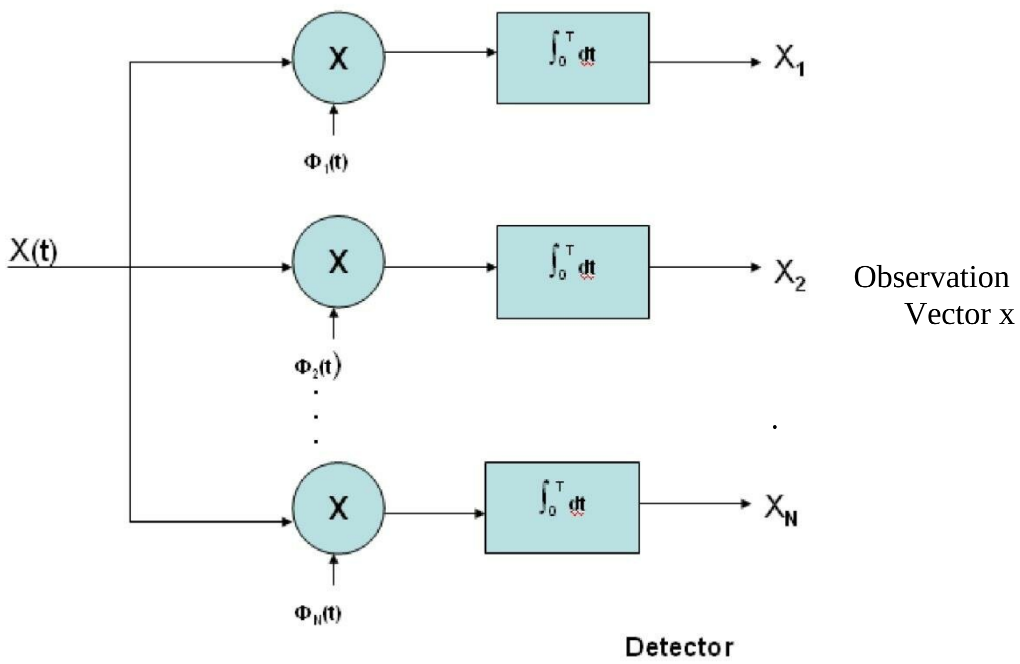


Fig 3.20 Correlative Receiver

For an AWGN channel and for the case when the transmitted signals are equally likely, the optimum receiver consists of two subsystems

1) .Receiver consists of a bank of M product-integrator or correlators $\Phi_1(t)$,$\Phi_2(t)$ …….$\Phi_M(t)$ orthonormal function
The bank of correlator operate on the received signal x(t) to produce observation vector x

Implemented in the form of maximum likelihood detector that operates on observation vector **x** to produce an estimate of the transmitted symbol m_i i = 1 to M, in a way that would minimize the average probability of symbol error.

The N elements of the observation vector **x** are first multiplied by the corresponding N elements of each of the M signal vectors $s_1, s_2 \ldots s_M$, and the resulting products are successively summed in accumulator to form the corresponding set of

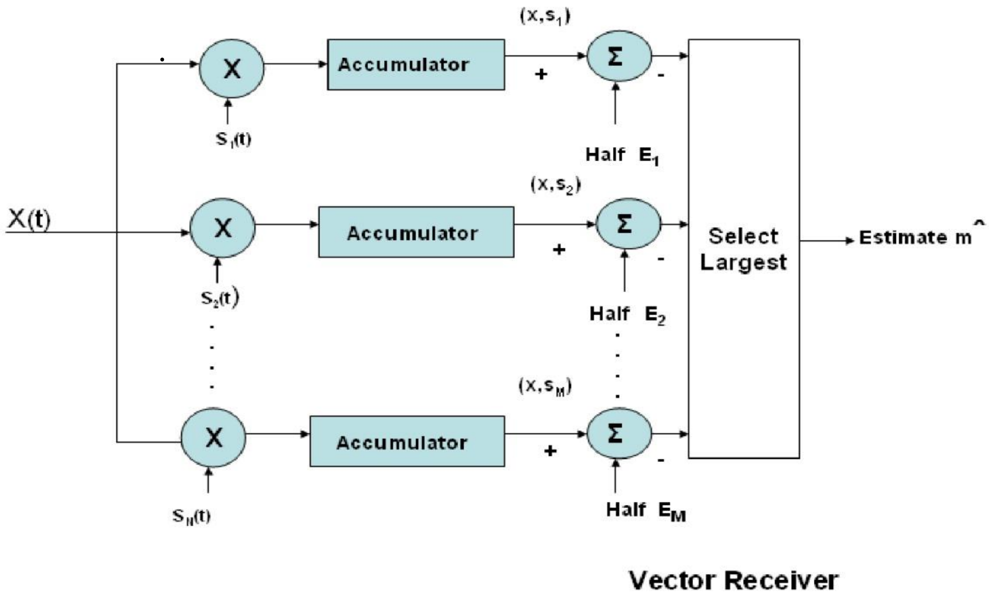


Fig 3.21 Vector Receiver

Inner products $\{(x, s_k)\}$ k= 1, 2 ..M. The inner products are corrected for the fact that the transmitted signal energies may be unequal. Finally, the largest in the resulting set of numbers is selected and a corresponding decision on the transmitted message made. The optimum receiver is commonly referred as a **correlation receiver.**