Unit I: Introduction to Data Communication [5 Hrs.]

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1.1 Introduction to data communication

Data Communications (DC):

Data communication refers to the exchange of data between a source and a receiver. Data communication is the process of transferring data between two or more devices using transmission medium such as cables or wireless signals.

For example, a common example of data communications is a computer connected to the Internet via a Wi-Fi connection, which uses a wireless medium to send and receive data from one or more remote servers.

1.2 Data Transmission Modes (simplex, half-duplex, and full-duplex)

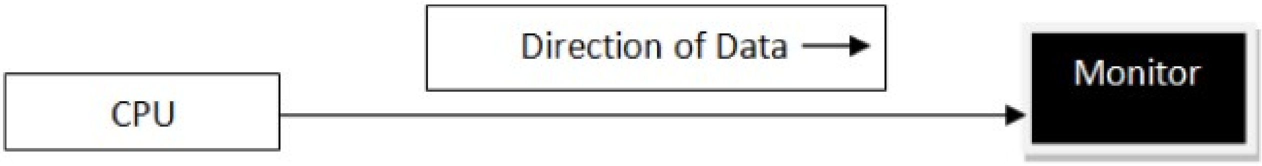
The way in which data is transmitted from one place to another is called data transmission mode. It is also called the data communication mode. It is indicates the direction of flow of information.

Types of **Data Transmission Modes :**
Different types of data transmission modes are as follows:
1. Simplex mode
2. Half-duplex mode
3. Full-duplex mode

1. Simplex Mode

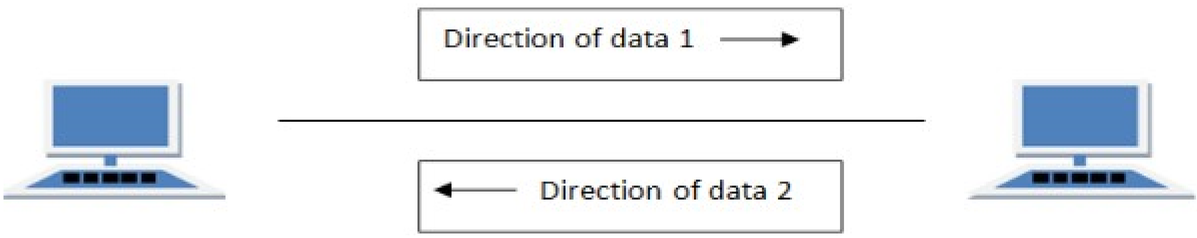
In simplex mode, data can flow in only one direction. In this mode, a sender can only send data and cannot receive it. Similarly,

a receiver can only receive data but cannot send it. Data sent from computer to printer is an example of simplex mode.
Examples of simplex Mode is loudspeaker, television broadcasting, television and remote, keyboard and monitor etc.



2. Half-Duplex Mode

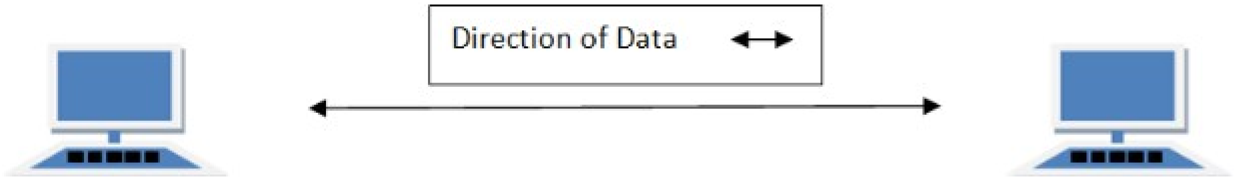
A half-duplex system can transmit data in both directions, but only in one direction at a time. Both the connected devices can transmit and receive but not simultaneously. When one device is sending the other can only receive and vice-versa. Data is transmitted in one direction at a time.
For example a walkie-talkie.



3. Full-Duplex Mode

In full duplex system we can send data in both directions as it is bidirectional. Data can be sent in both directions simultaneously. We can send as well as we receive the data. Full-duplex transmission, the channel capacity is shared by both communicating devices at all times.

Example of Full Duplex is a Telephone Network in which there is communication between two persons by a telephone line, through which both can talk and listen at the same time.



1.3 Fundamental Characteristics of Data Communication

The effectiveness of a data communications system depends on four fundamental characteristics:
delivery, accuracy, timeliness, and jitter.

1. Delivery: The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.

2. Accuracy: The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.

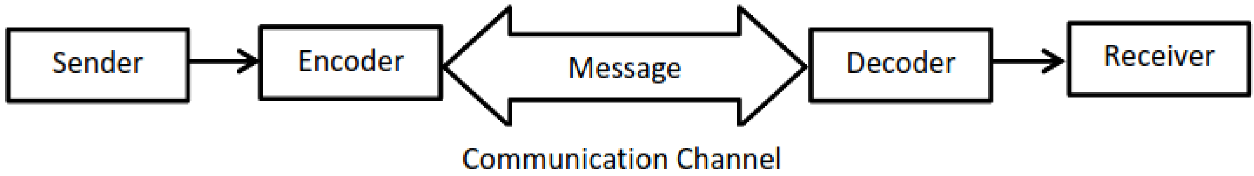
3. Timeliness: The system must deliver data in a timely manner. Data delivered late are useless.
In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called real-time transmission.

4. Jitter. Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets.

1.4 Components of Data Communication System

The basics components or elements of data communication system are as follows:

1. Message
2. Sender
3. Receiver
4. Medium or Communication Channel
5. Encoder and Decoder
6. Protocol



1. Message: - The message is the information or data that is to be communicated. It may consist of text, numbers, pictures, sounds, videos or any combination of these.

2. Sender: - A device that is used for sending messages (or data) is called sender. It is also called transmitter or source. The sender can be a computer, telephone, or a video camera etc.

3. Receiver: - A device that is used for receiving messages is called receiver. It is also known as sink. The receiver can be a computer, telephone set, printer, or a fax machine etc.

4. Medium: - The path through which data is transmitted (or sent) from one location to another is called transmission medium. It is

also called communication channel. It may be a wire, or fiber optic cable, etc.

5. Encoder and Decoder: - In communication systems, computers are used for senders and receivers. A computer works with digital signals. The communication channels usually use analog signals. The encoder and decoder are used in communication systems to convert signals from one from to another.

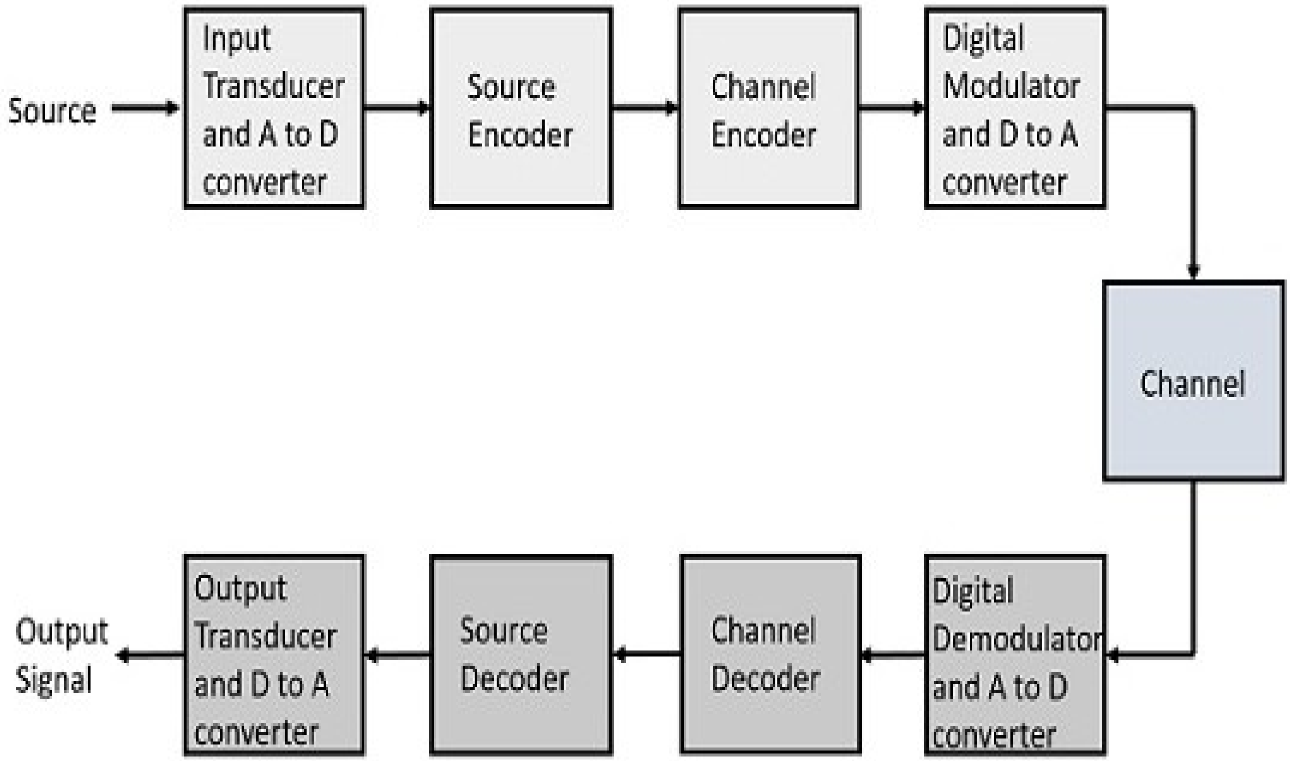
Encoder: - The encoder is an electronic device. It receives data from sender in the form of digital signals. It converts digital signals into a form that can be transmitted through transmission medium.

Decoder: - The decoder is an electronic device. It receives data from transmission medium. It converts encoded signals (i.e. analog signals) into digital form.

6. Protocol: - It is a set of rules that governs the data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating.

1.5 Block Diagram of Digital Communication System

The elements which form a digital communication system are:



Source
The source can be an analog signal. Example: A Sound signal

Input Transducer
This is a transducer which takes a physical input and converts it to an electrical signal (Example: microphone). This block also consists of an analog to digital converter where a digital signal is needed for further processes.

Source Encoder
The source encoder compresses the data into minimum number of bits. This process helps in effective utilization of the bandwidth. It removes the redundant bits (unnecessary excess bits, i.e., zeroes).

Channel Encoder
The channel encoder, does the coding for error correction. During the transmission of the signal, due to the noise in the channel, the signal may get altered and hence to avoid this, the channel encoder

adds some redundant bits to the transmitted data. These are the error correcting bits.

Digital Modulator

The signal to be transmitted is modulated here by a carrier. The signal is also converted to analog from the digital sequence, in order to make it travel through the channel or medium.

Channel

The channel or a medium, allows the analog signal to transmit from the transmitter end to the receiver end.

Digital Demodulator

The received signal is demodulated as well as converted again from analog to digital. The signal gets reconstructed here.

Channel Decoder

The channel decoder, after detecting the sequence, does some error corrections. The distortions which might occur during the transmission, are corrected by adding some redundant bits. This addition of bits helps in the complete recovery of the original signal.

Source Decoder

The resultant signal is once again digitized by sampling and quantizing so that the pure digital output is obtained without the loss of information. The source decoder recreates the source output.

Output Transducer

This converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output (Example: loud speaker).

Output Signal

This is the output which is produced after the whole process. Example − The sound signal received.

1.6 Introduction to Analog to Digital System

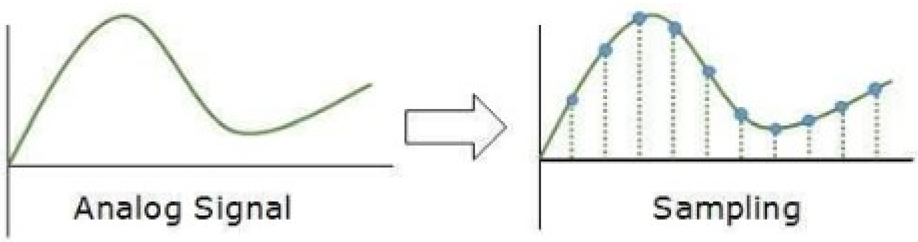
An Analog to Digital System (ADC system) refers to the process and technology that converts continuous analog signals (like sound, light, temperature, etc.) into a digital format (binary numbers: 0s and 1s) so that computers, micro controllers, and digital systems can process them. To convert analog wave into digital data, we use Pulse Code Modulation (PCM).

PCM involves three steps:

- Sampling
- Quantization
- Encoding.

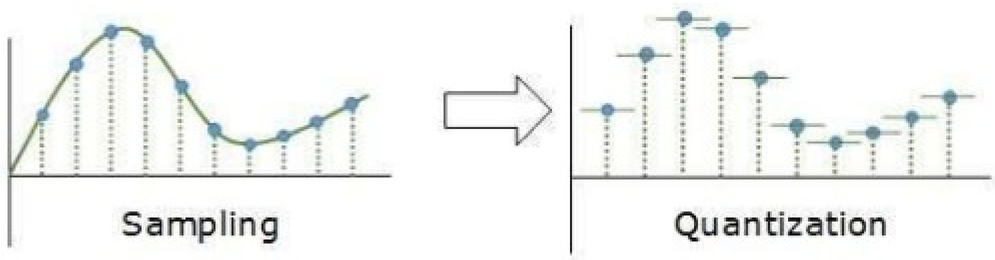
Sampling

The analog signal is sampled every T interval. Most important factor in sampling is the rate at which analog signal is sampled. According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.



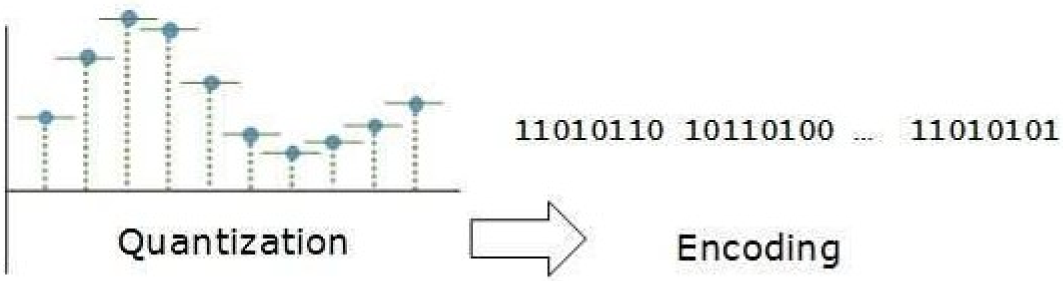
Quantization

Sampling yields discrete form of continuous analog signal. Every discrete pattern shows the amplitude of the analog signal at that instance. The quantization is done between the maximum amplitude value and the minimum amplitude value. Quantization is approximation of the instantaneous analog value.



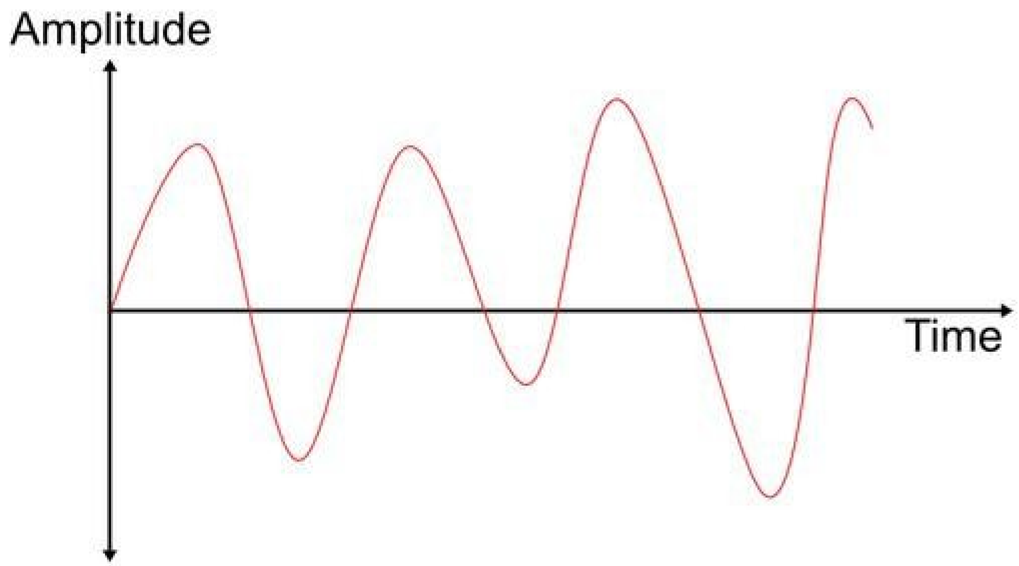
Encoding

In encoding, each approximated value is then converted into binary format.



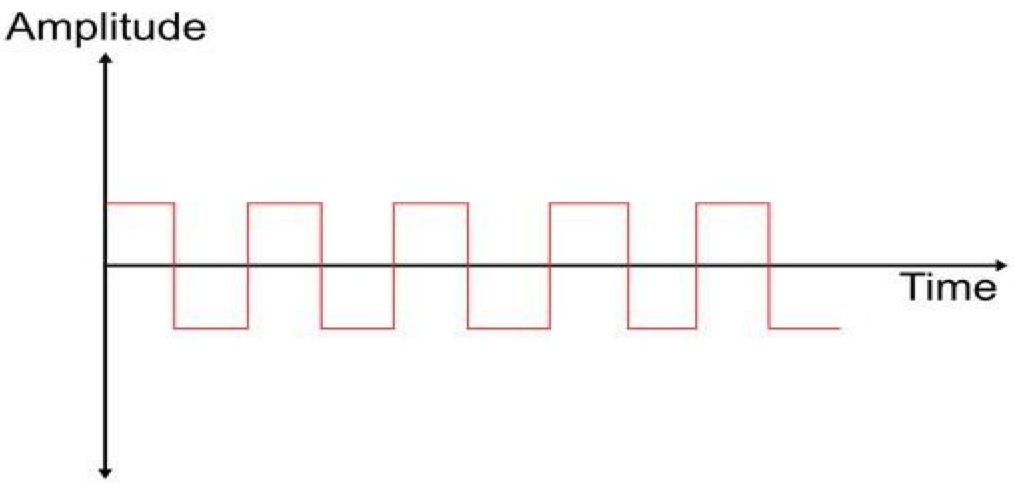
Analog System

Analog system is one that uses continuous time signal or analog signal which is a sinusoidal waveform. Analog system transmits the output in their raw form reducing the time of translation. The amplitude of the signal varies continuously with the time. Analog signals are used to represent sound, temperature, light intensity etc.



Digital System

A digital system is one whose signal has a finite number of discrete values. So, the digital system works on digital signals and is limited to binary values 0 or 1. Digital systems are used to process information in digital form. The digital system has wide applications in digital instruments like calculators, computers, Telephones, etc.



	Analog System	Digital System
Signal	Analog signal represents physical measurements.	Digital signals are discrete and generated by digital modulation.
Waves	Sine Waves	Square Waves
Representation	Continuous range of values to represent information	Uses discrete values to represent information
Technology	Records waveforms as they are.	Samples analog waveforms into a limited set of numbers and then records them.
Data transmissions	Affected by noise during transmission and write/read cycle.	Noise-immune during transmission and write/read cycle.
Response to Noise	More likely to get affected	Less likely to get affected
Flexibility	Hardware is not flexible.	Hardware is flexible.
Bandwidth	Less bandwidth.	More bandwidth to carry out the same information
Memory	Stored data in the form of wave signal	Stored data in the form of binary bit
Power	Consumes large power	Consumes negligible power
Uses	Best suited for audio and video transmission.	Best suited for Computing and digital electronics.
Cost	Cost is low	Cost is high
Example	Human voice in air, analog electronic devices.	Computers, CDs, DVDs

1.7 Advantages of digital transmission over Analog

1. Noise Resistance

Digital signals are much more resistant to noise and interference.
Even if noise affects the signal during transmission, it's easier to recover the original data in digital form.
In contrast, analog signals degrade continuously, and noise distorts them permanently.
Example: A digital call has clearer voice quality compared to an analog call over the same distance.

2. Higher Security

Encryption and error detection are much easier in digital transmission.
Digital data can be scrambled or encrypted to protect sensitive information.
Analog signals are harder to secure, making them more vulnerable to eavesdropping.

3. Efficient Data Compression

Digital transmission allows for data compression techniques.
This helps reduce bandwidth usage and storage requirements, making transmission faster and more efficient.
Example: MP3 files compress audio without losing much quality.

4. Easy Error Detection and Correction

Digital systems can implement techniques like parity bits, checksums, and Hamming codes.

This allows the detection and correction of errors that occur during transmission.
Analog systems do not have built-in mechanisms for error correction.

5. Integration with Computers and Digital Systems

Digital signals can be directly processed by computers, microcontrollers, and digital devices.
This simplifies design and increases compatibility with modern electronics.

6. Better Signal Quality Over Long Distance

In digital transmission, signals can be regenerated (reconstructed) perfectly using repeaters.
Analog signals degrade over distance, causing loss of quality and more noise.
Example: Optical fiber networks use digital transmission and maintain quality over hundreds of kilometers.

7. Scalability and Flexibility

Digital systems support multiplexing, allowing multiple digital signals to be sent over the same medium.
Adding new features, upgrading, or scaling digital systems is easier and cheaper.

8. Storage and Copying Without Quality Loss
Digital signals can be stored and copied repeatedly without any loss of quality.
Analog recordings (like cassettes or VHS) degrade with every copy or playback.

1.8 Sampling Theorem/Nyquist Sampling Theorem

The Sampling Theorem, also known as the Nyquist Sampling Theorem, is a fundamental principle in signal processing that defines how to sample a continuous analog signal so that it can be perfectly reconstructed from its samples in digital form.

Statement:
“If a continuous-time signal x(t) contains no frequency components higher than fmax Hz, then it can be completely reconstructed from its samples taken at a rate fs (sample per second) provided that :

$$f_s \geq 2f_{max}$$ ”

<u>Aliasing</u> occurs when the sampling rate is too low.

Nyquist rate: This minimum sampling rate at which signal can be converted into samples and can be recovered back without distortion is called the Nyquist rate.

Nyquist rate $f_{Nyquist} = 2f_{max}$.

Nyquist interval: The reciprocal of Nyquist rate is called as Nyquist interval.

Nyquist interval = $1 / 2f_{max}$ seconds.

Example:
An analog signal contains frequencies up to 8 kHz. What is the minimum sampling rate to avoid aliasing?
=> Here, $f_s \geq 2\times8$ kHz=16 kHz

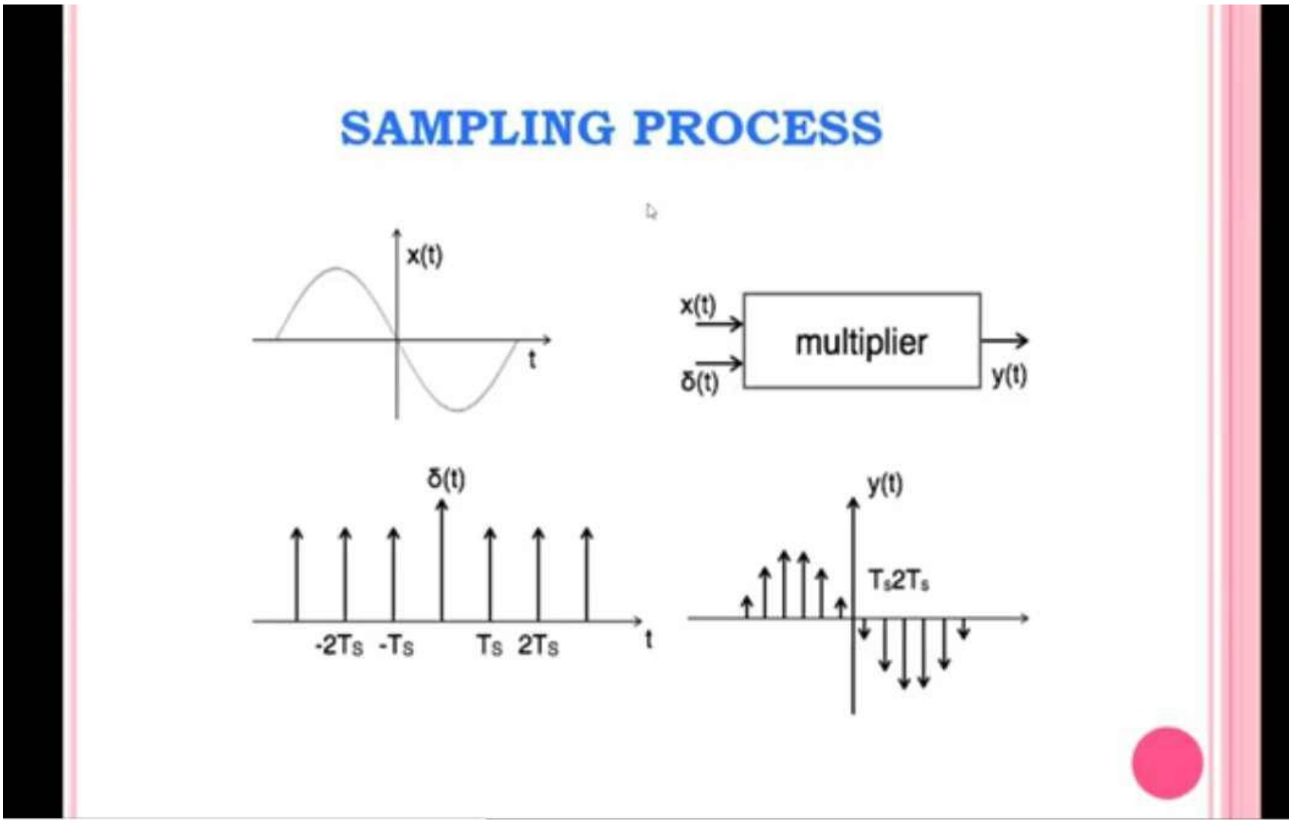
So, the signal must be sampled at least at 16,000 samples per second.

SAMPLING PROCESS:

The time interval between two subsequent sampling instants is called sampling interval.

In order to represent the original message signal faithfully (without loss of information), it is necessary to take as many samples of the original signal as possible.

Higher number of samples, closer is the representation.
The number of samples depends on the sampling rate and the maximum frequency of the signal to be sampled.



Q. A music signal has spectral components from 100 Hz to 8 kHz.
a. What is the Nyquist sampling rate for this signal?
b. If the signal is sampled at twice the Nyquist rate, how many samples are taken in 2 seconds?
Solution,
signal frequency range: 100 Hz to 8000 Hz
so,Maximum frequency fmax=8000 Hz
a.Nyquist Sampling Rate:

$$f_s=2\times f_{max}=2\times 8000=16000 \text{ Hz}$$

b. Samples at Twice the Nyquist Rate:

$$f_s=2\times 16000=32000 \text{ samples/sec}$$

therefore,Samples in 2 seconds=32000×2=64000 samples.

Q. A signal is sampled at **8 kHz**.
What is the **time interval** between two consecutive samples?

Solution:

given, fs = 8 Khz

Sampling period (T)=1/fs

=1/8000

=0.000125seconds

=125μs

Q. Let

$$x(t)=\cos(200\pi t)+\sin(600\pi t)$$

Find the **Nyquist sampling interval Ts**.

Solution,

comparing $\cos(200\pi t)$ by $A\cos 2\pi f_1 t$,we get

$A=1$

and $200\pi t = 2\pi f_1 t$

therefore,$f_1 = 100Hz$

comparing $\sin(600\pi t)$ by $B\sin 2\pi f_2 t$,we get

$B=1$

and $600\pi t = 2\pi f_1 t$

therefore,$f_1 = 300Hz$

$f_{max} = \max(f_1,f_2) = \max(100,300) = 300Hz$

Nyquist Rate

$f_s = 2f_{max}$

therefore, $f_s=2\times300=600$ Hz

Nyquist Interval

$T_s=1/f_s=1/600\approx1.667$ ms

Q.If the sampling rate is 44.1 kHz (used in audio CDs), what is the maximum frequency of the analog signal that can be sampled without aliasing?

Solution:

$f_s = 44.1$ KHz

$f_{max}= f_s/2 =44.1$ kHz$/2 =22.05$ kHz

1.10 Shannon Channel Capacity Theorem

Shannon Channel Capacity Theorem defines the maximum data rate (capacity) at which information can be transmitted over a noisy communication channel with arbitrarily low probability of error.

Statement:
"For a given communication channel with bandwidth B and a signal-to-noise ratio S/N, the maximum rate C (in bits per second) at which information can be transmitted with negligible error is:

$$\mathbf{C=Blog_2(1+S/N)}\text{ ''}$$

where,
C = Channel capacity in bits per second (bps)
B = Bandwidth of the channel in Hz
S = Signal power
N = Noise power

Q. A channel has a bandwidth of 3 kHz and an SNR of 30 dB.
Find the maximum data rate (channel capacity).
Solution:
SNR=30dB
SNR(linear)=10^(dB/10) =10^(30/10) =1000

Use Shannon formula

C=3000log2(1+1000) C=3000log2(1001) ≈3000×9.97
=29,910 bps

Example of Nyquist and Shannon Formulations (1 of 2)

- Spectrum of a channel between 3 MHz and 4 MHz ;

$SNR_{dB} = 24\text{ dB}$

$$B = 4\text{MHz} - 3\text{MHz} = 1\text{MHz}$$
$$SNR_{dB} = 24\text{ dB} = 10\log_{10}(SNR)$$
$$SNR = 251$$

- Using Shannon's formula

$$C = 10^6 \times \log_2(1+251) \approx 10^6 \times 8 = 8\text{Mbps}$$

Example of Nyquist and Shannon Formulations (2 of 2)

- How many signaling levels are required?

$$C = 2B\log_2 M$$
$$8 \times 10^6 = 2 \times \left(10^6\right) \times \log_2 M$$
$$4 = \log_2 M$$
$$M = 16$$

1.11 Bit Rate Vs Baud Rate

Bit Rate (Data Rate):

The number of bits transmitted per second.
Denoted as bps (bits per second)
Includes actual binary data (1s and 0s)
Measures how much data is transmitted

Bit Rate=Number of bits transmitted per second

Baud Rate (Symbol Rate):

The number of signal changes or symbols per second.
Denoted as baud
Each symbol can represent 1 or more bits
Measures how often the signal changes

Baud Rate=Number of symbols transmitted per second

The bit and baud rate have the connection:
Bit rate = baud rate x the number of bits per symbol

Example 1

What is the bit rate and baud rate for an analogue signal that carries 3 bits in each signal unit if 2000 signal units are sent per second?
Answer: Baud rate = 2000 baud per second, Bit rate = 2000 x 3 = 6000 bps

Example 2

What is the baud rate for an analogue signal if the bit rate of the signal is 2000 and each signal unit carries 4 bits?
Answer: Baud rate = 2000 / 4 = 500 baud

1.12 Numerical

Q1:A communication system transmits 1 bit per symbol at 2400 baud. What is the bit rate?
Solution:
Bit rate=Baud rate×Bits per symbol=2400×1=2400 bps

Q2: A system uses 8 distinct signal levels. It transmits at 1000 baud.
Calculate the bit rate.
Solution:
Bits per symbol=log2(8)=3
Bit rate=1000×3=3000 bps

Q3: A system transmits at 9600 bps using 4 bits per symbol. What is the baud rate?
Solution:
Baud rate=Bit rate/Bits per symbol=9600/4=2400 baud

Q4:
A system transmits at 4800 bps with a baud rate of 1200 baud. How many signal levels (M) are used?

Solution:
Bits per symbol=4800/1200=4

M=2bits per symbol=2^4=16 levels

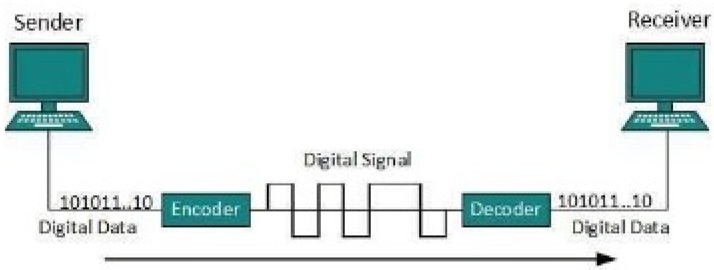
Q5: If a system uses QPSK (Quadrature Phase Shift Keying) which transmits 2 bits per symbol at a baud rate of 600, find the bit rate.
Solution:
Bit rate=600×2=1200 bps

1.13 Line Coding (Unipolar, Polar and Bi-Polar)

The process for converting digital data(bits) into digital signal(voltage or current) is said to be Line Coding.

Digital data is found in binary format. It is represented (stored) internally as series of 1s and 0s.

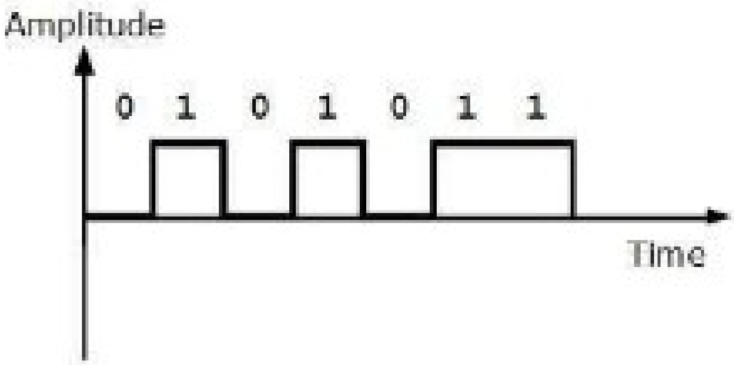
Digital signal is denoted by discreet signal, which represents digital data.



There are three types of line coding schemes available:

1.Uni-polar Encoding

Unipolar encoding schemes use single voltage level to represent data. In this case, to represent binary 1, high voltage is transmitted and to represent 0, no voltage is transmitted. It is also called Non-return-to-zero, because there is no rest condition i.e. it either represents 1 or 0.

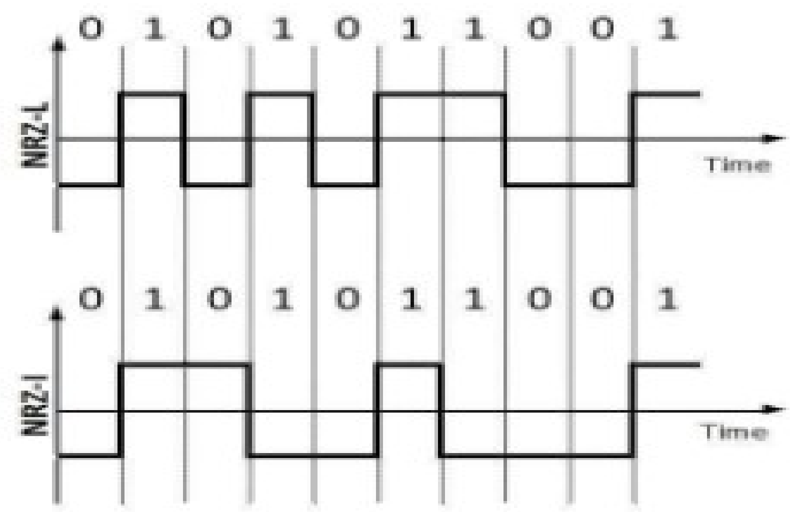


2. Polar Encoding

Polar encoding scheme uses multiple voltage levels to represent binary values. Polar encodings is available in four types:

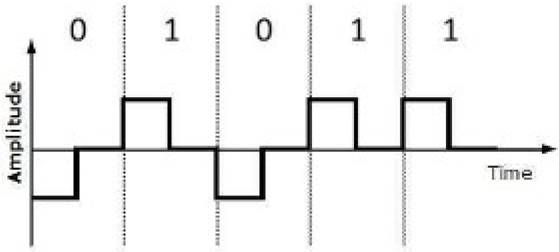
1. Polar Non-Return to Zero (Polar NRZ)

It uses two different voltage levels to represent binary values. Generally, positive voltage represents 1 and negative value represents 0. It is also NRZ because there is no rest condition.
NRZ scheme has two variants: NRZ-L and NRZ-I. NRZ-L changes voltage level at when a different bit is encountered whereas NRZ-I change voltage when a 1 is encountered.



2. Return to Zero (RZ)

Problem with NRZ is that the receiver cannot conclude when a bit ended and when the next bit is started, in case when sender and receiver’s clock are not synchronized. RZ uses three voltage levels, positive voltage to represent 1, negative voltage to represent 0 and zero voltage for none. Signals change during bits not between bits.



3. Manchester

This encoding scheme is a combination of RZ and NRZ-L. Bit time is divided into two halves. It transits in the middle of the bit and changes phase when a different bit is encountered.

4. Differential Manchester

This encoding scheme is a combination of RZ and NRZ-I. It also transit at the middle of the bit but changes phase only when 1 is encountered.

3. Bipolar Encoding

Bipolar encoding uses three voltage levels, positive, negative and zero. Zero voltage represents binary 0 and bit 1 is represented by altering positive and negative voltages.