Velammal College of Engineering and Technology, Madurai – 625 009

Department of Information Technology

UNIT II PULSE AND DATA COMMUNICATION

Pulse Communication: Pulse Amplitude Modulation (PAM) – Pulse Time Modulation (PTM) – Pulse code Modulation (PCM) - Comparison of various Pulse Communication System (PAM – PTM – PCM).

Data Communication: History of Data Communication - Standards Organizations for Data Communication- Data Communication Circuits - Data Communication Codes – Data communication Hardware - serial and parallel interfaces.

CO2: Demonstrate pulse and data communication techniques

INTRODUCTION

- Many Signals in Modern Communication Systems are digital. Also, analog signals are transmitted digitally.

- Reduced distortion and improvement in signal to noise ratios.

- PAM, PWM, PPM, PCM and DM.

- Data transmission, digital transmission, or digital communications is the physical transfer of data (a digital bit stream or a digitized analogue signal) over a point-to-point or point-to- multipoint communication channel.

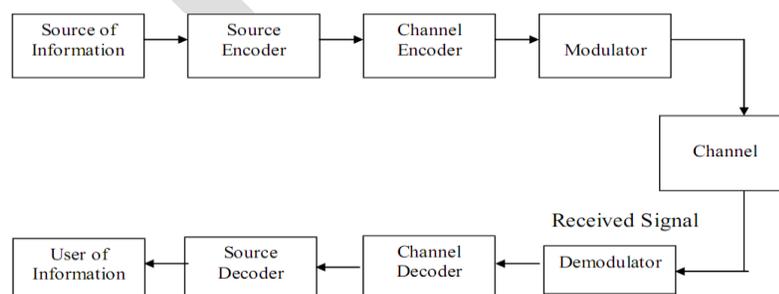Ex: optical fibers, wireless channels, computer buses....

ELEMENTS OF DIGITAL COMMUNICATION SYSTEMS

Fig. Block diagram of Digital Communication

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1. **Discrete Information Source:** It generates message to be transmitted. Examples are the data from computers, text data or tele type data.

2. **Source Encoder**: It assigns codes to the symbols (samples) generated from discrete information source. The code word having n number of bits. Each distinct sample having

   distinct(unique) code word. If code word length is 8 bit(n), we can have 256 distinct symbols(ie.,2^n).

3. **Channel Encoder**: We know that channel is the major source of notice due to that there are more chance of getting errors while propagating through channel. To avoid that channel encoding is required. In that extra bits are added to the binary sequence generated by the source encoder. These extra bits are called as redundant bits. These bits are defined with proper logic. The redundant will be helpful to detect the errors at the receiver bit sequence.

4. **Digital Modulator**: In digital modulator the message signal is digital data and carrier is analog one, in most cases we use sinusoidal waves. Some examples are ASK,FSK,PSK.MRI techniques.

5. **Channel**: It provides the link between transmitter and rceiver. Channel may be wired or wireless channel.

   ❖ Problems associated with channel:

1. Addictive Noise: This noise is occur due to internal solid state devices or resistors used in channel.

2. Ampltude and Phase Distortion: This noise is occurred due to non-linear characteristics of the channel.

3. Attenuation: This is due to internal resistance of the channel.

6. **Demodulator:** This device is used to detect the digital message signal from the modulated signal.

7. **Channel Decoder**: This is used to detect and correct the errors that occur in the digital message signal.

8. **Source Decoder**: This produces the sampling signal from the given digital message signal.

9. **Destination**: The sampled signal is converted into audio signal or video signal or any text signal depending on the signal.
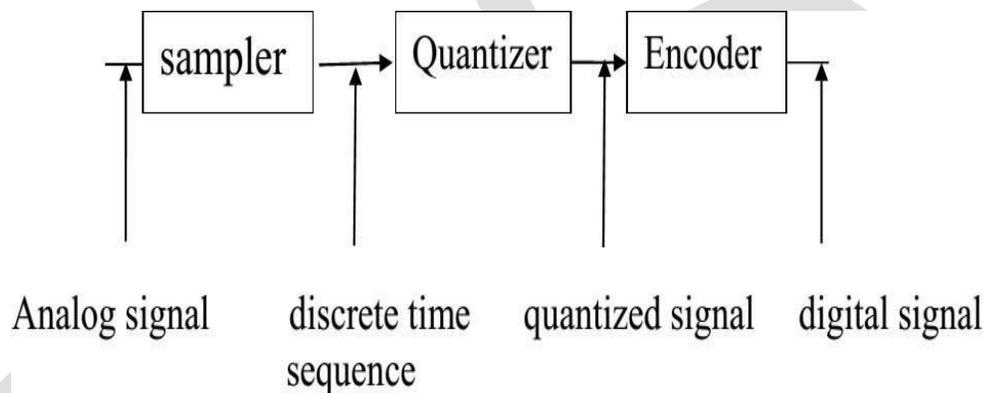

Fig. Basic block diagram of an A/D converter

**Advantages of digital communication systems**

1. Easy way of transmission of signals

2. Connection of more calls through one channel i.e., Multiplexing is possible using Digital Communication.

3. Source Encoding and Channel Encoding can be used to detect errors at the received signal.

4. Using repeaters between source and destination, we can reproduce the original signal with less distortions.

5. Security is the major advantage of digital communication compared to Analog Communication.

6. Transmitting analogue signals digitally allows for greater signal processing capability.

7. Digital communication can be done over large distances through internet and other things.

8. The messages can be stored in the device for longer times, without being damaged.

9. Advancement in communication is achieved through Digital Communication.

Disadvantages of digital communication systems

1. Sampling Error

2. Digital communications require greater bandwidth than analogue to transmit the same information.

3. The detection of digital signals requires the communications system to be synchronized, whereas generally speaking this is not the case with analogue systems.

4. Digital signals are often the approximation of voice signals, ie, we don"t get the exact analogue signal.

TYPES OF MODULATION – TREE DIAGRAM
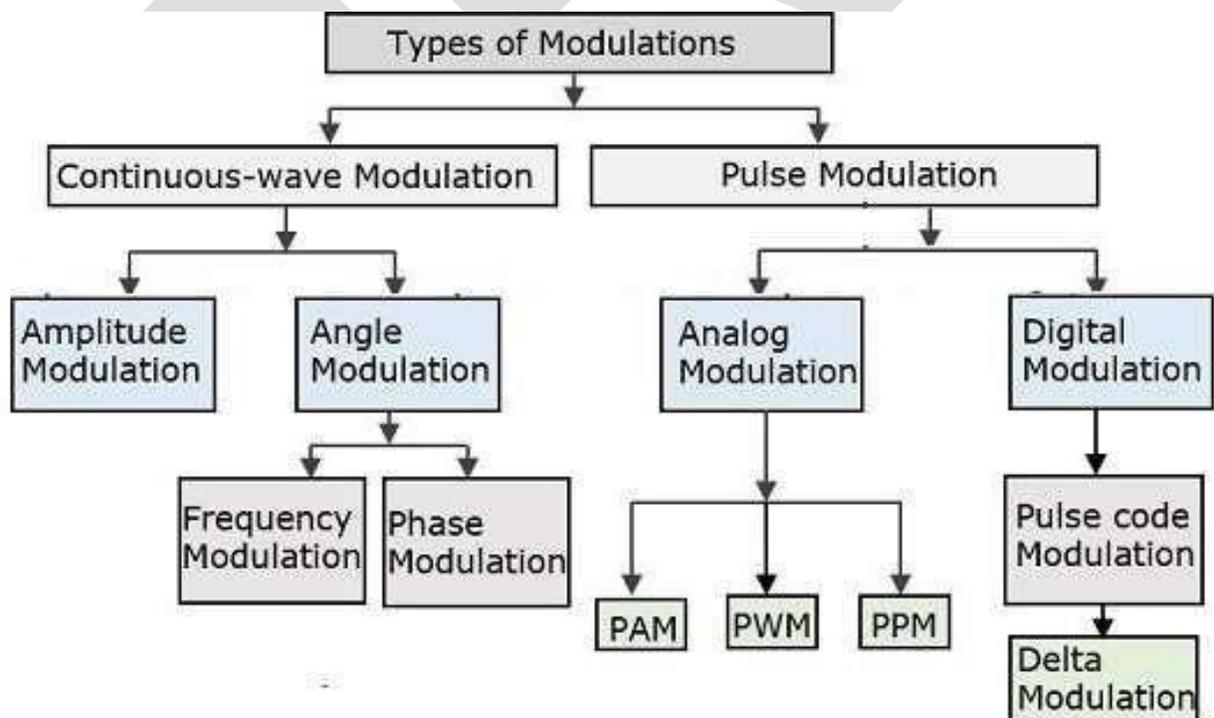
In Continuous Wave modulation schemes some parameter of modulated wave varies continuously with message.

In Analog pulse modulation some parameter of each pulse is modulated by a particular sample value of the message.

Pulse modulation of two types

1. Analog Pulse Modulation

    - Pulse Amplitude Modulation (PAM)

    - Pulse width Modulation (PWM)

    - Pulse Position Modulation (PPM)

2. Digital Pulse Modulation

    - Pulse code Modulation (PCM)

    - Delta Modulation (DM)

1. **Analog Pulse Modulation**

Analog pulse modulation results when some attribute of a pulse varies continuously in one-to-one correspondence with a sample value. In analog pulse modulation systems, the amplitude, width, or position of a pulse can vary over a continuous range in accordance with the message amplitude at the sampling instant, as shown in Figure 6.2. These lead to the following

Three types of pulse modulation:

    1. Pulse Amplitude Modulation (PAM)

    2. Pulse Width Modulation (PWM)

    3. Pulse Position Modulation (PPM)

PAM: In this scheme high frequency carrier (pulse) is varied in accordance with sampled value of message signal.

PWM: In this width of carrier pulses are varied in accordance with sampled values of message signal. Example: Speed control of DC Motors.

PPM: In this scheme position of high frequency carrier pulse is changed in accordance with the sampled values of message signal.
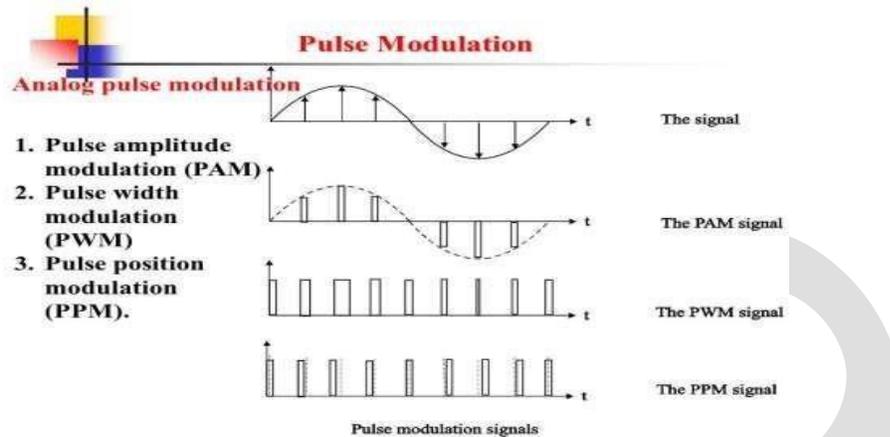

Fig. Representation of Various Analog Pulse Modulations

2. **Digital Pulse Modulation**

In systems utilizing digital pulse modulation, the transmitted samples take on only discrete values. Two important types of digital pulse modulation are:

    1. Delta Modulation (DM)

    2. Pulse Code Modulation (PCM)

ANALOG PULSE MODULATION

1. **Pulse Amplitude Modulation (PAM):**

In pulse amplitude modulation, the amplitude of regular interval of periodic pulses or electromagnetic pulses is varied in proposition to the sample of modulating signal or message signal. This is an analog type of modulation. In the pulse amplitude modulation, the message signal is sampled at regular periodic or time intervals and this each sample is made

proportional to the magnitude of the message signal. These sample pulses can be transmitted directly using wired media or we can use a carrier signal for transmitting through wireless.
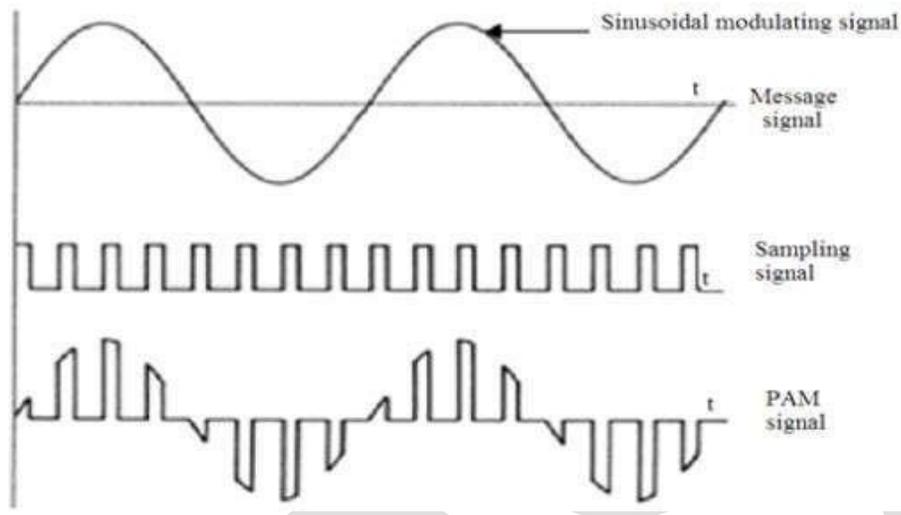

Fig. Pulse Amplitude Modulation Signal

There are two types of sampling techniques for transmitting messages using pulse amplitude modulation, they are

- **FLAT TOP PAM:** The amplitude of each pulse is directly proportional to instantaneous modulating signal amplitude at the time of pulse occurrence and then keeps the amplitude of the pulse for the rest of the half cycle.
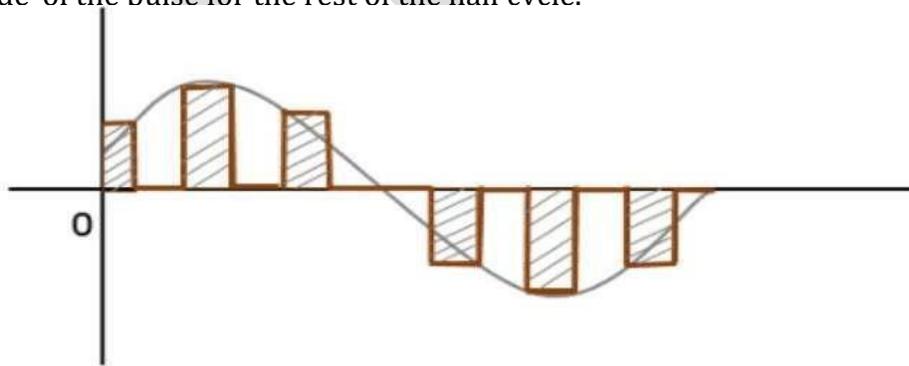

Fig. Flat Top PAM

- **Natural PAM:** The amplitude of each pulse is directly proportional to the instantaneous modulating signal amplitude at the time of pulse occurrence and then follows the amplitude of the modulating signal for the rest of the half cycle.
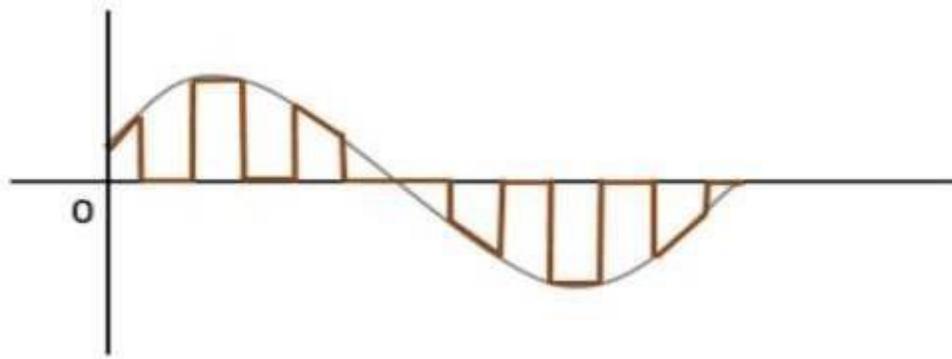
Fig. Natural PAM

Flat top PAM is the best for transmission because we can easily remove the noise and we can also easily recognize the noise. When we compare the difference between the flat top PAM and natural PAM, flat top PAM principle of sampling uses sample and hold circuit. In natural principle of sampling, noise interference is minimum. But in flat top PAM noise interference maximum. Flat top PAM and natural PAM are practical and sampling rate satisfies the sampling criteria.
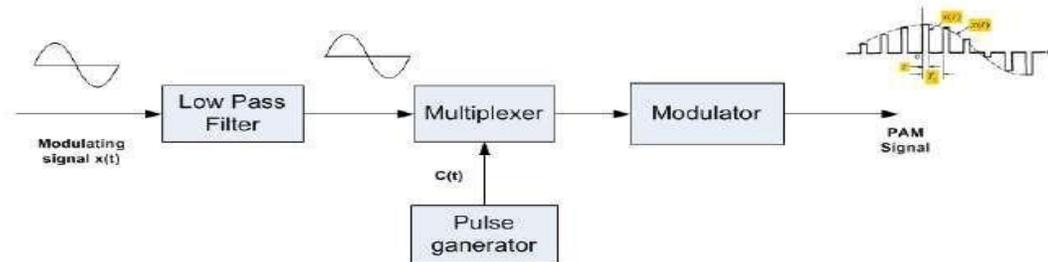
There are two types of pulse amplitude modulation based on signal polarity

1. Single polarity pulse amplitude modulation
2. Double polarity pulse amplitude modulation

In single polarity pulse amplitude modulation, there is fixed level of DC bias added to the message signal or modulating signal, so the output of modulating signal is always positive. In the double polarity pulse amplitude modulation, the output of modulating signal will have both positive and negative ends.

Carrier Pulses Train Used for Modulation

Analog Signal for Modulating

Pulse Amplitude Modulated Output
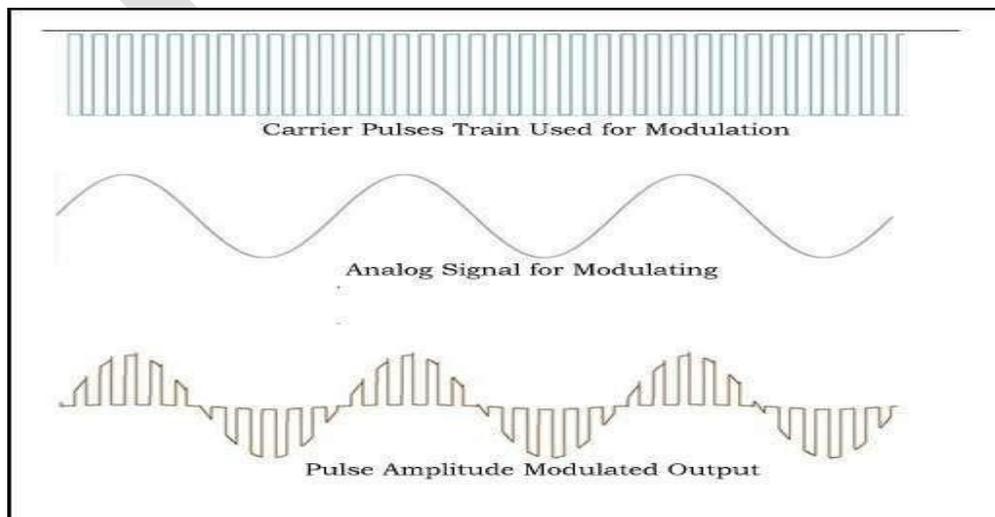
Block diagram of PAM generation

System for recovering message signal m(t) from PAM signal s(t).
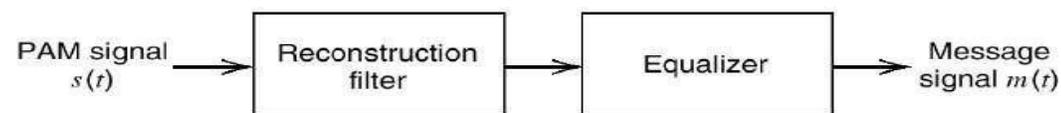
Advantages of Pulse Amplitude Modulation (PAM):

- It is the base for all digital modulation techniques and it is simple process for both modulation and demodulation technique.
- No complex circuitry is required for both transmission and reception. Transmitter and receiver circuitry is simple and easy to construct.
- PAM can generate other pulse modulation signals and can carry the message or information at same time.

Disadvantages of Pulse Amplitude Modulation (PAM):

- Bandwidth should be large for transmitting the pulse amplitude modulation signal. Due to Nyquist criteria also high bandwidth is required.
- The frequency varies according to the modulating signal or message signal. Due to these variations in the signal frequency, interferences will be there. So noise will be great. For PAM, noise immunity is less when compared to other modulation techniques. It is almost equal to amplitude modulation.

- Pulse amplitude signal varies, so power required for transmission will be more, peak power is also, even at receiving more power is required to receive the pulse amplitude signal.

Applications of Pulse Amplitude Modulation (PAM):

- It is mainly used in Ethernet which is type of computer network communication, we know that we can use Ethernet for connecting two systems and transfer data between the systems. Pulse amplitude modulation is used for Ethernet communications.
- It is also used for photo biology which is a study of photosynthesis.
- Used as electronic driver for LED lighting.
- Used in many micro controllers for generating the control signals etc.

DIGITAL PULSE MODULATION

Modulation is the process of varying one or more parameters of a carrier signal in accordance with the instantaneous values of the message signal.

1. PULSE CODE MODULATION(PCM)

The message signal is the signal which is being transmitted for communication and the carrier signal is a high frequency signal which has no data, but is used for long distance transmission.

There are many modulation techniques, which are classified according to the type of modulation employed. Of them all, the digital modulation technique used is **Pulse Code Modulation (PCM)**.

A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., **1s** and **0s**. The output of a PCM will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.


Sine Wave Input

PCM

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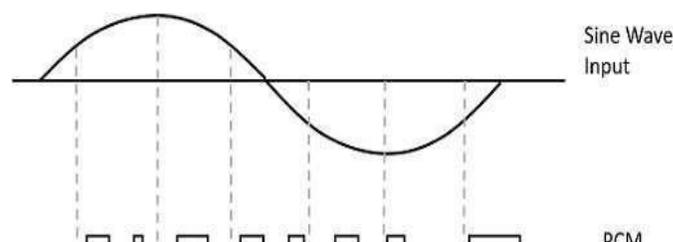
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as **digital**. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train. Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.
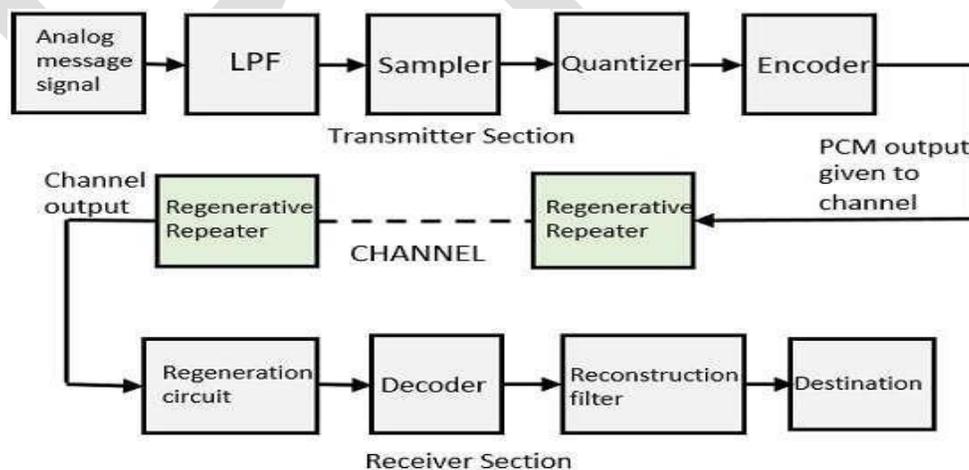
- ➢ **Low Pass Filter**

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the

message signal.

> ➤ **Sampler**

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem.

> ➤ **Quantizer**

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer reduces the redundant bits and compresses the value.

> ➤ **Encoder**

Encoder assigns code words to quantized sampled values. This coding techniques uses bits 0 and 1. If number of quantized levels are 16 then each sample is assigned with 4 bit code word.

> ➤ **Regenerative repeater:**

The PCM has an ability to control the distortion and noise caused by the transmission of bits along the channel. This ability is accomplished by several regenerative repeaters located at sufficient placing along channel.

Regenerative repeaters have three functions.

1. Equalizing
2. Timing circuits
3. Decision making device

Equalizer shapes the received pulse so as to compensate amplitude and phase distortion caused by the channel.

Timing circuits provides periodic pulse trains.

- Decision making device compares amplitude of equalized pulse plus noise to the pre-defined threshold levels to make decisions whether the pulse is present or not.

- If the pulse is present (i.e. decision is yes), clean new pulse is generated and transmitted through channel to next regenerative pulse. If the pulse is not present (i.e. the decision is no), it generates clean base line to next regenerative repeater, provided the noise too large caused bit error by taking the wrong decision

➢ **Decoder**

Decoder reboots all the received bits to make more words then it decodes as quantized PAM signals.

➢ **Reconstruction Filter:**

All coded words are passed through low pass filter so that analog signal can be reconstructed from quantized PAM signal. The cut off frequency of low pass filter is f_m Hz which is equal to band width of message signal.

➢ **Destination**

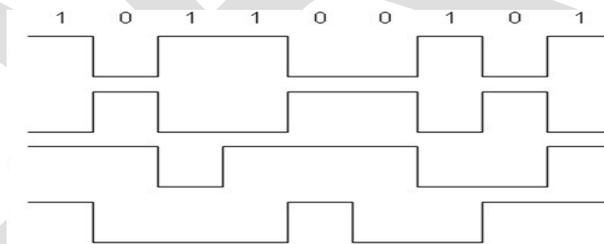It receives the signal from the reconstructive filter output is analog signal.

Fig. PCM waveform

Bit rate and bandwidth requirements of PCM :

➢ The bit rate of a PCM signal can be calculated form the number of bits per sample × the sampling rate. Bit rate $= nb \times fs$ The bandwidth required to transmit this signal depends on the type of line encoding used.

➢ A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Important Relations

- Quantization Noise $(Nq)=\Delta^2/2$

- Signal to Noise ratio

  $(SQNR)=32.22n$ or $SQNR$ in $dB=1.76+6.02n\cong(1.8+6n)dB$

- *Bit rate=No.of bits per sample×sampling rate=nfs*

- Bandwidth for PCM signal =n.fm

   Where,

   n – No. of bits in PCM code Fm

   – signal bandwidth

   fs – sampling rate

SAMPLING, QUANTIZATION AND CODING

1. **Sampling**

   - **Definition:** Sampling is defined as ―The process of measuring the instantaneous values of continuous-time signal in a discrete form.‖

   - **Sample** is a piece of data taken from the whole data which is continuous in the time domain.

When a source generates an analog signal and if that has to be digitized, having **1s** and **0s** i.e., High or Low, the signal has to be discretized in time. This discretization of analog signal is called as Sampling.

The following figure indicates a continuous-time signal **x (t)** and a sampled signal **xₛ (t)**. When **x (t)** is multiplied by a periodic impulse train, the sampled signal **xₛ (t)** is obtained.
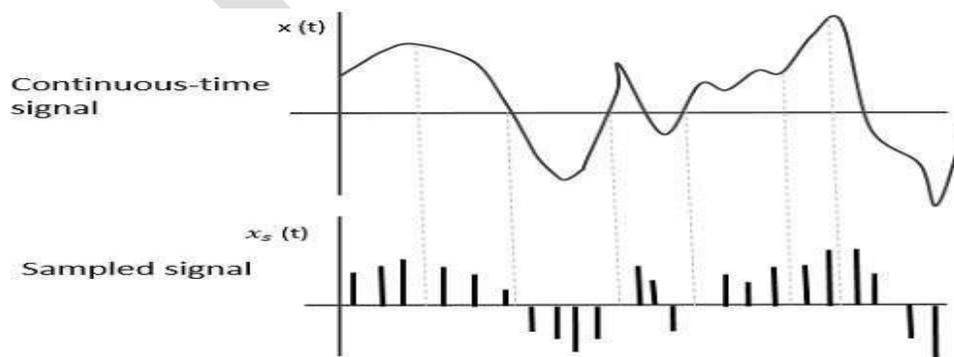
Sampling Rate

To discretize the signals, the gap between the samples should be fixed. That gap can be termed as a **sampling period T_s.**

$$\text{Sampling Frequency } fs=1/Ts$$

Where,

Ts is the sampling time

fs is the sampling frequency or the sampling rate

Sampling frequency -is the reciprocal of the sampling period. This sampling frequency, can be simply called as **Sampling rate**. The sampling rate denotes the number of samples taken per second, or for a finite set of values.

For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as Nyquist rate

Nyquist Rate

Suppose that a signal is band-limited with no frequency components higher than **W** Hertz. That means, **W** is the highest frequency. For such a signal, for effective reproduction of the original signal, sampling rate should be twice the highest frequency.
This means,

$$fs=2W$$

Where,

fs is the sampling rate

W is the highest frequency

This rate of sampling is called as **Nyquist rate**.

A theorem called, Sampling Theorem, was stated on the theory of this Nyquist rate.

Sampling Theorem

The sampling theorem, which is also called as **Nyquist theorem**, delivers the theory of sufficient sample rate in terms of bandwidth for the class of functions that are band limited.

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The sampling theorem states that, – a signal can be exactly reproduced if it is sampled at the rate f_s which is greater than twice the maximum frequency **W**.

To understand this sampling theorem, let us consider a band-limited signal, i.e., a signal whose value is **non-zero** between some **–W** and **W** Hertz.
Such a signal is represented as x(f)=0for|f|>W

For the continuous-time signal **x (t)**, the band-limited signal in frequency domain, can be represented as shown in the following figure.

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Band limited signal
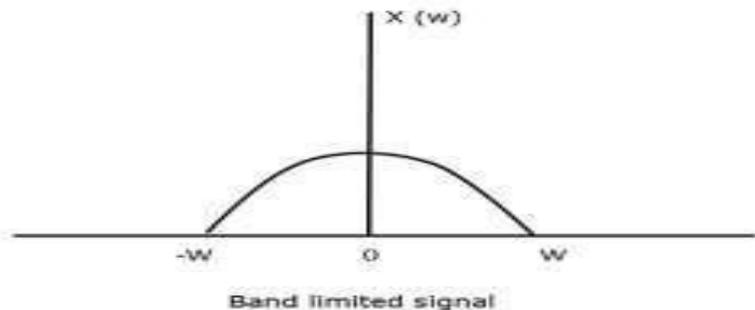
We need a sampling frequency, a frequency at which there should be no loss of information, even after sampling. For this, we have the Nyquist rate that the sampling frequency should be two times the maximum frequency. It is the critical rate of sampling.

If the signal **x(t)** is sampled above the Nyquist rate, the original signal can be recovered, and if it is sampled below the Nyquist rate, the signal cannot be recovered.
The following figure explains a signal, if sampled at a higher rate than **2w** in the frequency domain.
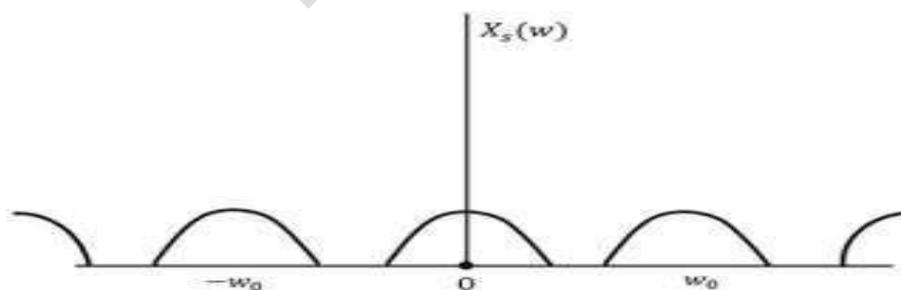
The above figure shows the Fourier transform of a signal xs(t). Here, the information is reproduced without any loss. There is no mixing up and hence recovery is possible.
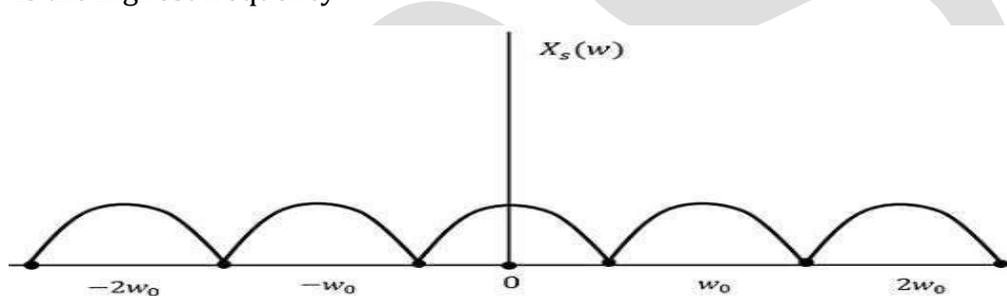
The Fourier Transform of the signal xs(t) is Xs (w)=1Ts∑n=−∞∞X(w−nw0)

Where Ts = **Sampling Period** and w0=2πTs

Let us see what happens if the sampling rate is equal to twice the highest frequency (**2W**) That means, Fs =2W

Where,

Fs is the sampling frequency

W is the highest frequency

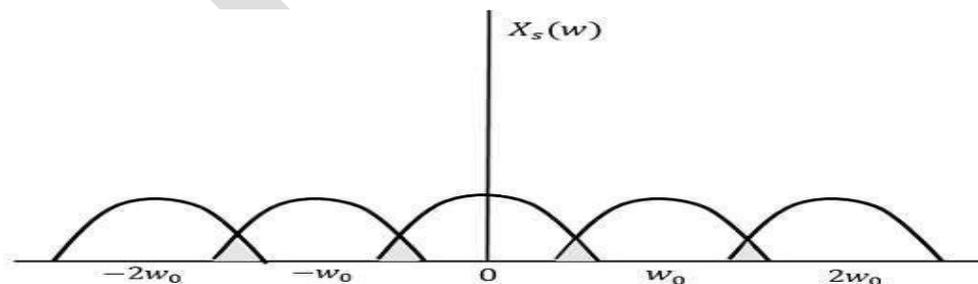The result will be as shown in the above figure. The information is replaced without any loss. Hence, this is also a good sampling rate.

Now, let us look at the condition,

Fs <2W

The resultant pattern will look like the following figure

We can observe from the above pattern that the over-lapping of information is done, which leads to mixing up and loss of information. This unwanted phenomenon of over-

lapping is called as Aliasing

Aliasing

Aliasing can be referred to as —the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a low-frequency component in the spectrum of its sampled version.

The corrective measures taken to reduce the effect of Aliasing are –

- In the transmitter section of PCM, a **low pass anti-aliasing filter** is employed, before the sampler, to eliminate the high frequency components, which are unwanted.
- The signal which is sampled after filtering, is sampled at a rate slightly higher than the Nyquist rate.

This choice of having the sampling rate higher than Nyquist rate, also helps in the easier design of the **reconstruction filter** at the receiver.

Scope of Fourier Transform

It is generally observed that, we seek the help of Fourier series and Fourier transforms in analyzing the signals and also in proving theorems. It is because –

- The Fourier Transform is the extension of Fourier series for non-periodic signals.
- Fourier transform is a powerful mathematical tool which helps to view the signals in different domains and helps to analyze the signals easily.
- Any signal can be decomposed in terms of sum of sines and cosines using this Fourier transform. The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.

Quantizing an Analog Signal

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital has to undergo sampling and quantizing
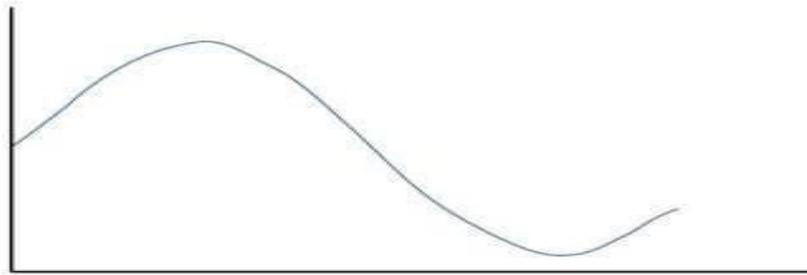
The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels.

Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.

Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as **representation levels** or **reconstruction levels**. The spacing between the two adjacent representation levels is called a **quantum** or **step-size**.

The following figure shows the resultant quantized signal which is the digital form for the given analog signal.
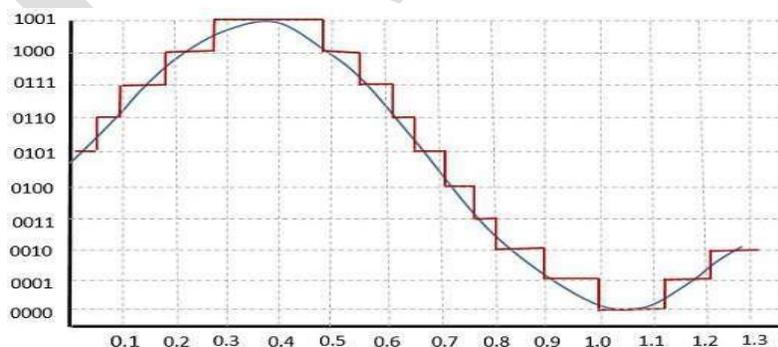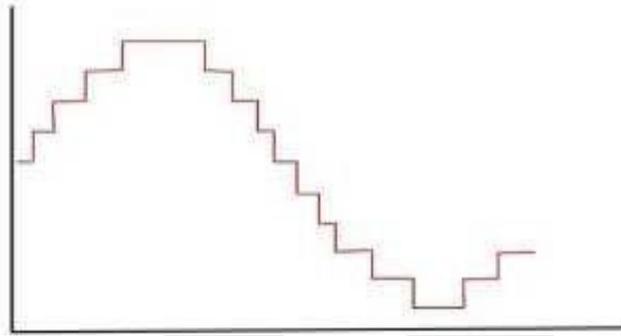

This is also called as **Stair-case** waveform, in accordance with its shape.

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

1. The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**.

2. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

There are two types of uniform quantization.

1. Mid-Rise type
2. Mid-Tread type.

The following figures represent the two types of uniform quantization
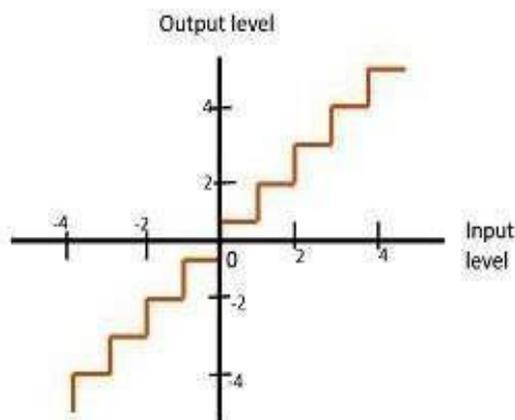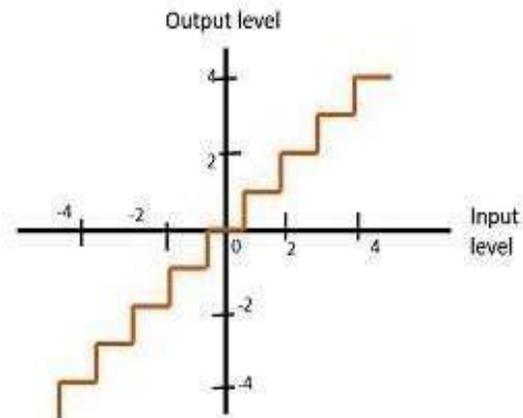

Fig 1 : Mid-Rise type Uniform Quantization

Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

1. The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair- case like graph. The quantization levels in this type are even in

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number.

2. The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.

Both the mid-rise and mid-tread type of uniform quantizer are symmetric about the origin.

$$\Delta = (max-min)L$$

$$nb = log2L$$

Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.The difference between an input value and its quantized value is called a **Quantization Error**.

A **Quantizer** is a logarithmic function that performs Quantization (rounding off the value). An analog-to- digital converter (**ADC**) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.

Original and Quantized Signal

Quantization Error
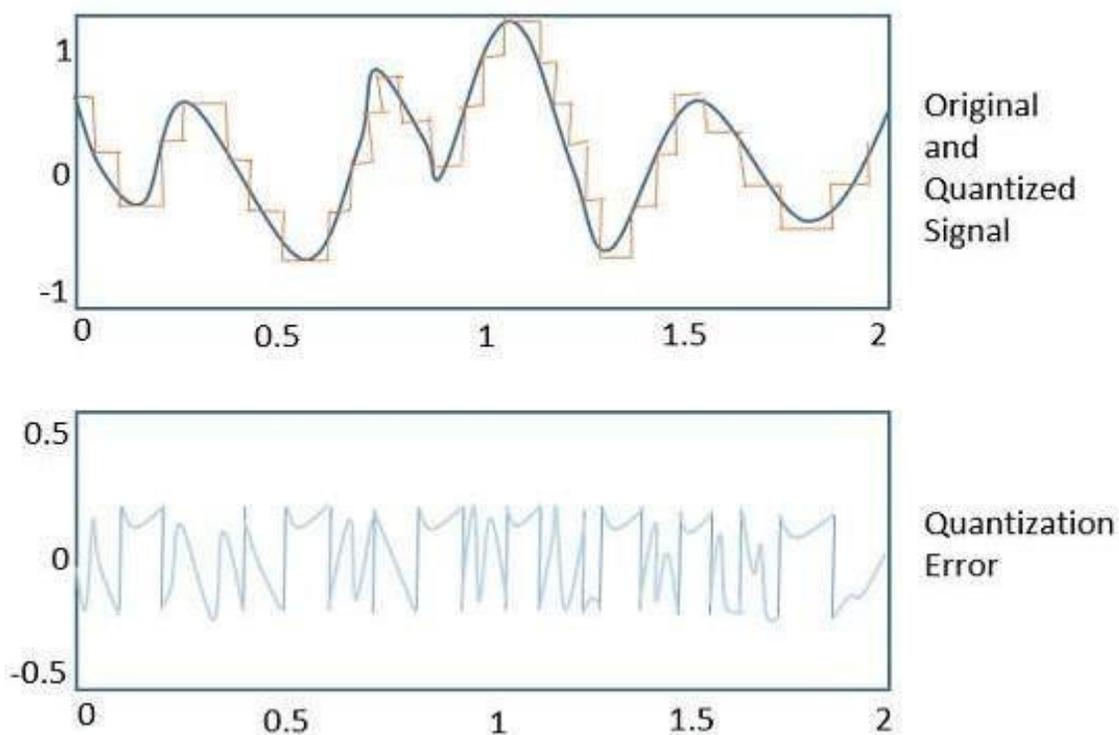
Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as Quantization Noise.

COMPANDING IN PCM SYSTEMS

The word **Companding** is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique
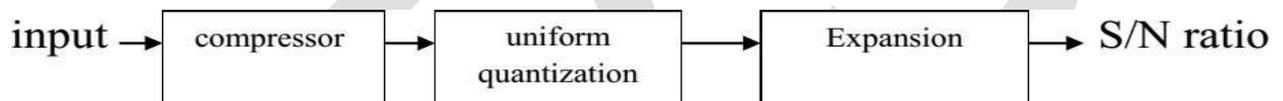
Fig. compounding

Companding means it amplifies the low-level signals as well as attenuate high level at the transmitter side. At the receiver side reverse operation done. It attenuates the low-level signals and amplifies the high-level signals you get the original signal. Non-uniform quantization cannot be applied directly by using companding technique.

Fig Companding curves for PCM

Companding is used to maintain constant Signal to Noise Ratio throughout dynamic quantization ratio

Fig. Non-Uniform Quantization

There are two types of Compounding techniques. They are –

1.A-law compounding Technique

    i. Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and no compression is done.

    ii. A-law has mid-rise at the origin. Hence, it contains a non-zero value.

    iii. A-law Companding is used for PCM telephone systems.

$$Y = \left. \frac{Ax}{1+\ln(A)} \right| \quad ; \text{ where } 0 \le x \le 1/A$$
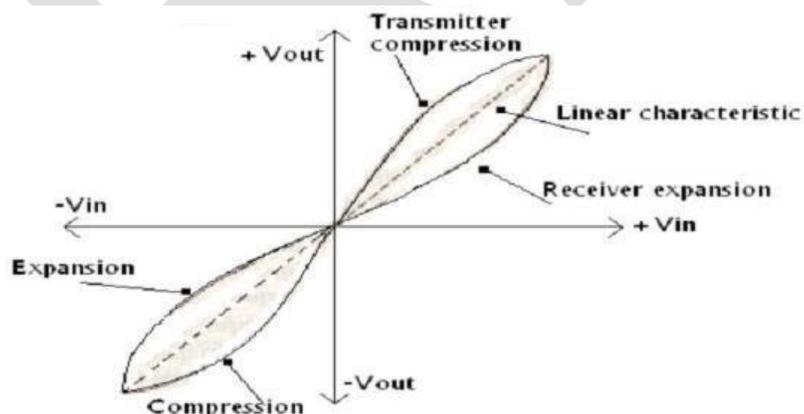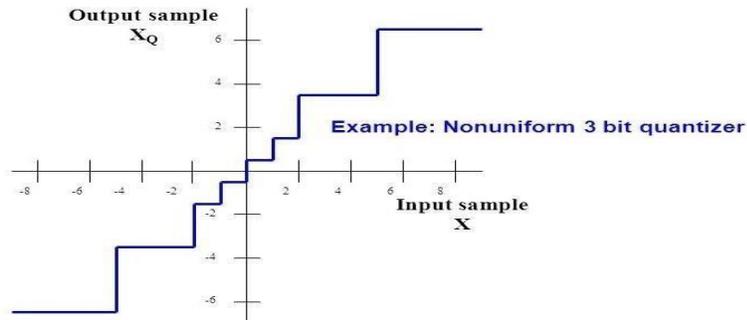
$$= \left. \frac{1+\ln Ax}{1+\ln(A)} \right| ; \quad 1/A \le x \le 1 \quad \text{practically } A=87.56$$

if A=1 we get uniform quantization

2.μ-law compounding Technique

    i. Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and no compression is done.

    ii. μ-law has mid-tread at the origin. Hence, it contains a zero value.

    iii. μ-law companding is used for speech and music signals.

$$Y = \pm \frac{\ln(1+\mu|x|)}{\ln(1+\mu)}; \quad |x| \le 1 \quad \text{Practically } \mu \text{ value is 256}$$

For the samples that are highly correlated, when encoded by PCM technique, leave redundant information behind. To process this redundant information and to have a better output, it is a wise decision to take a predicted sampled value, assumed from its previous output and summarize them with the quantized values. Such a process is called as **Differential PCM (DPCM)** technique.

Introduction to Data Communications:

In Data Communications, data generally are defined as information that is stored in digital form. Data communications is the process of transferring digital information between two or more points. Information is defined as the knowledge or intelligence. Data communications can be summarized as the transmission, reception, and processing of digital information. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs). The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

A data communications system has five components:
1. **Message:** The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.
2. **Sender:** The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.
3. **Receiver:** The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.
4. **Transmission medium:** The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.
5. **Protocol**: A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices.

Standards Organizations for Data Communications

An association of organizations, governments, manufacturers and users form the standards organizations and are responsible for developing, coordinating and maintaining the standards. The intent is that all data communications equipment manufacturers and users comply with these standards. The primary standards organizations for data communication are:

1. **International Standard Organization (ISO)**

   ISO is the international organization for standardization on a wide range of subjects. It is comprised mainly of members from the standards committee of various governments throughout the world. It is even responsible for developing models which provides high level of system compatibility, quality enhancement, improved productivity and reduced costs. The ISO is also responsible for endorsing and coordinating the work of the other standards organizations.

2. **International Telecommunications Union-Telecommunication Sector (ITU-T)**

   ITU-T is one of the four permanent parts of the International Telecommunications Union based in Geneva, Switzerland. It has developed three sets of specifications: the V series for modem interfacing and data transmission over telephone lines, the X series for data transmission over public digital networks, email and directory services;

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the I and Q series for Integrated Services Digital Network (ISDN) and its extension Broadband ISDN. ITU-T membership consists of government authorities and representatives from many countries and it is the present standards organization for the United Nations.

3. **Institute of Electrical and Electronics Engineers (IEEE)**

   IEEE is an international professional organization founded in United States and is comprised of electronics, computer and communications engineers. It is currently the world's largest professional society with over 200,000 members. It develops communication and information processing standards with the underlying goal of advancing theory, creativity, and product quality in any field related to electrical engineering.

4. **American National Standards Institute (ANSI)**

   ANSI is the official standards agency for the United States and is the U.S voting representative for the ISO. ANSI is a completely private, non-profit organization comprised of equipment manufacturers and users of data processing equipment and services. ANSI membership is comprised of people form professional societies, industry associations, governmental and regulatory bodies, and consumer goods.

5. **Electronics Industry Association (EIA)**

   EIA is a non-profit U.S. trade association that establishes and recommends industrial standards. EIA activities include standards development, increasing public awareness, and lobbying and it is responsible for developing the RS (recommended standard) series of standards for data and communications.

6. **Telecommunications Industry Association (TIA)**

   TIA is the leading trade association in the communications and information technology industry. It facilitates business development opportunities through market development, trade promotion, trade shows, and standards development. It represents manufacturers of communications and information technology products and also facilitates the convergence of new communications networks.

7. **Internet Architecture Board (IAB)**

   IAB earlier known as Internet Activities Board is a committee created by ARPA (Advanced Research Projects Agency) so as to analyze the activities of ARPANET whose purpose is to accelerate the advancement of technologies useful for U.S military. IAB is a technical advisory group of the Internet Society and its responsibilities are:

   I.   Oversees the architecture protocols and procedures used by the Internet.

   II.   Manages the processes used to create Internet Standards and also serves as an appeal board for complaints regarding improper execution of standardization process.

   III.   Responsible for administration of the various Internet assigned numbers

   IV.   Acts as a representative for Internet Society interest in liaison relationships with other organizations.

   V.   Acts as a source of advice and guidance to the board of trustees and officers of Internet Society concerning various aspects of internet and its

technologies.

8. Internet Engineering Task Force (IETF)

The IETF is a large international community of network designers, operators, vendors and researchers concerned with the evolution of the Internet architecture and smooth operation of the Internet.

9. Internet Research Task Force (IRTF)

The IRTF promotes research of importance to the evolution of the future Internet by creating focused, long-term and small research groups working on topics related to Internet protocols, applications, architecture and technology.

Layered Network Architecture

To reduce the design complexity, most of the networks are organized as a series of **layers** or **levels**, each one build upon one below it. The basic idea of a layered architecture is to divide the design into small pieces. Each layer adds to the services provided by the lower layers in such a manner that the highest layer is provided a full set of services to manage communications and run the applications. The benefits of the layered models are modularity and clear interfaces, i.e. open architecture and comparability between the different providers' components. A basic principle is to ensure independence of layers by defining services provided by each layer to the next higher layer without defining how the services are to be performed. This permits changes in a layer without affecting other layers. The basic elements of a layered model are services, protocols and interfaces. A **service** is a set of actions that a layer offers to another (higher) layer. **Protocol** is a set of rules that a layer uses to exchange information with a peer entity. These rules concern both the contents and the order of the messages used. Between the layers service interfaces are defined. The messages from one layer to another are sent through those interfaces.
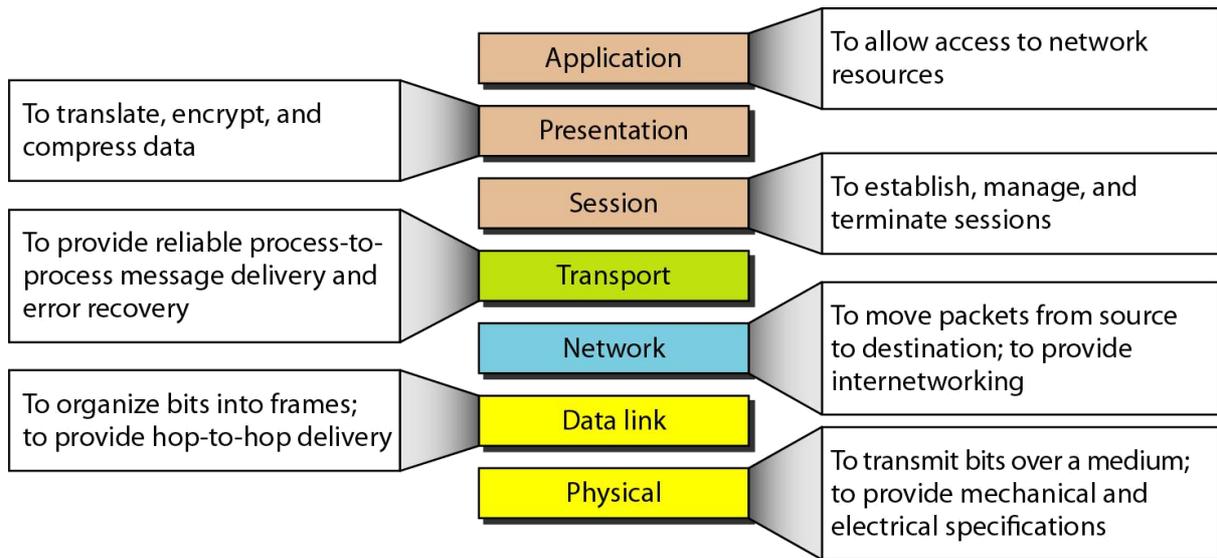
In a n-layer architecture, layer n on one machine carries on conversation with the layer n on other machine. The rules and conventions used in this conversation are collectively known as the layer-n protocol. Basically, a protocol is an agreement between the communicating parties on how communication is to proceed. Five-layer architecture is shown below**;** the entities comprising the corresponding layers on different machines are called **peers**. In other words, it is the peers that communicate using protocols. In reality, no data is transferred from layer n on one machine to layer n of another machine. Instead, each layer passes data and control information to the layer immediately below it, until the lowest layer is reached. Below layer-1 is the physical layer through which actual communication occurs.
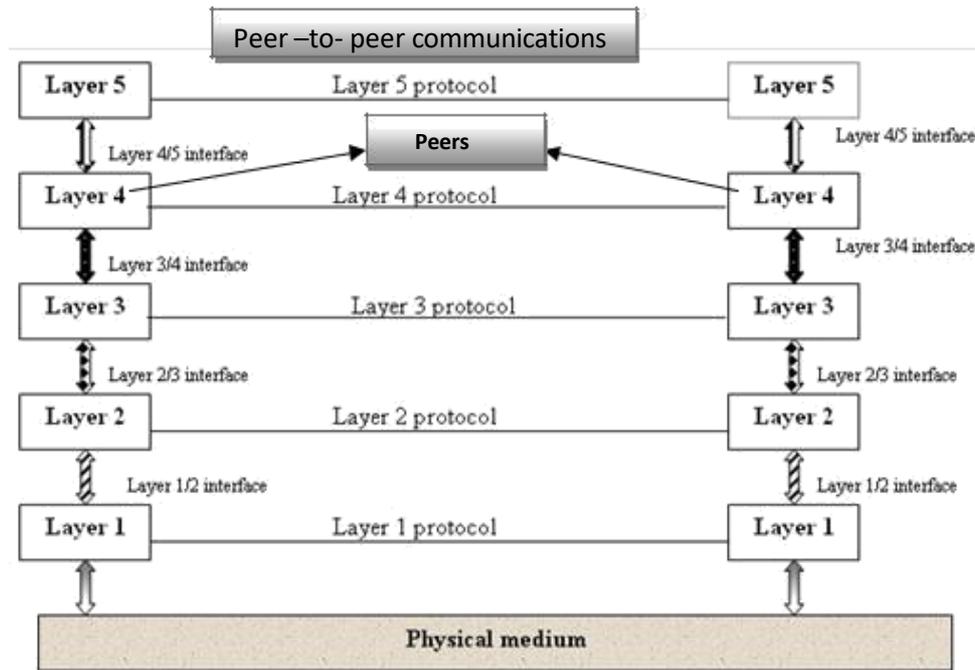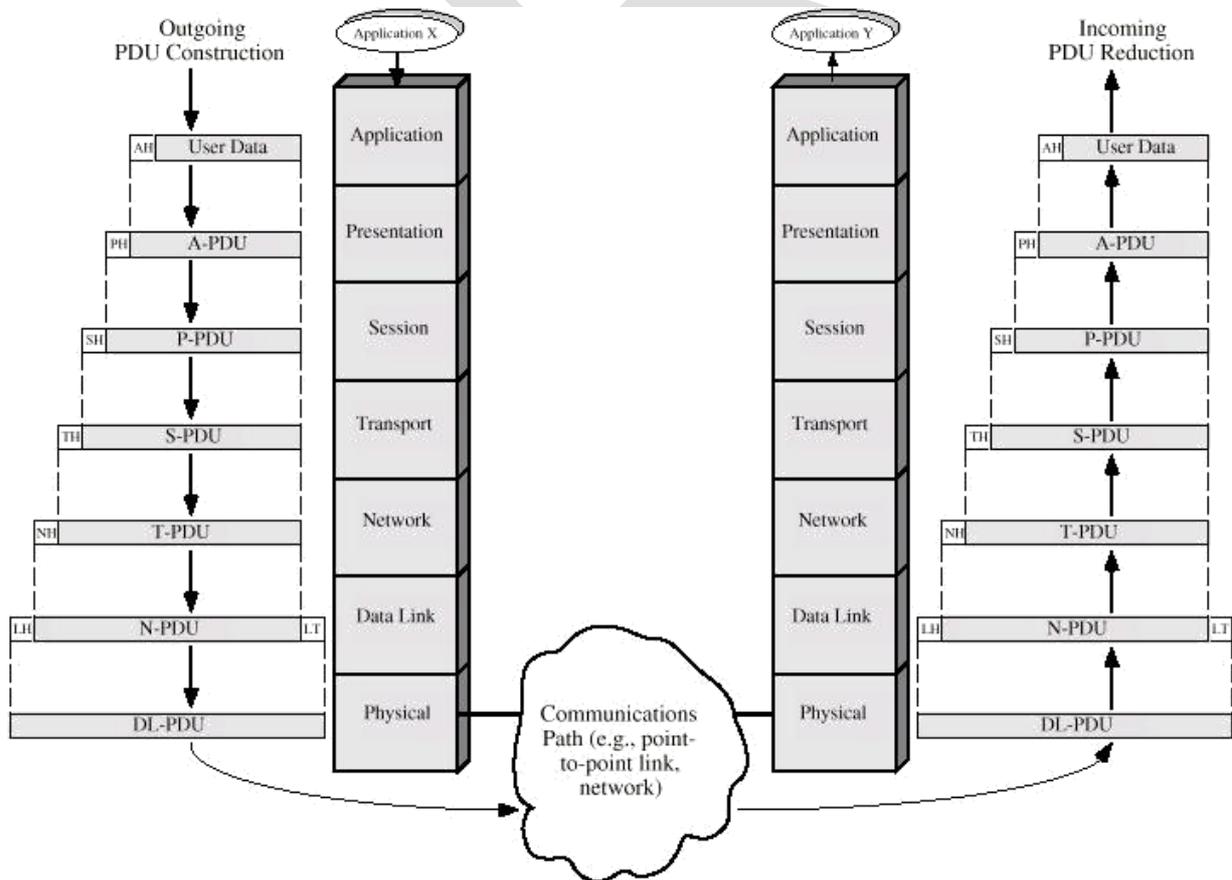
With layered architectures, communications between two corresponding layers requires a unit of data called a **protocol data unit (PDU)**. A PDU can be a header added at the beginning of a message or a trailer appended to the end of a message. Data flows downward through the layers in the source system and upwards at the destination address. As data passes from one layer into another, headers and trailers are added and removed from the PDU. This process of adding or removing PDU information is called **encapsulation/decapsulation.** Between each pair of adjacent layers there is an **interface**. The interface defines which primitives operations and services the lower layer offers to the upper layer adjacent to it. A set of layers and protocols is known as **network architecture**. A list of protocols used by a certain system, one protocol per layer, is called **protocol stack**.

Open Systems Interconnection (OSI)

International standard organization (ISO) established a committee in 1977 to develop architecture for computer communication and the OSI model is the result of this effort. In 1984, the Open Systems Interconnection (OSI) reference model was approved as an international standard for communications architecture. The term "open" denotes the ability to connect any two systems which conform to the reference model and associated standards. The OSI model describes how information or data makes its way from application programmes (such as spreadsheets) through a network medium (such as wire) to another application programme located on another network. The OSI reference model divides the problem of moving information between computers over a network medium into **SEVEN** smaller and more manageable problems. The seven layers are:

Layer	Function
Application	To allow access to network resources
Presentation	To translate, encrypt, and compress data
Session	To establish, manage, and terminate sessions
Transport	To provide reliable process-to-process message delivery and error recovery
Network	To move packets from source to destination; to provide internetworking
Data link	To organize bits into frames; to provide hop-to-hop delivery
Physical	To transmit bits over a medium; to provide mechanical and electrical specifications

The lower 4 layers (transport, network, data link and physical —Layers 4, 3, 2, and 1) are concerned with the flow of data from end to end through the network. The upper four layers of the OSI model (application, presentation and session—Layers 7, 6 and 5) are orientated more toward services to the applications. Data is Encapsulated with the necessary protocol information as it moves down the layers before network transit.

As with any layered architecture, overhead information is added to a PDU in the form of headers and trailers. Each layer provides a service to the layer above it in the protocol specification. Each layer communicates with the same layer's software or hardware on other computers.
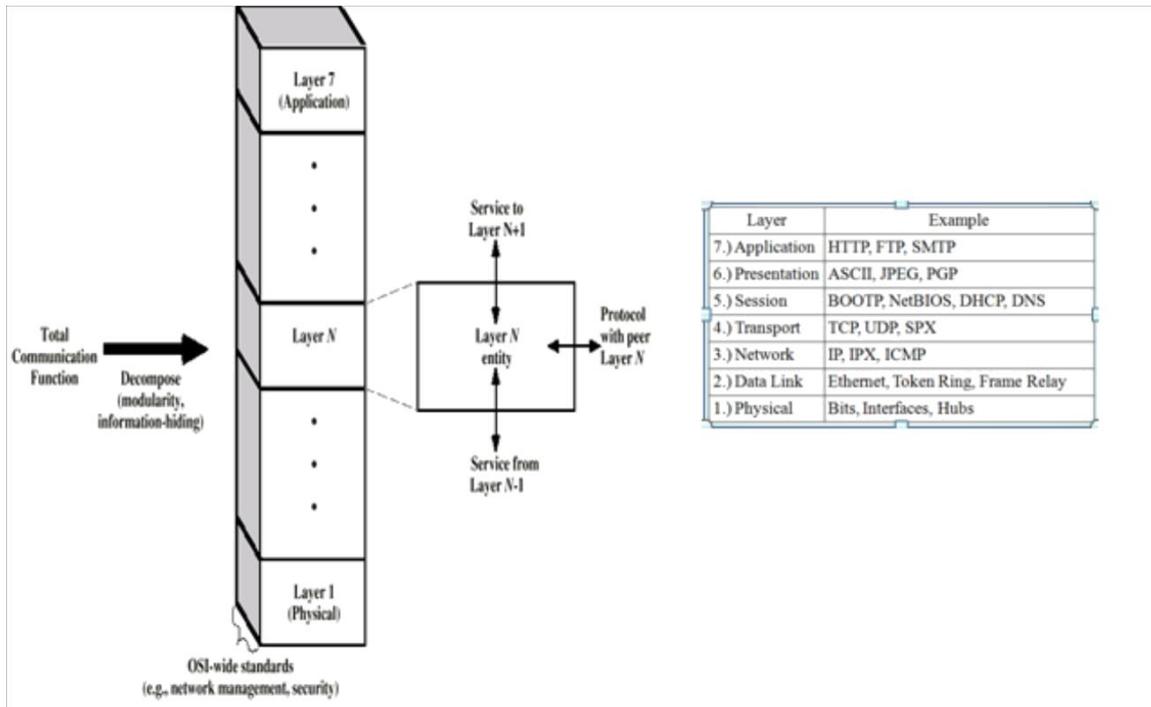
Physical Layer {the physical layer is responsible for transmitting individual bits from one node to the next}

The physical layer is the lowest layer of the OSI hierarchy and coordinates the functions required to transmit a bit stream over a physical medium. It also defines the procedures and functions that physical devices and interfaces have to perform for transmission occur. The physical layer specifies the type of transmission medium and the transmission mode (simplex, half duplex or full duplex) and the physical, electrical, functional and procedural standards for accessing data communication networks.
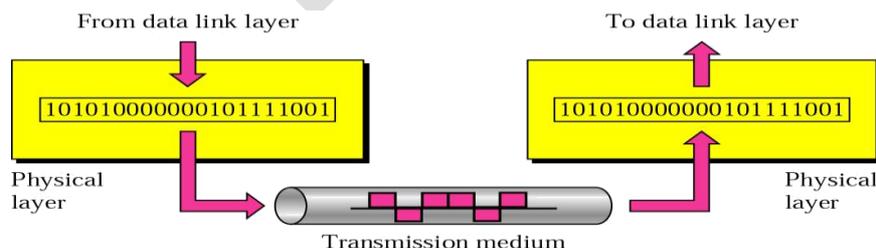

Transmission media defined by the physical layer include metallic cable, optical fiber cable or wireless radio-wave propagation. The physical layer also includes the carrier

system used to propagate the data signals between points in the network. The carrier systems are simply communication systems that carry data through a system using either metallic or optical fiber cables or wireless arrangements such as microwave, satellites and cellular radio systems.

Data-link Layer {the data link layer is responsible for transmitting frames from one node to the next}

The data link layer transforms the physical layer, a raw transmission facility, to a reliable link and is responsible for node-to-node delivery. It makes the physical layer appear error free to the upper layer (network layer).
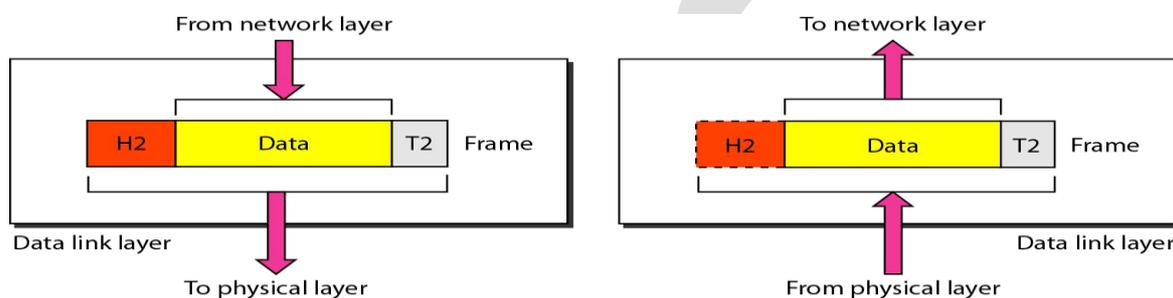
The data link layer packages data from the physical layer into groups called blocks, frames or packets. If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the physical address of the sender (source address) and/or receiver (destination address) of the frame. The data-link layer provides flow-control, access-control, and error-control.

Network Layer {is responsible for the delivery of individual packets from the source host to the destination host}

The network layer provides details that enable data to be routed between devices in an environment using multiple networks, subnetworks or both. This is responsible for addressing messages and data so they are sent to the correct destination, and for translating logical addresses and names (like a machine name FLAME) into physical addresses. This layer is also responsible for finding a path through the network to the destination computer.
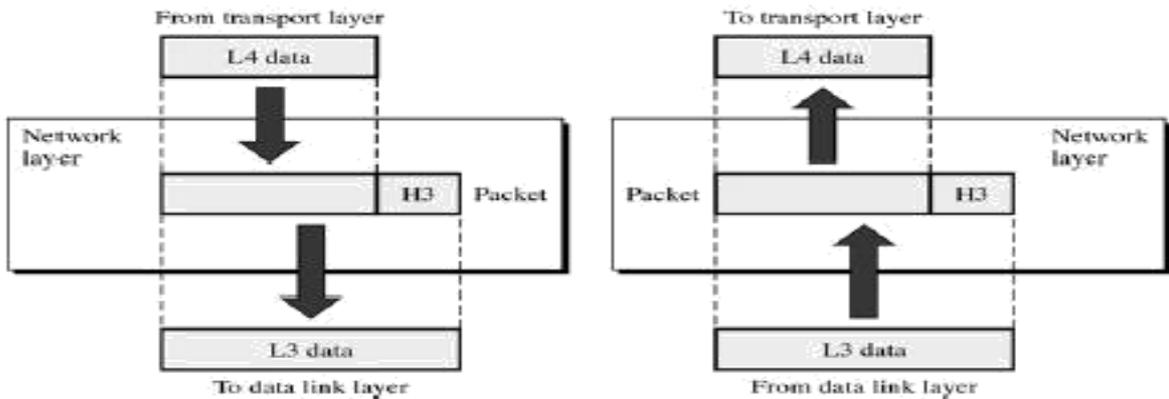
The network layer provides the upper layers of the hierarchy with independence from the data transmission and switching technologies used to interconnect systems. Networking components that operate at the network layer include routers and their software.

**Transport Layer** {is responsible for delivery of a message from one process to another}

The transport layer controls and ensures the end-to-end integrity of the data message propagated through the network between two devices, providing the reliable, transparent transfer of data between two endpoints.
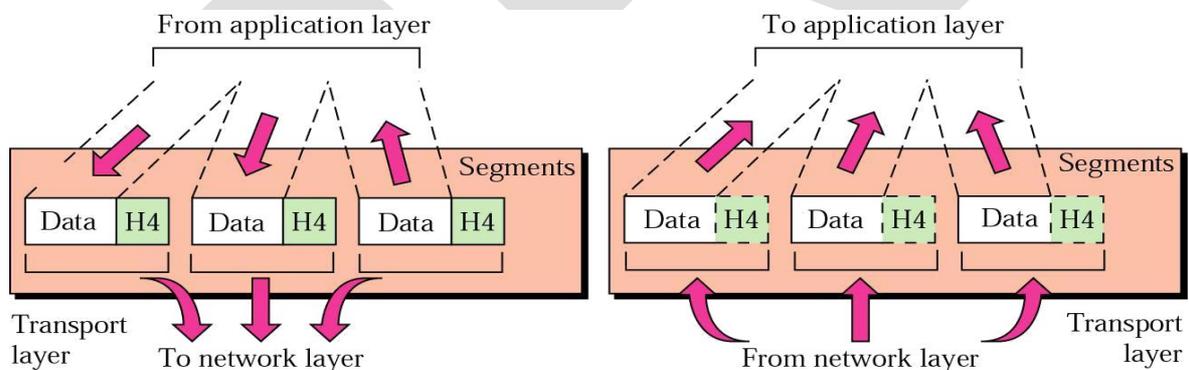
Transport layer responsibilites includes message routing, segmenting, error recovery and two types of basic services to an upper-layer protocol: connection oriented and connectionless. The transport layer is the highest layer in the OSI hierarchy in terms of communicatons and may provide data tracking, connection flow control, sequencing of data, error checking, and application addressing and identification.

**Session Layer** {responsible for dialog control and synchronization}

Session layer, some times called the dialog controller provides mechanism for controlling the dialogue between the two end systems. It defines how to start, control and end conversations (called sessions) between applications.
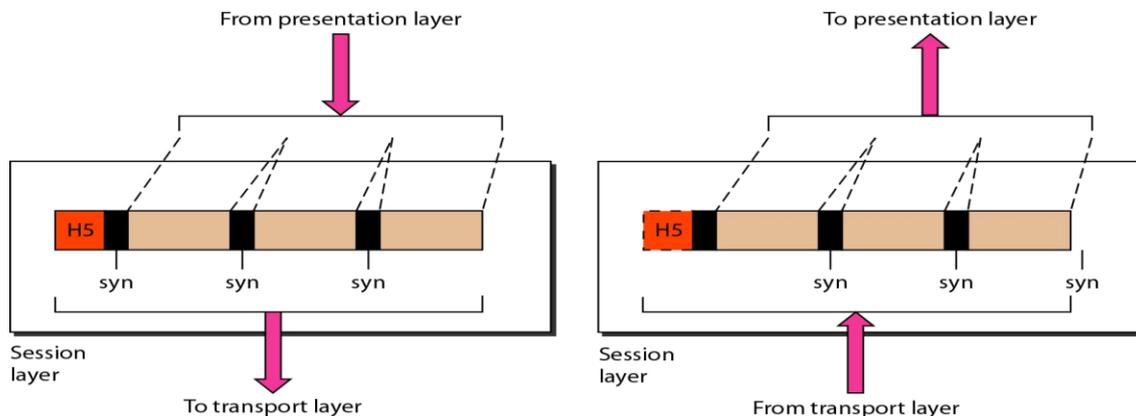
Session layer protocols provide the logical connection entities at the application layer. These applications include file transfer protocols and sending email. Session responsibilities include network log-on and log-off procedures and user authentication. Session layer characteristics include virtual connections between applications, entities, synchronization of data flow for recovery purposes, creation of dialogue units and activity units, connection parameter negotiation, and partitioning services into functional groups.

Presentation Layer {**responsible for translation, compression, and encryption**}

The presentation layer provides independence to the application processes by addressing any code or syntax conversion necessary to present the data to the network in a common communications format. It specifies how end-user applications should format the data.
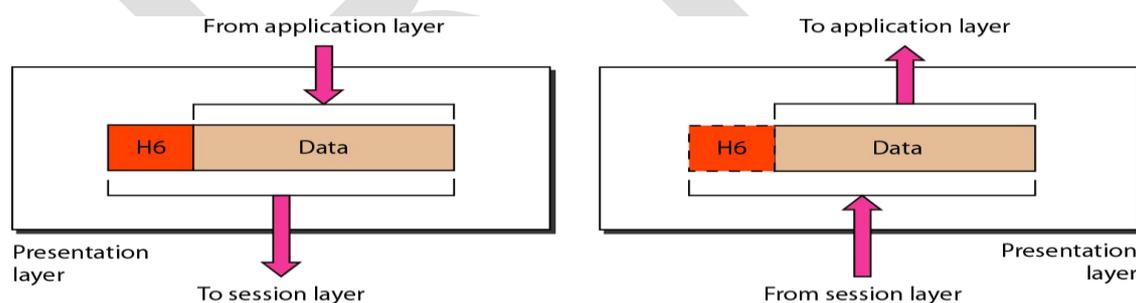
The presentation layer translated between different data formats and protocols. Presentation functions include data file formatting, encoding, encryption and decryption of data messages, dialogue procedures, data compression algorithms, synchronization, interruption, and termination.

Application Layer {**responsible for providing services to the user**}

The application layer is the highest layer in the hierarchy and is analogous to the

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general manager of the network by providing access to the OSI environment. The applications layer provides distributed information services and controls the sequence of activities within and application and also the sequence of events between the computer application and the user of another application.
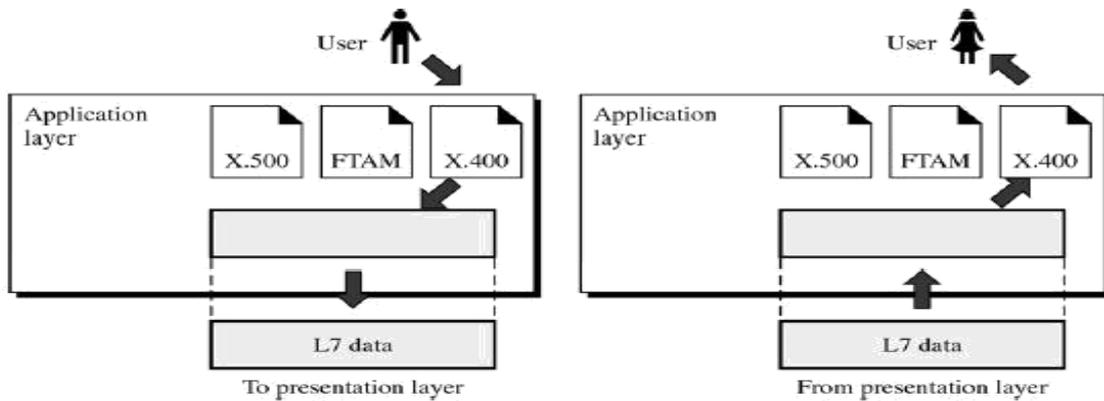

The application layer communicates directly with the user's application program. User application processes require application layer service elements to access the networking environment. The service elements are of two types: CASEs (common application service elements) satisfying particular needs of application processes like association control, concurrence and recovery. The second type is SASE (specific application service elements) which include TCP/IP stack, FTP, SNMP, Telnet and SMTP.

Data Communication Circuits

The underlying purpose of a digital communications circuit is to provide a transmission path between locations and to transfer digital information from one station (node, where computers or other digital equipment are located) to another using electronic circuits. Data communications circuits utilize electronic communications equipment and facilities to interconnect digital computer equipment. Communication facilities are physical means of interconnecting stations and are provided to data communications users through public telephone networks (PTN), public data networks (PDN), and a multitude of private
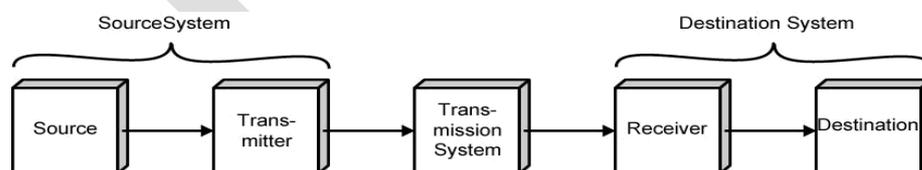
(a) General block diagram
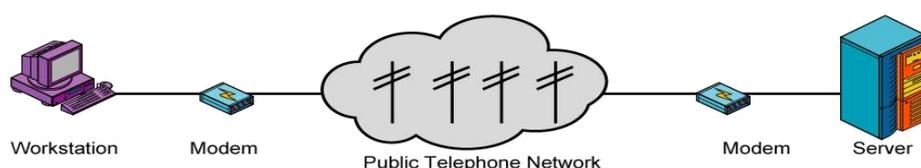
(b) Example

data communications systems.

The following figure shows a simple two-station data communications circuit. The main components are:

Source: - This device generates the data to be transmitted; examples are mainframe computer, personal computer, workstation etc. The source equipment provides a means for humans to enter data into system.

Transmitter: - A transmitter transforms and encodes the information in such a way as to produce electromagnetic signals that can be transmitted across some sort of transmission system. For example, a modem takes a digital bit stream from an attached device such as a personal computer and transforms that bit stream into an analog signal that can be handled by the telephone network.

Transmission medium: - The transmission medium carries the encoded signals from the transmitter to the receiver. Different types of transmission media include free-space radio transmission (i.e. all forms of wireless transmission) and physical facilities such as metallic and optical fiber cables.

Receiver: - The receiver accepts the signal from the transmission medium and converts it into a form that can be handled by the destination device. For example, a modem will

accept an analog signal coming from a network or transmission line and convert it into a digital bit stream.

Destination: - Takes the incoming data from the receiver and can be any kind of digital equipment like the source.

Serial and Parallel Data Transmission

There are two methods of transmitting digital data namely **parallel and serial** transmissions. In parallel data transmission, all bits of the binary data are transmitted simultaneously. For example, to transmit an 8-bit binary number in parallel from one unit to another, eight transmission lines are required. Each bit requires its own separate data path. All bits of a word are transmitted at the same time. This method of transmission can move a significant amount of data in a given period of time. Its disadvantage is the large number of interconnecting cables between the two units. For large binary words, cabling becomes complex and expensive. This is particularly true if the distance between the two units is great. Long multiwire cables are not only expensive, but also require special interfacing to minimize noise and distortion problems. Serial data transmission is the process of transmitting binary words a bit at a time. Since the bits time-share the transmission medium, only one interconnecting lead is required.
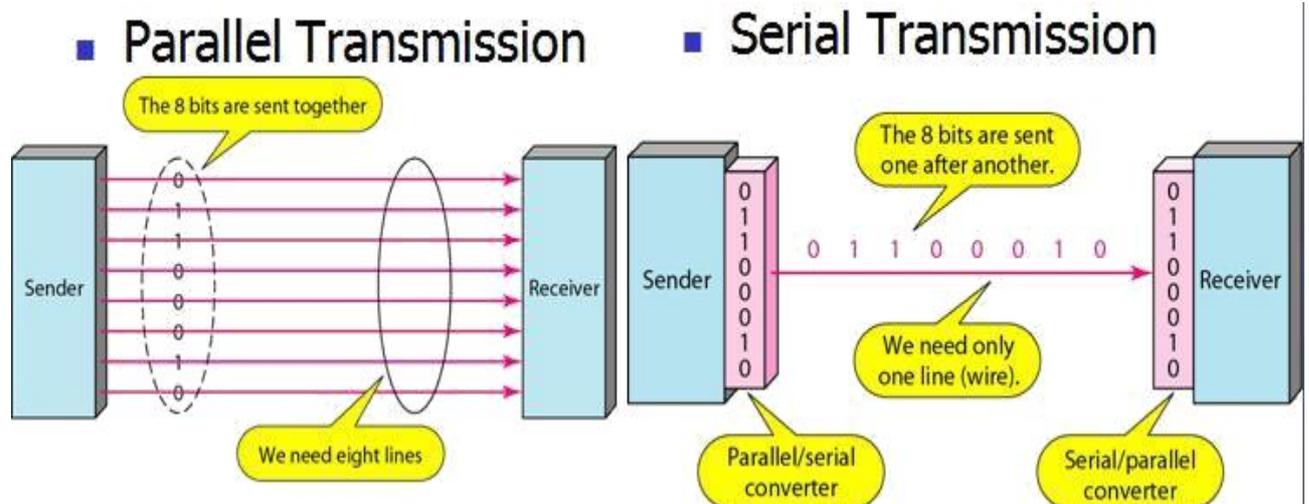
While serial data transmission is much simpler and less expensive because of the use of a single interconnecting line, it is a very slow method of data transmission. Serial data transmission is useful in systems where high speed is not a requirement. Parallel communication is used for short-distance data communications and within a computer, and serial transmission is used for long-distance data communications.

Data Communication Circuit Arrangements

A data communications circuit can be described in terms of circuit configuration and transmission mode.

Circuit Configurations

Data communications networks can be generally categorized as either two point or multipoint. A two-point configuration involves only two locations or stations, whereas a multipoint configuration involves three or more stations.
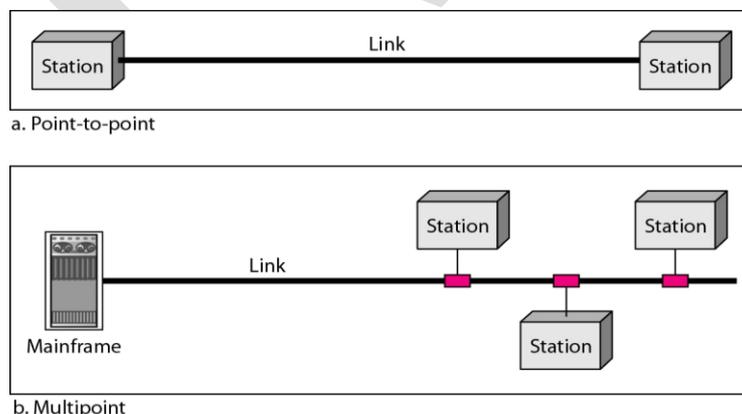
A two-point circuit involves the transfer of digital information between a mainframe

computer and a personal computer, two mainframe computers or two data communications networks. A multi-point network is generally used to interconnect a single mainframe computer (host) to many personal computers or to interconnect many personal computers and capacity of the channel is either Spatially shared: Devices can use the link simultaneously or Timeshare: Users take turns

Transmission Modes
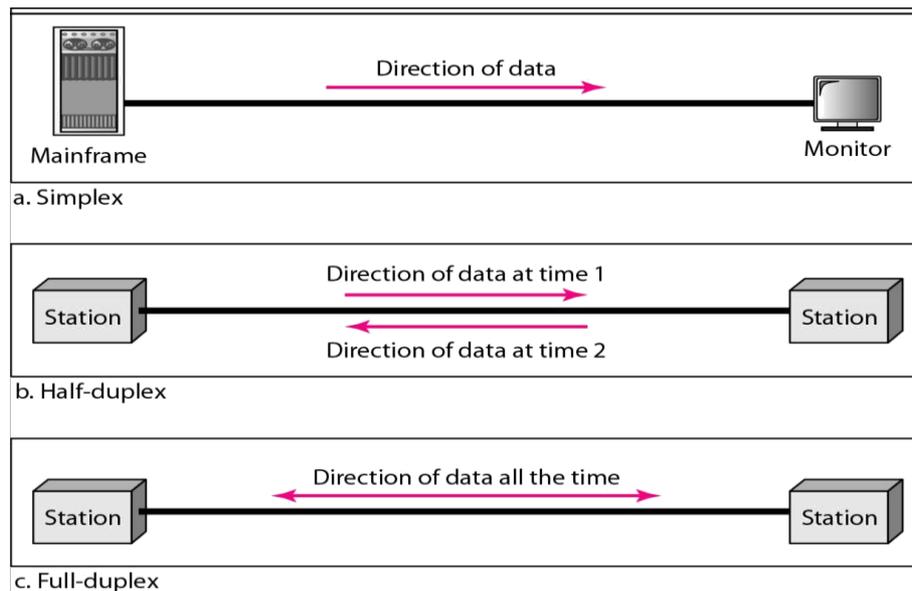
There are four modes of transmission for data communications circuits:

In **simplex mode(SX),** the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit; the other can only receive. Commercial radio broadcasting is an example. Simplex lines are also called receive-only, transmit-only or one-way-only lines.

In **half-duplex(HDX)** mode, each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa. The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction. Citizens band (CB) radio is an example where push to talk (PTT) is to be pressed or depressed while sending and transmitting.

In **full-duplex mode(FDX)** (called duplex), both stations can transmit and receive simultaneously. One common example of full-duplex communication is the telephone network. The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel must be divided between the two directions.

In **full/full duplex (F/FDX)** mode, transmission is possible in both directions at the same time but not between the same two stations (i.e. station 1 transmitting to station 2, while receiving from station 3). F/FDX is possible only on multipoint circuits. Postal system can be given as a person can be sending a letter to one address and receive a letter from another address at the same time.

21EC213 – ANALOG & DIGITAL COMMMUNICATION

Data Communications Networks

Any group of computers connected together can be called a data communications network, and the process of sharing resources between computers over a data communications network is called networking. The most important considerations of a data communications network are performance, transmission rate, reliability and security.

Network Components, Functions, and Features

The major components of a network are end stations, applications and a network that will support traffic between the end stations. Computer networks all share common devices, functions, and features, including servers, clients, transmission media, shared data, shared printers and other peripherals, hardware and software resources, network interface card (NIC), local operating system (LOS) and the network operating system (NOS).

Servers: Servers are computers that hold shared files, programs and the network operating system. Servers provide access to network resources to all the users of the network and different kinds of servers are present. Examples include file servers, print servers, mail servers, communication servers etc.

Clients: Clients are computers that access and use the network and shared network resources. Client computers are basically the customers (users) of the network, as they request and receive service from the servers.

Shared Data: Shared data are data that file servers provide to clients, such as data files, printer access programs, and e-mail.

Shared Printers and other peripherals: these are hardware resources provided to the users of the network by servers. Resources provided include data files, printers, software, or any other items used by the clients on the network.

Network interface card: Every computer in the network has a special expansion card called network interface card (NIS), which prepares and sends data, receives data, and controls data flow between the computer and the network. While transmitting, NIC passes frames of data on to the physical layer and on the receiver side, the NIC processes bits received from the physical layer and processes the message based on its contents.

Local operating system: A local operating system allows personal computers to access files, print to a local printer, and have and use one or more disk and CD drives that are located on the computer. Examples are MS-DOS, PC-DOS, UNIX, Macintosh, OS/2, Windows 95, 98, XP and Linux.

Network operating system: the NOS is a program that runs on computers and servers that allows the computers to communicate over a network. The NOS provides services to

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clients such as log-in features, password authentication, printer access, network administration functions and data file sharing.

Network Models

Computer networks can be represented with two basic network models: peer-to-peer client/server and dedicated client/server. The client/server method specifies the way in which two computers can communicate with software over a network.

Peer-to-peer client/server network: Here, all the computers share their resources, such as hard drives, printers and so on with all the other computers on the network. Individual resources like disk drives, CD-ROM drives, and even printers are transformed into shared, collective resources that are accessible from every PC. Unlike client-server networks, where network information is stored on a centralized file server PC and made available to tens, hundreds, or thousands client PCs, the information stored across peer-to-peer networks is uniquely decentralized. Because peer-to-peer PCs have their own hard disk drives that are accessible by all computers, each PC acts as both a client (information requestor) and a server (information provider). The peer-to-peer network is an appropriate choice when there are fewer than 10 users on the network, security is not an issue and all the users are located in the same general area.

The advantages of peer-to-peer over client-server NOSs include:
- No need for a network administrator
- Network is fast/inexpensive to setup & maintain
- Each PC can make backup copies of its data to other PCs for security.
- Easiest type of network to build, peer-to-peer is perfect for both home and office use.

Dedicated client/server network: Here, one computer is designated as server and the rest of the computers are clients. Dedicated Server Architecture can improve the efficiency of client server systems by using one server for each application that exists within an organization. The designated servers store all the networks shared files and applications programs and function only as servers and are not used as a client or workstation. Client computers can access the servers and have shared files transferred to them over the transmission medium. In some client/server networks, client computers submit jobs to one of the servers and once they process the jobs, the results are sent back to the client computer.

In general, the dedicated client/server model is preferable to the peer-to-peer client/server model for general purpose data networks.

Network Topologies

In computer networking, topology refers to the layout of connected devices, i.e. how the computers, cables, and other components within a data communications network are interconnected, both physically and logically. The physical topology describes how the network is actually laid out, and the logical topology describes how the data actually flow

through the network. Two most basic topologies are point-to-point and multipoint. A point- to-point topology usually connects two mainframe computers for high-speed digital information. A multipoint topology connects three or more stations through a single transmission medium and some examples are star, bus, ring, mesh and hybrid.

Star topology: A star topology is designed with each node (file server, workstations, and peripherals) connected directly to a central network hub, switch, or concentrator. Data on a star network passes through the hub, switch, or concentrator before continuing to its destination. The hub, switch, or concentrator manages and controls all functions of the network. It also acts as a repeater for the data flow.
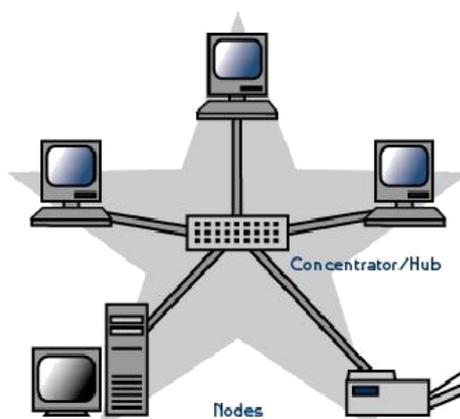
Advantages	Disadvantages
Easily expanded without disruption to the network	Requires more cable
Cable failure affects only a single user	A central connecting device allows for a single point of failure
Easy to troubleshoot and isolate problems	More difficult to implement