CHAPTER 2

Digital Modulation, Multiplexing and Spread Spectrum Techniques

Objectives...

- To study digital modulation techniques.
- To learn the concepts of ASK, FSK, PSK, QPSK, QAM.
- To understand Nyquist sampling theorem and PCM.
- To study delta modulation and adaptive delta modulation.

2.1 INTRODUCTION

- In previous chapter, the concepts of modulation, demodulation were discussed. In this chapter, various digital modulation techniques such as ASK, PSK, FSK, QPSK and QAM which play a vital role in transmitting digital data efficiently over communication channels are explained. These techniques ensure higher data rates, improved noise immunity and better bandwidth utilization.
- Multiplexing techniques including Frequency Division Multiplexing (FDM), Time Division Multiplexing (TDM), Code Division Multiplexing (CDM) and Orthogonal Frequency Division Multiplexing (OFDM) enable multiple signals to share a single communication medium simultaneously, thereby optimizing channel capacity and resource use.
- Spread spectrum techniques, such as Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS), further enhance communication reliability and security by spreading the signal over a wide frequency band using pseudo-random sequences. Together, these methods form the core of modern digital communication systems, ensuring efficient, secure and robust data transmission.
- All these methods are discussed in this chapter.

2.2 NYQUIST SAMPLING THEOREM

- In the analog communication system like AM, FM and PM, the entire modulating waveform is transmitted.
- However, in analog pulse modulation, periodic samples of the modulating waveform are taken. In the absence of noise and distortion it is possible to completely recover a continuous analog signal from its discrete samples, provided the samples are taken at greater than certain minimum rate.

- If the sampling frequency is high enough, that is if enough samples are transmitted, the modulating wave can be recovered at the receiver.
- It is found that, higher the sampling frequency, more accurate signal is reproduced at the receiver.
- The minimum sampling frequency required for reliable reproduction of the modulating signal is defined by Nyquist theorem.

Nyquist Theorem:

- To ensure that all of the information contained in the original signal is transferred, the sampling frequency in any pulse modulation system must not be less than twice the highest signal frequency.
- **Nyquist theorem** states that "*a signal can be represented by its samples and reconstructed if it is sampled at a rate at least twice its highest frequency component*".
- Mathematically, if the maximum frequency of a signal is f_{max}, then the sampling frequency f_s must satisfy $f_s \geq 2f_{max}$. This minimum rate, $2f_{max}$, is known as the **Nyquist rate**.
- Sampling below this rate causes aliasing, a distortion where different signals become indistinguishable from one another in the sampled data. That means, if the sampling rate is less than the Nyquist rate, the modulating signal cannot be accurately produced.
- The telephone speech may contain frequencies up to 3400 Hz. So, assuming that, highest frequency is 6400 Hz, then the telephone signal would have to be sampled at greater than 6800 samples per second. In practice, a sampling rate of 8000 Hz would be used.

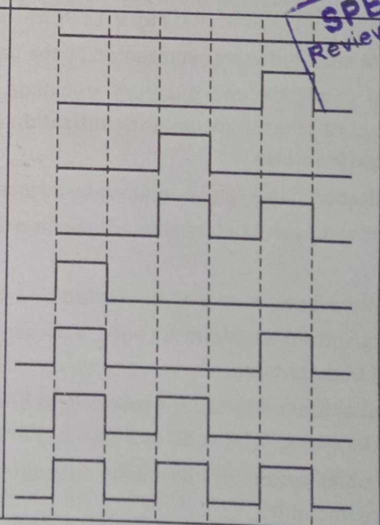
2.3 PULSE CODE MODULATION (PCM)

- Pulse code modulation is a digital system of communication.
- In PCM, the peak-to-peak amplitude range available for signal to be transmitted is divided into a number of standard values.
- The signal is sampled. The sample of signal is then transmitted not as its actual amplitude but the nearest standard amplitude.
- To transmit standard amplitude, binary code of six or seven digits is used.
- If signal is divided into 8 standard amplitudes, 3-bit code is used which gives more error.
- A seven-digit code is capable of transmitting $2^7 = 128$ standard amplitudes and in that case, error will be less.
- At the receiver, the code of pulses is translated back to the corresponding standard amplitude.
- For this, each received pulse is used to charge a capacitor.
- This capacitor is shunted by a resistor of such a value that the charge on the capacitor decreases to the half amplitude in the time corresponding to the interval between the pulses.
- At the end of the pulse group, the voltage present on the capacitor gives the standard amplitude that is represented by the code.
- Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals.
- It is the standard form of digital audio in computers, satellite communication, compact discs, digital telephony and other digital audio applications.

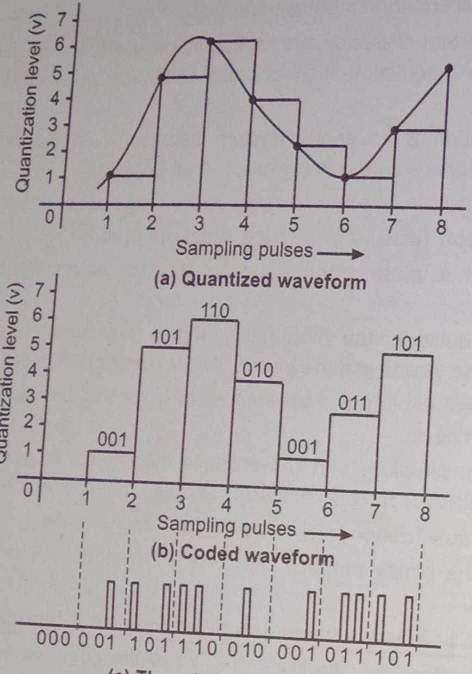
- Pulse modulation uses a continuous pulse train as a carrier and some characteristic of such pulse train is varied according to modulating signal.
- Pulse has three important characteristics, its amplitude, width and position which results into three types of pulse modulation such as pulse amplitude, pulse width and pulse position modulation.
- Such pulse modulation is an analog system because, even discrete valued samples of continuous analog signal are used in the modulation process.
- These discrete samples can take any value within the range of the continuous signal.
- For digital transmission, finite set of discrete values are used.
- Digital systems, such as pulse code modulation, also use pulses and hence is also a type of pulse modulation.
- In pulse code modulation, the modulating signal is sampled, the sample amplitude is converted into a binary code and the binary code is transmitted in groups as a train of pulses.
- In PCM, the sampled amplitude is transmitted as a specific binary number out of a limited range of binary numbers.
- To obtain this, each sample is first converted to the nearest standard amplitude, called the 'quantum'. This process of sample conversion is called 'quantizing the signal'.
- To represent eight quantization level, three-bit code can be used.
- Table 2.1 shows the binary number and 3-bit pulse code is represented by each of the quantization levels.

Table 2.1 : Binary number and pulse code for 8-bit quantization

Quantization level	Binary number	Pulse code waveform
0	000	
1	001	
2	010	
3	011	
4	100	
5	101	
6	110	
7	111	



- Fig. 2.1 shows PCM technique.



(a) Quantized waveform

(b) Coded waveform

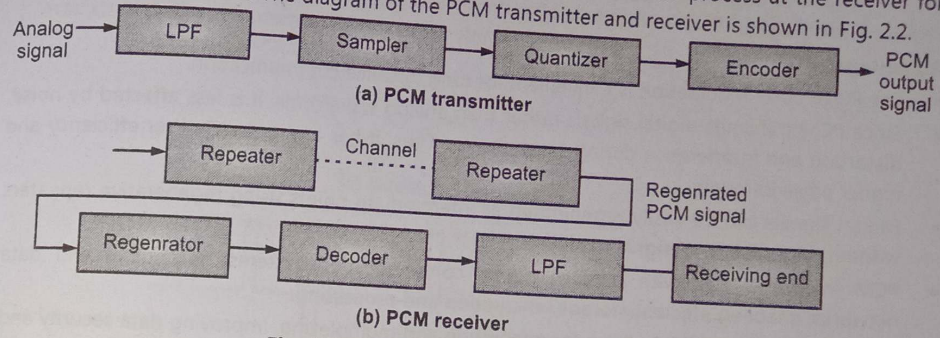
(c) Three-bit PCM pulse train

Fig. 2.1 : PCM waveforms

- The sample amplitudes are represented by the nearest quantum levels or standard amplitude.
- The actual amplitude and quantum amplitude may be slightly different which introduces certain amount of error known as **quantization error.** This generates some distortion known as **quantization noise.**
- The quantization noise can be reduced by increasing the number of quantization levels.
- After quantization and before transmission as a PCM signal, each sample is coded as a binary number.
- Eight quantum levels used are represented by 3-bit binary word.
- After the quantized waveform is coded, each sequential sample is transmitted as a pulse code by 3-bit PCM pulse train.
- For eight quantization level corresponding to 3-bit code, quantization error is more which can be reduced by using 8-bit code or 8-bit word which provides 256 quantum levels.
- Being the digital system of transmission, the primary advantage of PCM is its better noise and interference immunity.

2.3.1 PCM Transmitter and Receiver Block Diagram

- Pulse Code Modulation is a digital pulse modulation technique in which an analog signal is converted into a sequence of coded digital pulses. It involves three main stages at the transmitter: sampling, quantization and encoding and the reverse process at the receiver for signal reconstruction. The diagram of the PCM transmitter and receiver is shown in Fig. 2.2.



(a) PCM transmitter

(b) PCM receiver

Fig. 2.2 : PCM transmitter and receiver

PCM Transmitter :

- The PCM transmitter converts the input analog signal into a digital form suitable for transmission. The main blocks are:
  1. **Low-Pass Filter (LPF):** LPF filters the signal. The analog input signal first passes through a low-pass filter to limit its bandwidth and prevent aliasing during sampling. The filter ensures that, only frequency components below the Nyquist rate are retained.
  2. **Sampler:** The filtered analog signal is sampled at regular intervals according to the Nyquist Sampling Theorem. The output is a sequence of discrete-time samples representing the amplitude of the signal at specific time instants.
  3. **Quantizer:** Each sample is approximated to the nearest value from a finite set of discrete amplitude levels. This process introduces a small quantization error, but makes the signal suitable for digital representation.
  4. **Encoder:** The quantized values are then converted into binary code words (a sequence of bits). This digital output represents the amplitude of the original analog signal at each sampling instant.
- The resulting binary sequence is transmitted through the communication channel.

PCM Receiver :

- The PCM receiver performs the reverse operations to reconstruct the original analog signal from the received digital data. The main blocks are:
  1. **Repeater:** Used in long-distance communication, this block restores signal strength and corrects any bit errors caused by noise or attenuation in the transmission channel. This block is optional, if required then only it is used in receiver section.

2. **Decoder:** The received binary data is decoded back into a sequence of quantized amplitude levels corresponding to the original samples.
3. **Low-Pass Filter:** LPF basically reconstruct the signal. The discrete quantized samples are passed through a low-pass filter to smooth out the waveform and reconstruct the original analog signal.

Advantages of PCM:

- The Pulse Code Modulation is convenient for long distance communication.
- Since PCM transmits digital signals rather than analog waveforms, it is less affected by noise, distortion and interference during transmission. Thus, it has higher transmitter efficiency and higher noise immunity.
- Digital signals can be easily regenerated at intermediate points using regenerative repeaters without degradation of signal quality.
- PCM integrates well with modern digital communication systems, computers and data networks, enabling efficient storage, encryption and processing.
- The digital nature of PCM allows for encryption and multiplexing, improving data security and channel efficiency.
- Error detection and correction codes can be easily applied to digital PCM data to enhance reliability.

Disadvantages of PCM:

1. PCM requires a much larger bandwidth compared to analog transmission because each analog sample is represented by multiple bits.
2. The transmitter and receiver circuits are more complex due to the need for sampling, quantization, encoding and decoding.
3. The process of quantization introduces small errors, as the continuous amplitude values are approximated to discrete levels.
4. Digital processing and regeneration circuits often consume more power compared to simple analog systems.

Applications of PCM :

1. It is used in Telephony and Voice Communication. PCM is the standard method for digitizing voice signals in telephone and voice communication.
2. PCM is used in Audio and Video Recording. It is also used in compact discs (CDs), digital audio tapes and video broadcasting for high-quality sound and image reproduction.
3. PCM is used in data transmission systems, digital communication links, optical fiber systems and satellite communication.
4. PCM is used in Radar and Control systems. PCM ensures precise, noise-free transmission of signals in radar, telemetry and control applications.

2.4 CONCEPT OF DELTA MODULATION

- Delta modulation (DM), sometimes called slope modulation is a process of modulation in which a train of fixed width pulses is transmitted.

- Fig. 2.3 shows the waveforms associated with delta modulation, in which just one-bit (binary digit) per sample is transmitted. Pulse polarity is determined by, whether the modulating signal sample is rising or falling. .
- The input is made to rise or fall by a fixed step height at each pulse. The polarity of the pulse indicates whether the demodulator output should rise or fall.

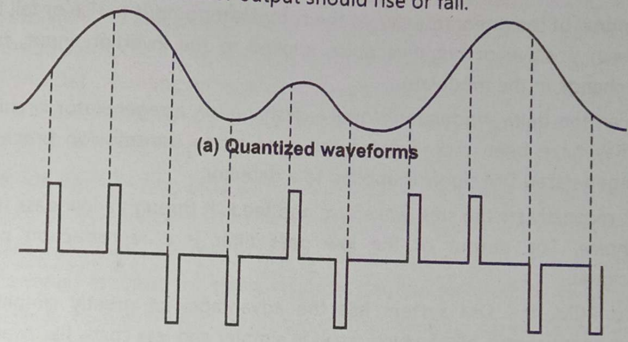


Fig. 2.3 : Delta modulation waveform

- Fig. 2.4 shows the block diagram of delta-modulation system. The modulating signal is applied to the non-inverting input of a high gain differential comparator and a digitally reconstructed version of the modulating signal is applied to the inverting input.
- The saturated output of the comparator, either positive or negative, will depend on the polarity of the difference voltage between the input signals. The output will thus represent ± binary 1.

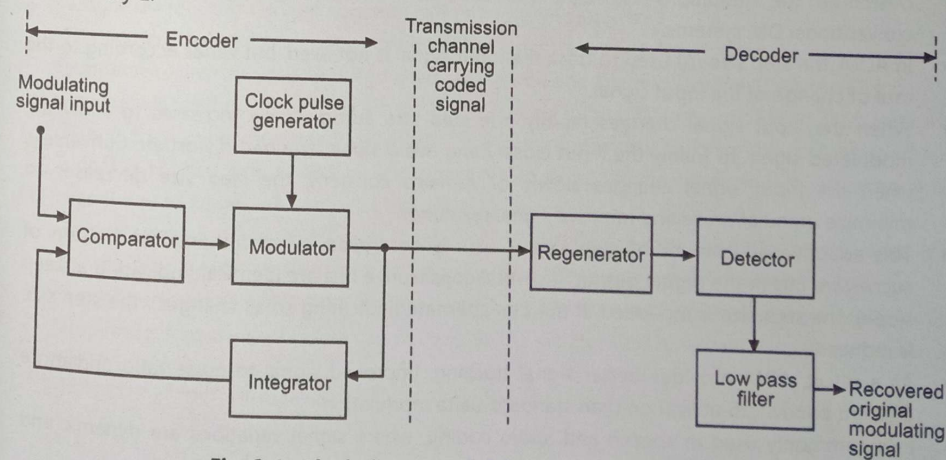


Fig. 2.4 : Block diagram of delta modulation system

- A certain sampling rate is decided, the train of pulse with desired sampling rate is applied to the modulator. The modulator either transmits these pulses directly for a + 1 or inverts their polarity for a – 1.
- The resulting signal is simultaneously applied to an integrator and transmitted as the output signal.
- The output signal of the integrator causes the comparator output to rise or fall by a fixed step height for each positive or negative pulse applied to the inverting input, thus causing a comparable change in the modulator.
- At the receiver, the delta modulated signal is reshaped in a regenerator circuit. Most of the noise that may have been picked up in the modulation transmission process is removed before the regenerated DM signal is applied to a detector.
- The detector reconstructs the step waveform and feeds it through a low pass filter to remove quantizing noise. The output of the low pass filter is a reproduction of the original modulating signal.
- Compared to PCM, the DM system has the advantages of greatly simplified encoding, decoding and quantization, all of which result in simpler and less costly hardware.
- However, the DM system cannot follow fast rising signal (since the integrator cannot follow a signal that has a very fast rise time, no problem for voice transmission since voice signal does not change abruptly). It has the disadvantage of requiring a higher sampling rate and a greater bandwidth than PCM.

Adaptive Delta Modulation :

- Adaptive Delta Modulation (ADM) is an improved form of Delta Modulation (DM) designed to overcome the limitations of slope overload distortion and granular noise present in conventional DM systems.
- In ADM, the step size (Δ) used to track the input signal is not fixed, but varies according to the rate of change of the input signal.
- When the input signal changes rapidly, the step size automatically increases to allow the modulated signal to follow the input closely and avoid slope overload distortion. Conversely, when the input signal changes slowly or remains constant, the step size decreases to minimize granular noise and improve signal resolution.
- This adaptive adjustment of step size is usually controlled by monitoring the pattern of successive bits in the digital output, if several consecutive bits are identical (indicating a steep slope), the step size is increased; if the bits alternate (indicating small changes), the step size is reduced.
- As a result, ADM provides better signal tracking, improved signal-to-noise ratio and more efficient bandwidth utilization than standard delta modulation.
- It is commonly used in speech and audio coding, where signal variations are dynamic and unpredictable.

2.5 DIGITAL MODULATION TECHNIQUES

- Digital Modulation is the process of transmitting digital information over an analog communication channel by varying one or more parameters of a carrier signal such as amplitude, frequency or phase in accordance with the digital data.
- It forms the backbone of all modern digital communication systems, including wireless networks, satellite links and data transmission over optical and radio channels.
- In digital modulation, the carrier signal is modified based on discrete digital symbols (usually binary 0s and 1s), resulting in improved noise immunity, higher data rates and efficient utilization of the available bandwidth compared to analog modulation.

  The most common digital modulation techniques include:

  1. **Amplitude Shift Keying (ASK):** The amplitude of the carrier signal is varied according to the digital data. It is simple to implement but more susceptible to noise.
  2. **Frequency Shift Keying (FSK):** The frequency of the carrier signal is changed between discrete values to represent binary symbols. It offers better noise immunity than ASK.
  3. **Phase Shift Keying (PSK):** The phase of the carrier is varied to represent digital data. Binary PSK (BPSK) and Quadrature PSK (QPSK) are widely used in modern communication systems.
  4. **Quadrature Amplitude Modulation (QAM):** A combination of amplitude and phase modulation, QAM efficiently transmits multiple bits per symbol, allowing higher data rates.
- Digital modulation techniques are selected based on factors such as channel conditions, required data rate, bandwidth availability and power efficiency. Together, these techniques ensure reliable, high-speed communication in digital systems.
- In most of the digital communication telephones, line network is used, in which, pairs of copper wires are situated between users' telephone and the telephone exchange office.
- The signals used on these lines are dc, in the limited frequency range of 300 to 3100 Hz using filters. Such conductors can carry 1 Mbs of information without any trouble, but when digital signal is applied to one end of line, the received signal at the other end shows attenuated signal due to capacitive and inductive effect of the cables.
- Thus, distortion occurs in the signal and data transmission speed is also degraded. Hence, it is necessary to handle dc signals by ac signaling with the process of modulation.
- A continuous tone in the 1000 to 2000 Hz range is introduced which is sine wave carrier. Its amplitude, frequency or phase can be modulated to transmit binary information as shown in Fig. 2.5 below.

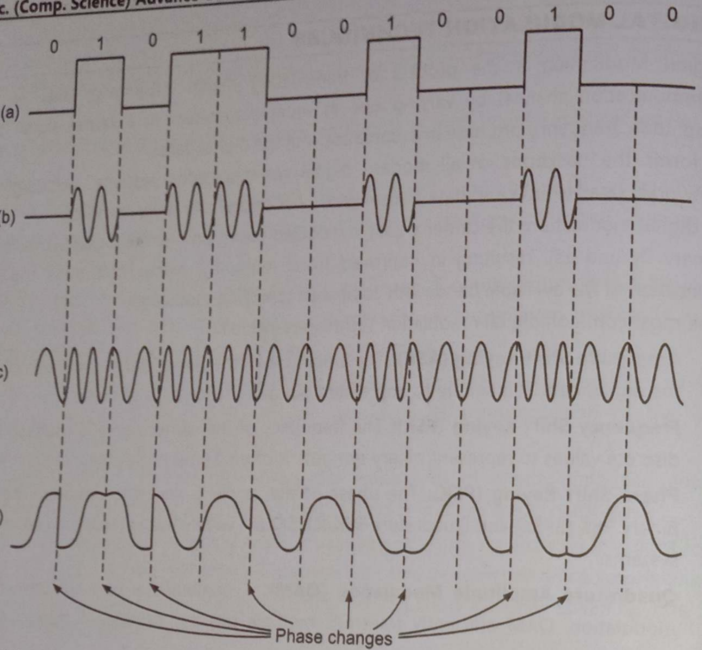


Fig. 2.5 : (a) A binary signal, (b) Amplitude modulation, (c) Frequency modulation, (d) Phase modulation

- In amplitude modulation, two different voltage levels are used to represent '0' and '1' respectively.
- In digital modulation this is known as Amplitude-Shift Keying (ASK). The term 'keying' refers to turning a transmitter 'ON' and 'OFF'.
- In frequency modulation, also known as Frequency-Shift Keying (FSK), two or more different tones are used.
- In most common form of phase modulation also known as (PSK), the carrier wave is systematically shifted to 45, 135, 225 or 315 degrees at uniformly spaced intervals. Each phase shift transmits 2 bits of information.

This concept of ASK, FSK and PSK is clearly shown in Fig. 2.5. Many of the times mixed forms are also used. All these concepts are discussed in brief in below section.

2.5.1 Frequency-Shift Keying (FSK)

- In Frequency-Shift Keying (FSK) two sine wave frequencies are used to represent binary logic levels '0' and '1'.
- Binary '0' has a frequency of 1070 Hz and is known as space, whereas binary '1' has a frequency of 1270 Hz and is known as a mark in a data communication system.

- These two frequencies are alternatively transmitted to create the serial binary data as shown in Fig. 2.6.

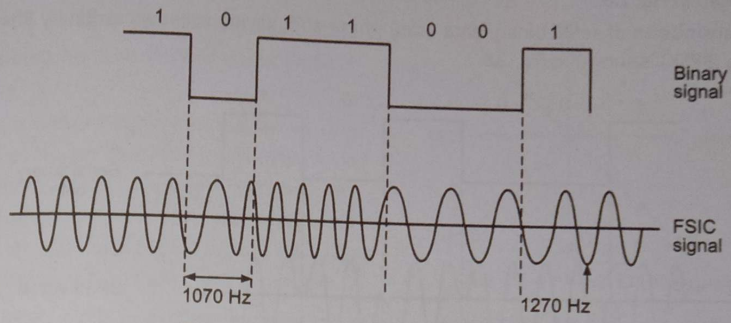


Fig. 2.6 : Frequency-shift keying (FSK)

- Here the two frequencies, 1070 Hz corresponding to '0' and 1270 Hz corresponding to '1' are within 300 Hz and 3000 KHz bandwidth generally associated with telephone network line.
- Transmitter block diagram of FSK is shown in Fig. 2.7.

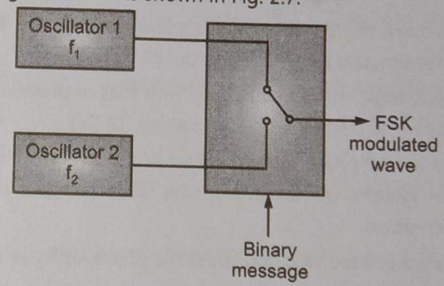


Fig. 2.7 : FSK transmitter

- The FSK transmitter uses two oscillators, each generating a distinct frequency, one for binary '1' and the other for binary '0'.
- These oscillators are controlled by an internal clock, which ensures that both signals remain synchronized. This synchronization is essential to prevent abrupt phase discontinuities in the output waveform when switching between frequencies during data transmission.
- The binary input data stream is applied to a switching circuit that selects the appropriate oscillator output based on whether the input bit is 0 or 1.
- As a result, the transmitted signal alternates smoothly between the two frequencies, representing the digital information without distortion or phase jumps.

2.5.2 Phase-Shift Keying (PSK)

- In Phase-Shift Keying (PSK) the binary signal to be transmitted changes the phase-shift of a sine wave character depending on binary '0' or '1' is transmitted.

- The phase-shift is a time difference between two sine waves of the same frequency as illustrated in Fig. 2.8.
- The transmission of serial binary data using phase-shift keying is known as Binary Phase-Shift Keying (BPSK) is shown in Fig. 2.8.

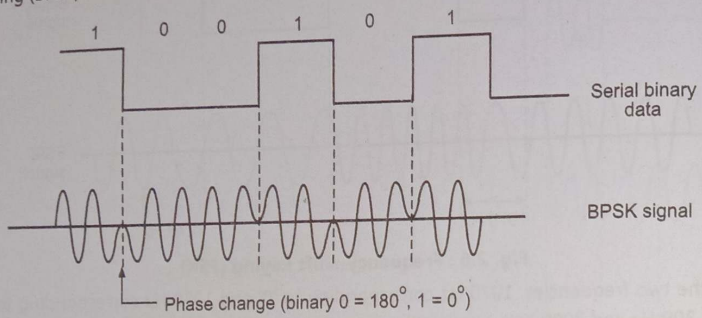
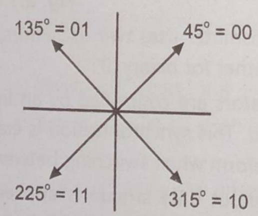


Fig. 2.8 : BPSK signal

- When binary logic '0' occurs, the carrier signal is transmitted in one phase and when binary logic '1' occurs the carrier signal is transmitted with 180° phase-shift.
- Usually, the phase of each logic level is compared with that of previous level rather than with constant reference is known as Delta Phase-Shift Keying (DPSK).
- Most DPSK systems use a four phase system called as Quadrature Phase-Shift Keying, QPSK or DQPSK. Such type of system. QPSK can transmit twice the data in the same bandwidth, than a single bit representation.
- The two-bit sequence is associated with four possible phase shifts as shown in Fig. 2.9 (a) (b) below.

Bit combination	Phase shift
00	45°
01	135°
10	225°
11	315°

(a)



(b)

Fig. 2.9 : QPSK modulation system

2.5.3 Quadrature Amplitude Modulation (QAM)

- Generally, amplitude modulation and phase modulation both are used as carrier and is called as Quadrature Amplitude Modulator or QAM system.

- Here with the phase shift, amplitude of the carrier is also varied with respect to binary data '0' or '1'. A simple 4-bit QAM is shown in Fig. 2.10 below.
- Similarly 8-QAM, 16-QAM signals can be generated resulting multibit transmission at a time.
- QAM is widely used in transmission of data from computer to another destinations via telephone line network within the bandwidth limit of 3 KHz with QAM data rates of 9600 bits/sec, which are normally achieved over the telephone line network.

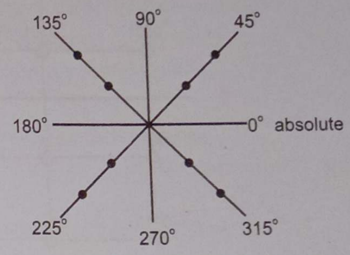


Fig. 2.10 : 4-bit QAM signal

- The BPSK, QPSK and QAM techniques are also widely used to transmit digital data in microwave, satellite communications.
- A variety of form of QAM are available and some common form include 8 QAM, 16 QAM, 32 QAM, 64 QAM, 128 QAM, 256 QAM etc.
- For domestic broadcast application for example, 64 - 256 QAM are used in digital cable TV and cable modem.
- 16 QAM and 64 QAM are currently used for digital terrestrial TV using Digital Video Broadcasting (DVB).
- A 16 QAM signal is generated by encoding 4 input bits at a time. The result is 8 phase shifts and two amplitude levels, for a total of 16 different PM/AM combinations.

Constellation Diagrams :

- A constellation diagram is the representation of a digital modulation scheme on the complex plane.
- The diagram is formed by choosing a set of complex numbers to represent modulation symbols. These points are usually ordered by the gray code sequence. Gray codes are binary sequences where two successive values differ in only one digit.
- The use of gray codes helps to reduce the bit errors. The real and imaginary axes are often called the in-phase and the quadrature. These points are usually arranged in a rectangular grid in QAM, though other arrangements are possible.
- The number of points in the grid is usually a power of two because in digital communications the data is binary.
- Upon reception of the signal, the demodulator examines the received symbol and chooses the closet constellation point based on Euclidean distance. It is possible to transmit more bits per symbols by using a higher-order constellation QAM, but this is more susceptible to noise because the points are closer together, resulting in a higher Bit Error Rate (BER).

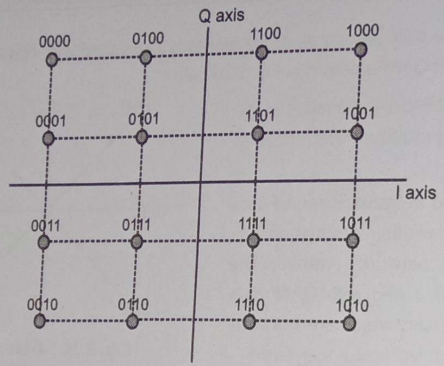


Fig. 2.11 : 16-QAM constellation diagram

- The 4-QAM implementation is easily obtained from the 16-QAM implementation by simplifying the lookup tables for digital to analog and analog to digital conversion as shown in Fig 2.12.

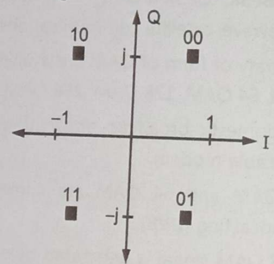


Fig 2.12 : 4-QAM constellation diagram

2.5.4 Amplitude Shift Keying (ASK)

- Amplitude modulation in which the carrier is switched between two different carrier levels is known as amplitude shift keying (ASK).
- A special form of ASK is one in which the carrier is simply switched ON or OFF, is shown in below Fig. 2.13.

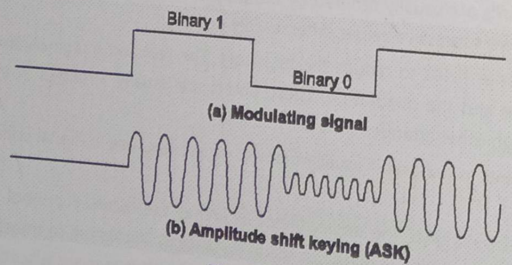


Fig. 2.13 : ASK

Applications of Digital Modulation Techniques :

1. **Amplitude Shift Keying (ASK):**
   - Used in optical fiber communication and low-speed digital data transmission.
   - Suitable for radio-frequency identification (RFID) systems.
   - Used in MODEMs for transmitting digital signals over telephone lines.
2. **Frequency Shift Keying (FSK):**
   - Widely used in radio telemetry, paging systems and wireless modems.
   - Common in low-frequency communication such as caller ID transmission and fax machines.
   - Used in emergency signaling systems and remote-control applications.
3. **Phase Shift Keying (PSK):**
   - Essential in satellite communication, Wi-Fi (IEEE 802.11) and cellular networks.
   - Used in digital television broadcasting and secure data transmission systems.
   - BPSK and QPSK are used in deep-space communication due to their robustness against noise.
4. **Quadrature Phase Shift Keying (QPSK):**
   - A type of PSK that transmits 2 bits per symbol, increasing data rate efficiency.
   - Used in DSL, 3G/4G/5G mobile communication and satellite links.
   - Also used in broadband wireless communication and digital video broadcasting.
5. **Quadrature Amplitude Modulation (QAM):**
   - Combines amplitude and phase modulation for high data-rate transmission.
   - Used in digital cable TV, DSL and broadband modems.
   - Integral to Wi-Fi (802.11n/ac/ax), LTE/5G and high-speed Ethernet systems.
   - Enables efficient use of bandwidth in high-capacity communication systems.

2.6 MULTIPLEXING TECHNIQUES

- Multiplexing is "the process of simultaneously transmitting two or more individual signals over a single communication channel".
- Multiplexing technique has effect of increasing number of communication channels. This facilitates to transfer more information.

Concept of Multiplexing :

- The multiplexers and demultiplexers studied in the last year are the basic elements to implement multiplexing. Fig. 2.14 shows the concept of multiplexing.
- Multiplexers are the circuits having many inputs and only one output, such multiplexers are used for multiplexing.
- Multiple input signals are combined by the multiplexer into a single composite signal which is then transmitted over the communication medium. At the other end of the communication link, a demultiplexer is used to separate the signals into their original form.

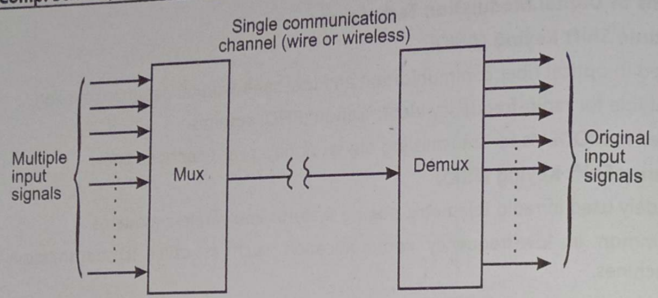


Fig. 2.14: Concept of multiplexing

- So, basically, multiplexing describes how several users can share a medium with minimum or no interference.

Need of Multiplexing :

- If more than one signal has to be transmitted from source to destination, one solution is to provide single communication channel for each. For example, each signal could be sent over a single pair of wires. Using multiple wires over a longer distance is an expensive process.
- In the telephony system there are millions of telephones in the country. If each telephone requires a two-wire path, imagine the number of wires required to make all the various connections.
- To minimize the number of wires, the multiplexing technique is used. In this technique, multiple telephone conversation is combined in such a way that, they can be transmitted over a single pair of wires or a single radio communication channel.
- Another example is, highways with several lanes. Many users (car drivers) use the same medium (the highways) with hopefully no interference (i.e. accident). This is possible due to provision of several lanes separating the traffic or different cars can use the same medium at different points in time, this simple example illustrates our everyday use of multiplexing.

The Main Needs for Multiplexing :

- Multiplexing enables the available bandwidth to be shared among several users or data sources, maximizing the use of the communication medium.
- By transmitting multiple signals through a single link instead of separate ones, multiplexing reduces infrastructure costs, cabling and maintenance.
- It allows many users or data streams to communicate simultaneously, thereby improving the overall capacity of the system.
- Fewer physical connections simplify system design and maintenance, particularly in large communication networks.
- Multiplexing supports the integration of various data types such as audio, video and text into one transmission channel, improving flexibility and scalability.

Multiplexing Techniques :

- For wireless communication, multiplexing can be carried out in four dimensions : space, time, frequency and code.
- The task of multiplexing is assigned specifically with these four aspects to each communication channel with maximum of medium utilization and minimum of interference.
- The term communication channel refers association of sender and receiver who want to exchange data.
- Multiple access schemes are used to allow many mobile users to share simultaneously a finite amount of radio spectrum.
- The sharing of spectrum is required to achieve high capacity by simultaneously allocating the available bandwidth or available number of channels to multiple users.

Basic Types of Multiplexing :

1. Space Division Multiplexing (SDM).
2. Time Division Multiplexing (TDM).
3. Frequency Division Multiplexing (FDM).
4. Code Division Multiplexing (CDM).
5. Orthogonal Frequency Division Multiplexing (OFDM)

- Generally, FDM and TDM systems are widely used. FDM systems are used to deal with analog information and TDM systems are used for digital information.

1. Time Division Multiplexing (TDM) :

- Time division multiplexing may be used with both digital and analog signal.
- In case of TDM, the multiple signals are transmitted in different time slots.
- In TDM, each signal can occupy the entire bandwidth of the channel. However, each signal is transmitted for only short interval of time. In other words, in TDM, signals are transmitted turn by turn.
- Fig. 2.15 shows the basic TDM concept. Here three signals are transmitted over a single channel.
- Each signal is allowed to use the channel for a fixed period of time one after the another. Once all the signals have been transmitted, the cycle repeats again and again.

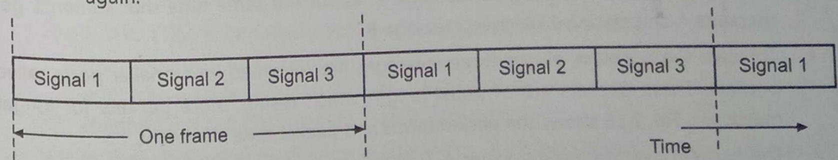


Fig. 2.15: Basic TDM concept

- Equal time slots are given for each signal. Signal transmission of each signal completes one operation called a frame.
- The cycle repetition rate is kept high; so that the original signal can be reassembled at the receiving end.
- In case of conventional method of data transfer, the base band signal is transmitted one after the other. Suppose three messages A, B and C of different length have to be transmitted, message A is transmitted first, followed in time by message B and then message C. Fig. 2.16 shows conventional transmission.

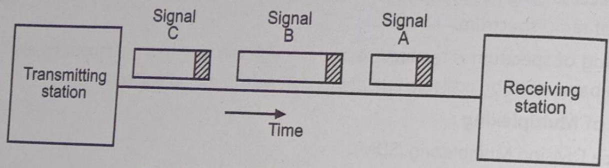


Fig. 2.16 : Conventional transmission

In case of conventional transmission, signal B has to wait till A gets transmitted and signal C has to wait till A and B get transmitted.

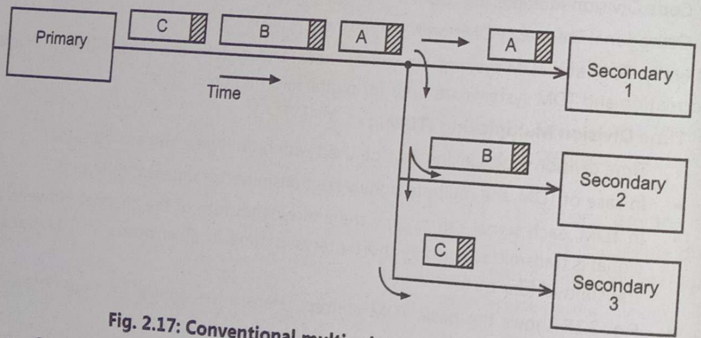
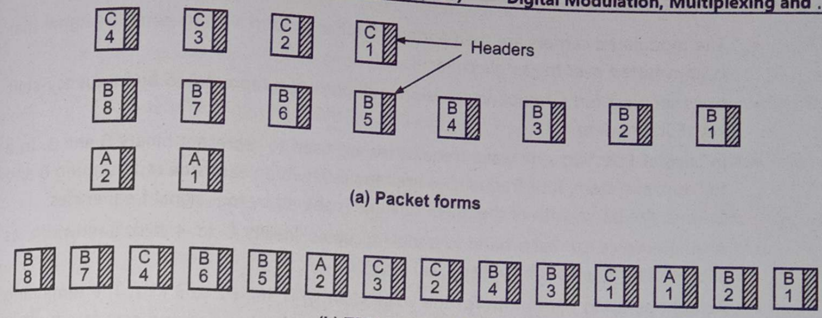


Fig. 2.17: Conventional multipoint communication

- Consider the situation of multipoint system as shown in Fig. 2.17, where the messages are destined for different secondaries. Secondary section 3 in Fig. 2.17 would like to start receiving its message at about the same time that station 1 got message A and station 2 received message B.
- To solve the problem, all the three messages are reformed into smaller ports called packets. These packets are of equal length which results more packets for longer messages. Fig. 2.18 shows the packet forms and packet time allotment.
- The packets constituting messages A, B and C are interleaved and assigned time slots as shown in Fig. 2.18.



(a) Packet forms

(b) TDM time slot frame

Fig. 2.18: Packet time allotment

- Each packet has a shaded area known as header which contains address and packet number information.
- The interleaved packets are transmitted and received by the secondaries. The appropriate packets which are determined by destination address in the header are extracted by each station as they are received and reassembled into their original message form.
- There are two forms of time division multiplexing : Synchronous TDM (STDM) and Asynchronous TDM (ATDM). STDM system assigns time slots of equal length to all packets regardless whether or not anything is to be sent by each station with an assigned time slot.
- The STDM is relatively easy to implement since the software allocates the time slots.
- The ATDM, also known as STAT MUX systems, are more complex. Here time slots are needed to be reassigned.
- The STAT MUX networks assign time slots only when they are to be used and delete them when they are idle. The total time used for a STAT MUX frame varies with the amount of traffic currently being handled.

2. Frequency Division Multiplexing (FDM) :

- Frequency division multiplexing (FDM) is based on the idea that the number of signals can share the bandwidth of common communication channel.
- In case of telephony, the telephone system is using the frequency range 300 Hz to 3 KHz. FDM allows many voice or base band channels (300 Hz to 3 KHz) to share space in much larger total band. This band of channels is then sent via radio, or fiber optic cable to a receiving station, where the band is demultiplexed into individual channels and are destined for specific end user.
- In case of frequency division multiplexing, each base band signal is modulated on separate carrier. Each carrier has different frequency.

- The modulated carriers are then added together to form a single complex signal that is transmitted over the single channel.
- In its simplest form, frequency division multiplexing is represented by frequency-shift key (FSK) modem.
- In case of FSK, two sine wave frequencies are used to represent binary 0 and 1. In a full duplex system, four frequencies are required, two on each side representing 0 and 1. Thus, the bandwidth of the telephone line is shared by four signal frequencies.
- A single voice or base band channel occupies ideally 0 to 4 KHz bandwidth as allocated for a telephone subscriber circuit.
- Actual operating bandwidth for the voice channel is 300 Hz to 3 KHz. The remaining band area i.e. 0 to 300 Hz and 3 KHz to 4 KHz remains unused and acts as a built-in guard band.
- Guard band is useful to avoid adjacent channel effect. A guard band is an area that acts as a buffer between adjacent channels to avoid one channel's data from crossing over into another channel.
- The first channel in FDM system occupies the frequency band from 0 to 4 KHz. A second channel could then begin at 4 KHz and end at 8 KHz.
- The second band also has 4 KHz bandwidth and contains the same guard band area. The total bandwidth of two channels will be 8 KHz.
- The original information signal such as sound (300 Hz to 3 KHz) is known as base band signal. Those systems that allow many channels to share the system's bandwidth are known as broad band systems.
- By definition, broad band is a communication channel having a bandwidth greater than a voice grade channel and is capable of much higher transmission rates.
- All voice grade channels have a bandwidth of 0 to 4 KHz. In order to transmit as a part of broad band signal, the voice channel is modulated on some carrier frequency.
- A mixer is used for this purpose. A mixer is a non-linear circuit. If mixer input frequencies are f_s and f_c, the output frequencies are f_s, f_c, $f_c - f_s$ and $f_c + f_s$. By using a filter circuit, it is possible to select only difference output frequency of mixer.
- Fig. 2.19 shows the principle of mixer block.

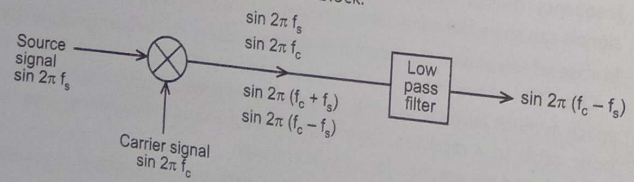


Fig. 2.19: Principle of mixer

3. Code Division Multiplexing (CDM) :

- Space Division Multiplexing (SDM) and Frequency Division Multiplexing (FDM) are most widely used schemes in a radio application from the early days, while Time Division Multiplexing (TDM) is used with many applications.
- In a commercial communication system, one of the relatively new schemes, referred as Code Division Multiplexing is used.
- At the beginning, CDM was used in military applications due to its inherent security feature, now-a-days it is also used in many civil wireless transmission processes.

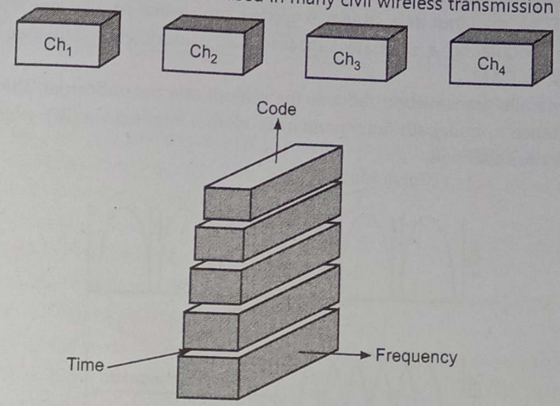


Fig. 2.20 : Code Division Multiplexing (CDM)

- For example, Fig. 2.20 shows all channels Ch 1 to Ch 4, use the same frequency at the same time for transmission.
- The separation is achieved by assigning a separate code to each channel and guard spaces are realized by using code with the necessary distance in a code space.
- The main advantage of CDM for wireless transmission is that, it gives good protection against tapping and interference. Different codes have to be assigned, but code space is huge compared to the frequency space.
- Assigning individual code to each sender do not cause problem. The disadvantage of this scheme is the relatively high complexity of the receiver.
- A receiver has to know the code and must be precisely synchronized with the transmitter to apply the decoding correctly.

4. Orthogonal Frequency Division Multiplexing (OFDM) :

- Orthogonal Frequency Division Multiplexing (OFDM) is an advanced multiplexing technique used in modern digital communication systems to achieve high data rates and robust transmission over bandwidth-limited channels.

- It is a special form of Frequency Division Multiplexing (FDM) in which multiple closely spaced subcarriers are transmitted simultaneously, each carrying a portion of the total data stream as shown in Fig. 2.21.
- The key feature of OFDM is the orthogonality of subcarriers, meaning that, even though the subcarriers overlap in the frequency domain, they do not interfere with each other because their peaks and nulls are mathematically aligned.
- This orthogonal relationship allows efficient utilization of bandwidth and minimizes inter-carrier interference (ICI).
- In OFDM, the input data stream is divided into several parallel low-rate data streams, each modulating a separate subcarrier using modulation schemes such as BPSK, QPSK, or QAM.
- The parallel transmission reduces the symbol rate per subcarrier, thereby increasing resistance to multipath fading and inter-symbol interference (ISI), which are common in wireless channels.

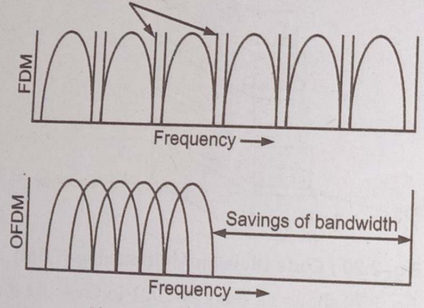


Fig 2.21 : FDM and OFDM

- Thus, in communication systems, OFDM has become an alternative to single-carrier modulation techniques such as Frequency Division Multiple Access (FDMA), Time-Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA).

2.7 SPREAD SPECTRUM TECHNIQUES

- Spread spectrum is modulation and multiplexing technique that distributes the signal and side band over a very bandwidth.
- As the name implies, Spread Spectrum (SS), spreading bandwidth needed to transmit data. The main advantage is the resistance to narrow band interference.
- It was developed primarily after World War II and was used in military communication application because of its secure communication techniques, i.e. immune to jamming.
- A special version of SS known as Code Division Multiple Access (CDMA) is now widely used in the newer digital cell phone. Some cordless phones also use SS. It offers numerous benefits so that it is used in many applications. The following Fig. 2.22 illustrates the concept of spread spectrum.

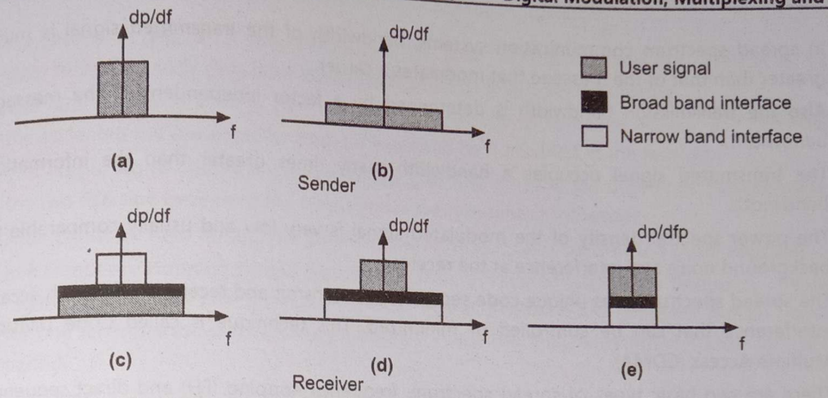


Fig. 2.22: Spread Spectrum (SS); spreading and despreading

- Fig. 2.22 (a) shows an ideal narrow band signal from sender to user data. (All figures are power density dp/df versus frequency f).
- The sender now spread the signal in Step - II as shown in Fig. 2.22 (b), i.e. it converts the narrow band signal into a broad band signal.
- The energy needed to transmit the signal (the area shown in diagram) is the same, but it is now spread over a larger frequency range.
- The power level of the spread signal can be much lower than that of the original narrow band signal without losing data.
- During the transmission, narrow band and broad band interference add to the signal in a step - III as shown in Fig. 2.22 (c).
- The sum of interference and user signal is received. The receiver now knows how to despread the signal, converting the spread user signal into a narrow band signal again, while spreading the narrow band interference and leaving the broad band interference as shown in Fig. 2.22 (d).
- In Fig. 2.22 (e), the receiver applies a bandpass filter to cut-off frequencies left and right of the narrow band signal. Finally, receiver can reconstruct the original data because the power level of the user signal is high enough i.e. the signal is much stronger than the remaining interferences.
- In radio communication and telecommunication, spread spectrum techniques are methods by which a signal to be transmitted in electrical, electromagnetic or acoustic form, generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth.
- Advantages of spread spectrum techniques are like establishment of secure communications, increasing resistance to natural interference noise, immunity of jamming signals, can hide signals, limits power flux density and improves multiple access communications.

- In spread spectrum communication systems, bandwidth of the transmitted signal is much greater than that of the message that modulates a carrier.
- Also the transmission bandwidth is determined by a factor independent of the message bandwidth.
- The transmitted signal occupies a bandwidth many times greater than the information bandwidth.
- The power spectral density of the modulated signal is very low and usually comparable to background noise and interference at the receiver.
- The spread spectrum uses unique code sequences to transmit and receive signals with access interference that can be controlled or minimized. This technique is called Code Division Multiple Access (CDMA).
- There are two basic types of spread spectrum, frequency hopping (FH) and direct sequence (DS).

2.7.1 Frequency Hopping Spread Spectrum

- Frequency hopping spread spectrum (FHSS) is a method of transmitting radio signals by rapidly changing the carrier frequency among many distinct frequencies occupying a large spectral band.
- In frequency hopping spread spectrum, signals are highly resistant to narrowband interference because the signal hops to a different frequency band.
- Frequency hopping spread spectrum (FHSS) transmission is the repeated switching of frequencies during radio transmission to reduce interference and avoid interception.
- Fig. 2.23 shows block diagram of frequency hopping spread spectrum (FHSS) transmitter.

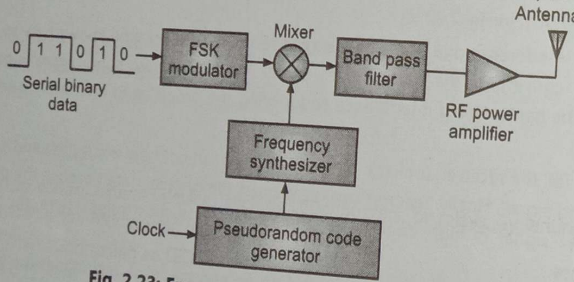


Fig. 2.23: Frequency hopping SS transmitter

- The serial binary data to be transmitted is applied to a conventional two tone FSK modulator.
- The modulator output is applied to a mixer.
- The synthesizer is driven by a digital circuit pseudorandom code generator.
- The pseudorandom code is a serial pattern of binary '0' and '1' that changes in a random fashion.

- As shown in Fig. 2.23, the pseudorandom code of the "1" and "0" generated from pseudorandom code generator drives mixer through frequency synthesizer.
- The randomness is enough to minimize the possibility of someone accidentally duplicating the code, but the predictability allows the code to be duplicated at the receiver.
- The output signal from the bandpass filter after the mixer is the difference between one of the two FSK sine waves and the frequency of the frequency synthesizer.
- After bandpass filter through RF power amplifier, the signal is transmitted by an antenna.
- In a Frequency Hopping Spread Spectrum system, the rate of synthesizer frequency change is higher than the data rate. Although the data bit and the FSK tone it produces remain constant for one data interval, the frequency synthesizer switches frequencies many times during this period.
- The time that the synthesizer remains on a single frequency is called the dwell time.
- The actual dwell time on any frequency varies with the application and data rate.
- As per Federal Communications Commission (FCC) regulation specifications there should be a minimum of 75 hopping frequencies and that the dwell time not exceed 400 µs.
- Fig. 2.24 shows frequency hopping spread spectrum receiver.

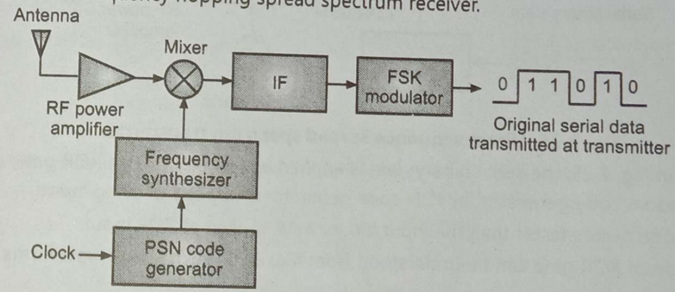


Fig. 2.24: Frequency hopping spread spectrum receiver

- The wideband signal is picked up by the antenna.
- Then it is feed up through a broadband RF amplifier and mixer to IF amplifier, as shown in Fig. 2.24.
- In this receiver, mixer is driven by a frequency synthesizer which is driven by PSN code generator same as in FHSS transmitter.
- Frequency synthesizer at the receiving end must have the same pseudorandom code sequence as that generated by the transmitter in order to receive the signal on the correct frequency.
- The signal is reformed through an IF amplifier to obtain the original FSK data.
- The IF signal is then applied to an FSK demodulator, which reproduces the original binary data train which was transmitted at FHSS transmitter.

Advantages of Frequency Hopping Spread Spectrum:

- FHSS receiver is used to acquire and synchronize the transmitted signal with the internally generated pseudorandom code.
- In FHSS, due to preamble signal and code at the beginning of the transmission, the two codes do not step with each other. Once synchronization has been established, the code sequences occur in step. Hence, FHSS technique is extremely secure and reliable. Any receiver not having the correct code cannot receive the signal.
- FHSS permits more signals to be packed in a given band than any other type of modulation or multiplexing.
- It permits a transmitter to selectively transmit to a single receiver without other receivers in the band being able to pick up the signal.

2.7.2 Direct Sequence Spread Spectrum

- Fig. 2.25 below shows block diagram of Direct Sequence Spread Spectrum transmitter (DSSS).

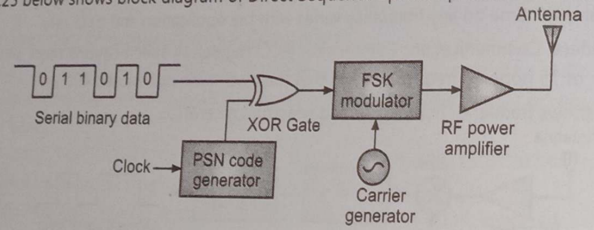


Fig. 2.25: Direct sequence spread spectrum transmitter

- As shown in Fig. 2.25, the serial binary data is applied to one input of an XOR gate, and serial pseudorandom code generated by PSN code generator is applied to other input.
- The PSN code occurs faster than the input binary data applied at XOR input.
- The function of XOR gate can be understood from Fig. 2.26 which shows waveforms operated at XOR gate.

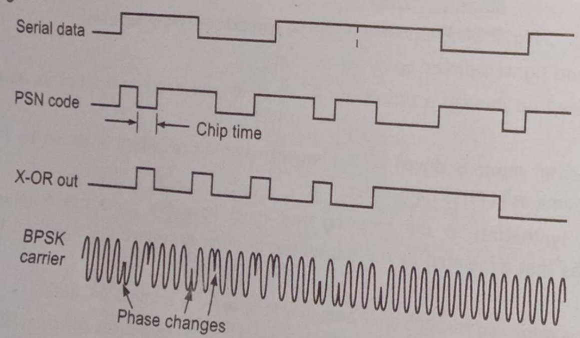


Fig. 2.26 : Waveform at XOR gate

- As shown in PSN code, One bit time for the pseudorandom code is called a chip and the rate of the code is called the chipping rate.
- As mentioned before, the chipping rate is faster than the input binary data rate.
- The output signal developed at output of the XOR gate is applied to a PSK modulator.
- As shown in Fig 2.26, output of XOR logic gate is set to logic "1" with serial data and PSN code are out of phase and to logic "0" when they are out of phase.
- The carrier phase is switched between 0° and 180° by the "1" and "0" of the XOR gate output. QPSK and other forms of PSK can also be used.
- The signal phase modulating the carrier is very much higher in frequency than the data signal.
- Hence, the modulator produces multiple, widely spaced sidebands whose strength is such that, the complete signal takes up a great deal of the spectrum.
- Hence, it results in getting the spread signal and provides a type of gain called processing gain to the signal.
- A block diagram of Direct Sequence Spread Spectrum (DSSS) receiver is shown in Fig. 2.27.

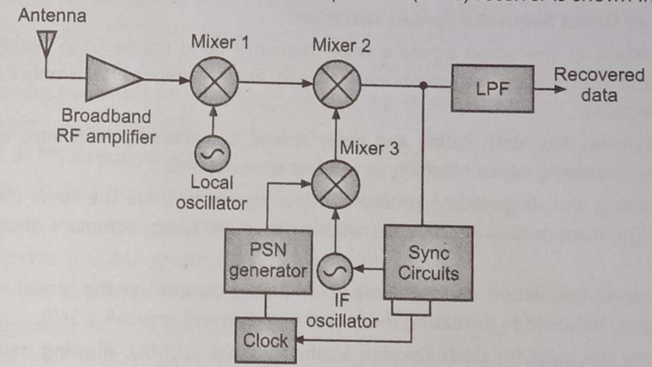


Fig. 2.27: DSSS receiver

- As shown in Fig. 2.27, the broadband spread spectrum signal received through antenna is amplified through broadband RF amplifier.
- This amplified signal is mixed with a local oscillator to translate the signal down to a lower IF in mixer 1 and then connected to mixer 2.
- As shown in Fig 2.27, mixer 3 produces a signal using PSN generated code sequence and mixed with mixer 2 to get another IF signal.
- The output of mixer 3 is identical to the output of mixer 1 but shifted in time. This comparison process, called correlation, takes place in mixer 2.
- If the two signals are identical, the correlation is 100% whereas if they are not alike the correlation is 0.

- The signal out of mixer 2 is connected to a synchronization circuit of the function to regenerate the exact frequency and phase of the carrier, which allows for demodulation to take place.
- The correlation process in the mixer produces a signal that is averaged in the low-pass filter at the output of mixer 2.
- The output signal will be a high average value if the transmitted and received PSN codes are alike.
- The output of mixer 3 is then filtered through low pass filter to recover the original serial binary data.
- The received signal at this level is called as despread signal.
- Direct Sequence Spread Spectrum is also called Code Division Multiple Access (CDMA) system.

Thus, Direct Sequence Spread Spectrum (DSSS) is a spread spectrum technique in which the digital data signal is multiplied by a high-rate pseudo-random (PN) sequence, spreading the signal energy over a much wider bandwidth than the original data.

Advantages of Direct Sequence Spread Spectrum:

1. The spreading of the signal across a wide frequency band makes DSSS highly resistant to narrowband interference and intentional jamming, as interference affects only a small portion of the spectrum.
2. DSSS minimizes multipath fading and delay spread by spreading reflections over time and frequency, improving signal reliability in wireless environments.
3. The spreading and despreading processes allow signals to share the same frequency band without significant mutual interference, enabling multiple-access communication (as in CDMA systems).
4. At the receiver, correlation with the same PN sequence compresses the spread spectrum back to its original bandwidth, increasing the processing gain and improving SNR.
5. DSSS forms the basis for Code Division Multiple Access (CDMA), allowing multiple users to transmit simultaneously within the same frequency band using distinct spreading codes.
6. Since the spreading code (PN sequence) appears as random noise to unintended receivers, DSSS provides a level of signal concealment and data security, making interception and decoding difficult without knowledge of the code.

2.7.3 Pseudo-random (PN) Sequence

- A Pseudo-Random (PN) Sequence is a deterministic binary sequence that appears randomly, but can be reproduced exactly when the initial conditions or seed are known.
- It plays a crucial role in spread spectrum communication systems, particularly in Direct Sequence Spread Spectrum (DSSS) and Code Division Multiple Access (CDMA) techniques.
- A PN sequence consists of a series of binary digits (1s and 0s) generated by a Linear Feedback Shift Register (LFSR) using a specific feedback logic.

- Although the sequence is generated algorithmically, it exhibits statistical properties similar to those of a truly random sequence such as an equal number of ones and zeros and minimal correlation between successive bits.
- In summary, a PN sequence is used to spread the transmitted signal across a wide frequency band at the transmitter and despread it at the receiver. It is predictable, yet noise-like characteristics make it essential for achieving secure, interference-resistant and multiple-access communication in modern digital systems such as CDMA, GPS and DSSS.

The Key Properties of PN Sequences :

1. A PN sequence can be exactly reproduced at the receiver using the same initial seed and generator polynomial as the transmitter.
2. Despite being generated deterministically, PN sequences exhibit randomness-like characteristics, making them suitable for spreading and despreading operations.
3. Over one complete cycle, the number of 1s and 0s in the sequence are nearly equal, ensuring signal balance.
4. PN sequences have desirable autocorrelation properties, meaning that, the correlation is high only when the sequences are aligned and low otherwise. This property allows the receiver to detect and synchronize the desired signal even in the presence of interference.
5. PN sequences generated by an n-stage shift register repeat after $(2^n - 1)$ bits, providing long, non-repetitive spreading codes.

Applications of PN Sequences:

- Spreading and despreading in DSSS communication.
- Synchronization between transmitter and receiver.
- Multiple access in CDMA systems using different PN codes.
- Encryption and secure communication.
- Error detection and correction techniques.

Exercise

[A] Multiple Choice Questions:

[I] Choose the most correct alternative for each of the following and rewrite the sentence.

1. Nyquist Sampling Theorem states that, a continuous-time signal can be reconstructed if it is sampled at a rate at least ....
   (a) Equal to the maximum frequency of the signal
   (b) Twice the maximum frequency of the signal
   (c) Half the maximum frequency of the signal
   (d) Four times the maximum frequency of the signal

2. In Pulse Code Modulation (PCM), which process converts discrete-time samples to a binary sequence?
   (a) Sampling (b) Quantization
   (c) Encoding (d) Filtering
3. Which of the following is an advantage of PCM?
   (a) High noise immunity (b) Low bandwidth requirement
   (c) Simple implementation (d) Minimal quantization error
4. Adaptive Delta Modulation (ADM) adjusts the step size based on ....
   (a) Signal frequency only (b) Amplitude of the first sample
   (c) Rate of change of the input signal (d) Number of quantization levels
5. In Amplitude Shift Keying (ASK), the digital information is represented by ....
   (a) Varying the carrier phase (b) Varying the carrier amplitude
   (c) Varying the carrier frequency (d) None of the above
6. In 4-QAM, each symbol represents ....
   (a) 1 bit (b) 2 bits
   (c) 3 bits (d) 4 bits
7. Which of the following is an application of FSK?
   (a) Wi-Fi networks (b) Caller ID and telemetry systems
   (c) Satellite television (d) CDMA mobile networks
8. The main purpose of multiplexing is to ....
   (a) Reduce noise in a channel
   (b) Combine multiple signals for transmission over a single channel
   (c) Increase the amplitude of the signal
   (d) Modulate the carrier with multiple frequencies
9. Time Division Multiplexing (TDM) works by ....
   (a) Assigning separate frequency bands to each signal
   (b) Transmitting signals sequentially in time slots
   (c) Using pseudo-random codes for spreading
   (d) Varying the carrier amplitude
10. Orthogonal Frequency Division Multiplexing (OFDM) is primarily used to ....
   (a) Increase noise immunity in wired networks
   (b) Transmit multiple low-rate data streams simultaneously using orthogonal subcarriers
   (c) Encrypt the transmitted signal
   (d) Reduce transmission power

11. Spread Spectrum techniques improve ....
   (a) Data rate only
   (b) Signal security and resistance to interference
   (c) Signal amplitude
   (d) Bandwidth efficiency only
12. In Frequency Hopping Spread Spectrum (FHSS), the carrier ....
   (a) Remains fixed
   (b) Changes amplitude randomly
   (c) Changes frequency according to a pseudo-random sequence
   (d) Is modulated using PCM
13. Direct Sequence Spread Spectrum (DSSS) spreads the signal using ....
   (a) Fast Fourier Transform (b) Pseudo-random (PN) sequences
   (c) Time-division slots (d) Orthogonal subcarriers
14. A Pseudo-Random (PN) sequence is ....
   (a) Truly random and unpredictable
   (b) A sequence of zeros only
   (c) A sequence of ones only
   (d) Deterministic but appears random to unauthorized users

Answers

1. (b)	2. (c)	3. (a)	4. (c)	5. (b)
6. (b)	7. (b)	8. (b)	9. (b)	10. (b)
11. (b)	12. (c)	13. (b)	14. (d)	15. (a)

[II] Fill in the blanks :

1. According to the Nyquist Sampling Theorem, a signal must be sampled at a rate at least .... times its maximum frequency.
2. In PCM, the process of approximating the sampled signal to the nearest discrete level is called ....
3. The PCM stage that converts quantized samples into binary code is known as ....
4. Delta Modulation suffers from .... distortion when the step size is too small to track rapid signal changes.
5. In Adaptive Delta Modulation, the step size .... according to the rate of change of the input signal.
6. In Amplitude Shift Keying (ASK), digital data is transmitted by varying the .... of the carrier signal.
7. Frequency Shift Keying (FSK) represents digital bits by varying the .... of the carrier signal.

8. Phase Shift Keying (PSK) transmits digital information by changing the .... of the carrier signal.
9. In 4-QAM, each symbol represents .... bits and is represented using a Constellation diagram.
10. The primary purpose of multiplexing is to allow .... signals to share a single communication channel.
11. In Time Division Multiplexing (TDM), different signals are transmitted in different ....
12. Orthogonal Frequency Division Multiplexing (OFDM) divides the data stream into multiple low-rate streams and transmits them on .... subcarriers.
13. Frequency Division Multiplexing (FDM) assigns each signal a separate .... band within the channel.
14. In Frequency Hopping Spread Spectrum (FHSS), the carrier frequency changes according to a .... sequence.

Answers

1. 2	2. Quantization	3. Encoding
4. Slope overload	5. varies/adapts	6. Amplitude
7. Frequency	8. Phase	9. Two
10. Multiple	11. Time slots	12. Orthogonal
13. Frequency	14. Pseudo-random	

[III] True/False :

1. According to the Nyquist Sampling Theorem, the sampling rate must be at least equal to the maximum frequency of the signal.
2. In PCM, quantization is the process of converting discrete-time samples into binary code.
3. PCM provides high noise immunity compared to analog transmission.
4. Delta Modulation can suffer from slope overload distortion if the step size is too small.
5. In Adaptive Delta Modulation (ADM), the step size remains constant regardless of signal variation.
6. ASK transmits digital data by varying the amplitude of the carrier signal.
7. FSK represents digital data by changing the phase of the carrier.
8. 4-QAM uses a constellation diagram with four points, each representing one bit.
9. The main purpose of multiplexing is to transmit multiple signals over a single communication channel.
10. In TDM, each signal is assigned a unique frequency band.
11. OFDM transmits multiple low-rate data streams on orthogonal subcarriers to improve bandwidth efficiency.
12. FDM assigns separate time slots to each signal in a channel.
13. Spread spectrum techniques improve signal security and resistance to interference.

14. In FHSS, the carrier frequency remains fixed throughout transmission.
15. DSSS spreads the data signal using a pseudo-random (PN) sequence.
16. A pseudo-random (PN) sequence is truly random and cannot be reproduced at the receiver.

Answers

1. False	2. False	3. True	4. True	5. False	6. True
7. False	8. False	9. True	10. False	11. True	12. False
13. True	14. False	15. True	16. False		

[B] Short Answer Questions :

1. State the Nyquist Sampling Theorem.
2. What is the main purpose of Pulse Code Modulation (PCM)?
3. List the main stages of a PCM transmitter.
4. Define Delta Modulation and mention one of its limitations.
5. How does Adaptive Delta Modulation (ADM) improve upon standard Delta Modulation?
6. Explain the basic principle of Amplitude Shift Keying (ASK).
7. What parameter of the carrier changes in Frequency Shift Keying (FSK)?
8. How many bits are represented by each symbol in 4-QAM?
9. Give one application of Phase Shift Keying (PSK).
10. Why is multiplexing necessary in communication systems?
11. What is the difference between FDM and TDM?
12. Briefly explain the concept of OFDM.
13. Mention one advantage of Time Division Multiplexing (TDM).
14. What is the main advantage of Spread Spectrum (SS) techniques?
15. Explain the basic concept of Frequency Hopping Spread Spectrum (FHSS).
16. How does Direct Sequence Spread Spectrum (DSSS) spread the signal?
17. What is a Pseudo-Random (PN) sequence and why is it used in SS systems?

[C] Long Answer Questions :

1. Explain the Nyquist Sampling Theorem. Why is it important in digital communication systems?
2. Draw transmitter and receiver block diagrams of PCM. Explain its working principle and explain how the original signal is reconstructed at the receiver.
3. Discuss the advantages, disadvantages and applications of PCM in digital communication.
4. Explain Delta Modulation (DM). What are its limitations and how does it differ from PCM?
5. Describe Adaptive Delta Modulation (ADM). How does it overcome the limitations of standard Delta Modulation?
6. Explain Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK) and Phase Shift Keying (PSK) with respect to concept, waveform and applications.

7. Discuss QPSK (Quadrature Phase Shift Keying), including the transmitter block diagram, working, waveform representation and applications.
8. Explain 4-QAM (Quadrature Amplitude Modulation), including phaser diagram, constellation diagram and its applications.
9. Compare ASK, FSK, PSK, QPSK and QAM in terms of bandwidth efficiency, noise immunity and application areas.
10. What is the necessity of signal multiplexing in communication systems? Give examples.
11. Explain the working of Frequency Division Multiplexing (FDM), Time Division Multiplexing (TDM) and Code Division Multiplexing (CDM) with suitable diagrams.
12. Explain the concept and working of Orthogonal Frequency Division Multiplexing (OFDM). Explain the use of orthogonal subcarriers, FFT/IFFT processing and cyclic prefix.
13. Discuss the advantages and applications of OFDM in modern communication systems.
14. What is Spread Spectrum (SS)? Explain its importance in communication systems.
15. Describe Frequency Hopping Spread Spectrum (FHSS), including the basic principle, working and advantages.
16. Explain Direct Sequence Spread Spectrum (DSSS). How does the PN sequence spread the signal and what are its advantages?
17. What is a Pseudo-Random (PN) sequence? Explain its properties, generation and applications in DSSS and FHSS systems.
18. Compare FHSS and DSSS in terms of working principle, advantages and typical applications.

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Syllabus ...

1. Introduction to Communication System (6 Hours)

- **Introduction to Communication System:** Elements of digital communication system (block diagram and explanation).
- **Characteristics of Communication Channel:** Signal, Signal types, Signal bandwidth, Channel bandwidth, Signal to noise ratio, Noise figure, Data rate, Baud rate, Channel capacity, Shannon-Hartley theorem. (Definition only).
- **Signal encoding:** Types of signal encoding formats, M-ary coding (Concept level),
- **Error Handling Codes:** Necessity of error control codes, Types of error handling codes, Hamming code (Error detection and correction).
- **Modulation and Demodulation:** Definition of modulation and demodulation, Need of modulation, Classification of Modulation.

2. Digital Modulation, Multiplexing and Spread Spectrum Techniques (8 Hours)

- **Pulse Modulation:** Nyquist sampling theorem, PCM (Transmitter and receiver block diagram, Advantages, disadvantages and application), Concept of Delta modulation and Adaptive delta modulation.
- **Digital Modulation Techniques:** ASK, PSK (Concept, waveform and application), FSK, QPSK, (Transmitter end block diagram, working, waveforms, application), 4-QAM (Phaser Diagram, constellation diagram and application.)
- **Multiplexing Techniques:** Necessity of signal multiplexing, FDM, TDM, CDM, OFDM (Conceptual diagram and working).
- **Spread Spectrum Techniques:** Introduction to Spread Spectrum (SS), Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS), Pseudo-random (PN) sequence.

3. Cellular and Satellite Communication (8 Hours)

- **Cellular Communication:** Cell and cellular telephony, Frequency reuse and hand-off, LTE, UMTS, 4G, 5G architecture network, Handovers in 5G, Future generation 6G.
- **Types of Antennas:** Working principle of dipole antenna and patch antenna.
- **Concept of Smart Antennas:** Importance and block diagram of MIMO, Concept of MU-MIMO and Massive MIMO.
- **Satellite Communication:** Segments, Orbits, Uplink and downlink (Block diagram and frequencies), and Applications.

4. Modern Communication Technology (8 Hours)

- **Wireless Sensor Network:** Sensing & Actuation (Concept only), WSN Architecture, WSN topologies, Types of nodes (Co-ordinator, Router and End Device).
- **Wireless Communication Protocols:** Bluetooth, Wi-Fi & RFID.
- **Data Acquisition**: Basic of Arduino platform (Pin diagram and significance of each pin), I/O control and data acquisition using Arduino.
- **Introduction of IoT:** Definition, Characteristics, Challenges and IoT applications.

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