CHAPTER 1

Introduction to Communication System

Objectives...

- To understand basic elements of digital communication system.
- To learn various characteristics of communication channel such as signal, signal bandwidth, channel bandwidth, baud rate channel capacity.
- To understand the signal encoding formats, M-ary coding.
- To know the error handing codes like Hamming codes.
- To understand the need of modulation and demodulation.

INTRODUCTION

- Communication is 'the basic process of exchanging information'. Communication is required to convey the thoughts, ideas, feelings, orders, news, etc.
- In initial years, information is transmitted over a short distance, usually through the spoken words. So, it has limitation.
- In early days, electromagnetic waves are used in the visible region for the communication. The idea was to send a light beam from one place to other and then to have its pre-coded interruptions to communicate information. This method was fast which leads to several modern techniques of communication. The only drawback was the range which was limited to line-of-sight distance.
- When electricity was discovered, Electronic communication has started with wire telegraphy. The telegraph was invented in 1844. Bell invented telephone in 1876.
- In 1887, radio was discovered and it was demonstrated in 1895. Radio communication was made possible by the invention of the triode tube. It subsequently becomes even more widely used through the invention and use of the transistor, ICs and other semiconductor devices.
- Well known forms of electronic communications such as the telephone, radio, television have increased our ability to share information. Today, they are major parts of our lives.
- Computers play an important role in communication nationwide or worldwide.
- Communication through computer in the form of internet has made a revolution in the electronic communication.

1.1 INTRODUCTION TO COMMUNICATION SYSTEMS

- Electronic communication systems are classified depending upon the nature of information signal transmitted by the system and the method of data flow.
- Depending upon the nature of signal, there are two types: Analog and Digital communication systems.
- Analog signal is continuous time-varying signal. The signal continuously vary with respect to time. The characteristics of the signal such as amplitude, frequency vary with respect to time. The example of analog signal is audio and video signals.
- Digital signals are represented in discrete values. The example of digital signal is square waveform which has two state, low and high.
- Digital signal is in the form of 0 or 1 which is known as bits. The data is transmitted in 0s and 1s for example 01011001, in the form of 8-bit data.
- Data used in computers is digital where binary codes representing numbers, letters and special symbols are transmitted by wire or radio.
- As digital signal transmission is more convenient and accurate, now-a-days even analog signal such as voice is first converted into digital form using analog to digital converter and then it is transmitted.

1.2 ELEMENTS OF DIGITAL COMMUNICATION SYSTEM

- Digital communication system uses digital data, digital modulation / demodulation techniques.
- The block diagram of a digital communication system is shown below:

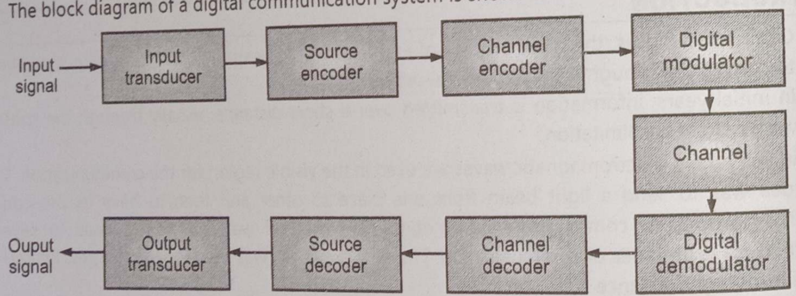


Fig. 1.1 : Digital communication system

- As shown in above block diagram, digital communication system consists of input transducer, source encoder, channel encoder, digital modulator, communication channel, digital demodulator, channel decoder, source decoder and output transducer connected in series.
- Digital communication is generally used as a conversion system from analog to digital.
- The input signal is usually an analog signal which is converted into digital by ADC (Analog to Digital Converter).
- The input transducer converts one form of energy to another. It converts the non-electrical energy to electrical energy and makes it suitable for transmission.

- The source encoder compresses the original data to the reduced number of bits and removes unnecessary bits. Thus, it helps in effective bandwidth utilization. The compressed data is digital data which is in the form of binary digits. Source encoder converts analog data into digital data.
- During the transmission, noise can be added into the original signal which can alter the original data. The channel encoder corrects the errors and enhances the transmission quality of the signal.
- Digital modulator modulates the signal by varying the transmitted signal's frequency, amplitude and phase.
- The signal is transmitted through communication channel. The data rate of the channel is measured in bits per second.
- Digital demodulator demodulates the signal and the source signal is recovered from the carrier signal.
- The channel decoder removes the parity bits which are added for error detection from the binary data and gives pure digital signal with no interference or noise.
- The source encoder works oppositely as that of the source encoder. It converts the binary data back to the waveforms.
- The output transducer converts the electrical signal back into its original form and makes the information suitable for the user at the output end.

Advantages of Digital Communication :

- It is fast, more accurate and more reliable than analog communication.
- The data can be transmitted at a longer distance.
- The error detection and correction are easy.
- The signal transmission speed is high.
- It is used in Image and video processing, Data compression, Digital signal processing, Speech processing, Satellites.

Disadvantages of Digital Communication :

- It consumes high power as the large number of components; higher bandwidth is used.
- Digital communication requires high transmission bandwidth to transmit the signals at high speed.
- The power loss in digital communication is higher than analog communication.

1.3 CHARACTERISTICS OF COMMUNICATION CHANNEL

- There are various characteristics of communication channel which will be discussed in below section :

1. Signal :

- In communication systems, a signal is a physical quantity that varies with time and carries information. It is used to represent data such as voice, video, text, or any other type of information that needs to be transmitted from a source to a destination.
- By definition, a signal is a function of one or more independent variables (such as time or space) that conveys information about the behavior or attributes of some phenomenon.

Signal Types :

- There are two types of signals analog signal and digital signal.

  (i) Analog Signal:
  - Analog signal is a continuous-time signal that varies smoothly over time (e.g., sound waves).
  - Analog signal uses less bandwidth as compared to the digital signal.
  - As amplifier is used in the analog communication system, it reduces the distortion and improves the signal quality.
  - Analog communication uses analog signal, so representation of signals is more accurate due to its continuous nature.
  - Analog signal can be easily modulated and demodulated using Amplitude Modulation (AM) and Demodulation.
  - Analog signals are easy to process as compared to the digital signals.

  (ii) Digital Signal :
  - Digital signal is a discrete-time signal that takes on only a finite number of distinct values (e.g., binary 0s and 1s).
  - Digital signal is fast, more accurate and more reliable than analog communication.
  - Digital data can be transmitted at a longer distance.
  - The error detection and correction are easy with digital signal.
  - The digital signal transmission speed is high.
  - Digital signal is used in Image and video processing, Data compression, Digital signal processing, Speech processing, Satellites.

2. Signal Bandwidth :

- Signal bandwidth refers to the range of frequencies that a signal occupies or requires for transmission.
- It is the difference between the highest and lowest frequencies contained in the signal.

$$\text{Bandwidth (BW)} = f_{high} - f_{low}$$

Where:

f_{high} = Highest frequency component of the signal

f_{low} = Lowest frequency component of the signal

- A narrowband signal has a small bandwidth (e.g., Audio telephone signals: ~3 KHz).
- A wideband signal has a large bandwidth (e.g., HD video signals: Several MHz to GHz).
- Bandwidth determines data rate.
- Higher bandwidth allows faster data transmission. It impacts signal quality and fidelity.
- It affects channel capacity in communication systems (as per Shannon's Theorem).

3. Channel Bandwidth :

- The frequency range between the lowest and highest frequencies that are passed through a component, circuit or system with acceptable attenuation is known as bandwidth for it.

- All transmission channels of any practical interest are of limited frequency bandwidth.
- The limitations arise from the physical properties of the channel or from deliberate limitations on the bandwidth to prevent interference from other sources.
- Most of the data communication systems seek to maximize the amount of data that can be sent on a channel primarily for economic reasons.
- Greater the bandwidth of a channel, the greater amount of information can be sent in a given time.
- Transmission of signals in binary form can acquire considerably more bandwidth than an equivalent analog signal.
- For the transmission of voice a bandwidth of about 3 KHz is sufficient whereas if the music is to be transmitted, a bandwidth of 15 to 20 KHz is required because of high frequencies and harmonics produced in the music with different musical instruments i.e. music contains more information than voice, which requires more bandwidth for transmission.
- A video signal or picture contains more information than voice and requires more bandwidth nearly in between 4 to 6 MHz.
- On the other hand, in digital transmission if higher the number of bits transmitted in a given time, the greater amount of information can be transmitted. But if bit rate is high, wider bandwidth is necessary for transmission of the data.
- Narrowing the bandwidth of a channel will cause the harmonics in the binary pulses to be filtered out.
- Thus, the channel bandwidth must be large enough to pass all harmonics to preserve the information in the digital data.
- The upper cut-off frequency of the channel is approximately equal to the channel bandwidth. It is this bandwidth that determines the information capacity of the channel.

4. Signal to Noise Ratio :

- Signal to noise ratio is a number that indicates the relative strength of the signal and the noise.
- However, in communication, noise is usually expressed as a power because; the received signal is also expressed in terms of power.
- If the signal and noise powers are known, the S/N ratio can be calculated.
- The S/N ratio is defined as, "the ratio of signal power to noise power at the same point".

Thus,

$$\frac{S}{N} = \frac{V_s^2/R}{V_n^2/R} = \left(\frac{V_s}{V_n}\right)^2$$

where,

V_s = Signal voltage

V_n = Noise voltage.

- The S/N ratio is usually expressed in decibels. If a receiver has an input signal power of 1.5 μW and the noise power is 0.2 μW then S/N ratio is,

$$S/N = 10 \log \left(\frac{1.5}{0.2}\right)$$
$$= 10 \log 2$$
$$= 10\ (0.3010) = 3.010 \text{ dB}$$

Usually, an effort is made to keep signal to noise ratio as high as practicable.

5. Noise figure :

- The noise figure NF is defined as, "the ratio of the signal to noise power at the input to the signal to noise power at the output".

$$NF = \frac{\text{Input S/N}}{\text{Output S/N}}$$

- The noise figure can be expressed as a number, but more often it is expressed in decibels.
- The noise figure is the difference in decibels (dB) between the noise output of the actual receiver to the noise output of an ideal receiver with the same overall gain and bandwidth.
- It is a number by which the performance of an amplifier or a radio receiver can be specified, with lower values indicating better performance.

6. Data Rate :

- The speed of data transfer is usually indicated by the number of bits per second (bit/s).
- Most data rates takes place at relatively slow speed (usually several thousand bits per second).
- The serial data rate in bits per second (bits/sec) is the reciprocal of the time duration of one bit (T_b).

$$\text{Data rate (bits/sec)} = \frac{1}{T_b}$$

- If the bit duration is 2 ms, the data rate is,

$$\text{Data rate} = \frac{1}{2 \times 10^{-3}} = 500 \text{ bits/sec}$$

- A bit time of 200 ns produces a data rate of,

$$\text{Data rate} = \frac{1}{200 \times 10^{-9}} = 5{,}000{,}000 \text{ bits/sec} = 5 \text{ MBPS}$$

- If the data rate is known, the bit time can be determined as,

$$T_b = \frac{1}{\text{bits/sec}}$$

- A 128 kbit/sec rate has bit time of

$$T_b = \frac{1}{128 \times 10^3} = 7.81\ \mu s$$

- In the internet connection with landline telephone system, the data at maximum 1200 to 2400 bits/sec can be communicated while on the fiber optics link several million bit/sec data can be transferred.

7. Baud rate :

- Baud rate is another term used to express the data transfer speed in data communication.
- *It is defined as 'the number of signaling elements or symbols that occur in a given unit of time generally 1 second'.*
- If the signaling element or symbol is only in logic levels of '0' or '1', baud rate is equal to the data rate in bits/sec.
- On the other hand, if the signaling element or symbol is of several discrete signals, amplitudes or phase shifts, each of which representing two or more data bits modulation techniques are to be applied.
- Several unique modulation techniques have been developed which use each symbol or baud to represent multiple bits.
- The number of symbol changes per unit of time is the number higher than the straight binary bit rate, but more bits per unit time are transmitted.
- Thus, higher bit rates can be transmitted over transmission line by using modulation techniques.
- An illustration will help to understand more clearly the concept.

Assume that we want to transmit decimal number 188. This can be represented in binary by 8-bit number 10110010. We can transmit binary equivalent of decimal 188 serially as a sequence of equal time interval pulses with binary logic '1' and '0' as shown in Fig. 1.2 below.

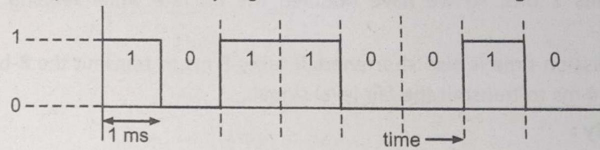


Fig. 1.2: Serial transmission of decimal number 188 (binary 10110010)

- Now, if each bit interval is 1 ms then the bit rate is 1000 bits/sec. The baud rate is also 1000 bits/sec or 1000 baud.
- Now, consider a scheme that represents 2-bit of data as different voltages levels. With four possible combinations of 2-bits we can assign a discrete voltage levels to each combination of 2-bits as below :

2-Bits Combination	Assigned Level
00	0 volt
01	1 volt
10	2 volt
11	3 volt

- Now instead of transmitting a binary signal with only two levels '0' and '1' as shown in Fig. 1.2 above, we transmit one of the four voltage levels 0, 1, 2 or 3 volts to corresponding 2-bit combination 00, 01, 10 and 11 respectively as shown in Fig. 1.3 below.

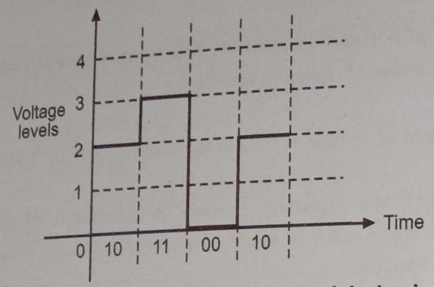


Fig. 1.3: Data transmission using two-bit code of decimal number 188

- Using four level systems, we could divide the word into 2-bit groups and transmit. The appropriate voltage level representing each our number is 10110010. It would be divided into group of

10	11	00	10
2 V	3 V	0 V	2 V

as shown in Fig. 1.3. Each is occurring at fix interval of time say 1 ms.

- Hence, the baud rate is 1000 baud because there is only one symbol per time interval (1 ms).
- However, the bit rate is 2000 bit/s because although only four symbols were transmitted, each symbol represents 2 bits, so we have doubled the bit rate while keeping the baud rate constant.
- The total transmission time is also shortened. It takes 8 ms to transmit the 8-bit binary word, but it only takes 4 ms to transmit the fair level signal.

8. Channel Capacity :

- The channel capacity C expressed in bits per second is twice the channel bandwidth in hertz.

  i.e. $\boxed{C = 2B}$

- The bandwidth B is usually same as the upper cut-off frequency of the channel; if there is no noise.
- The presence of noise reduces the amount of information that can be transmitted in a given bandwidth.
- If the bandwidth is increased to increase information rate, it allows to pass more noise.
- Hence, the choice of a bandwidth is compromise between information transmission rate and the amount of noise that can be transmitted in a given time.
- For example, assume a channel bandwidth of 5 KHz, the maximum theoretical bit capacity is

$$C = 2B = 2\,(5{,}000)$$
$$= 10{,}000 \text{ bits/sec.}$$

- The channel capacity expression can be modified by considering multiple level encoding scheme that permits more bits per symbol or baud to be transmitted.

- Thus, by multilevel scheme of binary number the bit rate can be increased with same baud rate.
- Now, let us think over bandwidth required to transmit the number 188 by the above two methods as binary one and four level one.
- Consider equation of channel capacity

$$C = 2B$$
$$\therefore \quad B = \frac{C}{2}$$

With binary transmission of decimal number 188, bit rate is 1000 bits/sec. Hence, the bandwidth

$$B = \frac{1000}{2} \text{ bits/sec}$$
$$= 500 \text{ Hz}$$

9. Shannon-Hartley theorem :

- The Shannon-Hartley theorem is related to random noise. It gives relation between bandwidth, channel capacity and noise.
- Scientist Claude Shannon derived a formula to prove that maximum capacity of an ideal channel whose only impairments are finite bandwidth and noise are randomly distributed over that finite bandwidth. That formula is

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

where
C = Channel capacity in bits/sec
B = Bandwidth in Hz
S/N = Signal-to-noise power ratio

- Shannon's value of C is normally not achievable because there are numerous impairments in every real channel besides those taken into account in Shannon's law.
- However, Shannon's law provides an upper theoretical limit to a binary channel.
- In this formula due to the nature of the function $\log_2$, the value of C in the formula can be increased more readily by increasing B than by increasing the S/N ratio.
- Now, consider one example, how noise effects the channel capacity and bandwidth. Assume that we are using general telephone network line with its bandwidth 3100 Hz; with S/N ratio 30 dB.
- Now, the maximum channel capacity is twice the channel bandwidth for two level binary system.

$$\therefore \quad C = 2B$$
$$\therefore \quad C = 2\,(3100) \qquad \text{(Now, bandwidth B = 3100 Hz)}$$
$$\therefore \quad C = 6200 \text{ bits/sec.}$$

Now, convert 30 dB S/N ratio into power ratio using conversion formula

$$dB = 10 \log P$$

where P is power ratio

$$\therefore \quad P = \text{antilog}\left(\frac{dB}{10}\right)$$

$$\therefore \quad P = \text{antilog}\left(\frac{30}{10}\right) \qquad \text{(Now, given S/N = 30 dB)}$$

$$\therefore \quad P = \text{antilog } 3$$

$$\therefore \quad P = 1000$$

$$\therefore \quad P = S/N \text{ ratio} = 1000$$

- Calculate channel capacity using Shannon law

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

Here, B = 3100 Hz, S/N = 1000

$$C = 3100 \log_2 (1 + 1000)$$

$$= 3100 \log_2 (1001)$$

$$= 3100 \times 9.97$$

$$\approx 31{,}000 \text{ bits/sec.}$$

- Thus, when noise is present, the channel capacity is changed from 6200 bits/sec to 31,000 bits/sec.
- Therefore, one has to use multilevel coding for transmission of data with channel capacity 31,000 bits/sec and bandwidth 3100 Hz in presence of noise.

1.4 SIGNAL ENCODING

- Signal encoding is the process of transforming data into a signal form suitable for transmission over a communication medium. It ensures that, the digital or analog data is accurately and efficiently represented in a signal (voltage, light, radio wave, etc.) for communication.
- The purpose of signal encoding is to ensure efficient transmission, maintain data integrity, enable synchronization between sender and receiver, to reduce errors and allow error detection/correction and to adapt to the characteristics of the transmission medium.

1.4.1 Types of Signal Encoding Formats

- Signal encoding formats refer to the methods used to represent data (digital or analog) in the form of signals for transmission over a communication channel. These formats vary depending on whether the data and signal are analog or digital.

Types of signal encoding formats are as follows :

1. **Digital-to-Digital Encoding (Line Coding) :**
   - It is used when digital data is transmitted as digital signals. It is essential for synchronization and accurate interpretation of binary data.

Following are the common formats of Digital-to-Digital encoding:

(a) **NRZ (Non-Return to Zero):** It represents '1' with a high voltage and '0' with a low voltage. It is simple, but no synchronization if long strings of 0s or 1s appear.

(b) **NRZ-I (Inverted):** In this, a transition occurs when the bit is '1' and no change for '0'. It has better synchronization than NRZ.

(c) **Manchester Encoding:** Each bit has a transition in the middle. '0' = high-to-low, '1' = low-to-high. It is self-synchronizing and widely used in Ethernet.

(d) **Differential Manchester Encoding:** In this, a transition is always in the middle. The transition at the start for '0'; no transition at the start for '1'. It is more robust to polarity reversal than Manchester.

(e) **4B/5B Encoding:** In this, groups of 4 bits are mapped to 5-bit code words. It ensures sufficient transitions for synchronization.

(f) **Bipolar Encoding (e.g., AMI, B8ZS, HDB3):** It uses three voltage levels as positive, negative and zero. It helps in error detection and synchronization.

2. **Analog-to-Analog Encoding (Analog Modulation) :**
   - It is used when analog data (like audio) is transmitted over an analog medium.

   Common formats are :

   (a) **Amplitude Modulation (AM):** It varies the amplitude of a carrier wave according to the analog signal. It is used in AM radio broadcasting.

   (b) **Frequency Modulation (FM):** It varies the frequency of the carrier wave based on the analog signal. It is used in FM radio, better noise resistance than AM.

   (c) **Phase Modulation (PM):** It varies the phase of the carrier wave according to the signal. It is less common alone but forms part of QAM and PSK.

3. **Digital-to-Analog Encoding (Digital Modulation) :**
   - It is used when digital data must be sent using an analog carrier signal (e.g., wireless or cable transmission).

   Common formats are :

   (a) **Amplitude Shift Keying (ASK):** In this, digital '1' and '0' are represented by different amplitudes. It is simple but sensitive to noise.

   (b) **Frequency Shift Keying (FSK):** It uses different frequencies for binary '1' and '0'. It is more noise-resistant than ASK.

   (c) **Phase Shift Keying (PSK):** It varies the phase of the carrier to represent digital bits. QPSK (Quadrature PSK) is a common variant transmitting 2 bits per symbol.

   (d) **Quadrature Amplitude Modulation (QAM):** It combines amplitude and phase variations to represent multiple bits per symbol. It is used in modern broadband systems like 4G/5G, Wi-Fi, cable TV.

4. **Analog-to-Digital Encoding :**
   - It is used to convert analog data (e.g., voice, music) into digital signals for processing, transmission, or storage.

Common formats are :

(a) **Pulse Code Modulation (PCM):** It samples the analog signal, quantizes the value and encodes it in binary. It is the standard format in audio CDs and telephony.

(b) **Delta Modulation (DM):** It encodes only the difference between successive samples. It is simpler and uses less bandwidth, but may suffer from signal distortion.

5. **M-ary Encoding (Multilevel Modulation) :**
   - It is a type of digital modulation where each symbol represents more than one bit, allowing higher data rates with limited bandwidth.

Common formats are :

(a) **M-ary ASK, FSK, PSK, QAM:** These formats extend ASK, FSK, PSK and QAM by using M distinct signal states (e.g., M = 4, 8, 16, 64). It is more efficient, but requires higher signal quality and complex receivers.

Example: 16-QAM transmits 4 bits per symbol (since $2^4 = 16$). It is used in modern communication like 4G LTE and cable modems.

1.5 M-ARY CODING

- M-ary coding is "a digital modulation technique where each symbol transmitted over the communication channel represents more than one bit of data by using M distinct signal elements (such as amplitude, frequency or phase levels)".
- The term "M" refers to the number of distinct symbols.
- Each symbol represents $\log_2 (M)$ bits.

$$\text{Bits per symbol} = \log_2 (M)$$

- M-ary coding is used to increase data rate without increasing the transmission bandwidth and to improve bandwidth efficiency by sending more bits per signal change.

1.5.1 Types of M-ary Coding / Modulation Formats

1. **M-ary Amplitude Shift Keying (M-ASK) :** It varies amplitude of the carrier. It is simple but sensitive to noise.
2. **M-ary Frequency Shift Keying (M-FSK) :** It uses M different frequencies for different symbols. It is more robust to noise but less bandwidth efficient.
3. **M-ary Phase Shift Keying (M-PSK) :** It uses M different phase shifts of the carrier wave.
   Example: QPSK (M = 4), 8-PSK.
4. **Quadrature Amplitude Modulation (M-QAM) :** It combines amplitude and phase changes. It is highly bandwidth efficient. It is used in modern systems like 4G/5G, Wi-Fi, Digital TV.

Advantages of M-ary Coding:

- Higher spectral efficiency.
- Sends more bits per symbol.
- Reduces symbol rate, which can reduce bandwidth.

Disadvantages:

- Requires higher signal quality (SNR) as M increases.
- More complex transmitters and receivers.
- Higher error probability in noisy channels for larger M.

Examples of M-ary Coding are given in below table:

M Value	Bits per Symbol	Example Format
M = 2	1 bit	Binary ASK, FSK, PSK
M = 4	2 bits	QPSK (Quadrature PSK)
M = 8	3 bits	8-PSK, 8-QAM
M = 16	4 bits	16-QAM
M = 64	6 bits	64-QAM (used in Wi-Fi, 4G)

1.6 ERROR HANDLING CODES

- When digital data is transmitted, bits flow from one point to another hence they are subject to unpredictable changes due to interference creating error at receiving end.
- Communication protocol contains some form of error checking method. The main objective in error detection is to ensure a fully correct transmission of the message.
- Many different methods have been used to ensure 100% accuracy in transmission.

1.6.1 Necessity of Error Control Codes

- In any communication system, errors can occur during transmission due to noise, interference, signal distortion, or channel imperfections.
- Error handling codes are crucial for detecting and correcting these errors, ensuring reliable and accurate data communication.

The following points highlight the importance of error control codes in communication systems :

- Error control codes ensures that, the received data is the same as the transmitted data.
- It prevents corruption of critical information, especially in financial, medical, or control systems.
- Physical transmission mediums (wireless, fiber, cables) are prone to thermal noise, electromagnetic interference, cross-talk. Error handling codes help identify and fix errors caused by noise.
- In wireless and satellite communication, error handling is essential due to long distances, high attenuation and environmental factors.
- Instead of retransmitting the whole message, error correction codes (like Hamming, Reed-Solomon) allow the receiver to correct certain errors without needing retransmission.
- Error detection and correction minimize the need for resending data, which saves bandwidth, reduces latency and increases throughput.
- Digital systems (like computers, mobile phones and the internet) demand high accuracy. Even 1-bit error in a binary instruction or file can lead to malfunction or data loss. So, error handling codes are required.

- Without error handling codes, communication systems would be unreliable, leading to data corruption, loss of information and system failures. These codes are essential building blocks for achieving robust and error-resilient communication.

1.6.2 Types of Error Handling Codes

- Error handling codes are used to detect and/or correct errors that occur during data transmission or storage.

These codes are classified into two broad categories:

1. **Error Detection Codes :** These codes can detect the presence of errors in the received data but cannot correct them. They are usually followed by a request for retransmission.
   Common Error Detection Codes:
   (a) **Parity Bit :** It adds a single bit to data (even or odd parity). It is simple, but detects only single-bit errors.
   (b) **Checksum :** It sums the binary values and appends the result. It is used in protocols like UDP, IP.
   (c) **Cyclic Redundancy Check (CRC) :** It uses polynomial division to generate a checksum. It is very effective at detecting burst errors. It is common in Ethernet, USB, etc.
2. **Error Correction Codes :** These codes detect and correct errors without requiring retransmission. They are useful in real-time and long-distance communications.
   Common Error Correction Codes:
   (a) **Hamming Code :** It can detect and correct single-bit errors and adds redundant parity bits at specific positions.
   (b) **Reed-Solomon Code :** It corrects multiple random or burst errors and used in CDs, DVDs, QR codes, satellite communication.
   (c) **BCH Code (Bose–Chaudhuri–Hocquenghem) :** It is powerful and flexible error correction code. It is used in storage devices and deep space communication.
   (d) **Convolutional Code :** It applies redundancy across multiple bits using shift registers and often decoded with the Viterbi algorithm. It is used in GSM, satellite and deep-space communication.
   (e) **LDPC (Low-Density Parity-Check) Code :** It is very efficient and near-Shannon limit performance and used in Wi-Fi (802.11n/ac/ax), 5G and digital TV.
   (f) **Turbo Code :** It combines two convolutional codes with interleaving. It is used in 3G/4G networks.

1.6.3 Hamming Code (Error Detection and Correction)

- Hamming code is "a set of error correction codes that can be used to detect and correct bit errors that can occur when digital data is transmitted or received".
- To determine exactly where the error is, a sufficient number of bits must be added. The minimum number of Hamming bits are computed with the expression.

$$2^n \geq m + n + 1$$

where
m = number of bits in data word
n = number of bits in Hamming code

Process of Encoding :

- The process of encoding a message using a hamming code by the sender includes 3 steps.
  Step 1: The first step is to calculate the number of redundant bits in a message.
  For example, if a message contains 'm' number of bits and 'n' number of redundant bits are added to the message, then 'mn' indicates (m + n + 1) different states.
  Where (m + n) represents the location of an error in every bit position 1 (extra state) represents no error. Since 'n' indicates 2^n states, which are equal to (m+n+1) states.
  Step 2: Place the redundant bits in exact/correct position.
  ∴ 'n' bits are inserted in the bit positions which are the power of 2 like 1, 2, 4, 8, 16 etc. These bit positions are indicated as n1 (position 1), n2 (position 2), n3 (position 3), etc.
  Step 3: Calculate the values of redundant bits.
  Here parity bits are used to calculate the values of redundant bits. Parity bits can make the number of 1's in a message either even or odd.
  If total number of 1's in a message is even, then even parity is used.
  If total number of 1's in a message is odd, then odd parity is used.

Process of Decrypting :

- The process of decrypting a message received from the sender by the receiver using the hamming code includes the following steps.
- This process is nothing but recalculation to detect and correct the errors in a message.
  Step 1: Count the number of redundant bits.
  The formula to encode the message using redundant bits is,
  $2^n \geq m + n + 1$
  Step 2: Correct the positions of all redundant bits. 'n' number of redundant bits are placed in a bit positions of power of 2 like 1, 2, 4, 8, 16, 32 etc.
  Step 3: Parity checking (odd parity and even parity)
  Parity bits are calculated based on the number of 1's in data bits and redundant bits.
  For example,
  Parity of n1 would be 1, 3, 5, 7, 9, 11, ...
  Parity of n2 would be 2, 3, 6, 7, 10, 11, ...
  Parity of n3 would be 4-7, 12-15, 20-23, ...

The hamming distance between the two different data streams of equal length is number of 1's.

The hamming distance between two data strings of equal length can be calculated by using the XOR operation.

For example, a = 11011001 , b = 10011101.

Hamming distance can be calculated as

11011001 ⊕ 10011101 = 01000100 (number of 1-bits are 2)

The hamming distance indicates the number of 1's in the resultant data stream.

So, d(11011001, 10011101) = 2

1.7 MODULATION AND DEMODULATION

- In a communication system, information signal such as voice, video or binary data is transmitted from source to the destination over some communication medium or channel. For example, voice signals are transmitted using wires in the telephone systems, twisted pair cables are used to carry binary data from one point to another in a computer network. However, when transmission distances are far, cables are sometimes impractical. In such cases, radio or wireless communication is used.
- In a radio system, an electromagnetic wave travels from transmitter to the receiver through space and antennas are required at both the ends for the purpose of coupling the transmitter and the receiver to the space link. Antennas convert electrical energy into electromagnetic waves and vice versa.

1.7.1 Definition of Modulation and Demodulation

Modulation :

- Modulation is defined as 'a process in which some characteristics of high frequency wave (known as carrier) such as amplitude, frequency and phase is altered in accordance with the instantaneous value of some other signal called the 'modulating signal'.
- The information signal which has to be transmitted is usually called the modulating signal, it may be voice, video or binary data.
- The high frequency signal, which is being modulated, is called the 'carrier'.
- The carrier is usually sine wave and considerably higher than information signal.
- The information signal may have any shape. The carrier after modulation is known as 'modulated wave'.
- The carrier voltage is represented by the expression

$$e_c = E_c (\cos \omega ct + \theta) \qquad \ldots (1.1)$$

where E_c is the peak amplitude of wave, ω is the angular frequency and θ is the initial phase.

- Three cases may arise, since there are three parameters of the carrier wave. Any one parameter can be varied keeping other two constant.
- When the amplitude E_c is varied in accordance with the modulating wave keeping ωc and θ constant, the process is called as 'amplitude modulation'.
- When ωc is varied in accordance with the modulating wave keeping E_c and θ constant, the process is known as 'frequency modulation'.
- When θ is varied in accordance with the modulating wave while ωc and E_c remain constant, the process is known as 'phase modulation'.
- In a continuous wave (CW) modulation, a sinusoidal wave is used as the carrier.
- When the amplitude of the carrier is varied in accordance with the base band signal, the process is known as 'amplitude modulation' and when the angle of the carrier is varied, it is known as 'angle modulation'.
- The angle modulation is further divided into frequency modulation and phase modulation.

- In case of frequency modulation, frequency of the carrier is varied and in case of phase modulation, phase of the carrier is varied in accordance with the base band signal.

Demodulation :

- At the broadcasting station, modulation is done to transmit the audio signal over to the receiver. When the modulated wave is received by the receiver, it is necessary to separate and recover the signal from the modulated carrier. This process is called as demodulation or detection.
- Thus, the process of recovering the original signal from the modulated wave is known as demodulation or detection.
- The detection is reverse of modulation. A detector or demodulator is the key circuit in any radio receiver.

1.7.2 Need of Modulation

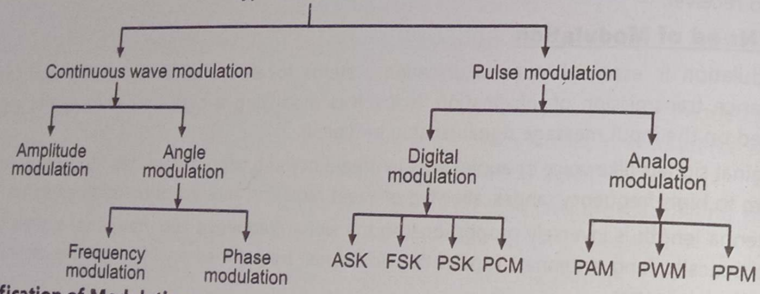
- Modulation is essential in communication systems to ensure efficient, reliable and long-distance transmission of information. It involves modifying a high-frequency carrier signal based on the input message signal (analog or digital).
- Original signals (like voice or audio) are low-frequency and cannot travel far. Modulation shifts them to high-frequency ranges, allowing efficient radiation and reception using antennas.
- Antenna length is inversely proportional to the signal frequency. Low-frequency signals need impractically long antennas. Modulating to higher frequencies enables the use of smaller, practical antennas.
- Modulation allows frequency division multiplexing (FDM). Multiple signals can share the same medium without interference by using different carrier frequencies.
- High-frequency carrier waves are less affected by noise. Certain modulation techniques (like FM) offer better noise immunity than baseband transmission.
- Different transmission media (optical fiber, coaxial cable, air) require signals in specific frequency ranges. Modulation adjusts signal properties to match the channel.
- Proper modulation helps in conserving bandwidth and efficient use of power. Techniques like QAM transmit multiple bits per symbol, increasing data rates.
- Modulation is the backbone of radio, television, mobile networks, satellite and Wi-Fi communication. Without modulation, wireless communication would be almost impossible.
- Benefits of Modulation are summarised in below table :

Benefit	Explanation
Long-distance transmission	Carrier wave carries the signal farther.
Smaller antennas	High-frequency signals need shorter antennas.
Multiplexing	Multiple signals on a single medium.
Noise reduction	Better resistance to distortion and noise.
Bandwidth and power efficiency	Optimized signal transmission.
Wireless communication capability	Enables radio, mobile, satellite communication.

1.7.3 Classification of Modulation

- Modulation is the process of varying a carrier signal in order to transmit information.
- There are basically two types of modulation: Continuous wave modulation and Pulse modulation.
- In a continuous wave (CW) modulation, a sinusoidal wave is used as the carrier. When the amplitude of the carrier is varied in accordance with the base band signal, the process is known as 'amplitude modulation', and when the angle of the carrier is varied, it is known as 'angle modulation'.
- The angle modulation is further divided into frequency modulation and phase modulation.
- In case of frequency modulation, frequency of the carrier is varied, and in case of phase modulation, phase of the carrier is varied in accordance with the base band signal.

Types of Modulation



Classification of Modulation :

- In pulse modulation, the carrier consists of a periodic sequence of rectangular pulses.
- In case of pulse amplitude modulation (PAM), amplitude of the pulses is varied according to the modulating signal amplitude.
- In case of pulse width modulation (PWM), width of the pulses will vary, whereas in case of pulse position modulation (PPM), positions of the pulses are varied according to the instantaneous value of modulating signal.
- In pulse amplitude, width or position can take any value within the range, that is why they are analog systems of communications.

Based on the nature of the input data (analog or digital) and the type of carrier signal (analog), modulation is broadly classified as follows:

1.7.3.1 Continuous Wave Modulation

[I] Amplitude Modulation (AM):

- It varies the amplitude of the carrier wave and it is used in AM radio broadcasting.
- In amplitude modulation, the amplitude of the carrier wave is varied in accordance with the instantaneous value of the modulating voltage keeping frequency and phase constant.

Let,

$$e_m = E_m \cos \omega_m t \qquad \text{... (1.1)}$$

represent the modulating voltage and

$$e_c = E_c \cos \omega_c t \qquad \text{... (1.2)}$$

represent the carrier voltage, where carrier frequency ω_c is much greater than ω_m.

Note that, the phase angle has been ignored in both expressions since it is unchanged by the amplitude modulation process. Even if we include phase angle, it will not affect the result but unnecessarily complicate the expressions, and hence for the implicity, θ has been dropped in equations (1.1) and (1.2).

- When a carrier wave is amplitude modulated, the amplitude of the carrier voltage waveform is caused to vary directly with the modulating voltage.

If A is the new amplitude, then according to the definition of amplitude modulation,

$$A = E_c + E_m \cos \omega_m t \qquad \text{... (1.3)}$$

The modulated carrier will have the expression,

$$e = A \cos \omega_c t \qquad \text{... (1.4)}$$

where e is the instantaneous voltage of the modulated signal.

Substituting equation (1.3) in equation (1.4),

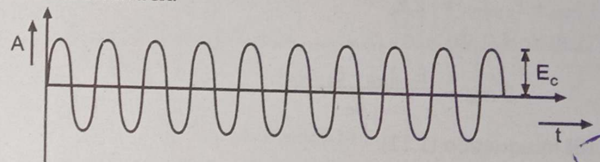
$$e = (E_c + E_m \cos \omega_m t) \cos \omega_c t \qquad \text{... (1.5)}$$

$$= E_c\left(1 + \frac{E_m}{E_c} \cos wm_t\right) \cos \omega_c t$$

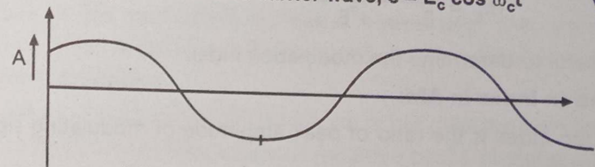
$$e = E_c (1 + m_a \cos \omega_m t) \cos \omega_c t \qquad \text{... (1.6)}$$

where $m_a = \frac{E_m}{E_c}$ ... (1.7)

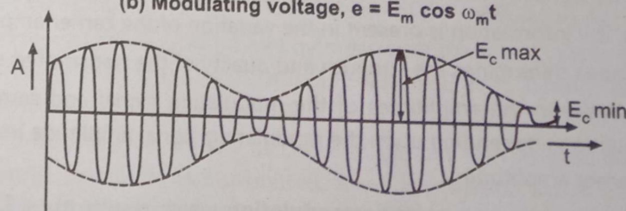
is known as **modulation index.**



(a) Unmodulated carrier wave, $e = E_c \cos \omega_c t$



(b) Modulating voltage, $e = E_m \cos \omega_m t$



(c) Amplitude modulated voltage, $e = E_c (1 + m_a \cos \omega_m t) \cos \omega_c t$

Fig. 1.4 : Amplitude modulation

- The modulation index is a number lying between 0 and 1 and is very often expressed as a percentage and called the percentage modulation.
- Fig. 1.4 (a) shows the waveform of unmodulated carrier voltage, Fig. 1.4 (b) that of sinusoidal modulating voltage and Fig. 1.4 (c) gives the waveform of the amplitude modulated carrier voltage.
- From the Fig. 1.4 (c), it is observed that the frequency of carrier remains unchanged but its amplitude varies in accordance with the variation of modulating voltage e_m. It is further seen that the maximum amplitude of modulating signal is,

$$E_{c\,(max)} = E_c(1 + m_a) \qquad \text{when } \cos\theta = 1$$

$$m_a = \frac{E_{c\,(max)} - E_c}{E_c} \qquad \dots (1.8)$$

and the minimum amplitude,

$$E_{c\,(min)} = E_c(1 - m_a) \qquad \text{when } \cos\theta = -1$$

Or

$$m_a = \frac{E_c - E_{c\,(min)}}{E_c} \qquad \dots (1.9)$$

Equating equations (1.8) and (1.9),

$$E_{c\,(max)} - E_c = E_c - E_{c\,(min)}$$

$$\therefore \quad E_{c\,(max)} + E_{c\,(min)} = 2\,E_c \qquad \dots (1.10)$$

Adding equations (1.8) and (1.9),

$$2\,m_a = \frac{E_{c\,(max)} - E_{c\,(min)}}{E_c} \qquad \dots (1.11)$$

Using equation (1.10) in equation (1.11),

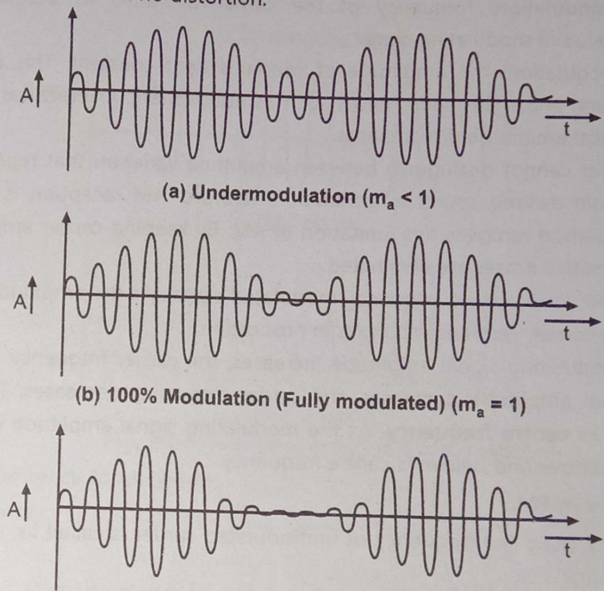
$$m_a = \frac{E_{c\,(max)} - E_{c\,(min)}}{E_{c\,(max)} + E_{c\,(min)}} \qquad \dots (1.12)$$

Equation (1.12) is useful to determine the modulation index.

Importance of Modulation Index in AM:

- In AM, the modulation index is the ratio of peak amplitude of modulating signal to the peak amplitude of carrier signal.
- After modulation, the information is present in the variation of the carrier amplitude.
- The modulation index determines the strength and quality of the transmitted signal.
- The relationship between the amplitudes of the modulating signal and carrier gives rise to three different situations depending upon the modulating signal amplitude less than, equal to or greater than carrier amplitude.
- If $E_m < E_c$, the situation is known as **undermodulation** which results $m_a < 1$, the amount of carrier amplitude variation is small and then the audio signal transmitted is weak.
- As the amplitude of the modulating signal is less than the carrier amplitude, distortion will not occur.

- Fig. 1.5 shows the modulated output voltage waveform for different values of modulation index.
- The ideal condition for AM is resulted when $E_m = E_c$ or $m_a = 1$, since it produces greatest output at the receiver with no distortion.



(a) Undermodulation ($m_a < 1$)

(b) 100% Modulation (Fully modulated) ($m_a = 1$)

(c) Overmodulation ($m_a > 1$)

Fig. 1.5 : Modulated output voltage waveform for various values of modulation index

- If the amplitude of the modulating voltage is higher than the carrier voltage, m_a will be greater than 1. This will cause severe distortion of the modulated waveform and the situation is known as **overmodulation**.
- Fig. 1.5 (c) shows overmodulation, the waveform is flat near the zero line. The received signal will produce an output waveform in which negative peaks of signal will be clipped off. This condition must be avoided as it results in distortion of the modulating signal.
- The modulation index can be determined by measuring the actual values of the amplitudes of modulation voltage and the carrier voltage. However, it is usually calculated from the modulated wave and using equation (1.12).
- In practice, it is desirable to use ma as close to 100 per cent as possible. So the maximum information signal amplitude is transmitted, more information signal power is transmitted, thereby producing a stronger, more intelligible signal.
- When the modulating signal amplitude varies, it is difficult to maintain 100% modulation. A voice signal, for example, changes amplitude as person speaks, then only peaks of the signal produce 100% modulation.

[II] Angle Modulation

1. Frequency Modulation (FM):

- It varies the frequency of the carrier wave. It is used in FM radio, audio transmission.
- In frequency modulation, frequency of the carrier is varied in accordance with the instantaneous value of modulating signal.
- In frequency modulation, the amplitude of carrier is kept constant. This is advantageous because all natural and man-made noises such as atmospheric or electrical machine sparks consist of electrical amplitude disturbances.
- The radio receiver cannot distinguish between amplitude variation that represent noise and those that contain desired sound information. Therefore, AM reception is generally noisy. Frequency modulation removes this limitation of AM. By keeping carrier amplitude constant, all amplitude sensitive noises are eliminated.
- Fig. 1.6 shows the waveform of frequency modulated signal. As the amplitude of information signal varies, the carrier frequency changes in proportion.
- In FM, as the modulating signal amplitude increases, the carrier frequency increases. As the modulating signal amplitude decreases, the carrier frequency decreases. The frequency of carrier is known as **centre frequency**. As the modulating signal amplitude varies, the carrier frequency varies above and below its centre frequency.

The Important Terms in FM:

1. **Centre frequency (f_c) :** The frequency of unmodulated carrier is called as 'centre frequency' or 'resting frequency'.
2. **Frequency deviation (f_d) :** The amount of change in carrier frequency produced by the modulating signal is called as frequency deviation. If f_{max} is the maximum and f_{min} is the minimum frequency of the modulated carrier after FM then,

$$f_d = f_{max} - f_c$$

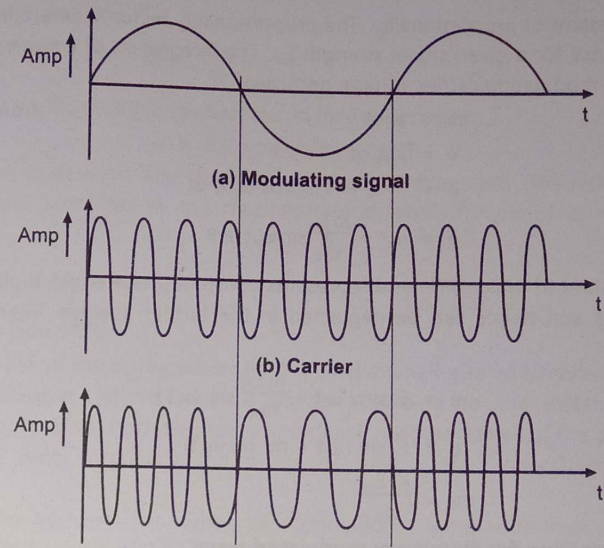
and

$$f_d = f_c - f_{min}$$

3. **Carrier swing :** The total variation in carrier frequency from minimum to maximum due to frequency modulation is known as carrier swing.

$$CS = f_{max} - f_{min}$$
$$= 2f_d$$

- As shown in the Fig. 1.6, the modulating signal is a low frequency sine wave. As sine wave goes positive, the frequency of carrier increases proportionally.
- At maximum amplitude of modulating signal, carrier frequency is maximum and at minimum amplitude of modulating signal, carrier frequency becomes minimum. When the modulating signal is at zero amplitude, the carrier frequency is the centre frequency.



(a) Modulating signal

(b) Carrier

(c) FM signal

Fig. 1.6 : Frequency modulation signal

Expression for Frequency Modulation:

Let the modulating voltage is given by the expression,

$$E_m = E_m \cos\omega_m t \quad \text{... (1.13)}$$

and the carrier voltage is given by,

$$e_c = E_c \sin(\omega_c t + \theta) \quad \text{... (1.14)}$$

Here the sine function is used to represent carrier for sake of mathematical simplicity. e_m and e_c represent instantaneous value of modulating and carrier voltage.

Let,

$$\phi = \omega_c t + \theta \quad \text{... (1.15)}$$

be the total instantaneous phase angle of the carrier voltage.

So equation (1.13) can be rewritten as,

$$e_c = E_c \sin\phi \quad \text{... (1.16)}$$

Angular frequency ω_c is related to phase angle ϕ by the relation,

$$\omega_c = \frac{d\phi}{dt} \quad \text{... (1.17)}$$

After FM, the frequency of the carrier varies in accordance with the instantaneous value of modulating voltage. Therefore, frequency of carrier after frequency modulation is,

$$\omega = \omega_c + K_f e_m$$

$$\omega = \omega_c + K_f E_m \cos\omega_m t \quad \text{... (1.18)}$$

where K_f is the constant of proportionality. The proportionality factor K_f determines the maximum variation in frequency for a given signal strength E_m. The integration of equation (1.18) gives the phase angle of the modulating carrier voltage because [from equation (1.17)]

$$d\phi = \omega_c dt$$

$$\phi = \int \omega_c \, dt$$

$$\phi = \int [\omega_c + K_f E_m \cos \omega_m t] \, dt$$

$$\phi = \omega_c t + \frac{K_f E_m}{W_m} \sin\omega_m t + \theta \quad \text{... (1.19)}$$

where θ is the constant of integration and represents constant phase angle. θ plays no part in the modulation process and hence can be neglected in the further analysis. Therefore, FM carrier voltage is,

$$e = E_c \sin\left(\omega_c t + \frac{K_f E_m}{\omega_m} \sin \omega_m t\right) \quad \text{... (1.20)}$$

$$e = E_c \sin(\omega_c t + m_f \sin \omega_m t)$$

where $m_f = \dfrac{K_f E_m}{\omega_m}$

is known as the **expression for frequency modulated wave.**

From equation (1.18), the instantaneous frequency of the frequency modulated carrier voltage in Hz is given by,

$$f = \frac{\omega}{2\pi}$$

$$f = f_c + \frac{K_f E_m}{2\pi} \cos\omega_m t \quad \text{... (1.21)}$$

The maximum value of frequency is,

$$f_{max} = f_c + \frac{K_f E_m}{2\pi} \quad \text{... (1.22)}$$

and the minimum value of frequency is,

$$f_{min} = f_c - \frac{K_f E_m}{2\pi} \quad \text{... (1.23)}$$

when $\cos \omega_m t$ is + 1 and −1 respectively. If f_c is the centre frequency, then the frequency deviation or the maximum variation in frequency is,

$$f_d = f_{max} - f_c = f_c - f_{min}$$

From equations (1.22) and (1.23),

$$f_d = \frac{K_f E_m}{2\pi} \quad \text{... (1.24)}$$

The modulation index for frequency modulation is defined as,

$$m_f = \frac{\text{Frequency deviation}}{\text{Modulating frequency}} = \frac{f_d}{f_m} \quad \text{... (1.25)}$$

Using equation (1.24) in (1.25),

$$m_f = \frac{K_f E_m}{2\pi f_m} = \frac{K_f E_m}{\omega_m} \quad \text{... (1.26)}$$

Using equation (1.26) in (1.20), the expression for FM becomes,

$$e = E_c \sin(\omega_c t + m_f \sin \omega_m t) \quad \text{... (1.27)}$$

Unlike AM, the modulation index for FM can be greater than unity. The ratio of the maximum frequency deviation permitted to the maximum modulating frequency is referred to as the **deviation ratio**.

$$\delta = \frac{f_d}{f_{m\,max}}$$

2. Phase Modulation (PM):

- It varies the phase of the carrier wave and it is used in analog satellite communication.
- In phase modulation, phase angle is varied according to the instantaneous value of modulating signal. Since, both frequency and phase are parameters of the carrier angle, which is function of time, frequency and phase modulation are collectively known as *angle modulation*.
- Thus, there are basically two types of modulation : Amplitude modulation and Angle modulation. It is found that FM and PM are closely related to each other. In fact, varying the phase shift of the carrier produces frequency modulation. Since, PM produces FM, PM is often referred to as indirect FM.
- In phase modulation, phase of the carrier is varied in accordance with the modulating signal.

  If $$e_c = E_c \sin(\omega_c t + \phi_c) \quad \text{... (1.28)}$$
  is the unmodulated carrier and
  $$e_m = E_m \sin \omega_m t \quad \text{(1.29)}$$
  be the modulating signal.

  In phase modulation, ϕ_c changes according to modulating signal. So after phase modulation, ϕ_c becomes ϕ_t such that,
  $$\phi(t) = \phi_c + Ke_m \quad \text{... (1.30)}$$
  where K is phase deviation constant. Usually ϕ_c is dropped from the equation since it is constant and does not affect the modulation. So equation (1.30) can be rewritten as,
  $$\phi(t) = K E_m \sin \omega_m t$$
  Or
  $$\phi(t) = m_p \sin \omega_m t \quad \text{... (1.31)}$$
  where $m_p = K E_m$ is known as **phase modulation index**. So the equation of phase modulated carrier becomes,
  $$e = E_c \sin(\omega_c t + m_p \sin \omega_m t) \quad \text{... (1.32)}$$
  Comparing equation (1.32) with that of frequency modulated carrier which is
  $$e = E_c \sin(\omega_c t + m_f \sin \omega_m t)$$
  shows that both equations are similar and hence similar to frequency modulated signal, phase modulated signal also contains infinite number of side bands. If the modulation index in the

two cases are same, the relative amplitudes of different side bands will also be same. Thus, the only difference in phase modulation and frequency modulation is the process by which modulation index is defined.

Now, the total phase in phase modulated signal is,

$$\phi = \omega_c t + m_p \sin \omega_m t \qquad \text{... (1.33)}$$

Since, $\omega = \frac{df}{dt}$, corresponding change in frequency is,

$$\omega = \omega_c + m_p \omega_m \cos \omega_m t$$

Thus, the maximum frequency deviation produced by phase modulation is $\frac{\omega_m m_p}{2\pi}$.

or $\quad (f_d)_p = f_m m_p \qquad \text{... (1.34)}$

whereas $\quad (f_d)_f = \frac{K_f E_m}{2\pi}$

Since, the modulation index m_p for phase modulation is proportional to the modulating signal amplitude, it is obvious from equation (1.34) that frequency deviation in phase modulated wave is proportional to both amplitude and frequency of the modulating signal. Whereas in case of frequency modulation, frequency deviation is proportional to only modulating signal amplitude and not its frequency.

Thus, in case of phase modulation, a frequency modulated wave is obtained in which frequency deviation is proportional to the modulating frequency.

The amplitude and frequency modulation is used in practical communication systems. Phase modulation as such is not used in practical analog transmission system since our ear cannot detect changes in the phase of the signal.

Comparison Between AM and FM :

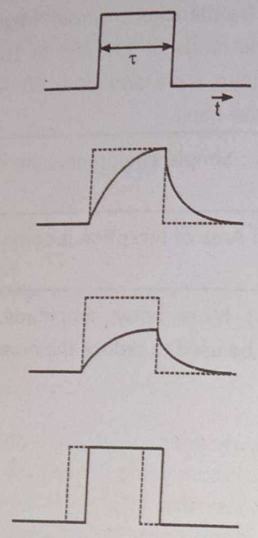
Amplitude Modulation	Frequency Modulation
1. **Definition :** In amplitude modulation, amplitude of the carrier is varied according to the instantaneous value of modulating signal.	1. In frequency modulation, frequency of the carrier is varied according to the instantaneous value of modulating signal.
2. **Side bands :** In AM, only two side bands are generated. Equation : $e = E_c (1 + m_a \cos \omega_m t) \cos \omega_c t$	2. In FM, infinite number of side bands are generated. $e = E_c \sin (\omega_c t + m_f \sin \omega_m t)$
3. **Power :** Only the power in side band is useful.	3. All the transmitted power in FM is useful.
4. **Bandwidth :** Bandwidth required for AM is twice the highest modulating frequency. Usually assigned bandwidth is 10 KHz.	4. Ideally, infinite bandwidth is required, more wider upto 10 times as that of AM.
5. **Carrier frequencies :** Carrier frequencies are in the range of 535 KHz to 1600 KHz for medium wave and upto 30 MHz for short wave band.	5. Carrier frequencies are in the range of 88 MHz to 108 MHz for FM radio broadcast.
6. **Circuits :** Simple equipments are required for AM.	6. More complex and costly circuits are required for FM.
7. **Range :** Area of reception is comparatively large.	7. Area of reception is limited due to line of sight propagation.
8. **Noise :** More noisy; amplitude limiters cannot be used to reduce the noise.	8. Less noisy; noise can be reduced by using amplitude limiters or by using special detectors, which does not respond to amplitude variations. AM broadcast operates in the upper VHF and UHF frequency range, at which there happens to be less noise than in the MF and HF ranges occupied by AM broadcast.
9. **Fidelity :** Fidelity is poor as maximum modulating frequency is 5 KHz.	9. Fidelity is better as maximum modulating frequency is 15 KHz.
10. **Pre-emphasis :** Pre-emphasis circuits are not used.	10. Pre-emphasis circuits are used which help to reduce high frequency noise.
11. **Guard band :** As there are no guard bands, there is possibility of interference.	11. Standard frequency allocation provides a guard band between commercial FM stations, so that there is less adjacent channel interference than in AM.

Contd. ...

1.7.3.2 Pulse Modulation

- 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 we must have finite set of discrete values.
- Digital systems, such as pulse code modulation, also use pulses and hence is also a type of pulse modulation. More details of pulse code modulation in article 2.3.

Pulse Transmission :



(a) A rectangular pulse

(b) A rectangular pulse not severely distorted by bandwidth limitations

(c) A rectangular pulse distorted by transmission due to very low bandwidth

(d) By applying bias pulse can be reconstructed

Fig. 1.7 : Pulse transmission distortion

- Fig. 1.7 shows a pulse. A rectangular pulse as shown has a fixed time duration or width τ and a constant amplitude during that time period.
- A fourier analysis of a square pulse shows that a square pulse is composed of continuous frequencies.
- For pulse transmission in communication we have bandwidth limitation which is equivalent to a filter. So if such a square pulse is passed through a low pass filter, then all the frequencies above the filter cut-off frequency will be eliminated and the pulse at the output will be rounded off as shown in Fig. 1.7 (b). Such a generated pulse has constant amplitude with lead and tail edges.
- Since pulse generating equipment normally generates such type of pulse only which is far from perfect. So if the rise and fall times of the edges are very much shorter than the pulse width τ, we can treat the pulse as a perfect one and ignore the rise and fall times.
- If the bandwidth of the transmission channel is very narrow, it induces excessive phase distortion. The pulse gets smeared and reduced in amplitude as shown in Fig. 1.7 (c).
- If the distortion is too severe, it is impossible to detect the pulse at the receiving end. Noise added during transmission may produce an unwanted pulse or cancel a wanted pulse which results in error in transmission. By adding bias to the received pulse it can be reconstructed to its original shape.
- Even though square pulse contains all frequency components, it is found that the most of the spectrum is contained in the frequency interval from zero to $1/\tau$, where τ is the pulse width.

- For satisfactory pulse transmission in a communication system, the bandwidth of the channel should be atleast $1/\tau$. We can observe the distortion produced by bandwidth restriction on oscilloscope. However, to display accurately the pulse waveform in the oscilloscope, the oscilloscope bandwidth should be atleast four times $1/\tau$ in order to pass more of the high frequency detail of the spectrum.

[I] Digital Modulation :

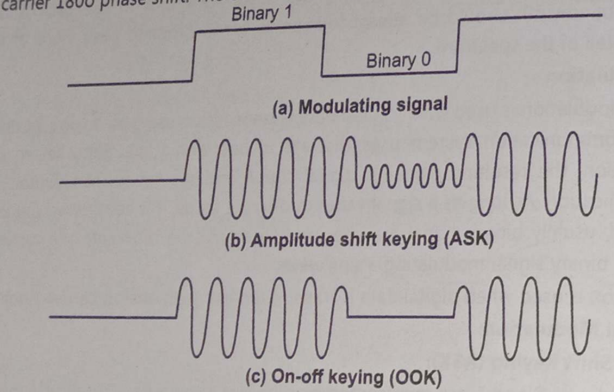
- Digital modulation is used when digital data is transmitted using an analog carrier signal.
- If the communication system uses discrete values within the finite set to represent information, the resulting signal is digital signal and system is then digital system of communication. As long as a signal varies in discrete steps, it is considered digital.
- A digital, usually binary signal may be used to amplitude modulate the carrier. Fig. 1.8 shows a binary signal modulating a sine wave.

Digital modulation is used when digital data is transmitted using an analog carrier signal.

Types of Digital Modulation:

1. **Amplitude Shift Keying (ASK):**
   - In ASK, digital 0 and 1 are represented by different amplitudes. Amplitude modulation in which the carrier is switched between two different carrier levels is known as amplitude shift keying (ASK).
     As shown in the Fig. 1.8 (b), binary 1 level produces maximum carrier amplitude and binary 0 produces lower amplitude.
   - A special form of ASK is one in which the carrier is simply switched ON or OFF. The binary level 1 turns the carrier ON and the binary 0 level turns the carrier OFF. This is called as ON-OFF keying (OOK).
2. **Frequency Shift Keying (FSK):**
   - It uses different frequencies to represent binary data. To transmit binary data on telephone line, a technique known as frequency shift keying is used in which two different frequencies are assigned to represent 0 and 1.
   - In FSK, two sine wave frequencies are used to represent binary 0 and 1. For example, a binary 0 usually called a space in data communication has a frequency of 1070 Hz. A binary 1, referred to as a mark is 1270 Hz.
   - These two frequencies are alternately transmitted to create the serial binary data.
   - To permit full duplex operation, another set of frequencies are defined. A binary 0 or space is 2025 Hz and a binary 1 or mark is 2225 Hz. These tones are within the voice frequencies (300 to 3000 Hz). In the communication system at the transmitter, FSK modulator is used and at the receiver, FSD demodulator is used. The **modulators** and **demodulators** are known as modems.
3. **Phase Shift Keying (PSK):**
   - PSK varies the phase to represent bits (e.g., BPSK, QPSK).
   - In phase shift keying (PSK), the binary signal to be transmitted changes the phase shift of a sine wave depending upon whether a binary 0 or binary 1 is to be transmitted.

- The phase shift is a time difference between two sine waves of the same frequency.
- In binary PSK, binary 0 is transmitted with one phase and binary 1 is transmitted with a carrier 180o phase shift. The carrier sine wave usually has frequency 1600 Hz or 1700 Hz.

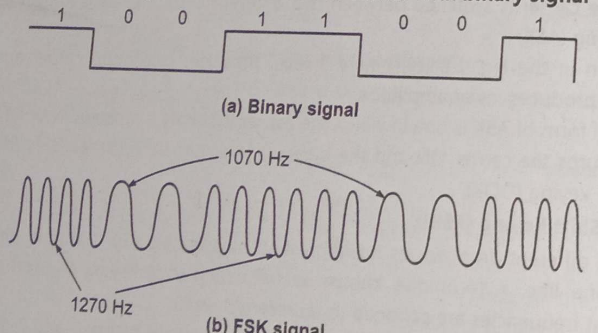


(a) Modulating signal

(b) Amplitude shift keying (ASK)

(c) On-off keying (OOK)

Fig. 1.8 : Amplitude modulation of carrier with binary signal



(a) Binary signal

(b) FSK signal

Fig. 1.9 : Frequency shift keying

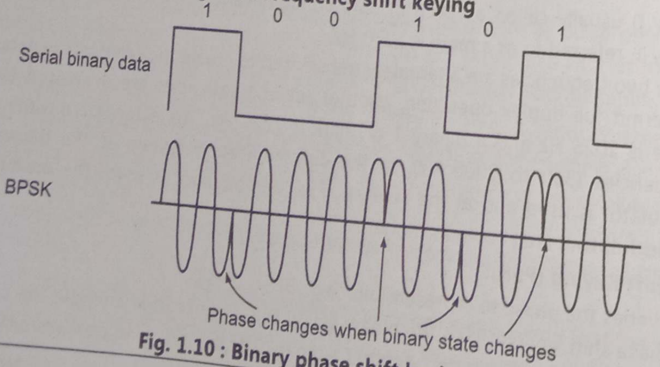


Fig. 1.10 : Binary phase shift keying

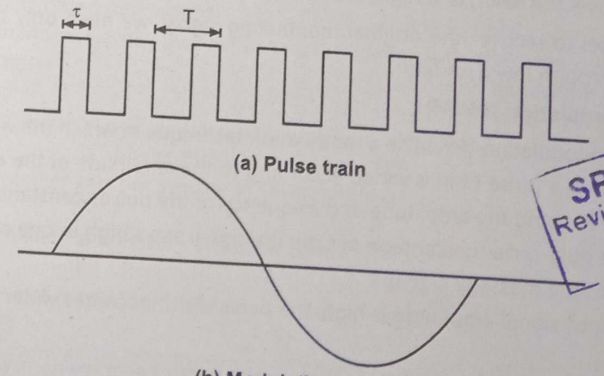
4. Pulse Code Modulation (PCM) :

- 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.

[II] Analog Modulation :

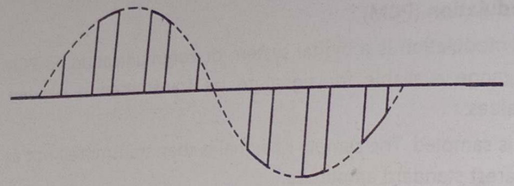
1. Pulse Amplitude Modulation (PAM) :

- In Pulse Amplitude Modulation, carrier used is not the sine wave but a train of pulse.
- In PAM, amplitude can take any value and hence it is analog against ASK in which amplitude can take only two values and hence it is digital communication.
- In pulse amplitude modulation, the amplitude of the pulses in the pulse train is varied according to the instantaneous value of the modulating signal.
- The overall effect is multiplying the pulse train by the modulating signal. Fig. 1.11 shows pulse train, a modulating sine wave and positive resulting PAM waveforms.

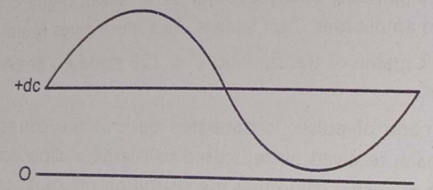


(a) Pulse train

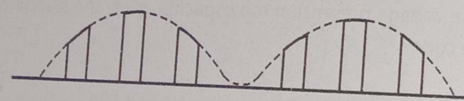
(b) Modulating signal



(c) Dual polarity PAM waveform



(d) Modulating signal with dc offset



(e) Single polarity PAM waveform

Fig. 1.11 : Pulse amplitude modulation waveforms

- From the Fig. 1.11 (c) we see that the pulse amplitude is made to vary in proportion to the amplitude of lower frequency modulating signal [Fig. 1.11 (b)].
- The resultant dual polarity waveform is a series of pulses whose amplitude corresponds to the original modulating signal [Fig. 1.11 (c)].
- If the modulating signal is offset by a dc level sufficient to ensure that the pulses are always positive, the result is single polarity waveform.
- In either case, to recover the original modulating signal, we need only to pass the PAM waveform through low pass filter.

2. Pulse Width Modulation (PWM) :

- Pulse Width Modulation (PWM) is a modulation technique in which the width (duration) of each pulse in a pulse train is varied according to the amplitude of the analog input signal, while keeping the amplitude and frequency of the pulses constant.
- In PWM, the duty cycle (percentage of time the signal stays high in one cycle) is varied in proportion to the message signal.
- When the input signal amplitude is high, the pulse width becomes wider (higher duty cycle).
- When the input signal amplitude is low, the pulse width becomes narrower (lower duty cycle). Thus, the information is represented by variation in pulse width.

3. Pulse Position Modulation (PPM) :

- Pulse Position Modulation (PPM) is a type of pulse modulation technique in which the position (timing) of each pulse, with respect to a reference time, is varied according to the amplitude of the analog input signal.
- The amplitude and width of the pulses remain constant, but their positions change in time to represent the modulating signal.
- In PPM, each analog sample value is used to shift the position of a corresponding pulse within a fixed time frame.
- A higher amplitude of the modulating signal causes the pulse to shift further in time. A lower amplitude causes the pulse to shift closer to the reference. Thus, the timing of the pulses carries the information of the message signal.

ASK, PSK, FSK, PCM will be discussed in detail in next chapter.

Exercise

[A] Multiple Choice Questions:

[I] Choose the Most Correct Alternative for Each of the Following and Rewrite the Sentence :

1. Which of the following is not a basic element of a digital communication system?
   (a) Transmitter (b) Noise (c) Receiver (d) Antenna
2. The function of the modulator in a communication system is to ....
   (a) Increase data rate (b) Remove noise from signal
   (c) Superimpose information signal on carrier (d) Convert analog to digital
3. Which term describes the ratio of signal power to noise power?
   (a) Bandwidth (b) Signal strength
   (c) SNR (Signal to Noise Ratio) (d) Baud rate
4. What is the unit of signal bandwidth?
   (a) dB (b) Hertz (Hz) (c) Bits (d) Volts
5. In digital communication, baud rate refers to ....
   (a) Bits transmitted per second (b) Signal frequency
   (c) Symbols transmitted per second (d) Channels used
6. According to Shannon-Hartley theorem, channel capacity increases with:
   (a) Increasing noise (b) Decreasing bandwidth
   (c) Increasing SNR (d) Lowering frequency
7. Which of the following is a digital-to-digital encoding technique?
   (a) ASK (b) Manchester (c) FM (d) PSK
8. What is the number of bits per symbol in 16-ary coding?
   (a) 2 (b) 3 (c) 4 (d) 5
9. Which of the following is an error correction code?
   (a) CRC (b) Checksum (c) Parity (d) Hamming Code
10. The main purpose of modulation is to ....
    (a) Reduce noise
    (b) Make signal compatible with transmission channel
    (c) Convert analog to digital
    (d) Store signals

11. Which is NOT a type of modulation?
    (a) Frequency Modulation (b) Phase Shift Keying
    (c) Delta Modulation (d) Synchronous Transmission
12. What is the process of recovering the original signal at the receiver end called?
    (a) Modulation (b) Sampling (c) Demodulation (d) Quantization
13. Channel capacity is expressed as ....
    (a) C = 2B (b) C = B (c) C = 2/B (d) C = 4B
14. Which of the following is not a signal encoding format ?
    (a) Digital-to-digital (b) Digital-to-analog
    (c) Analog-to-digital (d) Digital-to-pulses
15. Frequency Shift Keying (FSK) is a .... type of modulation.
    (a) Digital (b) Analog (c) Phase (d) Pulse

Answers

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

[II] Fill in the Blanks :

1. The component responsible for converting the original message into a suitable format for transmission is called the ______.
2. The process of removing noise and recovering the original signal at the receiver is called ______.
3. In digital communication, information is transmitted in the form of ______.
4. The range of frequencies contained in a signal is called ______.
5. The range of frequencies a channel can transmit is called ______.
6. In M-ary coding, each symbol represents ______ bits.
7. Parity bits, checksum and CRC are examples of ______ codes.
8. ______ code can detect and correct single-bit errors.
9. ______ is the process of varying a carrier wave according to the message signal.
10. The reverse process of modulation at the receiver is called ______.
11. ASK, FSK, PSK and QAM are types of ______ modulation.
12. ______ refers to the maximum data rate that a channel can support.
13. The relationship between bandwidth, SNR and capacity is given by the ______ theorem.
14. The ratio of signal power to noise power is known as ______.
15. The number of signal changes or symbols per second is called ______.

Answers

1. Encoder	2. Decoding	3. binary digits / bits
4. signal bandwidth	5. channel bandwidth	6. $\log_2(M)$
7. error detection	8. Hamming	9. Modulation
10. Demodulation	11. Digital	12. Channel capacity
13. Shannon-Hartley	14. signal-to-noise ratio	15. baud rate

[III] True/False :

1. The main components of a digital communication system include transmitter, channel and receiver.
2. In a digital communication system, noise is completely eliminated.
3. Digital signals vary continuously over time.
4. Noise figure quantifies the distortion added by the receiver.
5. Shannon-Hartley theorem relates data rate with bandwidth and signal-to-noise ratio.
6. Parity bits can both detect and correct errors.
7. Modulation is the process of varying a carrier wave to carry information.
8. AM, FM and PM are types of digital modulation.
9. M-ary coding is used to reduce data rate.
10. Signal encoding converts data into a format suitable for transmission.
11. Modulation allows the use of small-size antennas.
12. Demodulation is performed at the transmitter end.
13. CRC is a type of error correction code.
14. Error control codes are not required in communication systems.
15. Baud rate and data rate are always the same.

Answers

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

[B] Short Answer Questions :

1. List the basic elements of a digital communication system.
2. What is the role of the transmitter in a communication system?
3. Explain the function of the channel in a digital communication system.
4. Why is the receiver important in a communication system?
5. Define signal bandwidth.
6. What is meant by channel bandwidth?
7. Define signal-to-noise ratio (SNR).
8. What is noise figure?
9. Define data rate.
10. What is baud rate? How is it different from data rate?
11. What is channel capacity?
12. State the Shannon-Hartley theorem.
13. List any four types of digital-to-digital encoding formats.
14. Explain the concept of M-ary coding.
15. How does M-ary coding improve bandwidth efficiency?
16. Give one example of M-ary coding.
17. Differentiate between error detection and error correction.

18. List any three types of error detection codes.
19. Name two error correction codes used in digital communication.
20. What is Hamming code used for?
21. Define modulation.
22. What is demodulation?
23. Differentiate between analog and digital modulation.
24. List any two types of digital modulation techniques.
25. Classify modulation techniques based on the type of message signal.

[C] Long Answer Questions :

1. Draw a neat block diagram of a digital communication system and explain the function of each block in detail.
2. Describe the major components of a digital communication system. How does information flow from the sender to the receiver?
3. Explain the role of the transmitter, channel and receiver in a communication system with suitable examples.
4. Write short notes on the following terms:
   (a) Signal-to-noise ratio (SNR)
   (b) Noise figure
   (c) Data rate
   (d) Baud rate
   (e) Channel capacity
5. Explain Shannon-Hartley theorem and its significance in determining the channel capacity.
6. What is signal encoding? Describe various types of digital-to-digital encoding formats with diagrams or examples.
7. Explain the concept of M-ary coding. How does it improve the performance of digital communication systems? Give suitable examples.
8. Compare binary encoding and M-ary encoding with respect to bandwidth efficiency and complexity.
9. Why are error control codes necessary in digital communication systems? Discuss with examples.
10. Classify and explain different types of error handling codes. Give examples of each.
11. Explain the working of Hamming code for single-bit error detection and correction. Show the encoding and decoding process with an example.
12. Define modulation and demodulation. Explain their importance in communication systems with suitable diagrams.
13. Why is modulation needed in a communication system? List and explain at least four reasons.
14. Classify various types of modulation techniques. Explain analog and digital modulation schemes with examples.
15. List advantages and disadvantages of analog and digital communication

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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