

# SYLLABUS

## DIGITAL COMMUNICATION

Tutorial	Practical	Theory	Sessional	Total	Time
1	0	75	25	100	3 Hrs.

### UNIT I:

**Information Theory:** Introduction, Entropy, Huffman Coding, Channel Capacity, Channel Coding, Linear Block Codes, Matrix Description, Syndrome Decoding, Hamming Code, Cyclic Code, Convolution Code generation and Viterbi decoding.

### UNIT II:

**Pulse Modulation System:** Model of digital communication systems, Sampling theorem for baseband and bandpass signals: natural sampling, Flat top sampling, Signal recovery & holding, Quantization of signal, Quantization error, Source coding & companding, Pulse code modulation (PCM), Noise in PCM system, Differential pulse code modulation (DPCM), Adaptive pulse code modulation (ADPCM), Delta modulation (DM), Comparison of PCM, DPCM and DM, Adaptive delta modulation, Quantization noise, Time division multiplexed system (T & E type systems), Calculation of O/P signal power, The effect of thermal noise, O/P signal to noise ratio in delta modulation.

### UNIT III:

**Base Band Pulse Transmission:** Matched filter and its properties average probability of symbol error in binary enclosed PCM receiver, Intersymbol interference, Nyquist criterion for distortionless base band binary transmission, ideal Nyquist channel raised cosine spectrum, correlative level coding Duo binary signalling, tapped delay line equalization, adaptive equalization, LMS algorithm, Eye pattern.

### UNIT IV:

**Digital Pass Band Transmmision:** Pass band transmission model; gram Schmidt orthogonalization procedure, geometric Interpretation of signals, Response of bank of correlators to noise input, detection of known signal in noise, Hierarchy of digital modulation techniques, BPSK, DPSK, DEPSK, QPSK, systems; ASK, FSK, QASK, Many FSK, MSK, Many QAM, Signal space digram and spectra of the above system, effect of intersymbol interference, bit symbol error probabilities, synchronization.

## Topics Covered/Syllabus

- Passband transmission model
- Gram Schmidt orthogonalization procedure
- Geometric interpretation of Signals
- Response of bank of correlators to noise input
- Detection of known signal in noise
- Hierarchy of digital modulation techniques
- BPSK                      • DPSK
- DEPSK                  • QPSK
- ASK                      • FSK
- QASK                    • M-ary FSK
- MSK
- Effects of intersymbol interference
- Bit versus error probabilities
- Synchronization

OSWAL  
PUMPS & MOTORS

# DIGITAL PASS BAND TRANSMISSION

## 4.1. PASSBAND TRANSMISSION MODEL.

Q.1: Explain Passband transmission model with the help of block diagram.

Ans. Bandpass modulation is the process by which an information signal is converted into a sinusoidal waveform.

For digital modulation, such a sinusoidal waveform of duration  $T$  is referred to as a digital symbol.

Bandpass modulation can be defined as the process in which the amplitude, frequency, or phase of an RF carrier, or a combination of them, is varied in accordance with the information to be transmitted.

A model of passband transmission system is depicted in Fig. 4.1.

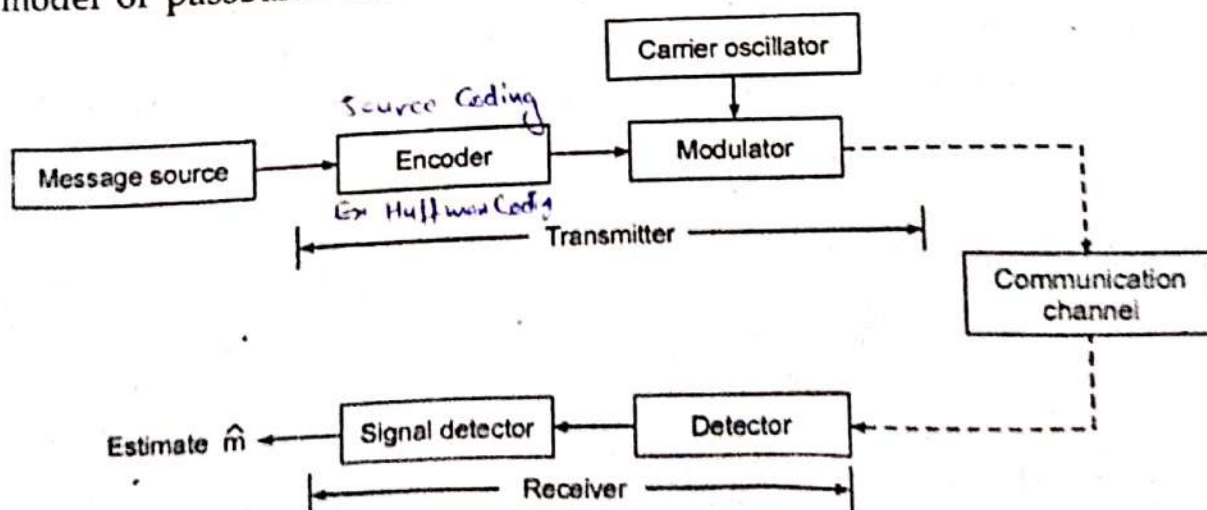


Fig. 4.1: Passband transmission model.



The model has three parts. They are:

- (i) Transmitter
- (ii) Communication channel
- (iii) Receiver.

The message source emits one symbol per  $T_b$  seconds. Let these symbols be denoted by  $m_1, m_2, \dots, m_M$ . The probabilities of these symbols are  $P(m_1), P(m_2), \dots, P(m_M)$ .

If all the  $M$  symbols from the message source are equally likely, then

$$P(m_1) = P(m_2) = \dots = P(m_M) = \frac{1}{M}$$

The message source output is then given to the encoder. The encoder produces a corresponding vector  $S_i$  made up of  $N$  real elements.

The modulator produces a distinct signal  $S_i(t)$  of duration  $T_b$  seconds. This signal represents the symbol  $m_i$  produced by the message source.

The signal  $S_i(t)$  is then transmitted through the communication channel. The communication channel is a bandpass channel.

The communication channel is a linear channel which have a sufficient bandwidth.

The receiver end consists of a detector and a signal detector.

The receiver operation reverse to the operation performed by the transmitter.

The receiver is designed in such a way that it minimize the effect of channel noise.

The estimate  $\hat{m}$  (of transmitted signal  $m_i$ ) is recovered from the output of the signal detector.

#### 4.2 GRAM SCHMIDT ORTHOGONALIZATION PROCEDURE

Q.2. What is Gram schmidt procedure? Illustrate how this procedure is used to construct orthogonal waveforms. (KU 2017)

Q.3. Discuss Gram schmidt orthogonalization procedure.

Ans. Suppose there is a set of  $M$  energy signals denoted by  $S_1(t), S_2(t), \dots, S_M(t)$ . Let  $S_1(t)$  be a set chosen arbitrarily, the first basis function is defined by

$$\phi_1(t) = \frac{S_1(t)}{\sqrt{E_1}}$$

where  $E_1$  is the energy of the signal  $S_1(t)$ .

$$\begin{aligned} \text{then } S_2(t) &= \sqrt{E_2} \phi_2(t) \\ &= S_{21} \phi_1(t) \end{aligned}$$

where  $S_{21} = \sqrt{E_2}$  and  $\phi_2(t)$  has unit energy as required.

Now, using signal  $S_2(t)$ , coefficient  $S_{21}$  is defined as

$$S_{21} = \int_0^T S_2(t) \phi_1(t) dt$$

Let  $g_2(t)$  be an intermediate function, such that

$$g_2(t) = S_2(t) - S_{21} \phi_1(t) \quad \dots(1)$$

which is orthogonal to  $\phi_1(t)$  over the interval  $0 \leq t \leq T$  by virtue of the definition of  $S_{21}$  and the fact that the basic-function  $\phi_1(t)$  has unit energy.

Now, the second basis function is

$$\phi_2(t) = \frac{g_2(t)}{\sqrt{\int_0^T g_2^2(t) dt}} \quad \dots(2)$$

Substituting eq. (1) in eq. (2), we get

$$\phi_2(t) = \frac{S_2(t) - S_{21} \phi_1(t)}{\sqrt{E_2 - S_{21}^2}} \quad \dots(3)$$

where  $E_2$  is the energy of the signal  $S_2(t)$ .

From eq. (2), it is seen that

$$\int_0^T \phi_2^2(t) dt = 1$$

and from eq. (3)

$$\int_0^T \phi_1(t) \phi_2(t) dt = 0$$

So,  $\phi_1(t)$  and  $\phi_2(t)$  form an orthonormal pair as required.

Continuing the procedure in the same manner, a general expression may be written as

$$g_i(t) = S_i(t) - \sum_{j=1}^{i-1} S_{ij} \phi_j(t)$$

where the coefficients  $s_{ij}$  are defined by

$$S_{ij} = \int_0^T S_i(t) \phi_j(t) dt, \quad j = 1, 2, \dots, i-1$$

For  $i = 1$ , the function  $g_i(t)$  reduces to  $S_i(t)$ .

Given the  $g_i(t)$ , the set of basis functions may be defined as

$$\phi_i(t) = \frac{g_i(t)}{\sqrt{\int_0^T g_i^2(t) dt}}, \quad j = 1, 2, \dots, N$$

which forms an orthogonal set.  
The dimensions  $N$  is less than or equal to the number of given signals,  $M$ , depending on one of the two possibilities.

- (i) The signal  $S_1(t), S_2(t), \dots, S_M(t)$  form a linearly independent set, in which case  $N = M$ .
- (ii) The signal  $S_1(t), S_2(t), \dots, S_M(t)$  are not linearly independent, in which case  $N < M$  and the intermediate function  $g_i(t)$  is zero for  $i > N$ .

#### 4.3 GEOMETRIC INTERPRETATION OF SIGNALS

Q.4. Illustrate the geometric interpretation of signals.

Ans. Each signal in the set  $\{s_i(t)\}$ ,  $i = 1, 2, \dots, M$  may be expanded as

$$S_i(t) = \sum_{j=1}^N S_{ij} \phi_j(t) \quad 0 \leq t \leq T \quad \dots(1)$$

$$i = 1, 2, \dots, M$$

The coefficients of the expansion,  $S_{ij}$ , are defined as

$$S_{ij} = \int_0^T S_i(t) \phi_j(t) dt \quad i = 1, 2, \dots, M \quad \dots(2)$$

$$j = 1, 2, \dots, N$$

Accordingly, each signal in the set  $\{S_i(t)\}$  is completely determined by the vector of its coefficients, as shown by

$$S_i = \begin{bmatrix} S_{i1} \\ S_{i2} \\ \vdots \\ S_{iN} \end{bmatrix} \quad i = 1, 2, \dots, M \quad \dots(3)$$

The vector  $S_i$  is called the signal vector.

If the conventional notion of two and three dimensional Euclidean spaces be extended to an  $N$ -dimensional Euclidean space, then the set of signal vectors  $\{S_i\}$ ,  $i = 1, 2, \dots, M$  may be visualized as the corresponding set of  $M$  points in an  $N$ -dimensional Euclidean space, with  $N$  mutually perpendicular axes labeled  $\phi_1, \phi_2, \dots, \phi_N$ . This  $N$ -dimensional Euclidean space is called the signal space.

In an  $N$ -dimensional Euclidean space, length of vectors and angles between vectors may be defined.

It is customary to denote the length or norm of a signal vector  $S_i$  by the symbol  $\|S_i\|$ . The squared length of any signal vector  $S_i$  is defined to be the inner product or dot product of  $S_i$  with itself.

Fig. 4.2 illustrates the case of a two dimensional signal space with three signals, i.e.,  $N = 2$  and  $M = 3$ .

In case of  $N = 2$  or  $3$ ,

$$\begin{aligned} \|S_i\|^2 &= (S_i, S_i) \\ &= \sum_{j=1}^N S_{ij}^2 \end{aligned} \quad \dots(4)$$

where  $S_{ij}$  are the elements of  $S_i$ .

For larger values of  $N$ , length is defined in the same way.

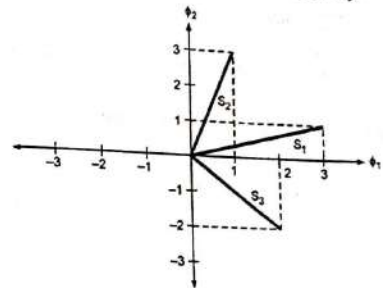


Fig. 4.2: Illustration of geometric interpretation of signals for the case when  $N = 2$  and  $M = 3$ .

The cosine of the angle between two vectors is defined as the inner product of the two vectors divided by the product of their individual norms. That is, the cosine of the angle between the vectors  $S_i$  and  $S_j$  equals the ratio  $(S_i, S_j) / (\|S_i\| \|S_j\|)$ .

The two vectors  $S_i$  and  $S_j$  are thus orthogonal or perpendicular if their inner product  $(S_i, S_j)$  is zero.

The energy of a signal  $S_i(t)$  of duration  $T$  seconds is

$$E_i = \int_0^T S_i^2(t) dt \quad \dots(5)$$

Therefore, substituting (1) in (5), we get

$$E_i = \int_0^T \left[ \sum_{j=1}^N S_{ij} \phi_j(t) \right] \left[ \sum_{k=1}^N S_{ik} \phi_k(t) \right] dt \quad \dots(6)$$

Interchanging the order of summation and integration

$$E_i = \sum_{j=1}^N \sum_{k=1}^N S_{ij} S_{ik} \int_0^T \phi_j(t) \phi_k(t) dt \quad \dots(7)$$

But since the  $\phi_j(t)$  form an orthonormal set, then

$$E_i = \sum_{j=1}^N S_{ij}^2 \quad \dots(8)$$

Eq. (4) and (8) show that the energy of a signal  $S_k(t)$  is equal to the squared length of the signal vector  $S_k$  representing it.

In case of a pair of signals  $S_i(t)$  and  $S_k(t)$ , represented by the signal vectors  $S_i$  and  $S_k$  respectively, then

$$\begin{aligned} \|S_i - S_k\|^2 &= \sum_{j=1}^N (S_{ij} - S_{kj})^2 \\ &= \int_0^T [S_i(t) - S_k(t)]^2 dt \end{aligned}$$

where  $\|S_i - S_k\|$  is the Euclidean distance between the points represented by signal vector  $S_i$  and  $S_k$ .

#### 4.4 RESPONSE OF BANK OF CORRELATORS TO NOISY INPUT

Q.5. Explain the response of bank of correlators to noise input. (KU 2017)

Ans. Suppose that the input to the bank of  $N$  product integrators or correlators is not the transmitted signal  $S_i(t)$  but rather received signal  $x(t)$  is defined in accordance with the AWGN channel as

$$x(t) = S_i(t) + W(t) \quad \begin{cases} 0 \leq t \leq T \\ i = 1, 2, \dots, M \end{cases}$$

where  $w(t)$  is a sample function of the white Gaussian noise process  $W(t)$  of zero mean and power spectral density  $N_0/2$ .

The output of correlator  $j$  is the sample value of a random variable  $X_j$ , where sample value is defined as

$$\begin{aligned} x_j &= \int_0^T x(t) \phi_j(t) dt \\ &= S_{ij} + W_j, \quad j = 1, 2, \dots, N \end{aligned}$$

The first component,  $S_{ij}$ , is the deterministic component of  $x_j$  due to the transmitted signal  $S_i(t)$  and is equal to

$$S_{ij} = \int_0^T S_i(t) \phi_j(t) dt$$

The second component,  $w_j$ , is the sample value of a random variable  $W_j$  due to the channel noise  $w(t)$  and is equal to

$$w_j = \int_0^T W(t) \phi_j(t) dt$$

Consider a stochastic process whose sample function  $x'(t)$  is related to the received signal  $x(t)$  as

$$x'(t) = x(t) - \sum_{j=1}^N x_j \cdot \phi_j(t)$$

or

$$\begin{aligned} x'(t) &= S_i(t) + w(t) - \sum_{j=1}^N (S_{ij} + w_j) \phi_j(t) \\ &= w(t) - \sum_{j=1}^N w_j \phi_j(t) = w'(t) \end{aligned}$$

The sample function hence depends on the channel noise  $w(t)$ . Received signal may be expressed as

$$\begin{aligned} x(t) &= \sum_{j=1}^N x_j \cdot \phi_j(t) + x'(t) \\ &= \sum_{j=1}^N x_j \cdot \phi_j(t) + w'(t) \end{aligned}$$

$w'(t)$  may be viewed as a remainder that must be included on the right hand side of the equation to preserve equality.

It may be noticed that the expansion of  $x(t)$  is random (stochastic) due to the channel noise at the receiver input.

#### 4.5 DETECTION OF KNOWN SIGNAL IN NOISE

Q.6. How can a known signal be detected in noise. Explain in detail.

Ans. Assume that in each time slot of duration  $T$  seconds, one of the  $M$  possible signals  $S_1(t), S_2(t), \dots, S_M(t)$  is transmitted with equal probability of  $\frac{1}{M}$ .

Then for AWGN channel, the sample function  $x(t)$  is given by

$$x(t) = S_i(t) + w(t) \quad 0 \leq t \leq T$$

$i = 1, 2, 3, \dots, M$

where  $w(t)$  is the sample function of the white Gaussian noise process  $W(t)$  with zero mean and power spectral density  $N_0/2$ .

The receiver has to observe the signal  $x(t)$  and make the best estimate of the transmitted signal  $S_i(t)$ .

The transmitted signal  $S_i(t)$ ,  $i = 1, 2, \dots, M$  is applied to a bank of correlators, with a common input and supplied with an appropriate set of  $N$  orthonormal basis functions, the resulting correlator outputs define the signal vector  $S_i$ .

$S_i(t)$  may be represented by a point in a Euclidean space of dimensions  $N \leq M$ . Such a point is referred as transmitted signal point or message point.

The collection of message point in the Euclidean space is known as signal constellation.



When the received signal  $x(t)$  is applied to the bank of  $N$  correlators, the output of the correlator define a new vector  $x$  called observation vector.

The observation vector  $x$  differs from the signal vector  $S_i$  by a random noise vector  $w$ .

The vectors  $x$  and  $w$  are sampled values of the random vectors  $X$  and  $W$  respectively.

The noise vector  $w$  represents the portion of the noise  $w(t)$  which will interfere with the detected process.

Based on the observation vector  $x$ , the received signal  $S(t)$  is represented by a point in the same Euclidean space. This point is referred to as received signal point.

The relation between them is shown in Fig. 4.3.

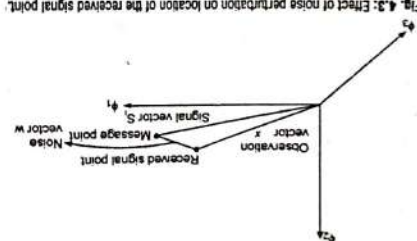


Fig. 4.3: Effect of noise perturbation on location of the received signal point.

In the detection problem, the observation vector  $x$  given, a mapping is to be performed from  $x$  to an estimate of the transmitted symbol, in such a way that would minimize the average probability of symbol error in the decision.

The maximum likelihood detector provides solution to this problem.

#### 4.6 HIERARCHY OF DIGITAL MODULATION TECHNIQUES

Q.7. Explain the different types of digital modulation techniques.

Ans.

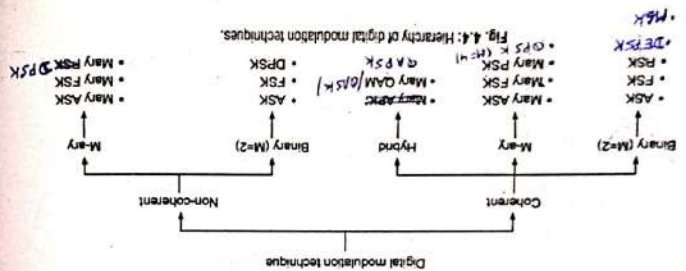


Fig. 4.4: Hierarchy of digital modulation techniques.

#### 4.7 BPSK SYSTEMS

Q.8. What do you mean by coherent Binary FSK? Discuss their generation and detection techniques.

Ans. BPSK system is binary phase shift keying. In BPSK, phase of the sinusoidal carrier is changed according to the data bit to be transmitted.

In BPSK, the binary symbols, '1' and '0' modulate the phase of the carrier. Let us assume that the carrier is

$$S(t) = A \cos 2\pi f_c t$$

where,  $A$  = peak value of sinusoidal carrier.

For the standard 1Ω load resistor, the power dissipated would be

#### 4.17 EFFECT OF INTER SYMBOL INTERFERENCE

Q.18. What do you mean by intersymbol interference? What are the effects of intersymbol interference.

Ans. Intersymbol Interference: Refer Unit 3, Section 3.3

##### Effects of Intersymbol Interference

Intersymbol interference affects the performance of digital systems due to the limited bandwidth occupancy of the systems.

When ISI is present, the correlator receiver or matched filter receiver no more acts as an optimum filter.

This degrades the actual error rate.

The effects of intersymbol interference can be studied with the help of Eye pattern.

Eye pattern: Refer Unit 3, Section 3.12.

#### 4.18 BIT VERSUS SYMBOL ERROR PROBABILITIES

Q.19. Derive an expression for the relation between the average probability of symbol error and the bit error rate.

Ans. The symbol error rate is an appropriate figure of merit when the messages of length  $m = \log_2 M$  are transmitted.

But whenever a binary data is to be transmitted, a figure of merit known as the probability of bit error rate (BER) is more meaningful to be used.

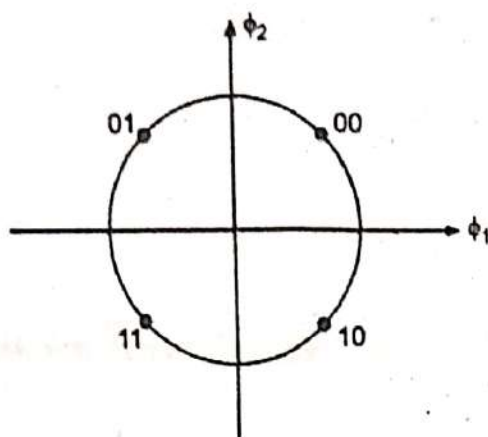
The relation between the average probability of symbol error  $P_s$  and the bit error rate (BER) can be determined as:

##### Case 1

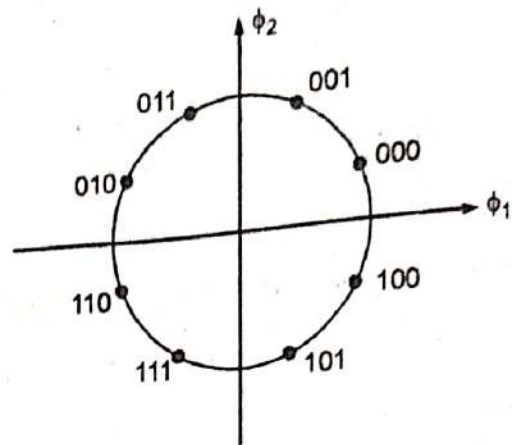
In this case, a mapping is performed from binary to M-ary symbols in such a way that the two binary M-tuples corresponding to any pair of adjacent symbols in an M-ary modulation scheme differ in only one bit position.

This is shown in the signal constellations of M-ary PSK for  $M = 4$  and  $M = 8$  in Fig. 4.41(a) and (b) respectively.





(a)  $M = 4$



(b)  $M = 8$

Fig. 4.41: Signal constellation of M-ary PSK.

When the symbol error probability  $P_e$  is small then the possibility of mistaking one symbol for its nearest symbols is greater than any other type of symbol error.

Due to the use of gray code, for a given symbol rate, the most probable number of bit errors is only one.

There are  $\log_2 M$ -bits per symbol.

Therefore, the relation between BER and symbol error probability  $P_e$  is as follows:

$$\text{BER} = \frac{P_e}{\log_2 m}, M \geq 2$$

#### Case 2

In this case, let us assume that all the symbol errors are equally likely and they occur with a probability given as:

$$\frac{P_e}{M-1} = \frac{P_e}{2^k - 1}$$

where  $P_e$  = the average probability of symbol error

$$K = \log_2 m.$$

For this case, the relation between BER and  $P_e$  is given as

$$\text{BER} = \left[ \frac{2^k}{2^k - 1} \right] P_e$$

where,

$$K = \log_2 M$$

### 4.19 SYNCHRONIZATION

**Q.20. Write a short note on synchronization.**

**Ans.** The coherent reception of a digitally modulated signal requires that the receiver be synchronous with the transmitter.

So synchronization may be defined as:

'Two sequences of related events performed separately, one in the transmitter and the other in the receiver, are said to be synchronous relative to each other when



the events in one sequence and the corresponding events in the other occur simultaneously, except for some finite delay.'

There are two basic modes of synchronization:

- (i) Carrier Synchronization
- (ii) Symbol Synchronization

#### (i) Carrier Synchronization

When coherent detection is used in signaling over AWGN channels via the modulation of a sinusoidal carrier, knowledge of both the frequency and phase of the carrier is necessary. The process of estimating the carrier phase and frequency is called **carrier recovery** or **carrier synchronization**.

#### (ii) Symbol Synchronization

To perform demodulation, the receiver has to know the instants of time at which the modulation in the transmitter changes its state *i.e.*, the receiver has to know the starting and finishing times of the individual symbols, so that it may determine when to sample and when to quench the product-integrators. The estimation of these times is known as **clock recovery** or **symbol synchronization**.

Depending on whether some form of aiding is used or not, synchronization schemes may be classified as:

- (i) Data aided synchronization
- (ii) Non data aided synchronization

#### (i) Data aided synchronization

In data-aided synchronization schemes, a preamble is transmitted along with the data bearing signal in a time multiplexed manner on a periodic basis.

The preamble contains information about the symbol timing, which is extracted by appropriate processing of the channel output at the receiver.

Such an approach is commonly used in digital satellite and wireless communications, where the motivation is to minimize the time required to synchronize the receiver to the transmitter.

Limitations of data-aided synchronization are:

- (a) Reduced data throughput efficiency, which is incurred by assigning a certain portion of each transmitted frame to the preamble.
- (b) Reduced power efficiency, which results from the allocation of a certain fraction of the transmitted power to the transmission of the preamble.

#### (ii) Non data aided synchronization

In this scheme, the use of preamble is avoided and the receiver has the task of establishing synchronization by extracting the necessary information from the noisy distorted modulated signal at the channel output.

Throughout and power efficiency are improved in this approach but the time taken to establish synchronization is increased.



### Algorithmic Approach to Synchronization

Given the received signal, the maximum likelihood method is used to estimate two parameters. Carrier phase and symbol timing.

In the algorithmic approach, the symbol-timing recovery is performed before phase recovery. This is done because once the envelope delay incurred by signal transmission through a dispersive channel is known, then one sample per symbol at the matched filter output may be sufficient for estimating the unknown carrier phase.

Computational complexity of the receiver is minimized by using synchronization algorithms that operate at the symbol rate  $1/T$ .

The algorithmic approach to synchronization is as follows:

- (i) Through processing the received signal corrupted by channel noise and channel dispersion, the likelihood function is formulated.
- (ii) The likelihood function is maximized to recover the clock.
- (iii) With clock recovery achieved, the next step is to maximize the likelihood function to recover the carrier.

### QUICK REVIEW

- Bandpass modulation is the process by which an information signal is converted into a sinusoidal waveform.
- The digital modulation techniques can be classified into 2 categories:
  - (i) Coherent techniques
  - (ii) Non coherent techniques
- In coherent modulation techniques, a phase synchronized carrier is to be generated at the receiver to recover the information source.
- In non-coherent modulation techniques, no phase synchronization of local carrier is needed at the receiver end.
- In binary phase shift keying (BPSK), phase of the sinusoidal carrier is changed according to the data bit to be transmitted.
- In amplitude shift keying (ASK), only one unit energy carrier is used which is switched on or off depending upon the input binary sequence.
- In frequency shift keying (FSK), the frequency of a sinusoidal carrier is shifted according to the binary symbol.
- Two sequences of related events performed separately, one in the transmitter and the other in the receiver, are said to be synchronous relative to each other when the events in one sequence and the corresponding events in the other occur simultaneously, except for some finite delay."
- There are two basic modes of synchronization
  - (i) Carrier synchronization
  - (ii) Symbol synchronization
- Depending on whether some form of aiding is used or not, synchronization schemes may be classified as
  - (i) Data aided synchronization
  - (ii) Non data aided synchronization

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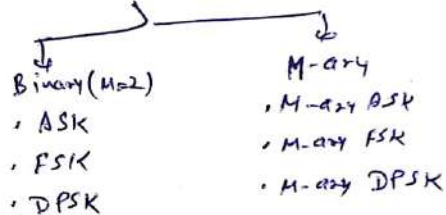
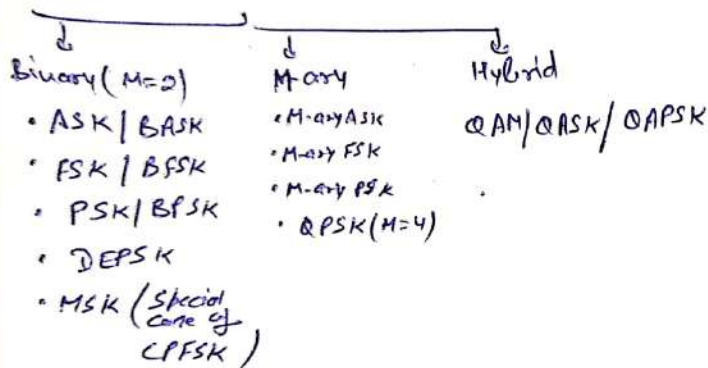


# HIERARCHY

Digital Mod<sup>y</sup> Tech.

Coherent

Non-Coherent



\* Do only Coherent Techniques

\* FSK & M-ary FSK are only Scheme which are mostly used Non-Coherent by Practically

## SOME POINTS

### Some Points:

- \*  $\text{erfc}$  is the monotonically decreasing function.  
e.g.  $P_e = \frac{1}{2} \text{erfc}\left(\frac{d}{2\sqrt{N_0}}\right)$ , if  $\frac{d}{2\sqrt{N_0}}$  increases,  $P_e$  falls rapidly.  
The maximum value of  $P_e = \frac{1}{2}$  when  $\frac{d}{2\sqrt{N_0}}$  is very small.  $P_e = \frac{1}{2}$  means receiver will make incorrect decision for half No. of times.
- \* Coherent Detection exhibits less  $P_e$  than Non Coherent. But Coherent detection needs more hardware.



Scanned with CamScanner

Name	Definition	Equation	Waveform	Signal space diagram	Bit error probability	Spectral density	TX	RX
ASK / BPSK or ASK or ON-OFF Keying	A Carrier is switched on & off depending upon digital signal.	$s(t) = \sqrt{\frac{E_b}{T_b}} \cos(2\pi f_c t)$ for '1' $s(t) = 0$ for '0' $s(t) = \sqrt{\frac{E_b}{T_b}} \cos(2\pi f_c t)$ $s(t) = 0$ then $P(t) = \frac{\sqrt{E_b}}{\sqrt{2}} \cos(2\pi f_c t)$			$P_e = \frac{1}{2} \exp\left(-\frac{E_b}{2N_0}\right)$	Since in BPSK $BW = 2f_c$		
FSK Non-Coherent	Two carrier of different frequencies are transmitted depending upon binary sequence. $f_1 = M f_c$ $f_2 = M f_c$ Twice the carrier frequency of $f_c$ is taken such that $f_1$ & $f_2$ are orthogonal.				$P_e = \frac{1}{2} \exp\left(-\frac{E_b}{2N_0}\right)$ It is more than BPSK.			
FSK Coherent								
BPSK Non-Coherent					$P_e = \frac{1}{2} \exp\left(-\frac{E_b}{2N_0}\right)$ Non-coherent orthogonal BPSK			
QPSK Non-Coherent								
QPSK Coherent								



SHEET-15

M-ary PSK vs M-ary FSK  
 In M-ary PSK, need more BW  
 as  $f_c$  of M-ary PSK is less than M-ary FSK

M-ary PSK vs M-QAM (M=16)  
 In PSK, need some BW  
 as M-ary PSK have more  $P_b$  as  
 points are closer

- 1) Explain Phase Shift Keying (PSK) modulation.
- 2) Explain Binary PSK (BPSK) modulation.
- 3) Explain Quadrature PSK (QPSK) modulation.
- 4) Explain M-ary PSK modulation.
- 5) Explain M-ary FSK modulation.
- 6) Explain M-ary QAM modulation.
- 7) Compare PSK and FSK in terms of bandwidth efficiency.
- 8) Compare PSK and QAM in terms of power efficiency.
- 9) Write a note on Synchronization.
- 10) (Project Handwritten)

Power Bandwidth

	M-ary PSK	M-ary QAM
$d$	$d = \sqrt{0.1 E_b}$ or $d = \sqrt{E_b}$	$d = 2 \sqrt{1 E_b}$ (for M=16)
Bandwidth	$\frac{2 f_c}{N}$	$\frac{2 f_c}{N}$

Name	Definition	Equation	Waveform	Signal-space diagram & d	Bit error probability	Spread BW	Receiver
M-ary PSK	In M-ary PSK, there are total of M signals each having $M/2$ bits. In M-ary PSK, the signals have the same frequency but different phases.	$S(t) = \sqrt{\frac{2E_b}{T}} \cos(\omega_c t + \theta)$ where $\theta = 0, \dots, M-1$ $T = M T_b$ $E_b = M E_s$	—	$d = \sqrt{N E_b}$	Less than M-ary PSK	$BW = \frac{2 f_c}{N}$ (2 in max. M-ary PSK)	Receiver block diagram showing demodulation and decision.
QAM	Phase & Amplitude are changed.	$S(t) = \sqrt{\frac{2E_b}{T}} \cos(\omega_c t + \theta) - \sqrt{\frac{2E_b}{T}} \sin(\omega_c t + \theta)$ for M-ary QAM, $L = \sqrt{M}$ then $(b_1, b_2) = \{(1, 1), (1, -1), (-1, 1), (-1, -1)\}$	—	16-QAM signal-space diagram showing 16 points in a square constellation.	$d = 2 \sqrt{0.1 E_b}$ (This is for M=16) $E_b = N E_s$	$BW = \frac{2 f_c}{N}$	No need