Better-to-Know

Assalam-o-alikum!
This is Dr. Eerie, a student at VU, holds many certifications in the domain of cybersecurity & networking, intermediate in Python3, founder of “**De Technocrats**” etc. This’s what I heard about myself from people around me ;)

Anyways I compiled this PDF because I noticed that there are no officially provided notes from VU for CS601. But there’s a reading section after every lecture. So, I compiled it by copying & pasting that reading section stuff including pics. I hope that this will save student time during learning as well as it’ll save them from headache during exams’ seasons which they used to get due to leaping around the whole lecture sections on VULMS in exams season to revise the course content in a short period of time.

I didn’t not only copy and paste the whole lectures to compile these notes, but also rectified some structural errors that were in reading section there.
Also added some extra point(s) under the heading of FYI (i.e. For Your Information), these points would be helpful to students and VU as well !!

NOTE:
There’s no copyright infringement intended. All the stuff that I copied & pasted here belongs to VU CS601 reading sections and VU is the institute who deserves the credit for this.

I just invested my time in compiling these notes for pubic (students) welfare. If you find these notes helpful so you can appreciate it via joining me at De Technocrats on YouTube , GitHub , Telegram (use VPN because telegram is banned in Pakistan). There you’ll find more exclusive free educational stuff related to computer sciences in PDF, video, audio, and other formats.

You can contact me (I may slow to respond or may not be able to do so):
There’s a WhatsApp number is YouTube channel description. Or Telegram contact is my personal **GitHub** profile.

Regards
Wishing you the best for your studies

Lecture # 1

Welcome to the exciting world of Data Communication. This foundational course has been carefully designed to equip undergraduate students with a broad comprehension of the essential principles and techniques that serve as the foundation for contemporary communication systems. Today's world is characterized by rapid technological advancements, and proficiency in transmitting, receiving, and exchanging data effectively and securely holds the utmost significance. This course is an entry point for students to delve into the complex system of technologies and protocols that provide uninterrupted communication on a global scale.

During this course, students will explore the fundamental principles of data communication, covering the ideas of data transfer, networking technologies, and internet protocols. Students will acquire knowledge and understanding of the essential hardware and software components that constitute the foundational infrastructure of contemporary communication networks. By learning a foundational grasp of data encoding and delving into advanced subjects like network security and wireless communication, students will develop the requisite knowledge and abilities, to effectively traverse the intricate nature of the digital communication domain.

FYI: This lecture is just an introduction to the course. Nothing special to learn here.

Lecture # 2

Communication: Sharing of Information (Local or remote).

Telecommunications: Communication at a Distance (includes telephony, telegraph, and television etc.)

Data communications: Exchange of data between two devices via some form of transmission media.

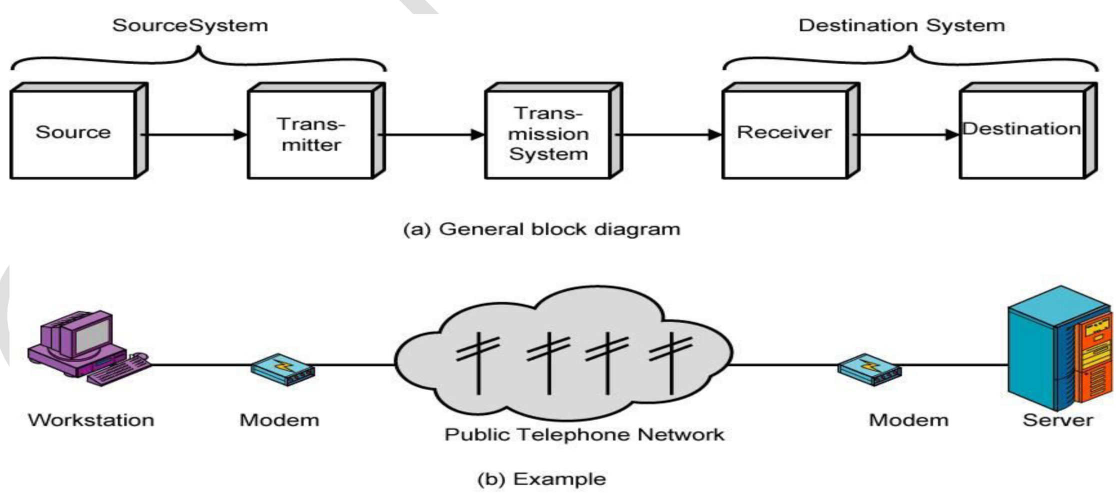


Figure 1.1 Simplified Communications Model

Lecture # 3

Effectiveness of a Data Communication System:

The effectiveness of a Data Communication System depends on the following factors:

- **Delivery:**

  When data is sent from one place to another correctly and successfully, this is called delivery.

- **Accuracy:**

  When data is sent, it must be accurate, which means that there must be no errors.

- **Timeliness:**

  For data to be considered timely, it must be sent within a reasonable amount of time.

- **Jitter:**

  The difference in packet arrival times in a network, which affects how consistently data is sent at regular intervals, is called jitter.

Lecture # 4

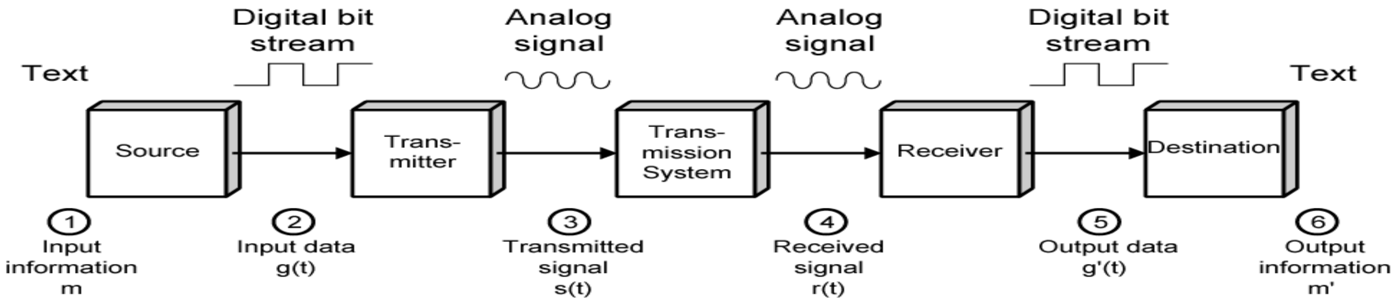


Figure 1.2 Simplified Data Communications Model

FYI: This lecture covers the same reading content as of previous lecture i.e. lecture # 3, but only includes this extra picture.

Lecture # 5

A data communications system has five components:

- Sender
- Receiver
- Transmission medium
- Protocol
- Message

Lecture # 6

Following are the different forms of Information.

- Text
- Numbers
- Images
- Audio
- Video

Data Flow between two devices in the following modes:

Simplex: In simple terms, simplex is like talking on a one-way street. While in simplex mode, data can only go from one device to another. The device that receives the data cannot send it back. Like a TV remote control, you can tell it what to do (like changing stations), but it doesn't get any information from the TV.

Half Duplex: It is like talking on a walkie-talkie. Data can be sent and received in this mode, but not at the same time. Like switching places in a chat. People listen to each other when they talk and talk to each other when they listen. Changing between sending and getting is possible, but not both at the same time.

Full Duplex: Communication that works both ways is like a normal phone call. There is time for both people to talk and listen. Like having a street that goes both ways, data can flow in both directions at the same time. This is like how we talk to each other in real life—we can both talk and listen at the same time, which makes dialogue smooth and natural.

Lecture # 7

- **Network:** Interconnection of a set of devices capable of communication

- **Host:** In computer networks, a host is any device that is connected to the network, like a computer or printer. Each host has its own unique IP address, which lets other devices on the network talk to each other and share data.

- **Connecting Device:** In networks, a connecting device is either hardware or software that lets devices talk to each other by controlling network traffic and managing data transfers easier.
  A network must be able to meet a certain number of criteria such as:

- Performance
- Reliability
- Security

Lecture # 8

Physical Network Attributes:

- **Link**: It refers to a way for two or more devices to talk to each other or a connection that lets them share info. Different types of media, like wired (like Ethernet cables and fiber-optic lines) or wireless (like Wi-Fi and Bluetooth) technologies, can be used to make links.

- **Type of Connection**
    - **Point-to-Point**

      In networking, point-to-point connections let two devices talk to each other directly over a fixed link, making it easy for them to share data.

    - **Multipoint**

      Multipoint links, on the other hand, let more than two devices share the same communication link, though they do have to compete for the available bandwidth. This lets more than two devices talk to each other over a shared network.

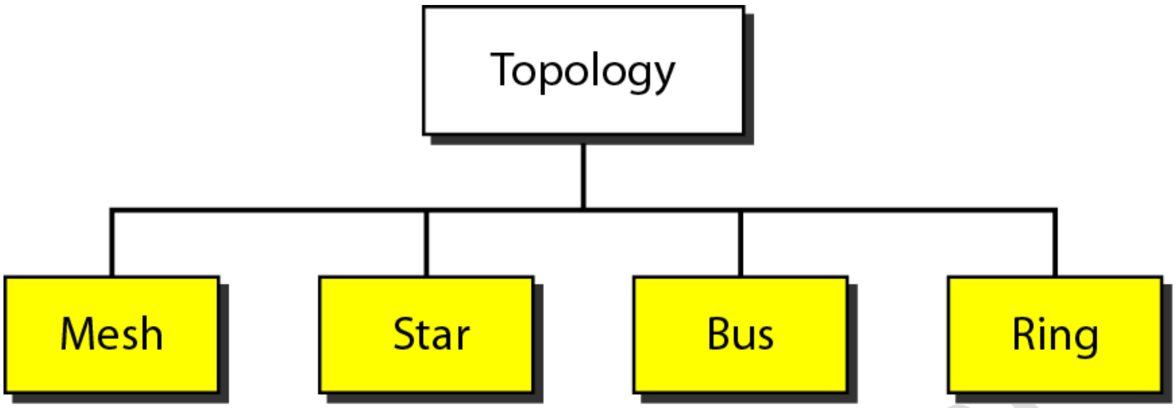
Lecture # 9

Physical Topologies

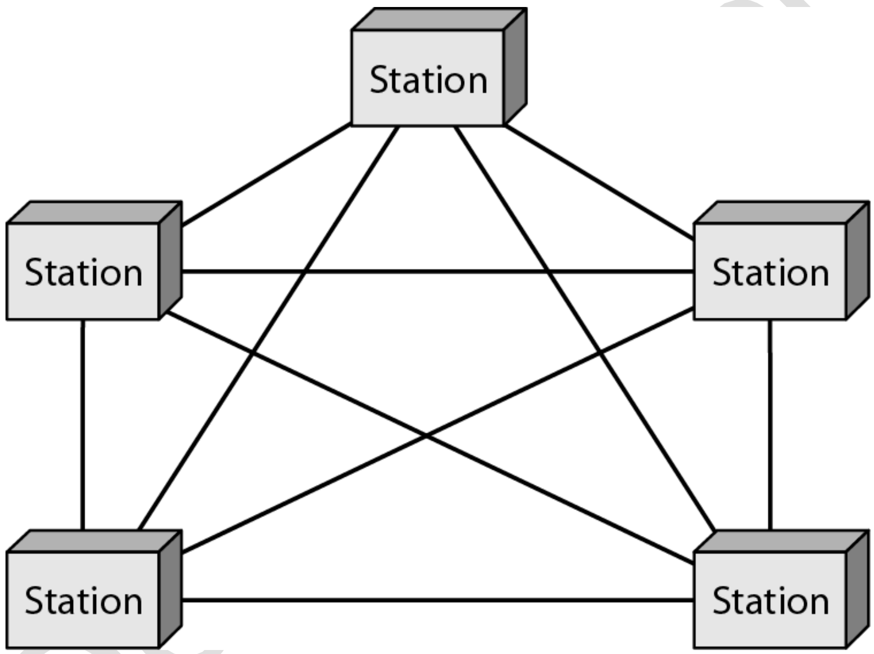
- Links + Nodes = Topology
- **Topology:** Physical Layout of Network
- **Physical Topologies:**
    - **Mesh:** A network setup where each device is connected to every other device through individual point-to-point links.
    - **Star:** In a star network topology, every device is connected through a dedicated point-to-point link solely to a central controller, often referred to as a hub.
    - **Bus:** A single extended cable serves as a backbone to connect all devices within a network in a multipoint configuration.
    - **Ring:** Every device maintains an exclusive point-to-point link with only the two neighboring devices on either side of it.

Physical Topologies

Mesh Topology:



A network setup where each device is connected to every other device through individual point-to-point links.

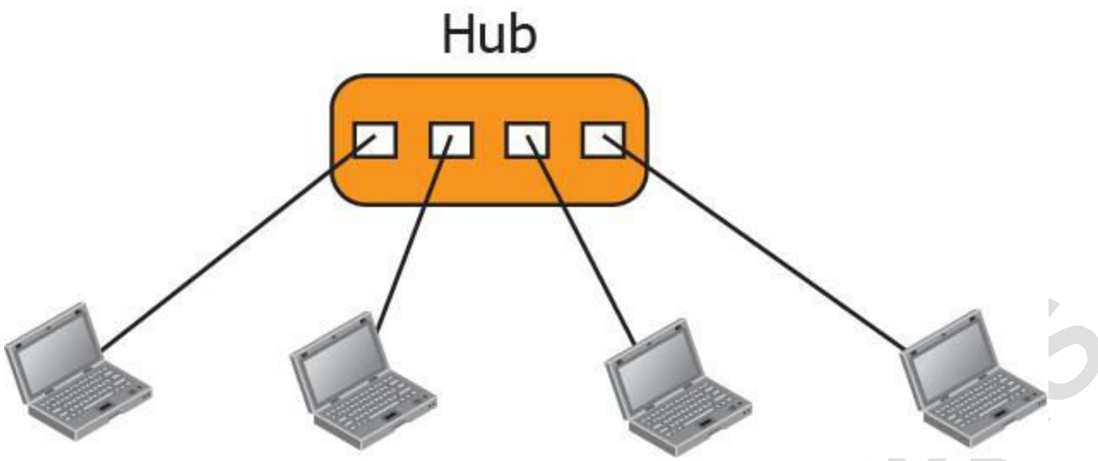


<u>Lecture # 10</u>

<u>**FYI:**</u> It contains the above pics and the below one:

Star Topology:

In a star network topology, every device is connected through a dedicated point-to-point link solely to a central controller, often referred to as a hub.

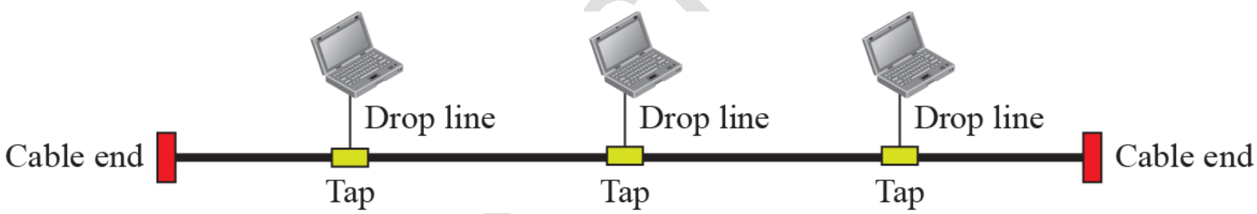


Lecture # 11

Physical Topologies

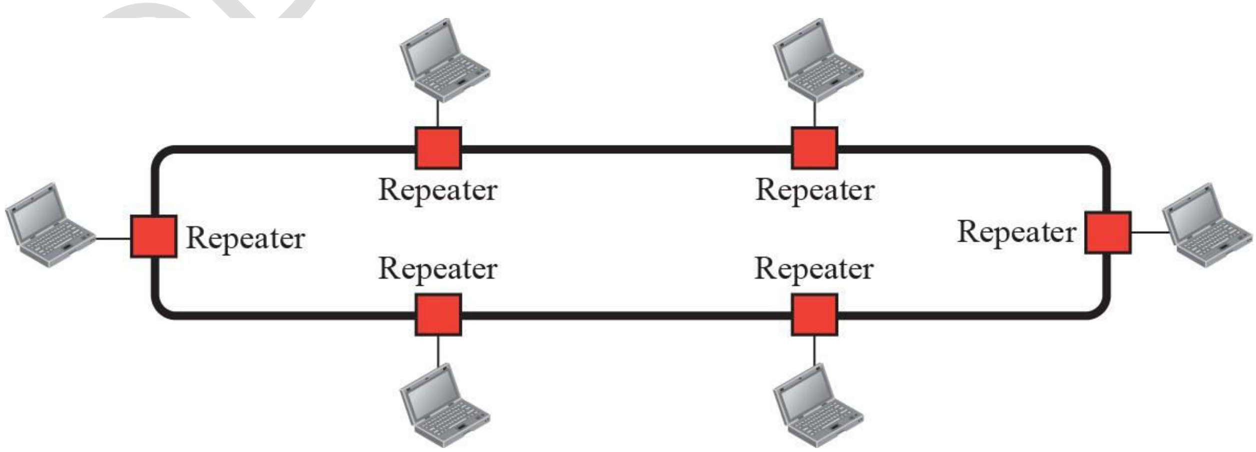
Bus Topology:

A single extended cable serves as a backbone to connect all devices within a network in a multipoint configuration.



Ring Topology:

Every device maintains an exclusive point-to-point link with only the two neighboring devices on either side of it.



Lecture # 12

Network Types

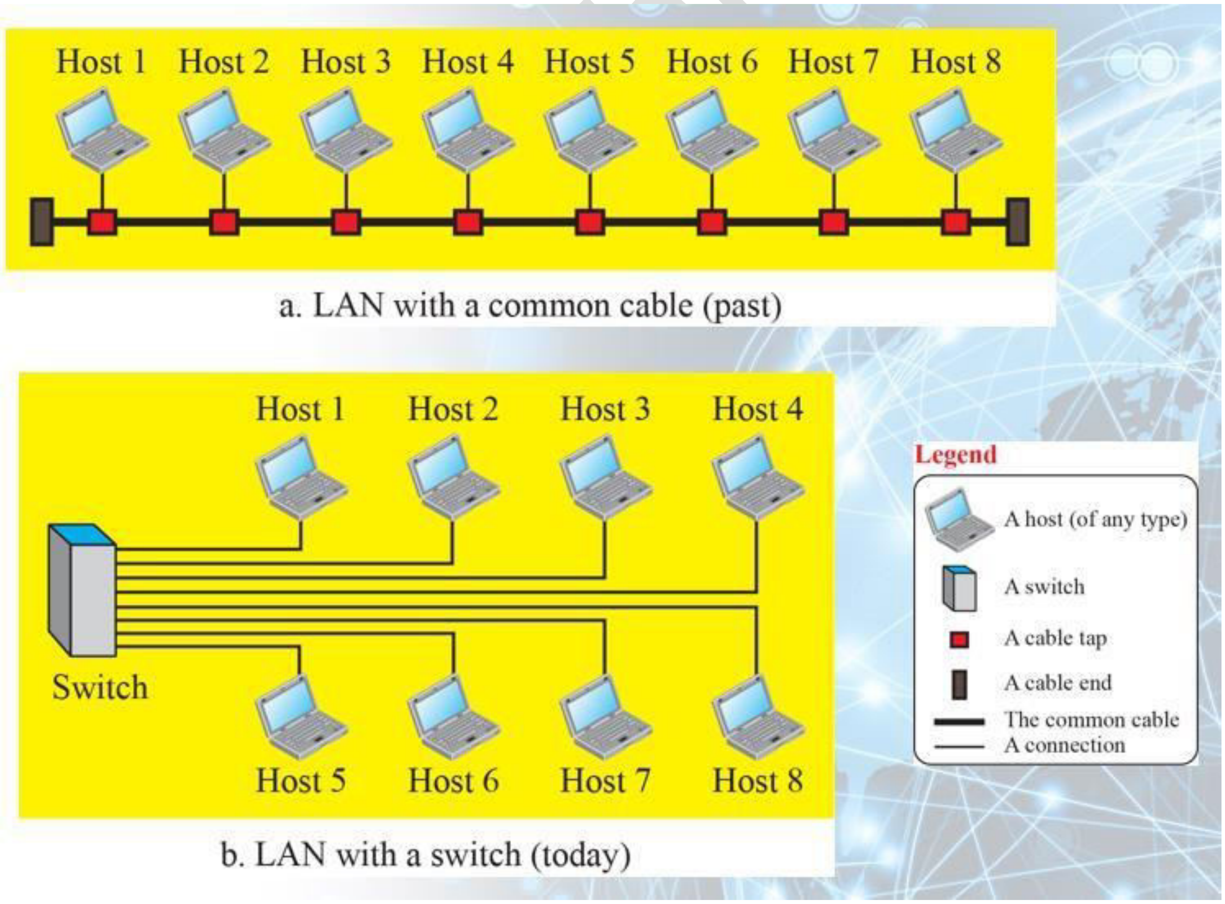
Network classification:

Networks can be classified in various ways based on different criteria.

- Size
- Geographical Coverage
- Ownership

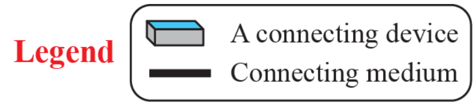
Local Area Networks

- Usually Privately owned.
- Connects some hosts in a single office, building, or campus.
- Can be as simple as two PCs and a printer in someone's home office
- Can extend throughout a company
- Host Address



Lecture # 13

Wide Area Network



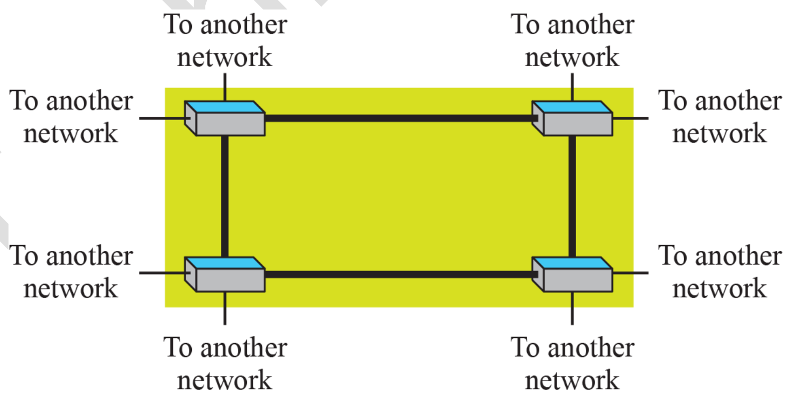
- Wider geographical span than a LAN
- Spans a town, a state, a country, or even the world
- Interconnects connecting devices such as switches, routers, or modems
- Normally created and run by communication companies
- Point-to-Point WAN
- Switched WAN
- Internetwork

Point-to-Point WANs:

A Point-to-Point Wide Area Network (WAN) is a type of network connection that establishes a direct link or communication path between two specific locations or endpoints.

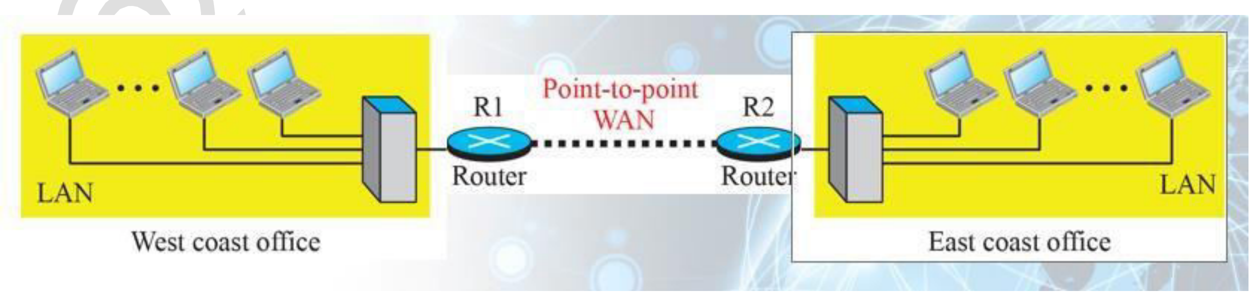
Switched WANs:

A switched WAN typically refers to a Wide Area Network (WAN) that utilizes switching technology for data transmission



Internetwork:

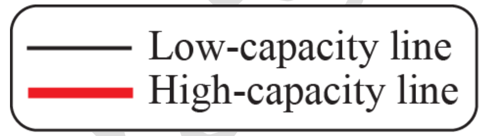
is a network that connects multiple individual networks or subnetworks together, allowing them to communicate and share information across a larger, interconnected system.



Lecture # 14

Switching

- Circuit-Switched Network: A dedicated link, referred to as a circuit, remains constantly established between the two end systems, while the switch can solely control its activation or deactivation.
- Packet- Switched Network: A packet-switched network is a type of data communication network in which digital data is divided into small packets for transmission.



Circuit Switched Network:

A circuit-switched network is a type of telecommunication network in which a dedicated communication path or circuit is established between two devices for the duration of their

conversation.

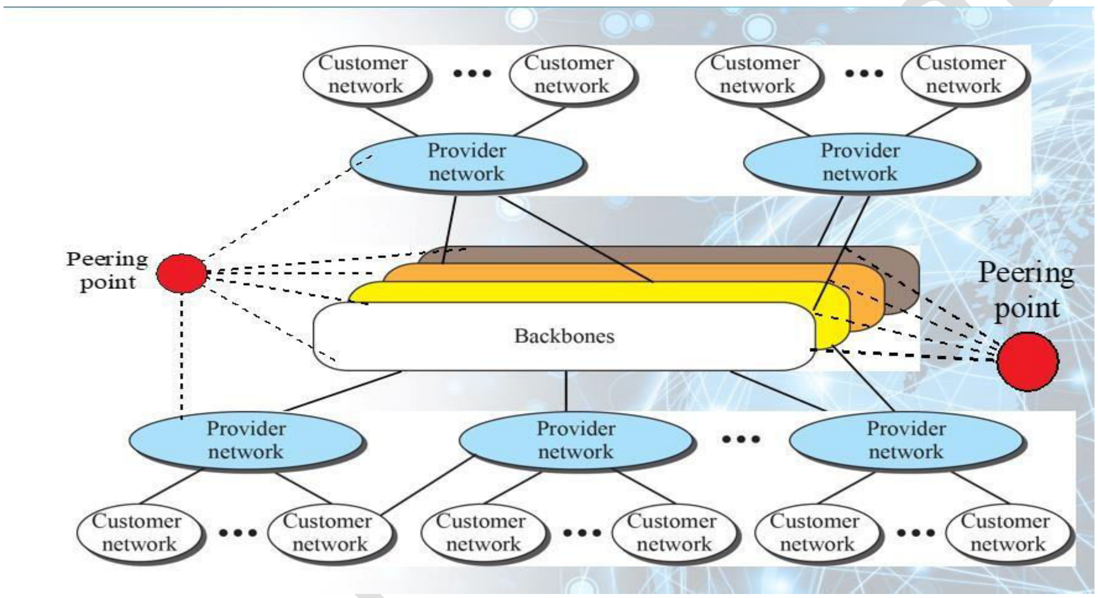
Packet Switched Network:

A packet-switched network is a type of data communication network that transmits digital information in discrete packets of data.

Lecture # 15

The Internet

- An internet (note the lowercase i) is two or more networks that can communicate with each other
- The Internet (uppercase I), and is composed of thousands of interconnected networks.
- Accessing the Internet



Lecture # 16

Internet History

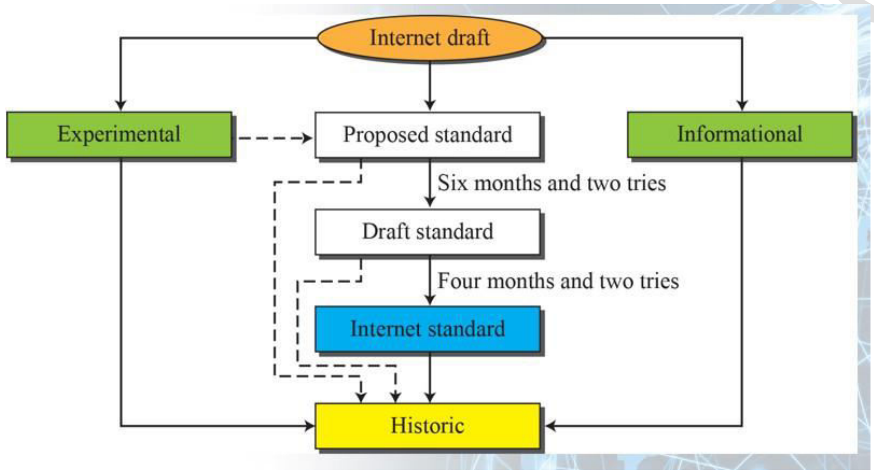
- Telegraph and Telephone networks, before 1960: Constant-rate communication only
- ARPANET: short for Advanced Research Projects Agency Network, was one of the earliest and most significant computer networks that laid the foundation for the development of the modern internet.
- MILNET: short for Military Network, was a computer network used by the United States Department of Defense (DoD) in the 1980s.
- CSNET: short for Computer Science Network, was a computer network that played a significant role in the early development of the internet.
- NSFNET: The National Science Foundation Network, commonly known as NSFNET, was a pivotal computer network that played a crucial role in the development and expansion of the internet in the United States.

Lecture # 17

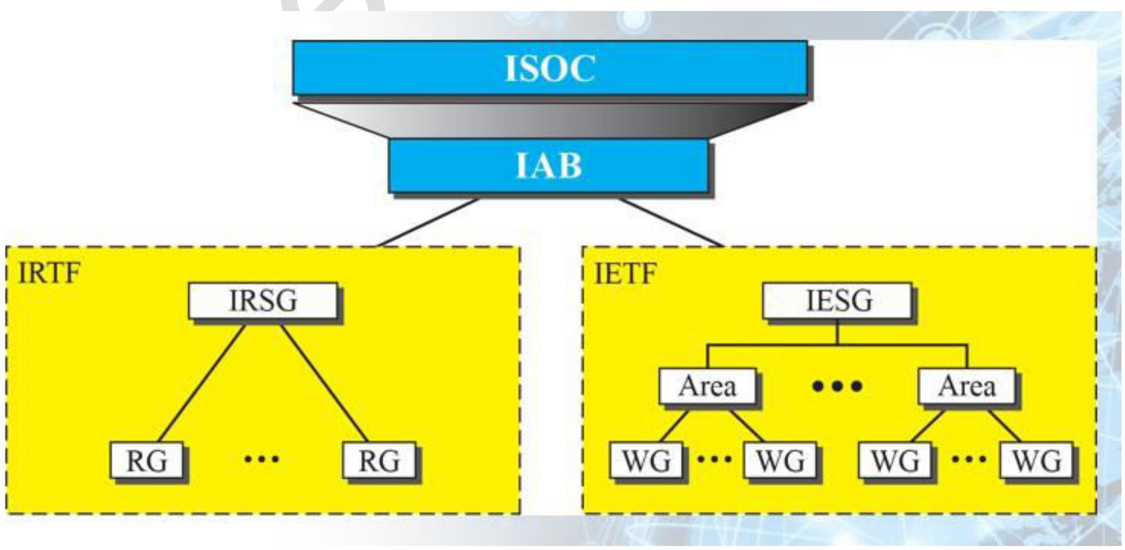
Internet Standards and Administration

- Internet draft: An Internet draft is a provisional document, still under development, and lacking official status, typically having a lifespan of approximately six months.
- Request for Comments (RFC): Based on guidance from internet authorities, a draft could be released as a Request for Comment (RFC).
    - Proposed Standard
    - Draft Standard
    - Internet Standard
    - Historic
    - Experimental
    - Informational

Internet Standards



Internet Administration



Lecture # 18

Protocol Layering - Introduction

Protocol

Rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively.

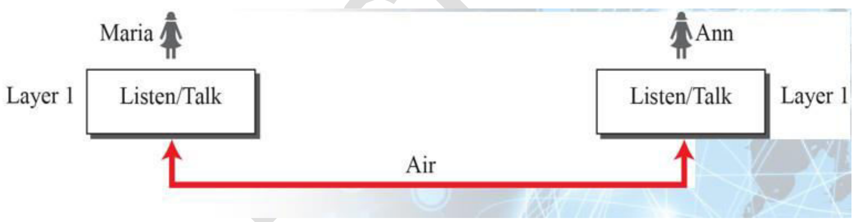
Protocol Layering

Simple Communication: Exchange of information between two or more individuals or entities using straightforward and easily understood language or methods. We may need only one simple protocol.

Complex Communication: The task is divided between different layers, in that case we need a protocol at each layer, or protocol layering.

Protocol Layering - Example Scenario 1

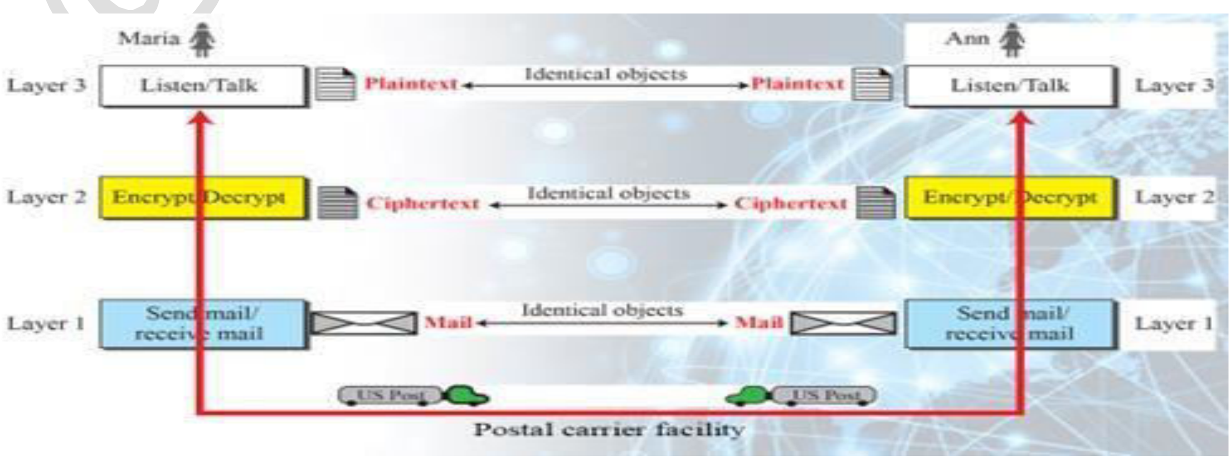
In the picture below, communication between Maria and Ann takes place in one layer, face to face, in the same language.



Lecture # 19

Protocol Layering - Example Scenario 2

In the below scenario, Maria and Ann are using encryption/decryption technique so their ideas do not disclose to other people



Lecture # 20

Protocol Layering - Advantages and Disadvantages

Advantages

- Modularity
- Separation of Service & Implementation
- Reduced Complexity & Cost

Disadvantages

- **Complexity:** Layering introduces multiple protocols, each responsible for a specific function or task.
- **Overhead:** Each layer adds overhead in the form of headers, trailers, and additional processing.

Lecture # 21

Protocol Layering - Principles

- Bidirectional Communication → Each Layer performs two opposite tasks in each direction.
- Two objects under each layer at both sites should be identical.

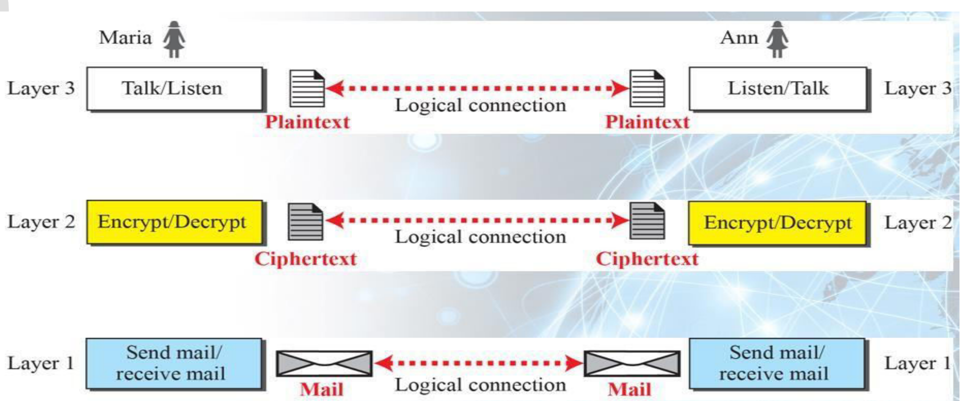
Protocol Layering - Logical Connections

Logical Connections

- Imaginary connection between each layer

Protocol Layering

Now Maria and Ann can imagine that there is a logical connection at each layer through which they can send their data created from that layer.

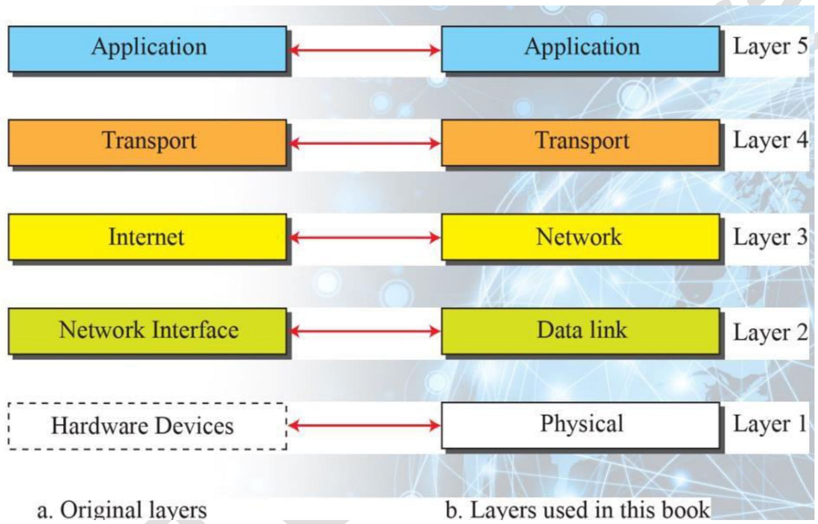


Lecture # 22

TCP/IP Protocol Suite

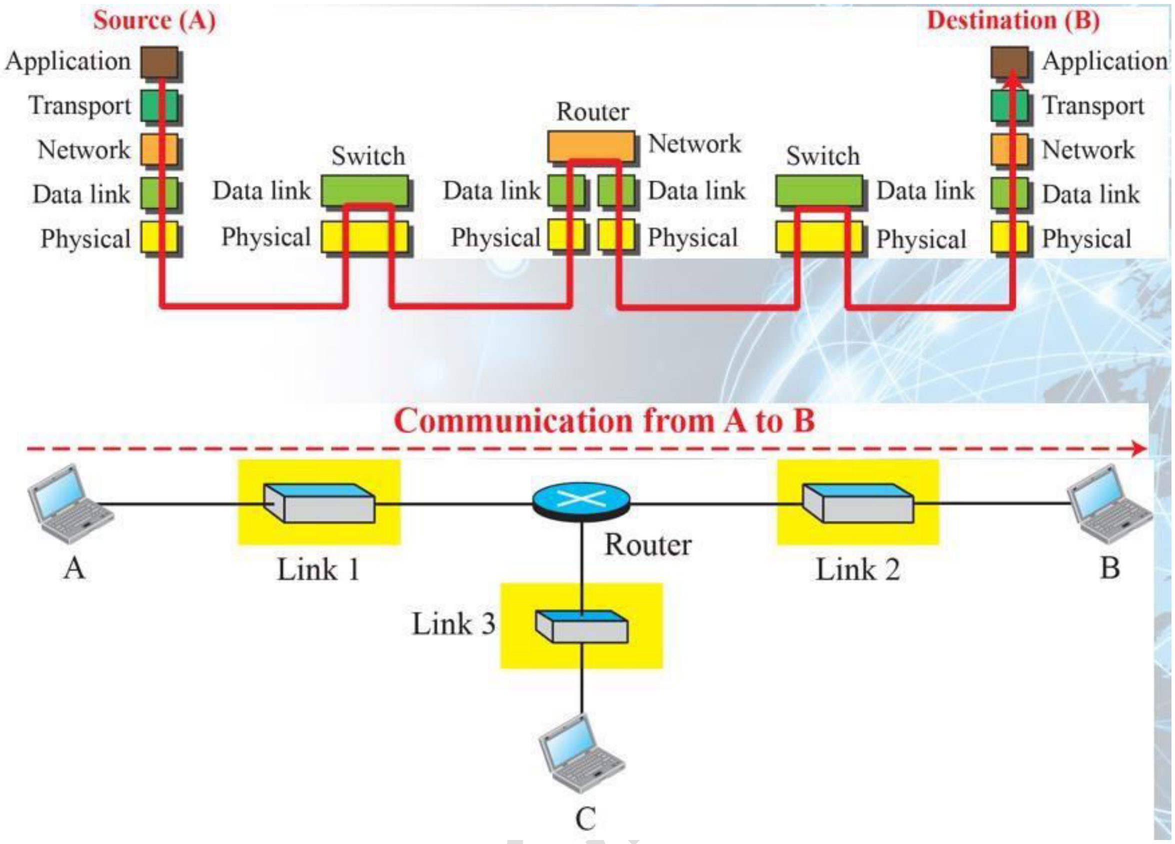
TCP/IP Protocol Suite

- Protocol suite used on the Internet today.
- Each Layer provides specific functionality.
- Transmission Control Protocol/Internet Protocol (TCP/IP) is a hierarchical protocol suite used for data communication in computer networks.
- Presented in 1973 and chosen to be the official protocol of Internet in 1983



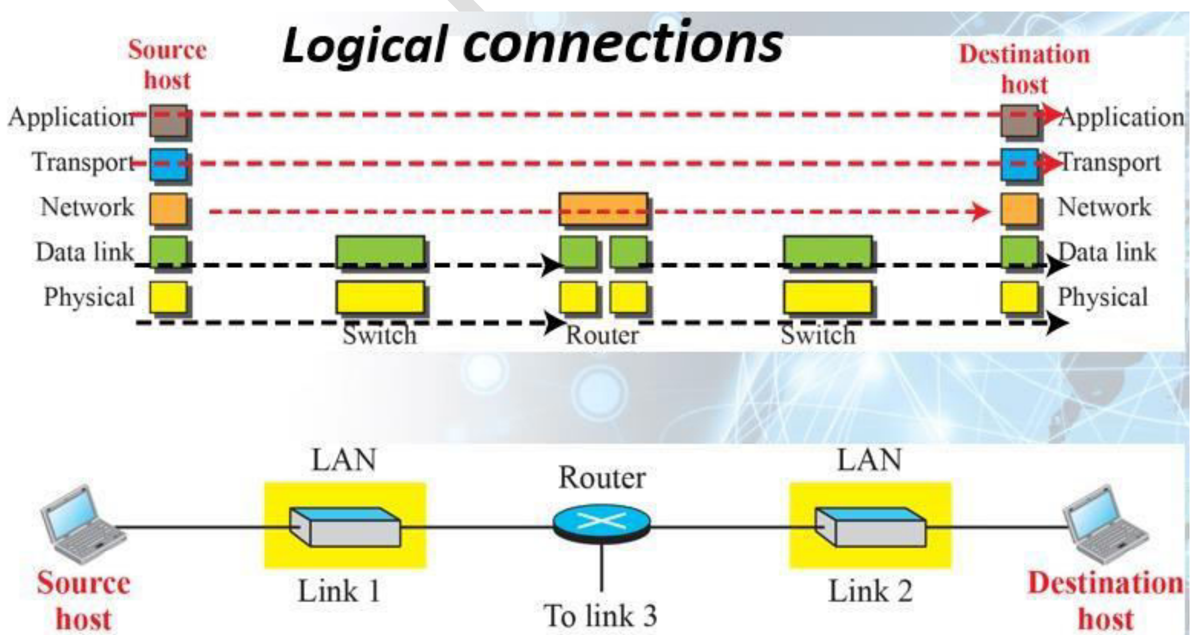
TCP/IP Protocol Suite - Layered Architecture

- In the below figure, let us assume that computer A communicates with computer B.
- Both hosts participate in all five layers of the communication process.
- The router is involved in only three layers, there is no transport or application layer in a router as the router is used only for routing.
- A link-layer switch is involved only in two layers, datalink and physical.



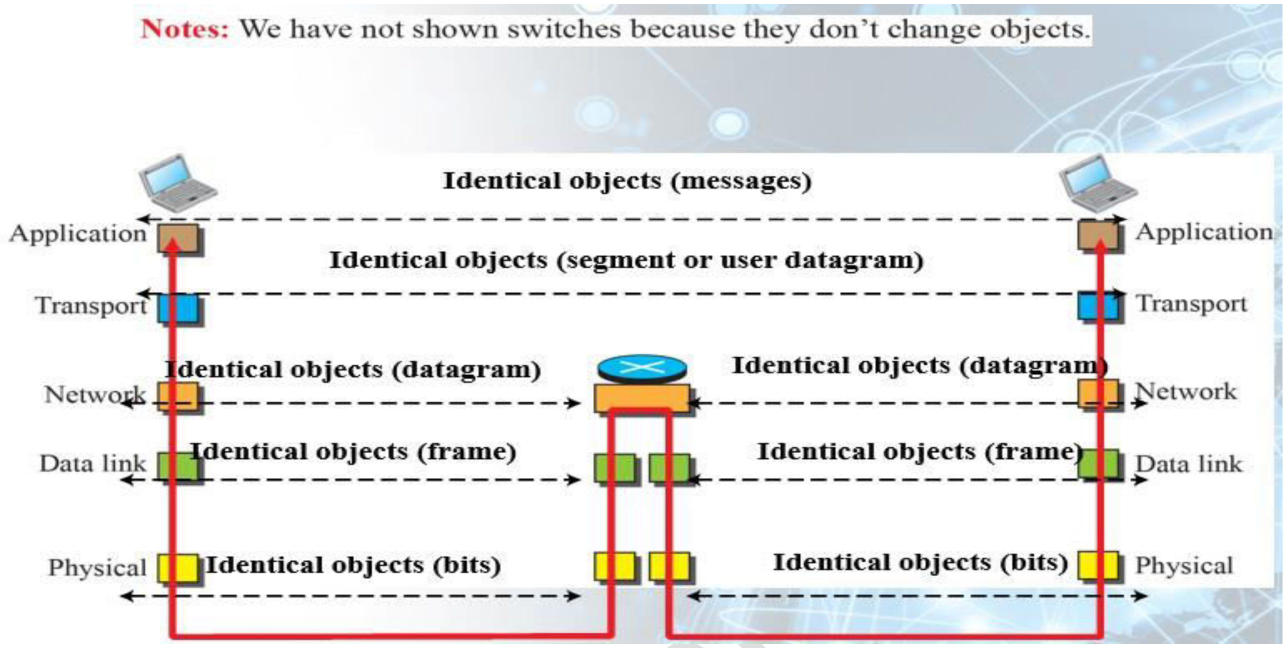
Lecture # 23

The following diagram shows the **Logical Connections between TCP/IP Layers.**



Now the other figure shows the identical objects of each layer:

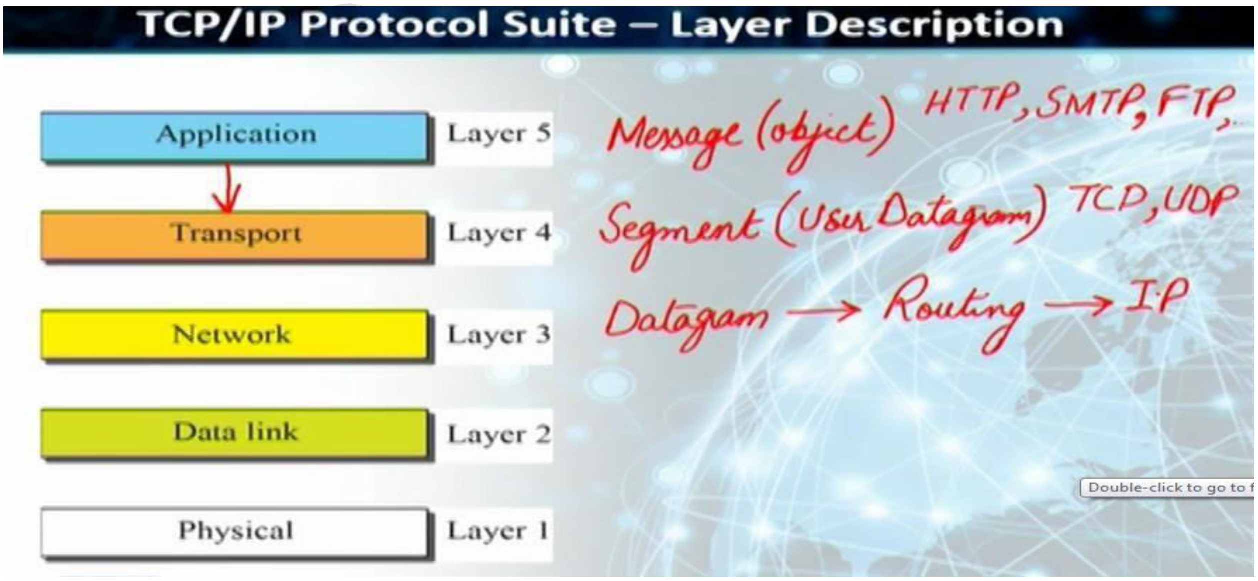
For example,
Application layers exchanges the identical objects as "messages" and Physical layers as "bits".



Lecture # 24

The following figure shows the protocols and the objects used in Layer 3-5.

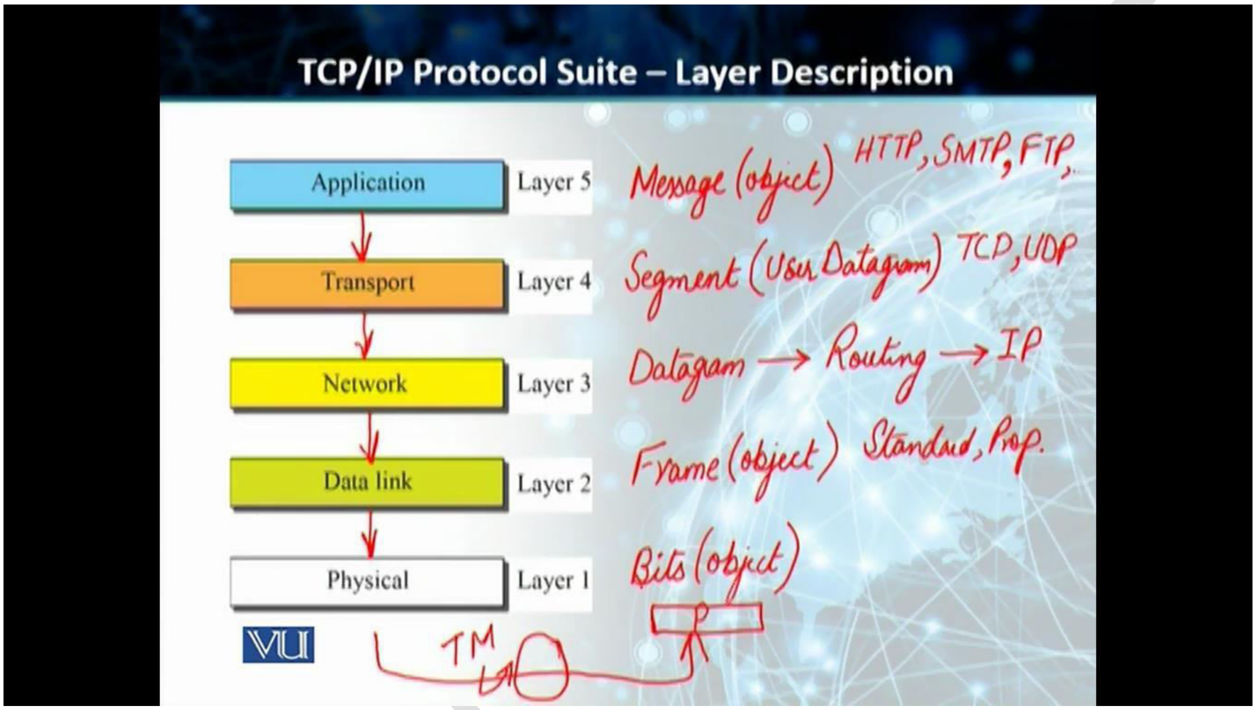
- HTTP, SMTP and FTP protocols are used in Application layer
- TCP and UDP in Transport layer
- IP protocol in Network layer



Lecture # 25

The following figure shows the protocols and the objects used in **Layer 3-5.**

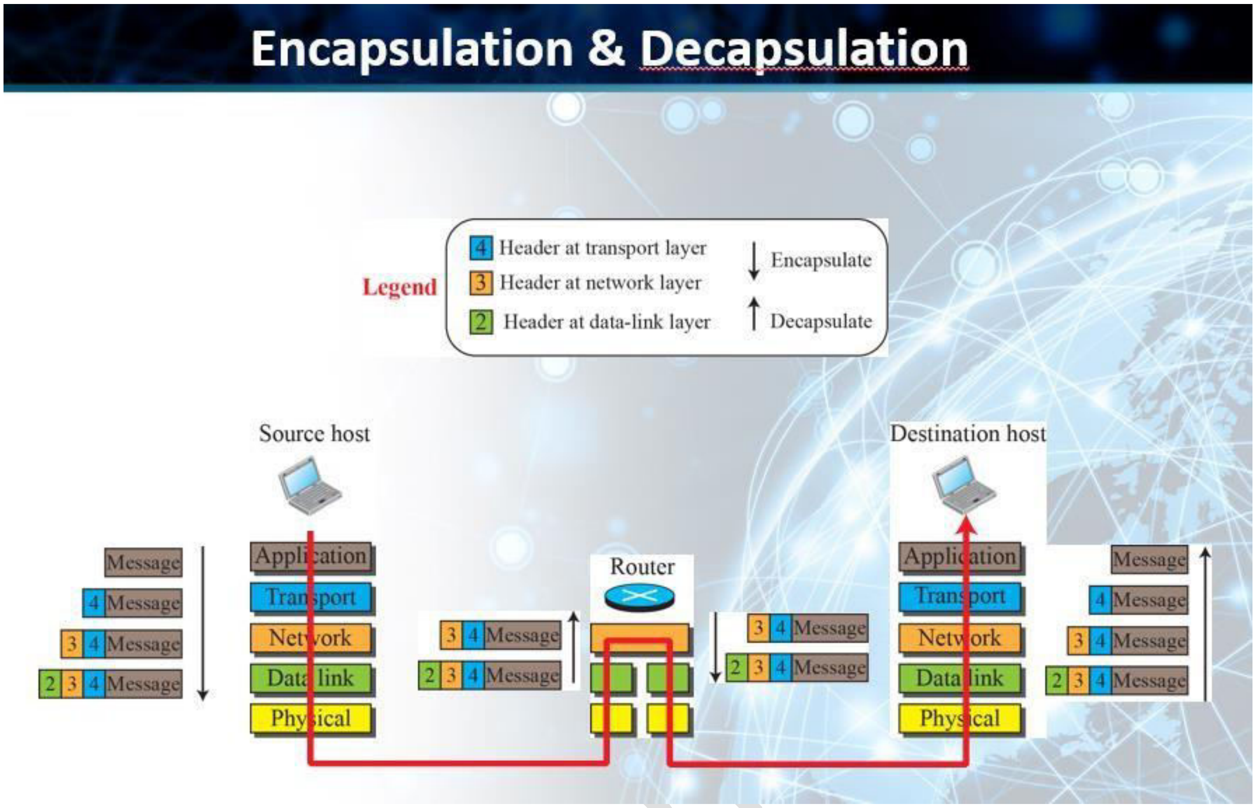
- HTTP, SMTP and FTP protocols are used in Application layer
- TCP and UDP in Transport layer
- IP protocol in Network layer



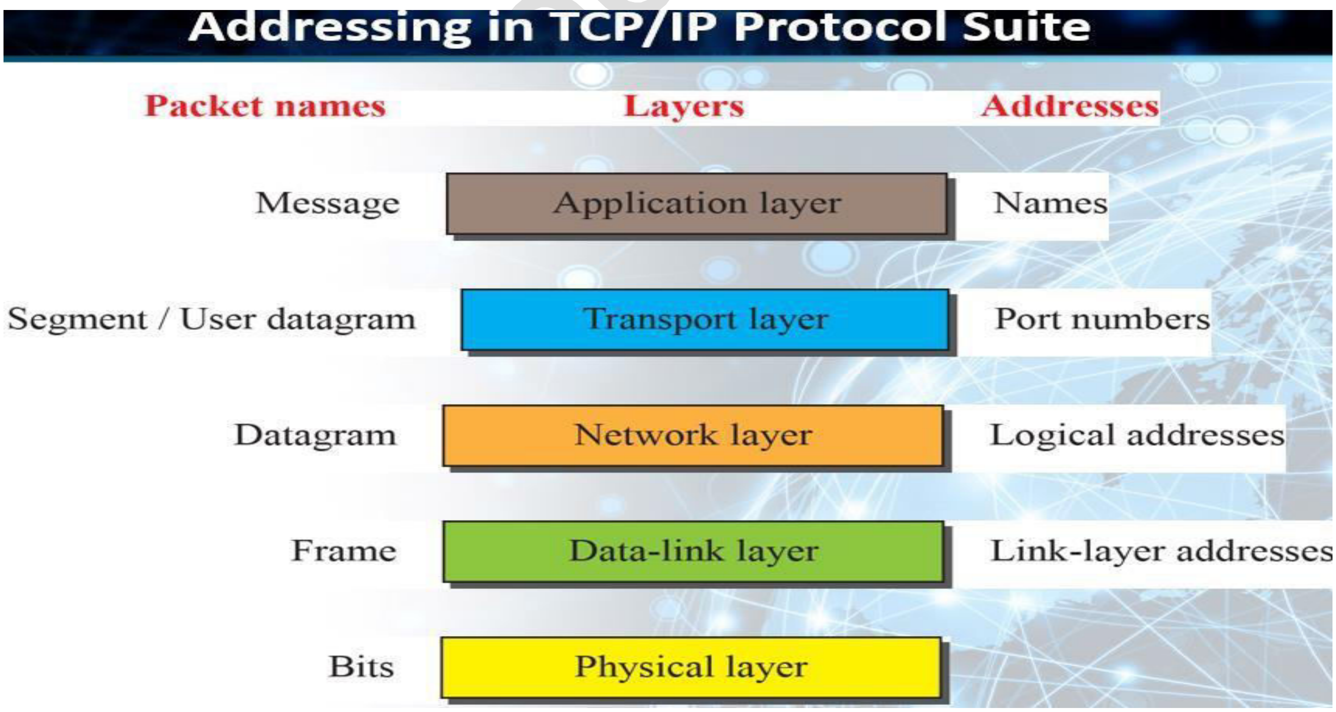
Lecture # 26

Encapsulation is the process of enclosing data from higher network levels with headers from lower layers to facilitate transmission, similar to placing a letter inside an envelope.

Decapsulation refers to the procedure of unsealing the envelope, eliminating the headers, and gaining access to the initial data upon reception. These operations guarantee the appropriate encapsulation and decapsulation of data as it traverses various network layers.



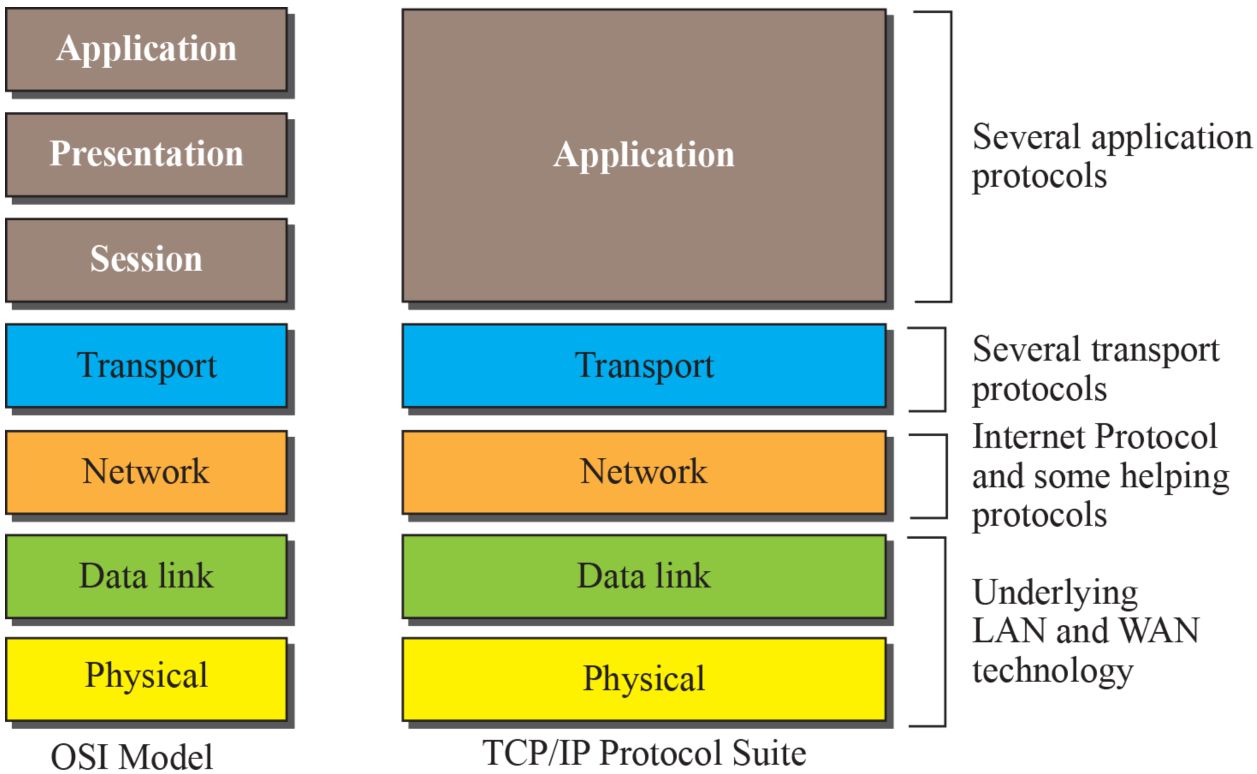
The following shows the addressing done at each layer:



Lecture # 27

The number and layers are both different in TCP/IP protocol suite as compared with OSI model.

The following figure clearly depicts the difference:

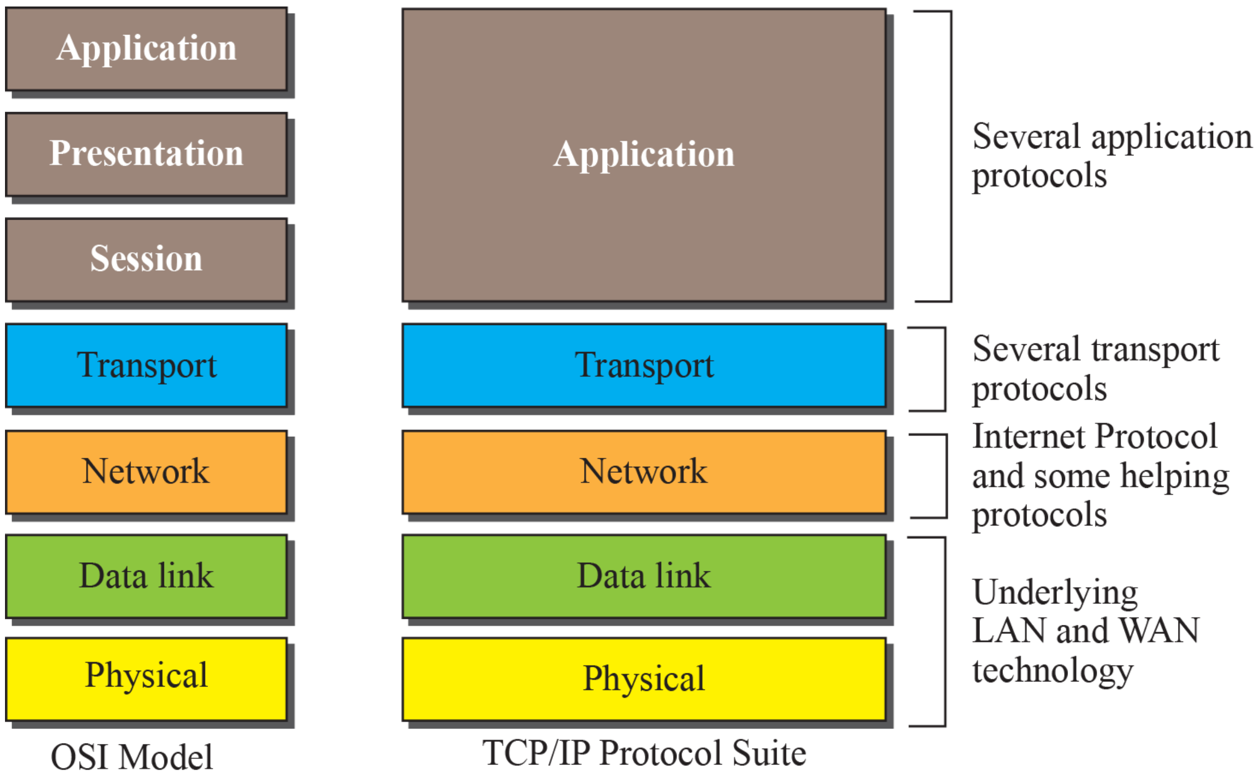


There are several reasons OSI did not replace TCP/IP, three of them are mentioned below:

- OSI was completed when TCP/IP was fully in place
- Some layers in OSI not fully defined
- Performance of TCP/IP better than that of OSI

Lecture # 28

We will discuss the bottom two layers i.e**. the Physical and Datalink layers in our course.**

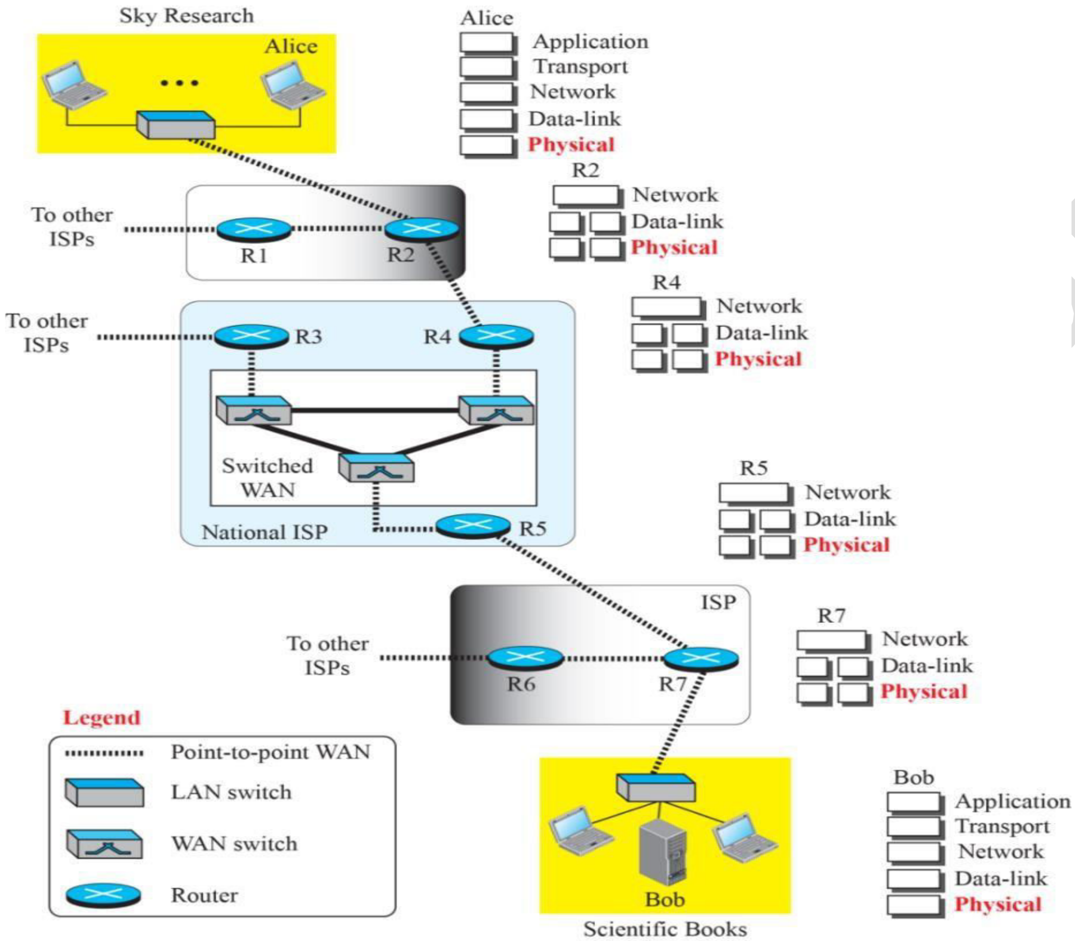


The following topics in these two layers will be covered:

- Analog & Digital Transmission
- Transmission Media
- Switching
- Error Detection and Correction
- Media Access and Data Link Control
- Wired and Wireless LANs

Lecture # 29

The following figure shows the layers involved in the communication between two users i.e. Alice and Bob.



The data signals can take two forms: Digital and Analog

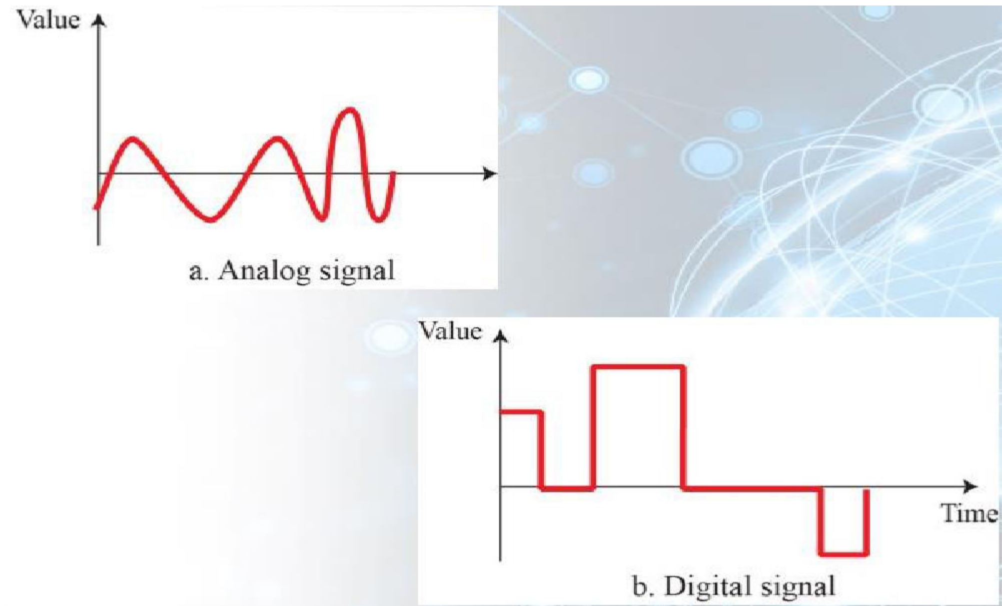
Digital Data:
Digital data is information that is stored in a precise numerical form, most often in the binary system, where data is stored as groups of 0s and 1s. In a digital representation, each number (bit) stands for a different piece of data. Digital data is accurate, simple to change, and can be sent, saved, and processed quickly by computers and other electronic systems. Noise and degradation are less likely to happen during transfer, so it can be used for long-term storage and accurate communication.

Analogue Data:
On the other hand, analogue data uses constant signals to show information. Within a certain time frame, these signals can have a lot of different meanings. Natural sounds, human voices, and temperatures in the surroundings are all examples of analogue data. An analogue signal is different from digital signals because it is constant and could have any value within a certain range. In audio and video signals, for example, analogue signals are used when the information they carry changes smoothly and constantly. During transfer and conversion, noise and degradation can happen to analogue data.

Lecture # 30

Analog Signal:
A continuous, smooth waveform that uses an infinite range of values to describe information is called an analogue signal. Like nature's sounds or changes in temperature, it changes all the time.

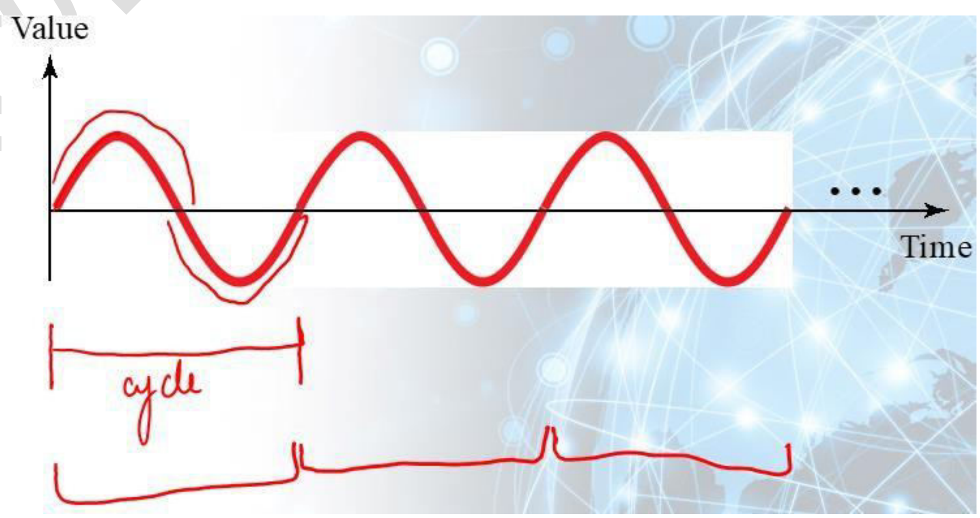
Digital Signal:
A digital output only has a limited number of values, which are usually shown as 0s and 1s. In computers and other modern electronics, digital signals are used to reliably and accurately send and store data.



Lecture # 31

Periodic analogue signals:
Periodic analogue signals are signals that exhibit a recurring pattern within a defined time frame. They demonstrate consistent and repetitive behaviour, which allows for easy analysis within the field of signal processing.

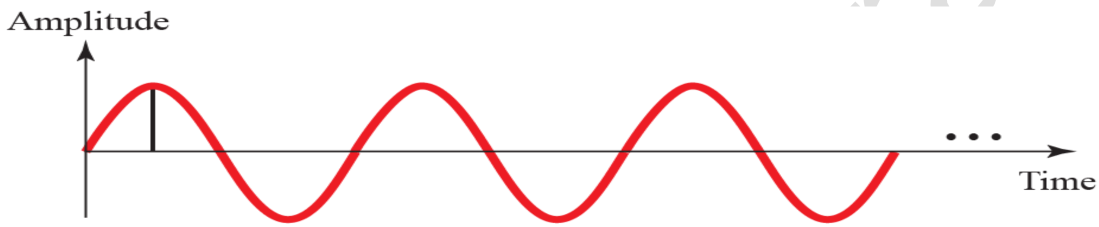
Let's look at a simple periodic analogue signal, specifically a sine wave.



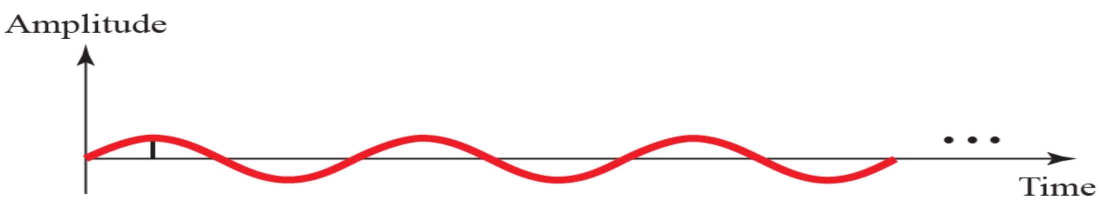
A simple periodic analog signal, visualized as a sine wave, displays a consistent rhythmic motion at regular intervals. Sine waves are fundamental in explaining natural phenomena like sound waves and electrical currents. They have a singular frequency and amplitude, making them vital in signal processing and mathematics.

Composite Periodic Analog Signal:

A composite periodic analog signal is complex, composed of multiple sine waves with different frequencies and amplitudes. These waves combine to form intricate waveforms, representing complex real-world signals like music or speech. Professionals use this breakdown method in fields like telecommunications and audio engineering for effective analysis and manipulation.



a. A signal with high peak amplitude



b. A signal with low peak amplitude

Lecture # 32

Frequency refers to the number of cycles that a wave completes within a given time period, usually measured in Hertz (Hz). It determines various characteristics, such as pitch in sound waves and colour in light waves.

The formula for **frequency (f)** is:

$$f = 1/t$$

Where:

- f represents the frequency in Hertz (Hz).
- t represents the period of the wave, which is the time it takes to complete one full cycle, measured in seconds (s).

Period		*Frequency*	
Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^3 Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^6 Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^9 Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

Lecture # 33

Examples (**FYI:** Numerical problem) of frequency calculation:

The power we use at home has a frequency of 60 Hz. The period of this sine wave can be determined as follows:

The period (TT) of a wave is the reciprocal of its frequency (ff). In this case, where the frequency is 60 Hz, the period can be calculated using the formula:

$$T = 1/f$$

Substituting the given frequency:

T=1/60 Hz

T=0.0166 seconds or 16.66 msec

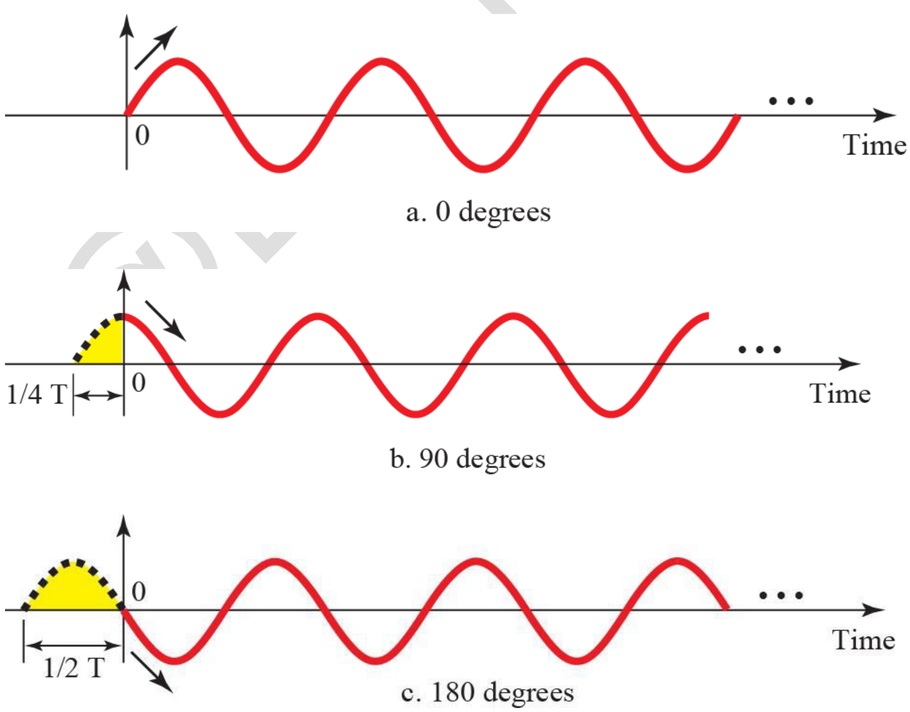
Lecture # 34

Phase or Phase Shift refers to the position of a point on a waveform in relation to a reference point, usually measured in degrees or radians. This is an important concept in signal processing and telecommunications, as it helps with wave synchronization and alignment.

Formula:

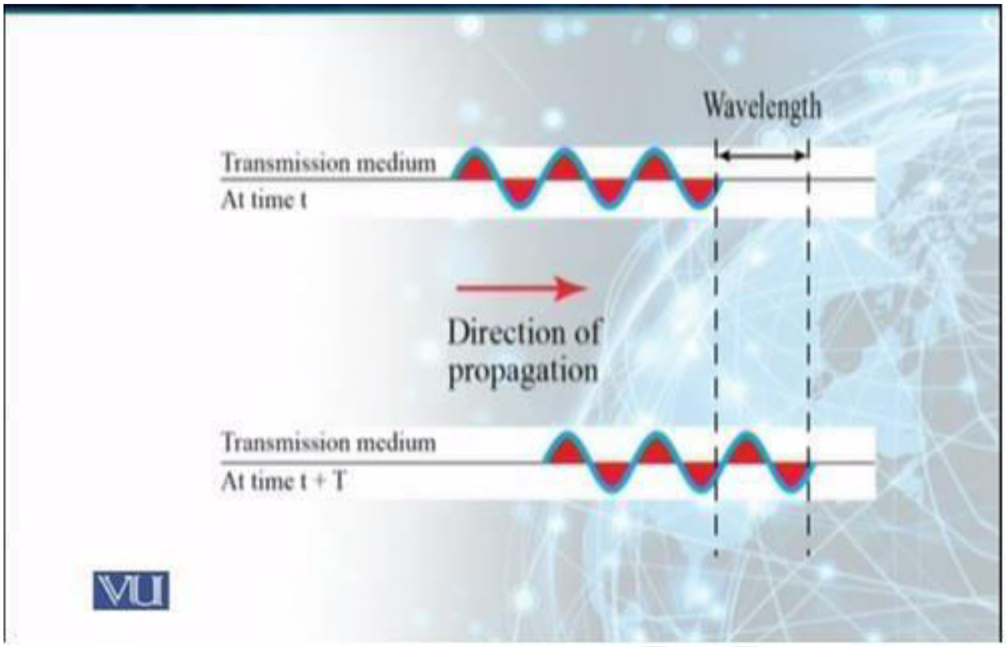
$360\ degrees = 2\pi\ radians$

$1\ degree = 2\pi/360\ radians$



Lecture # 35

Wavelength is another characteristic of a signal traveling through a transmission medium. Wavelength binds the period or the frequency of a simple sine wave to the propagation speed of the medium.



The formula for **wavelength (λ)** is:

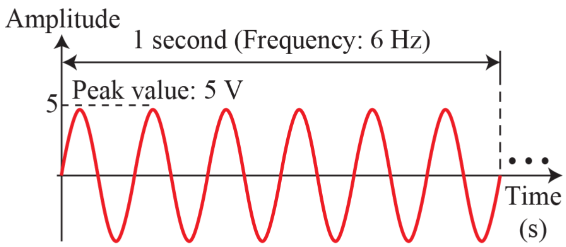
$$\lambda = f v$$

Where:

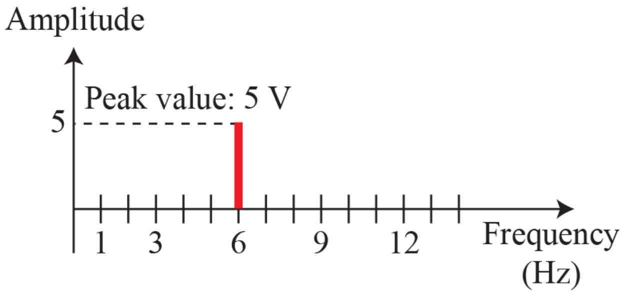
- λ is the wavelength in meters (m).
- v is the velocity of the wave in meters per second (m/s).
- f is the frequency of the wave in Hertz (Hz).

Lecture # 36

- A sine wave is comprehensively defined by its amplitude, frequency, and phase.
- We have been showing a sine wave by using what is called a time domain plot.
- The time-domain plot shows changes in signal amplitude with respect to time (it is an amplitude-versus-time plot).
- Phase is not explicitly shown on a time-domain plot.

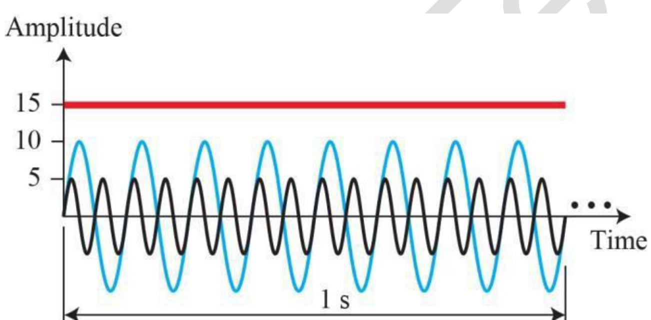


a. A sine wave in the time domain

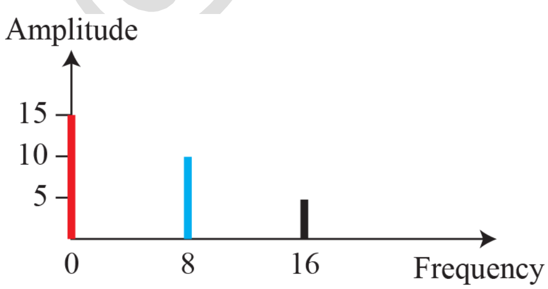


b. The same sine wave in the frequency domain

The frequency domain is more compact and useful when we are dealing with more than one sine wave. For example, Figure 3.9 shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.



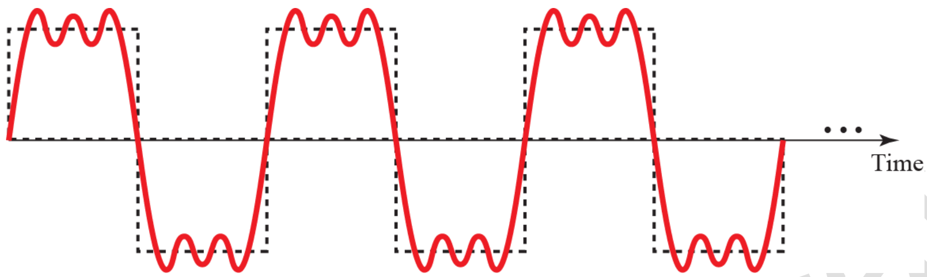
a. Time-domain representation



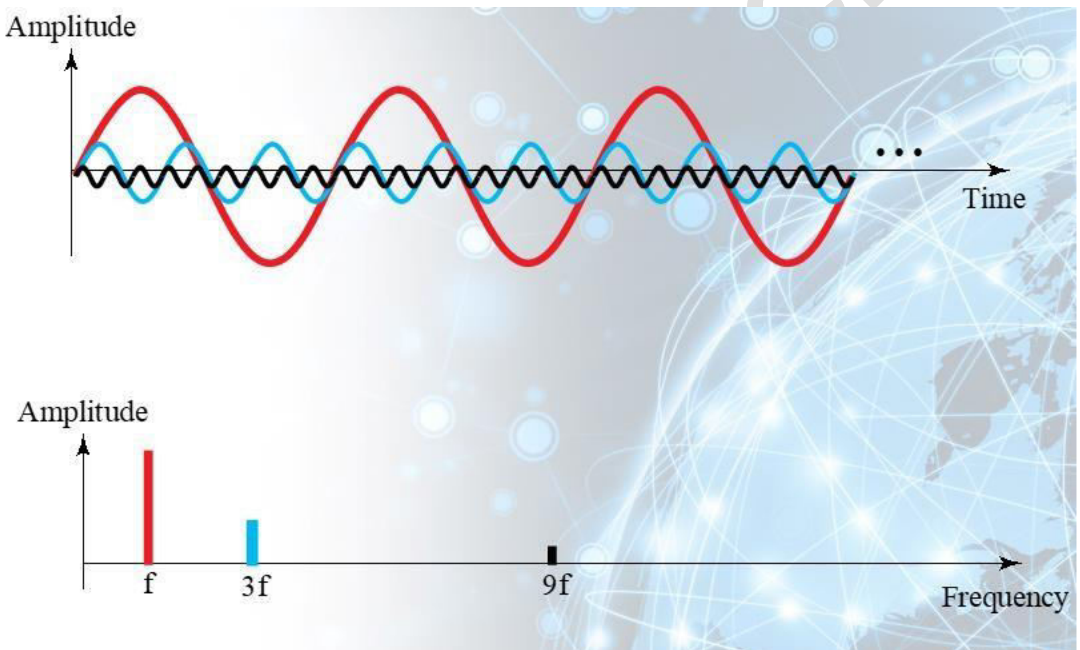
b. Frequency-domain representation

Lecture # 37

- Single Sine Wave can only carry limited information
- Composite Signal is made up of multiple simple sine waves
- Can be periodic or non-periodic



Following figure shows the decomposition of composite periodic signal:



b. Frequency-domain decomposition of the composite signal

Lecture # 38

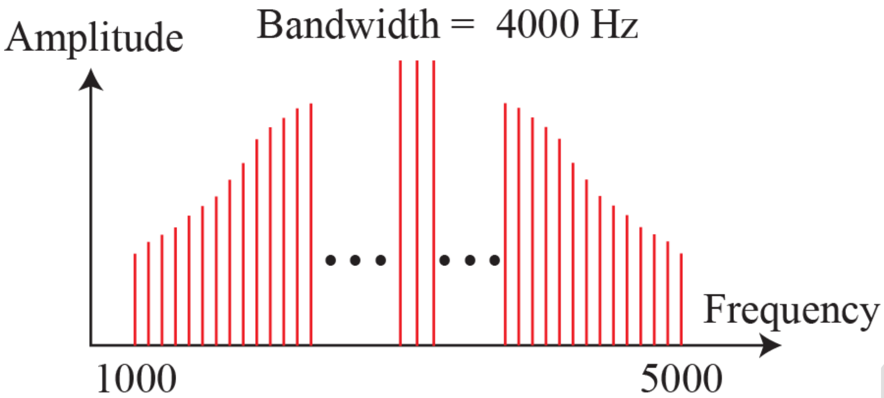
A communication channel or network's bandwidth is its frequency range. It is the difference between the higher and lower frequencies in a continuous band, measured in Hertz.

In radio transmission, a channel's bandwidth determines its frequency range. In signal processing, a wide bandwidth transmits more data but uses more resources. Thus, understanding bandwidth is crucial to optimising communication systems to ensure the channel or network can handle data transmission frequencies.

Bandwidth can be used in two different contexts with two different measuring values:

- Bandwidth in Hertz
- Bandwidth in bits per second

Bandwidth of a composite signal:

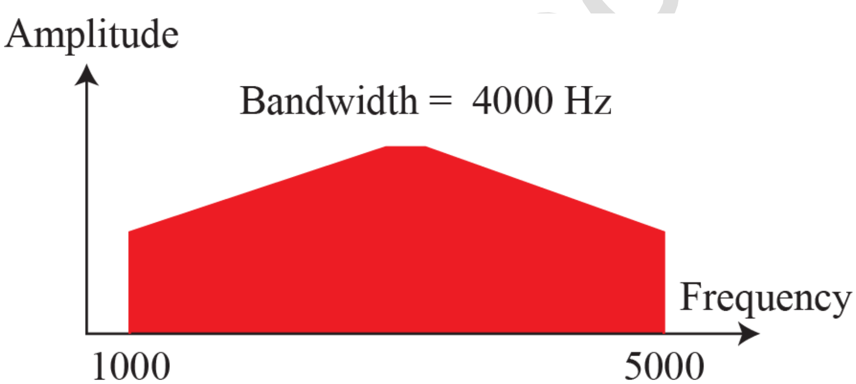


a. Bandwidth of a periodic signal

$B = f_h - f_l$

$B = 5000 - 1000$

$B = 4000$ Hz



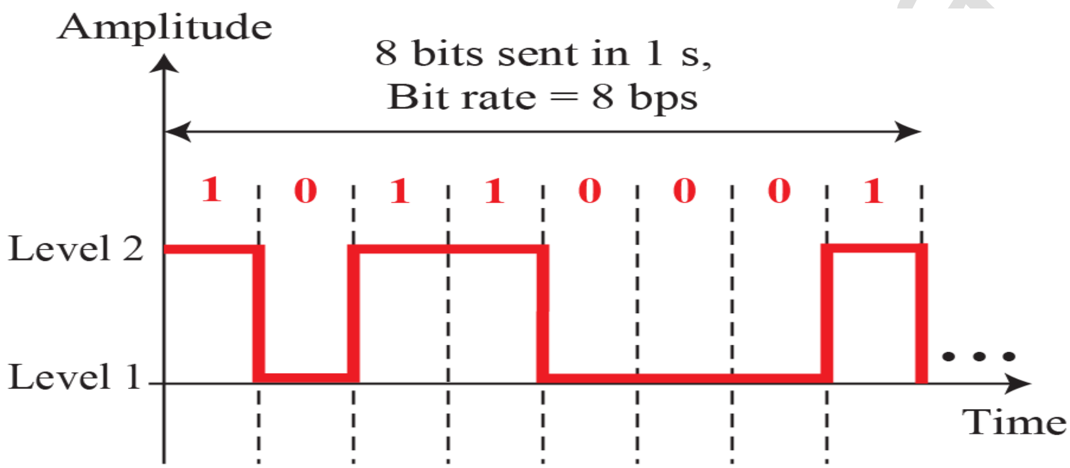
b. Bandwidth of a nonperiodic signal

Lecture # 39

In addition to being represented by an analog signal, information can also be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels.

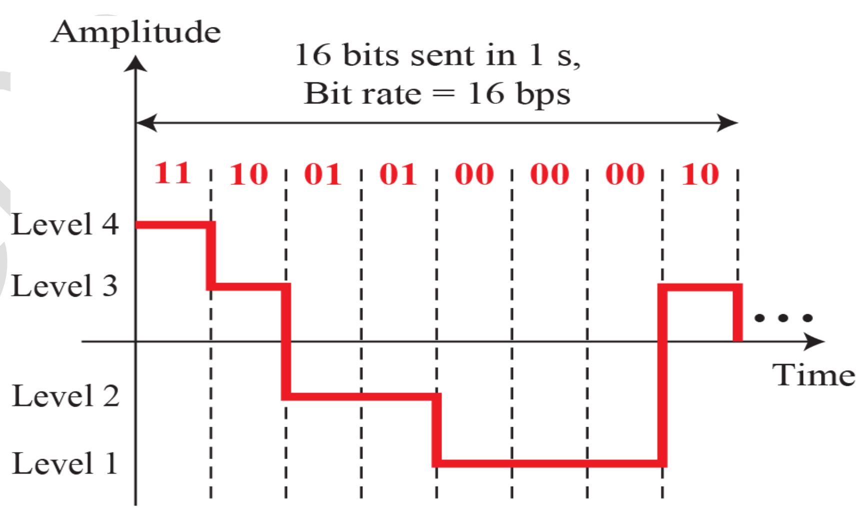
Information can also be represented by a digital signal

- For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage
- A digital signal can have more than two levels so that we can send more than one bit for each level



a. A digital signal with two levels

In general, if a signal has L levels, each level needs log2 L bits. So, we can send $\log_2 2 = 1$ bit per level



b. A digital signal with four levels

$\log_2 4 = 2$ bits

Lecture # 40

(**FYI:** Numerical problems) Example:

1) A digital signal has eight levels. How many bits are needed per level?

Number of bits per level:

$Log_2 L$

$Log_2 (8)$

=3 **ANSWER**

Hence 3 bits are needed per level.

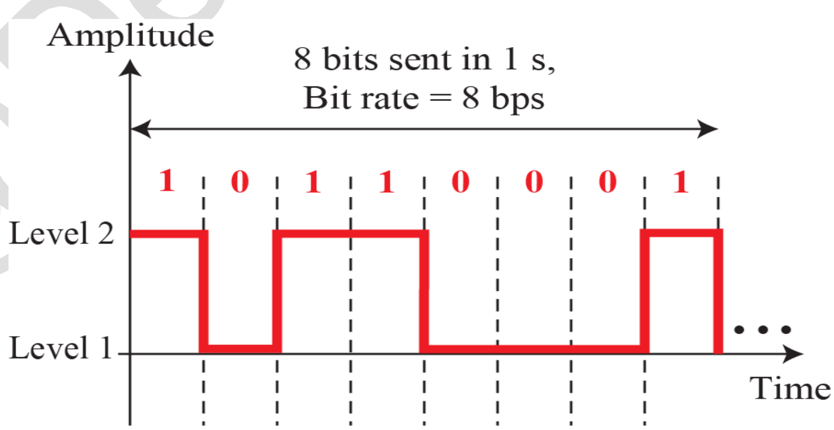
2) A digital signal has nine levels. How many bits are needed per level? We calculate the number of bits by using the formula.

We calculate the number of bits by using the formula. Each signal level is represented by 3.17 bits. However, this answer is not realistic. The number of bits sent per level needs to be an integer as well as a power of 2. For this example, 4 bits can represent one level.

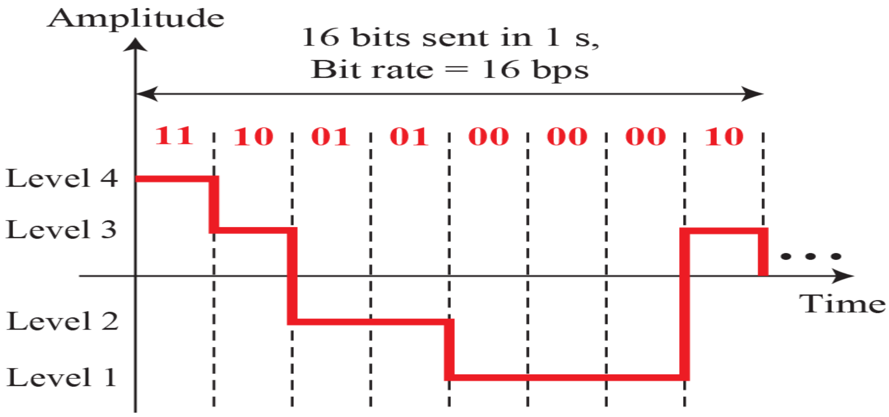
Lecture # 41

Bit Rate

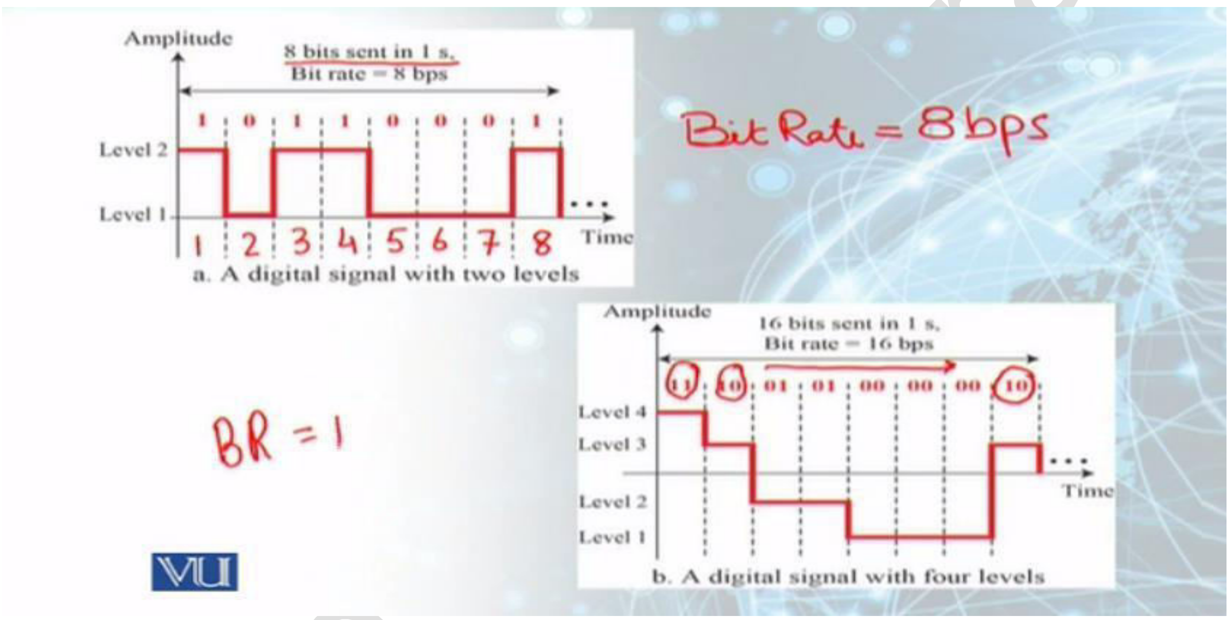
- Number of bits sent in 1 second.
- Bit Rate is expressed in bits per second (bps)
- Most digital signals are non-periodic, and thus period and frequency are not appropriate characteristics.



a. A digital signal with two levels



b. A digital signal with four levels

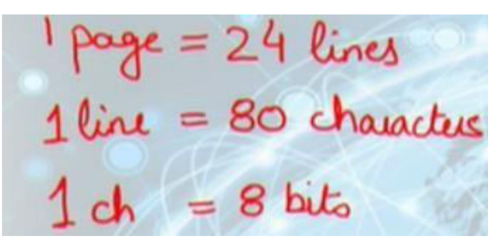


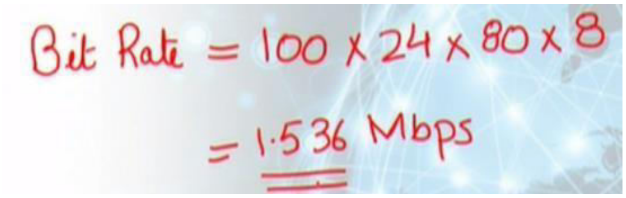
Bit Rate

Most digital signals are nonperiodic, and thus period and frequency are not appropriate characteristics. Another term—bit rate (instead of frequency)—is used to describe digital signals. The bit rate is the number of bits sent in 1s, expressed in bits per second (bps). Figure 3.17 shows the bit rate for two signals.

(FYI: Numerical problem) Example

Assume we need to download text documents at the rate of 100 pages per second. What is the required bit rate of the channel?

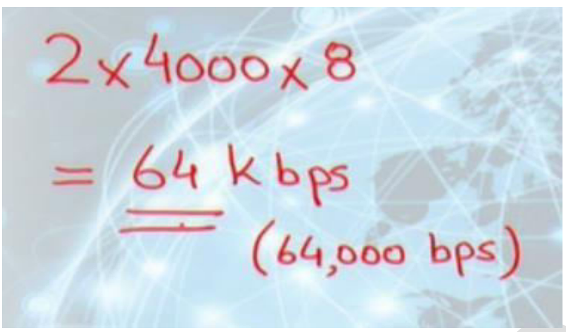




$100 \times 24 \times 80 \times 8 = 1{,}536{,}000 \text{ bps} = 1.536 \text{ Mbps}$ ANSWER

Example

A digitized voice channel is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?



A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate

$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$

Bit Length

We discussed the concept of the wavelength for an analog signal: the distance one cycle occupies on the transmission medium. We can define something similar for a digital signal: the bit length. The bit length is the distance one bit occupies on the transmission medium.

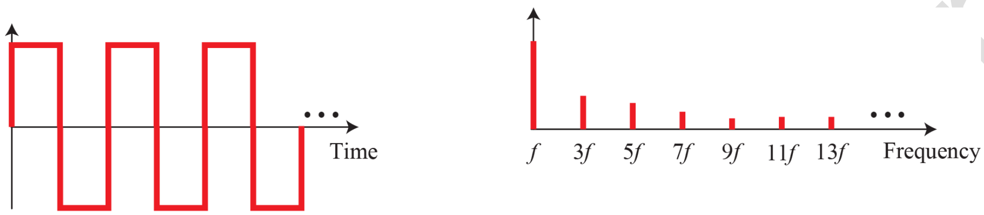
$$\text{Bit length} = \text{propagation speed} \times \text{bit duration}$$

Lecture # 42

Digital Signal as Composite Analog Signal

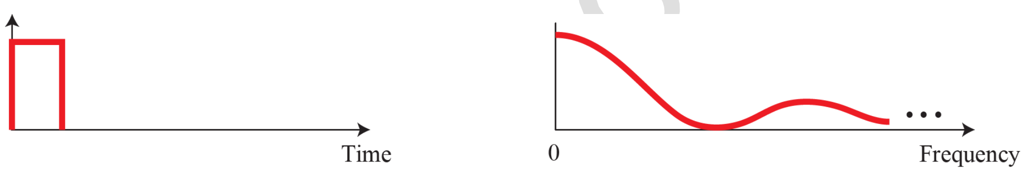
- Based on Fourier analysis, a digital signal is a composite analog signal
- A digital signal, in the time domain, comprises connected vertical and horizontal line segments.
- A vertical line in the time domain means a frequency of infinity.
- A horizontal line in the time domain means a frequency of zero.
- Going from a frequency of zero to a frequency of infinity implies all frequencies in between are part of the domain.

Digital Signal as Composite Analog Signal



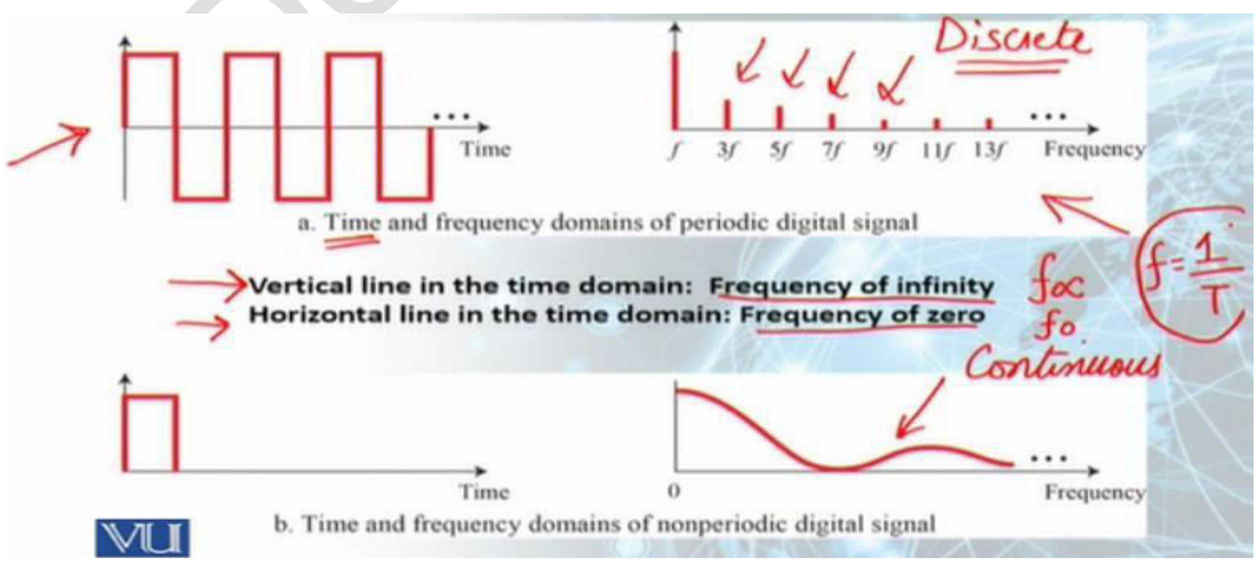
a. Time and frequency domains of periodic digital signal

Vertical line in the time domain: Frequency of infinity Horizontal line in the time domain: Frequency of zero



b. Time and frequency domains of nonperiodic digital signal

Digital Signal as Composite Analog Signal



Lecture # 43

Transmission of Digital Signals

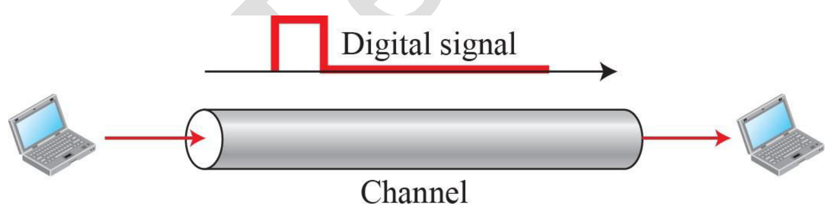
- Digital signal, periodic or non-periodic, is a composite analog signal with frequencies between zero and infinity (Infinite Bandwidth)

- Two approaches for transmission:
    - Baseband Transmission: Baseband transmission is a method of transmitting digital signals over a communication medium, such as a cable or a fiber optic line, in which the entire bandwidth of the medium is used to send a single stream of data.
    - Broadband Transmission: Broadband transmission is a method of sending multiple signals simultaneously over a communication medium, such as a cable, fiber optic line, or wireless channel, by dividing the available bandwidth into multiple frequency channels.

Transmission of Digital Signals

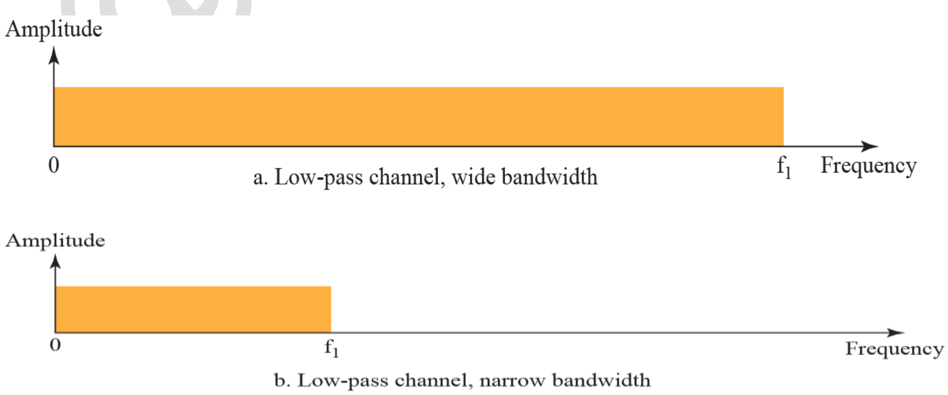
- A vertical line in the time domain means a frequency of infinity
- A horizontal line in the time domain means a frequency of zero.
- Going from a frequency of zero to a frequency of infinity implies all frequencies in between are part of the domain.

Baseband Transmission

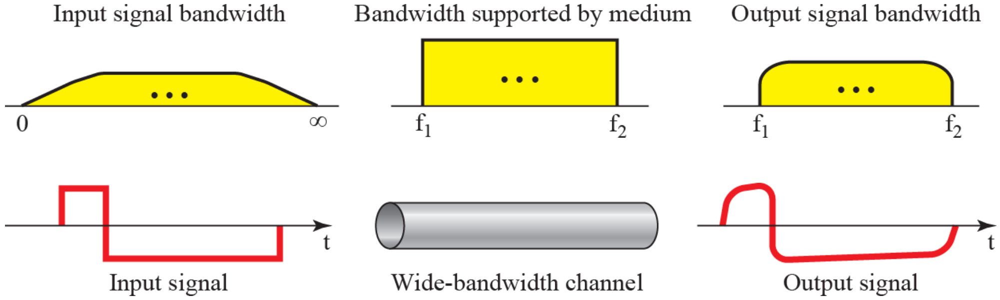


Sending a Digital Signal without changing it to an Analog Signal

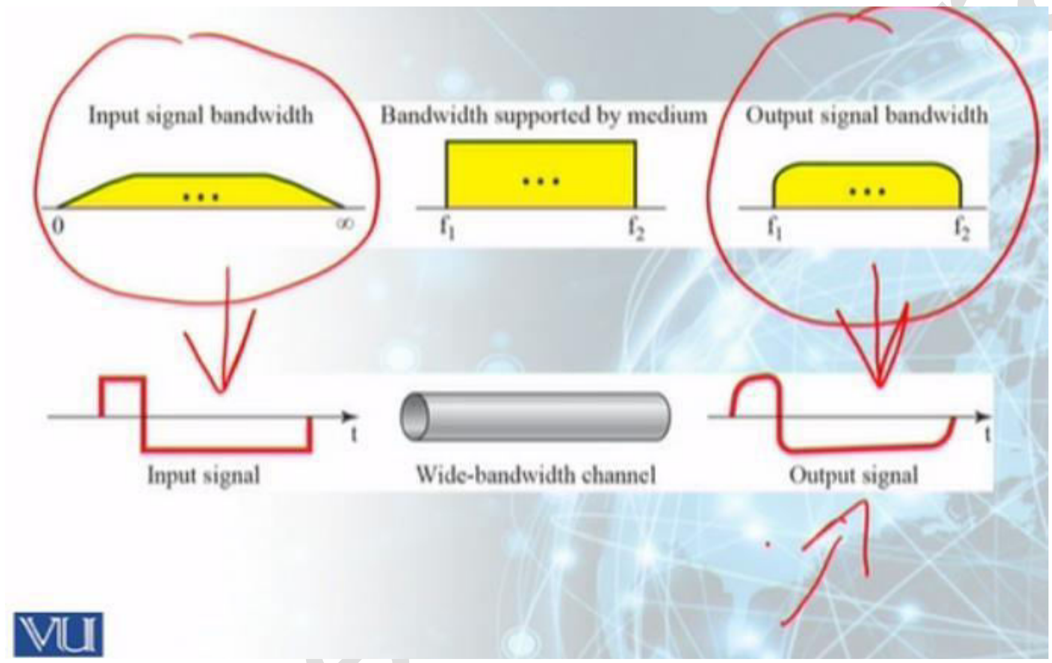
Baseband Transmission



Baseband Transmission



Baseband Transmission



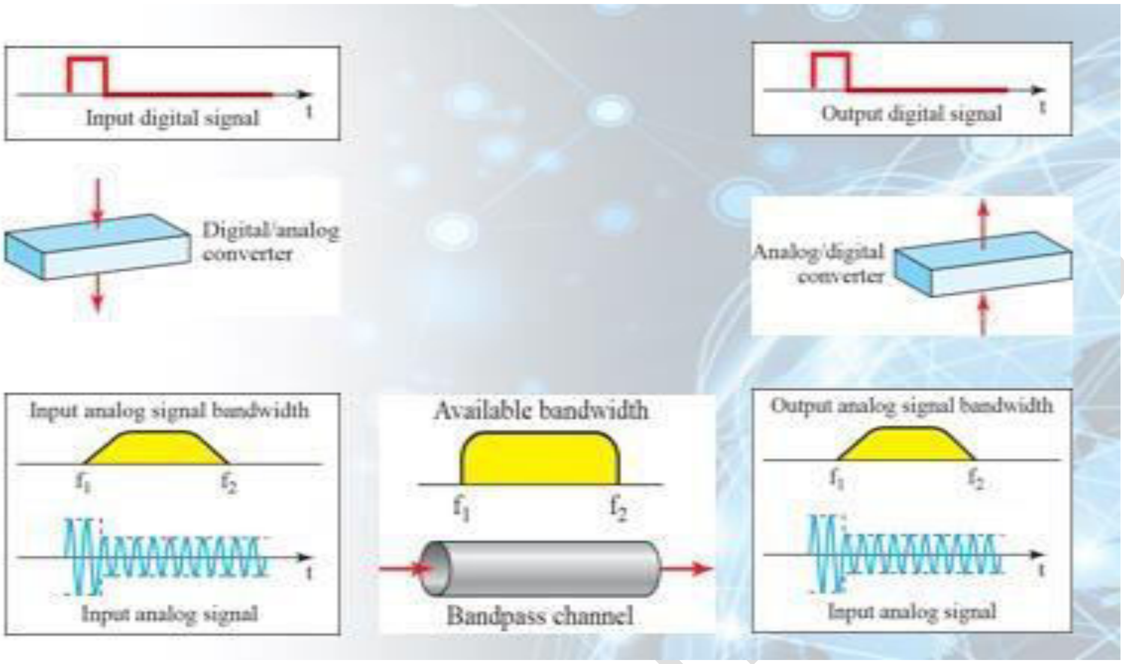
Lecture # 44

Broadband Transmission (Modulation)

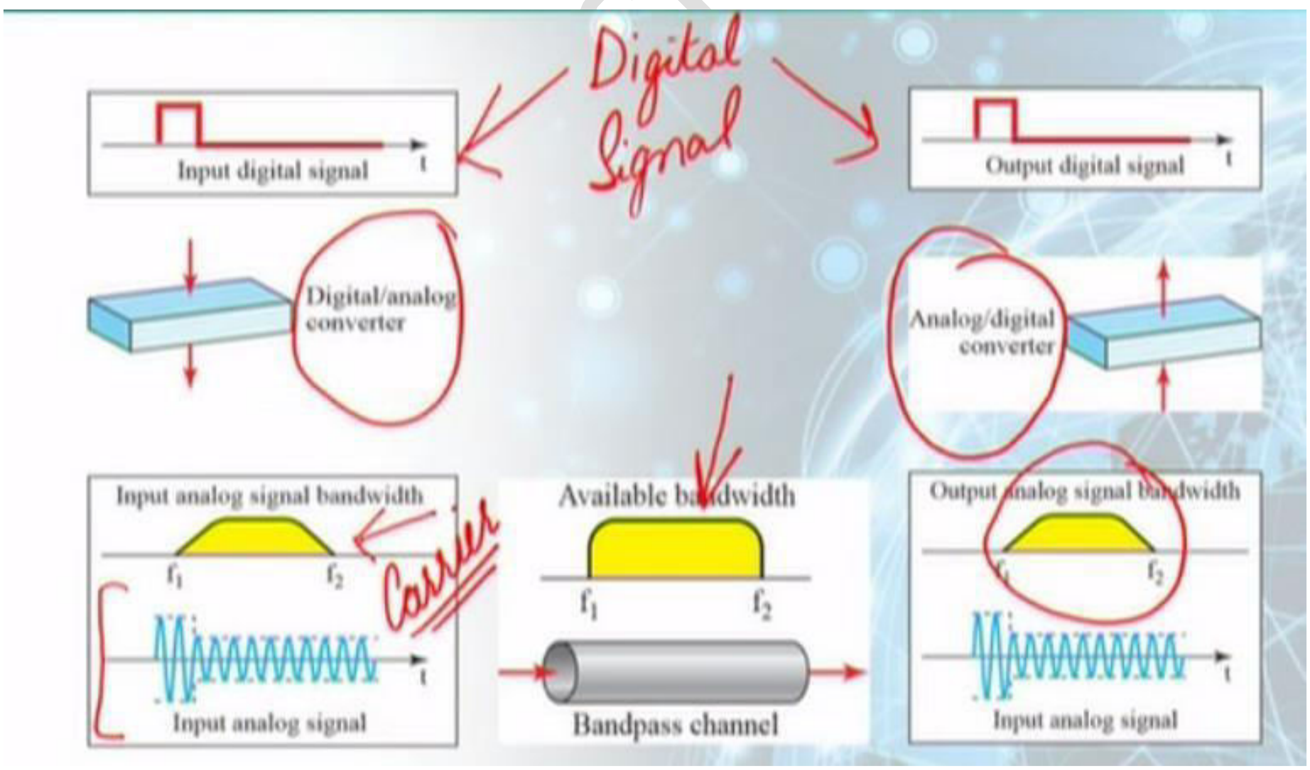
- Changing the Digital signal to an Analog signal for transmission
- Modulation allows us to use a bandpass channel—a channel with a bandwidth that does not start from zero
- More available than a low-pass channel

Broadband Transmission (Modulation)

Broadband Transmission (Modulation)



Broadband Transmission (Modulation)

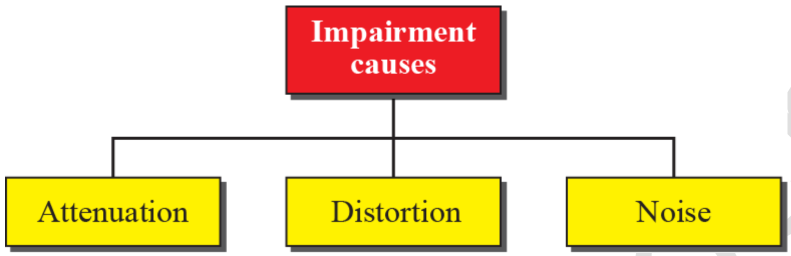


Lecture # 45

Transmission Impairments

- Transmission media are not perfect
- Cause Signal impairments
- Signal sent is not the same as the signal received

Causes of Transmission Impairment



Transmission Impairment
What is sent is not what is received. Three causes of impairment are attenuation, distortion, and noise

Transmission Impairment

Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium.

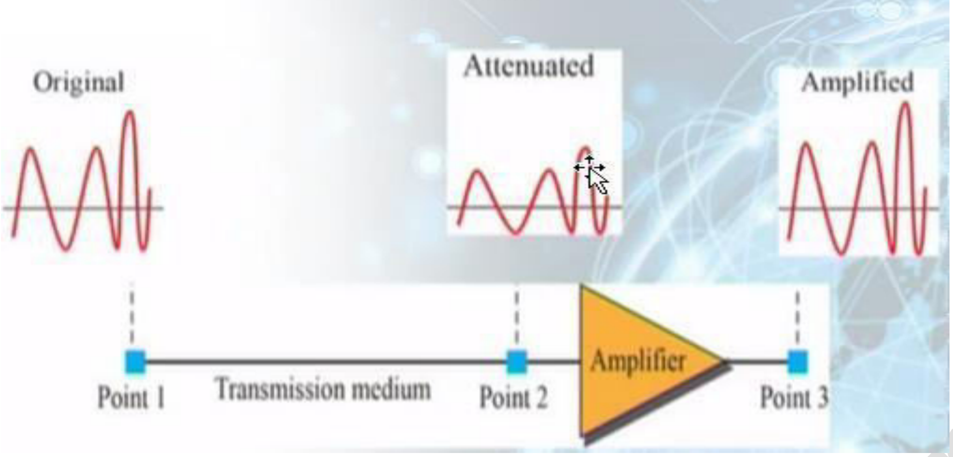
Attenuation

Attenuation means a loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium. That is why a wire carrying electric signals gets warm, if not hot, after a while. Some of the electrical energy in the signal is converted to heat.

To compensate for this loss, amplifiers are used to amplify the signal. Figure 3.27 shows the effect of attenuation and amplification.

Attenuation and amplification

Figure 3.27



Example 3.26

Suppose a signal travels through a transmission medium and its power is reduced to one half. This means that P2 = 0.5 P1. In this case, the attenuation (loss of power) can be calculated as

$$\boxed{10 \log_{10} P_2/P_1 = 10 \log_{10} (0.5\, P_1)\, /P_1}$$

$$\boxed{= 10 \log_{10} 0.5 = 10 \times (-0.3) = -3 \text{ dB.}}$$

A loss of 3 dB (−3 dB) is equivalent to losing one-half the power.

Example 3.27

A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 =$ 10P1. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1} =$$

$$10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

Example 3.29

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as dB_m and is calculated as dBm = 10 log10 Pm, where Pm is the power in milliwatts. Calculate the power of a signal if its $dB_m = -30$.

Solution

We can calculate the power in the signal as

$$dB_m = 10\log_{10} \longrightarrow dB_m = -30 \longrightarrow$$

$$\log_{10} P_m = -3 \longrightarrow P_m = 10^{-3}\,\mathrm{mW}$$

Lecture # 46

Attenuation and Amplification - Decibel

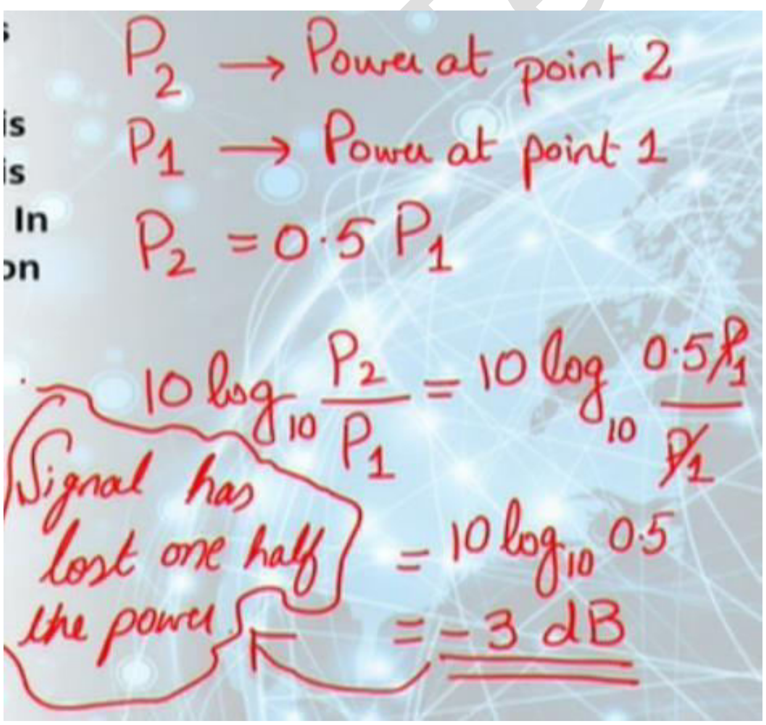
- Unit of Signal strength is Decibel or dB
- Decibel (dB) measures the relative strengths of two signals or one signal at two different points

$$10\log_{10} P_2/P_1$$

- Decibel is negative if a signal is attenuated and positive if signal is amplified

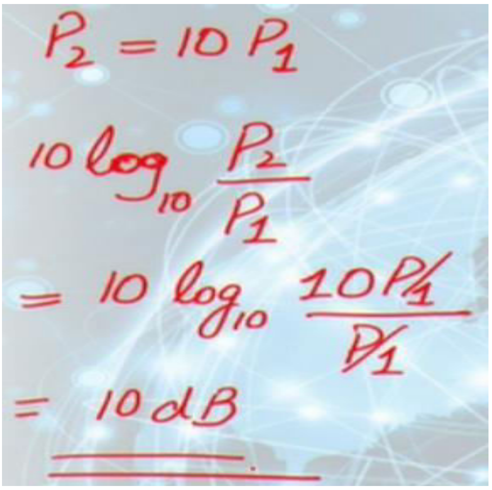
Example

Suppose a signal travels through a transmission medium and its power is reduced to one half. This means that P2 = 0.5 P1. In this case, the attenuation (loss of power) can be calculated as



Example

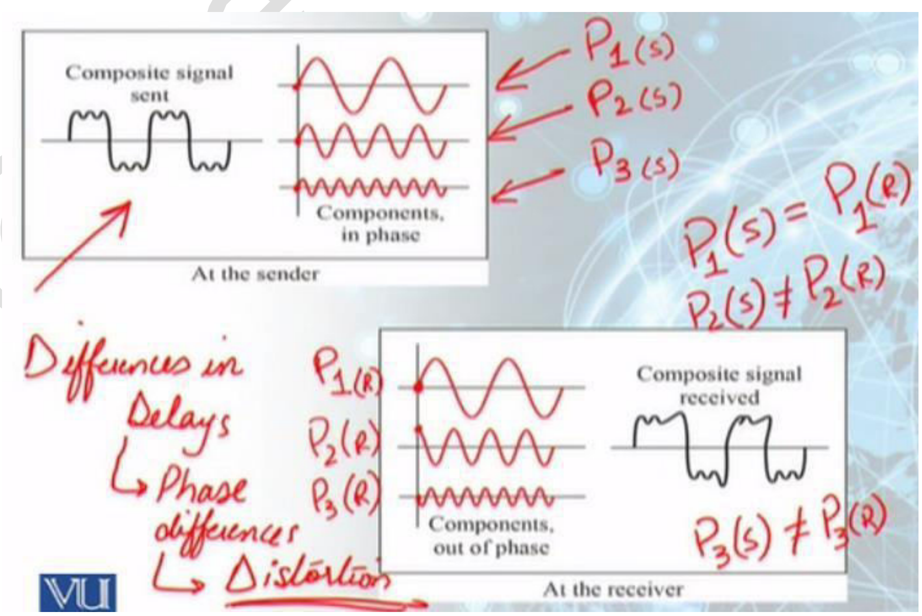
A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P1$. In this case, the amplification (gain of power) can be calculated as



Lecture # 47

Distortion

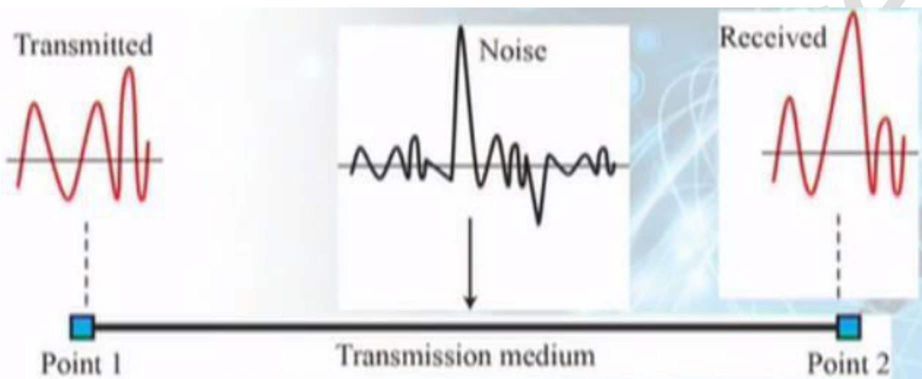
- Distortion means that the signal changes its form or shape.
- Distortion can occur in a composite signal made of different frequencies.
- Each signal component has its own propagation speed (see the next section) through a medium and, therefore, its own delay in arriving at the destination.
- Differences in delay may create a difference in phase if the delay is not the same as the period duration.



Lecture # 48

Noise

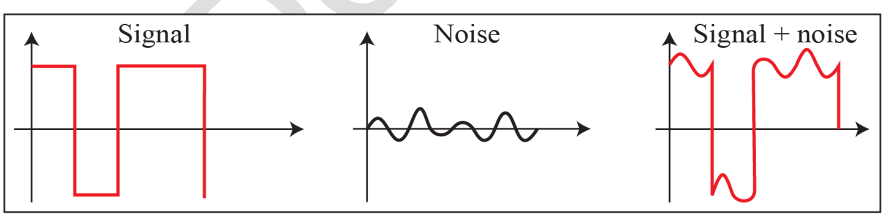
- Noise is another cause of impairment.
- Several types of noise, such as thermal noise, induced noise, crosstalk, and impulse noise, may corrupt the signal.
- Thermal noise is the random motion of electrons in a wire, which creates an extra signal not originally sent by the transmitter.
- Induced noise comes from sources such as motors.
- Crosstalk is the effect of one wire on the other.



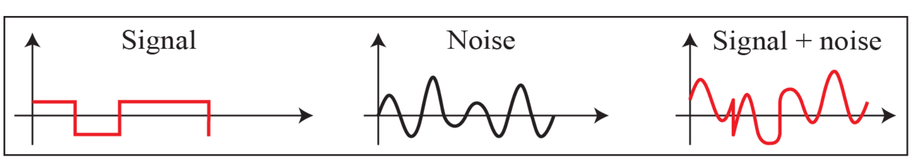
Noise – Signal to Noise Ratio (SNR)

- Signal to Noise Ratio (SNR) is used to find the theoretical bit rate limit of a signal

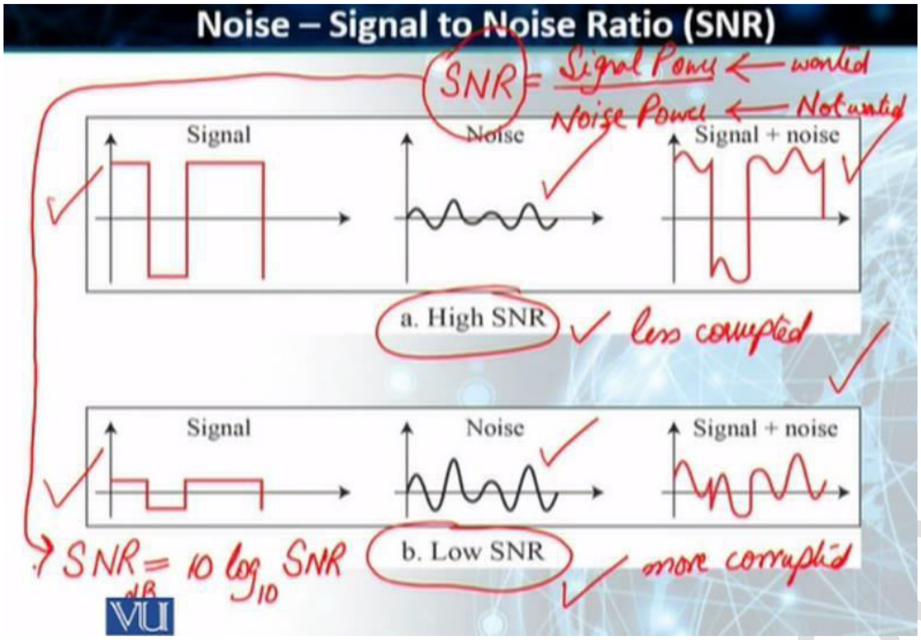
$$\mathrm{SNR} = \frac{\text{average signal power}}{\text{average noise power}}$$



a. High SNR

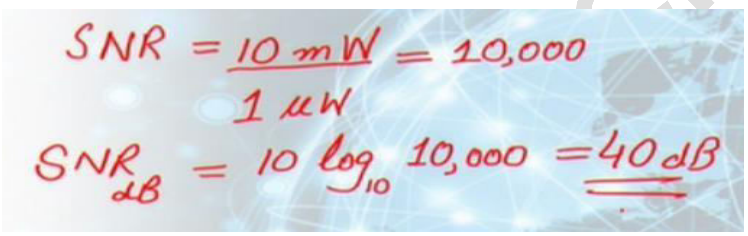


b. Low SNR



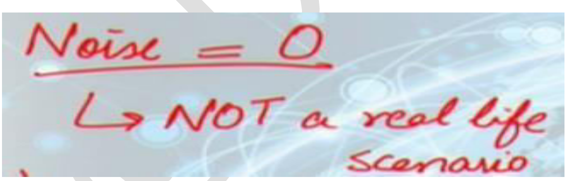
Example

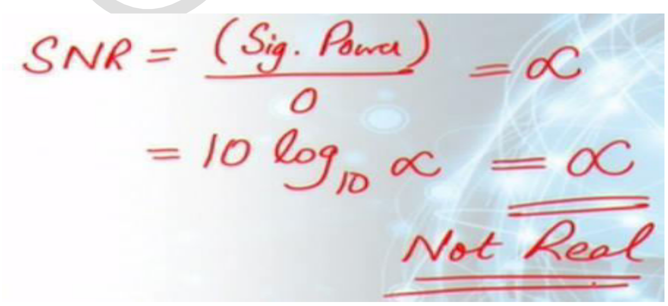
The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and $SNR_{dB?}$



Example

The values of SNR and SNR_{dB} for a noiseless channel are calculated as





Lecture # 49

Data Rate Limits

- How fast we can send data, in bits per second, over a channel?
- Data Rate depends on 3 factors:
  - The Bandwidth available
  - The level of the signals we use
  - The level of noise

- Two theoretical formulas developed to calculate the data rate:
  - one by Nyquist for a noiseless channel
  - another by Shannon for a noisy channel

Noiseless Channel: Nyquist Rate

- For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate
- Finding balance between Bit rate and System Reliability

$$\text{Bit Rate} = 2 \times \text{Bandwidth} \times \log_2 L$$

BW = BW of channel

L = No. of signal levels

BR = bps

$$\boxed{\text{Bit Rate} \propto L}$$

$L \uparrow \implies \text{Bit Rate} \uparrow$

$(L \implies 0, 1)$ $L = 20$ levels

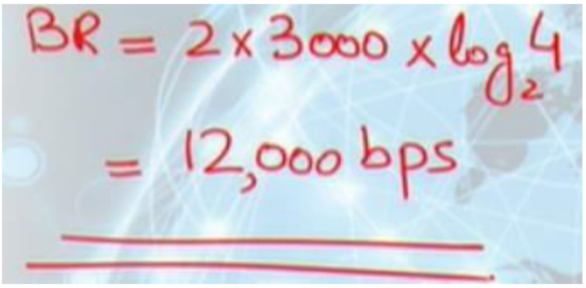
Example

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$BR = 2 \times 3000 \times \log_2 2$$
$$= 6000 \text{ bps}$$

Example

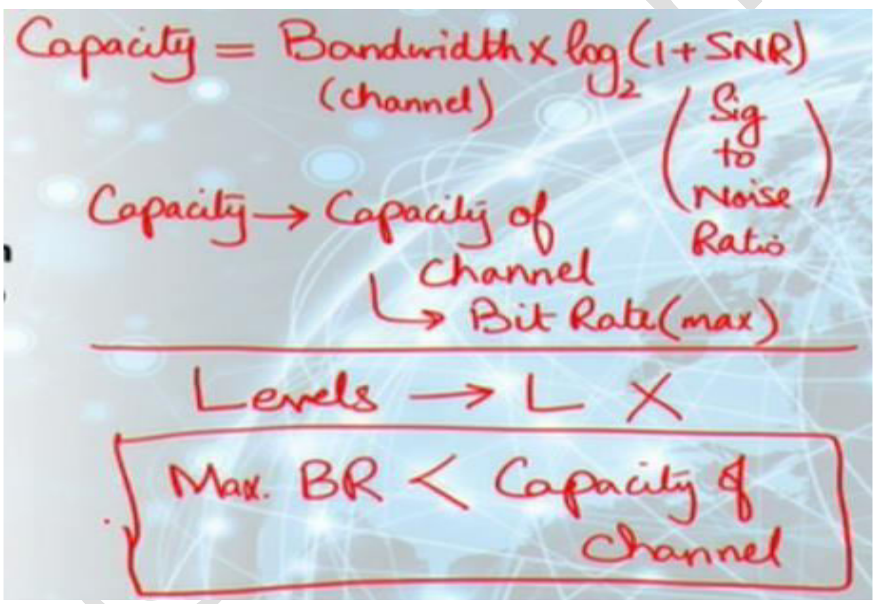
Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as



Lecture # 50

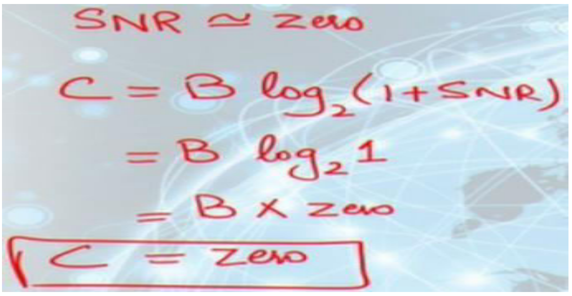
Noisy Channel : Shannon Capacity

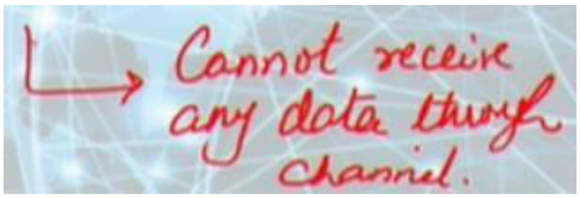
- We cannot have a noiseless channel; the channel is always noisy
- In 1944, Claude Shannon introduced a formula, to determine the theoretical highest data rate for a noisy channel:



Example

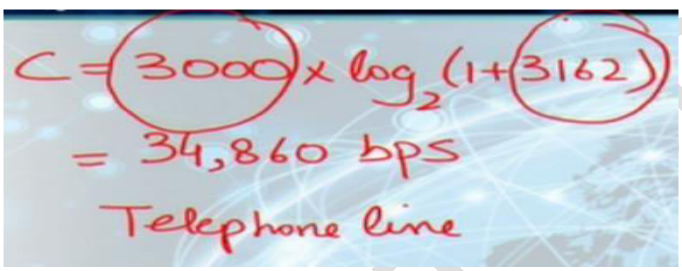
Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as





Example

Theoretical highest bit rate of a Telephone line with a Bandwidth of 3000 Hz assigned for data communication. SNR is usually 3162. The capacity is calculated as:



Using Both Limits

- In practice, we need to use both methods to find the limits and signal levels
- Shannon's formula gives us the upper limit while the Nyquist formula gives us the signal levels

Example

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as

$$C = B \log_2 (1 + \text{SNR}) = B \log_2(1 + 0)$$

$$= B \log_2 1 = B \times 0 = 0$$

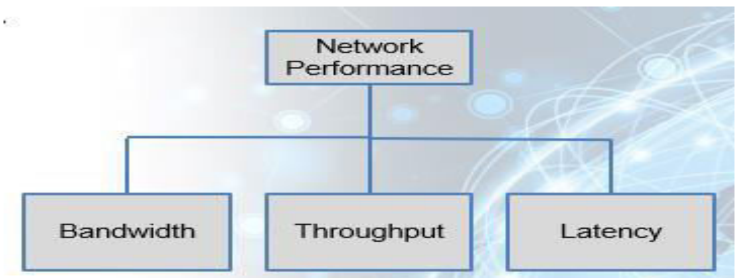
This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

Lecture # 51

Network Performance

- Data transmission (in form of Signal) over a network and how the network behaves is important.
- More important is the performance of the network.

Network Performance

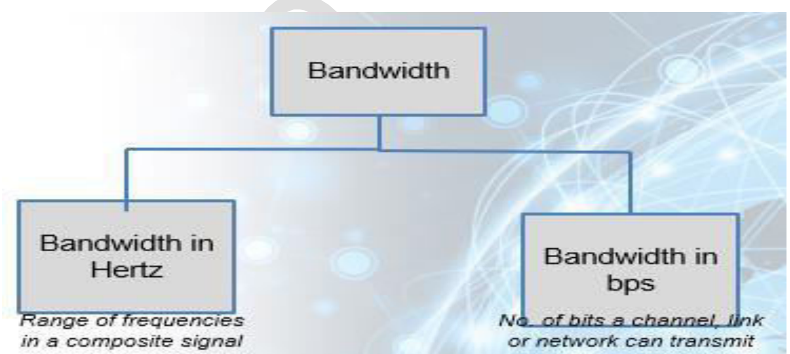


There are 3 characteristics of network performance.

Bandwidth

- An important characteristic that measures Network Performance
- Bandwidth can be used in two different contexts with two different measuring values:
  - Bandwidth in Hertz
  - Bandwidth in bits per second

Bandwidth



Throughput

- Measure how fast we can send data through a network.
- Bandwidth is not the same as Throughput
- A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B

Example

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

$$T = \frac{(12,000 \times 10,000)}{60} = \underline{2 \text{Mbps}}$$

$$\boxed{\begin{aligned} T &= 2 \text{Mbps} \\ B &= 10 \text{Mbps} \end{aligned}}$$

Throughput

The throughput is a measure of how fast we can send data through a network. Although, at first glance, bandwidth in bits per second and throughput seem the same, they are different. A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B.

Latency

The latency or delay defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source.

We can say that latency is made of four components: propagation time, transmission time, queuing time and processing delay.

Latency = propagation time + transmission time + queuing time + processing delay

Example

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

Solution

We can calculate the throughput as

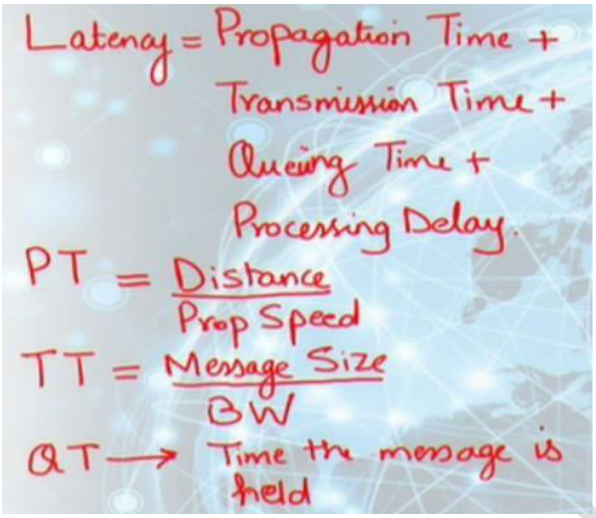
Throughput = (12,000X10,000)/60 = 2Mbps

The throughput is almost one-fifth of the bandwidth in this case.

Lecture # 52

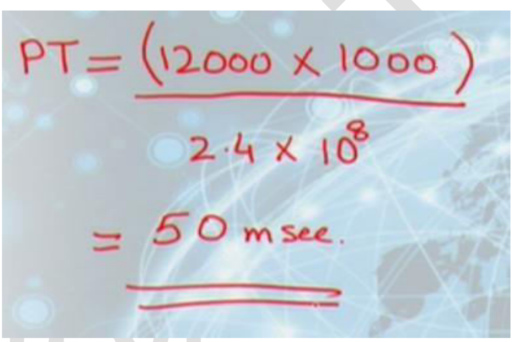
Latency or Delay

Latency or delay defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source



Example

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be 2.4×10^8 m/s in cable.



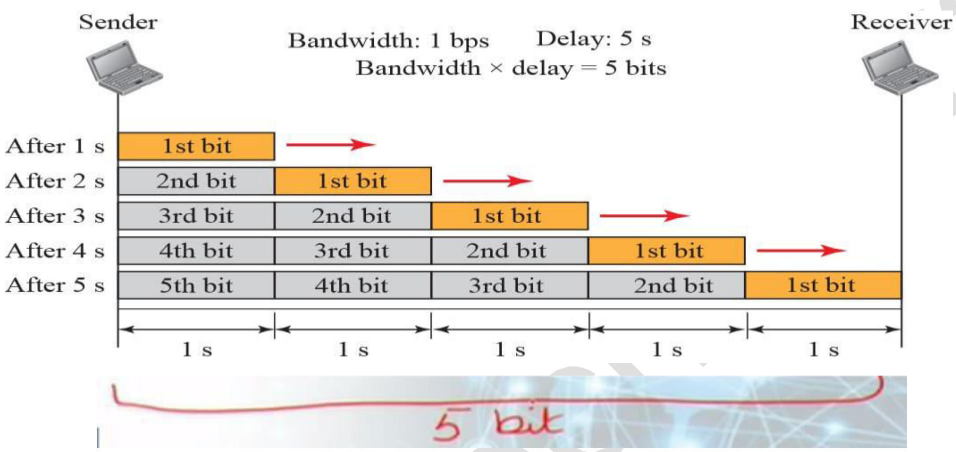
Lecture # 53

Delay – Bandwidth Delay Product

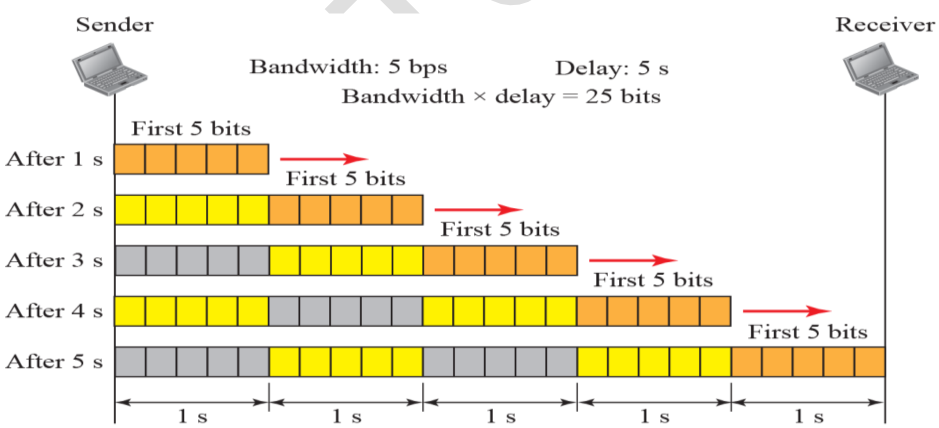
- Bandwidth and delay are two performance metrics of a link
- Product of the two, The Bandwidth-Delay Product defines the number of bits that can fill a link

Bandwidth-Delay Product

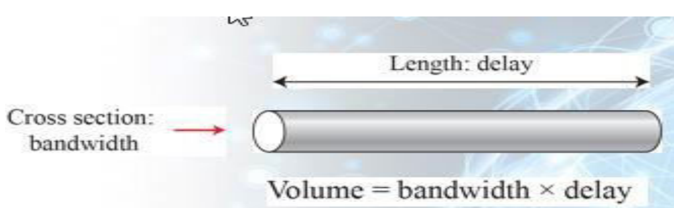
Case 1



Case 2



Bandwidth-Delay Product



We can think about the link between two points as a pipe. The cross section of the pipe represents the bandwidth, and the length of the pipe represents the delay.

We can say the volume of the pipe defines the bandwidth-delay product, as shown above.

Delay - Jitter

Jitter is a problem if different packets of data encounter different delays and the application using the data at the receiver site is time-sensitive (audio and video data, for example)

Lecture # 54

When it comes to digital communication, one important process is digital-to-digital conversion. This process involves converting data from one digital format to another. Understanding this conversion process is crucial for effectively transmitting digital information across different communication channels.
There are various methods for converting digital signals to digital format:

- **Line coding** is a crucial process in which binary data is transformed into digital signals, enabling smooth transmission over a communication channel. It guarantees seamless coordination between the sender and receiver.

- **Block coding** involves the grouping of a set of bits and encoding them into a specific code. This method improves the integrity of the data and allows for the detection and correction of errors.

- **Scrambling** is a valuable technique that helps to enhance the efficiency of data transmission by randomising the data. By doing so, it minimises the chances of having long sequences of identical bits and optimises the use of the communication channel.

Let's discuss the data and signal elements:

The smallest unit of data representation is a bit. The value can be either 0 or 1, serving as the foundation for digital information processing and storage.

In digital communication, a signal element refers to the smallest unit of a digital signal. It represents a particular voltage level or phase change and is utilized to transmit data over a communication channel.

Lecture # 55

Understanding data and signal rates is essential for optimising efficiency and speed in digital communication.

The **data rate**, which is measured in bits per second (bps), represents the quantity of data elements that are transmitted within a one-second interval. Understanding digital information interchange requires a grasp of the fundamental building blocks: bits represented by 0s and 1s. A higher data rate enables faster communication and data transfer by transmitting a larger amount of data in a shorter period of time.

The **signal rate**, measured in baud, refers to the quantity of signal elements that are transmitted within a single second. Signal elements consist of pulses, phases, and modulated signals. Signal rate is primarily concerned with the physical alterations in the signal that transmits the data, rather than the actual content being transmitted. The signal rate is also known as the pulse or modulation rate.

Lecture # 56

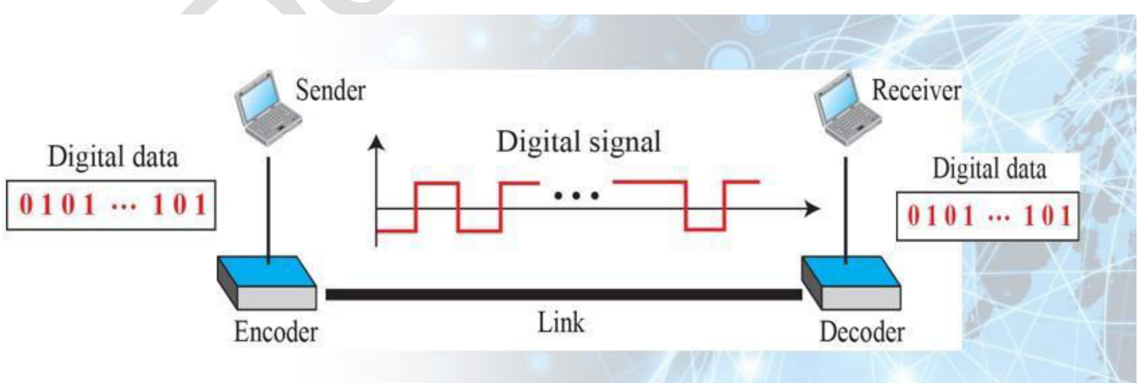
Line coding converts digital data into digital signals, a key digital communication procedure. Computer memory stores text, numbers, graphics, music, and video as sequences of bits (0s and 1s). Digital information is built from these bits.

Line coding ensures efficient and reliable data transfer over communication lines.

Line coding schemes:

Following are the 5-line coding schemes:

- Unipolar
- Polar
- Bipolar
- Multilevel
- Multitransition



Lecture # 57

Line coding schemes:

Following are the 5-line coding schemes:

- Unipolar
- Polar
- Bipolar
- Multilevel
- Multitransition

Unipolar Scheme:

Unipolar line coding involves placing all digital signal levels on one side of the time axis, above or below. In binary, 1s are represented by a positive or non-zero voltage level, while 0s are zero. Avoiding signal time axis crossing simplifies signal detection and synchronisation.

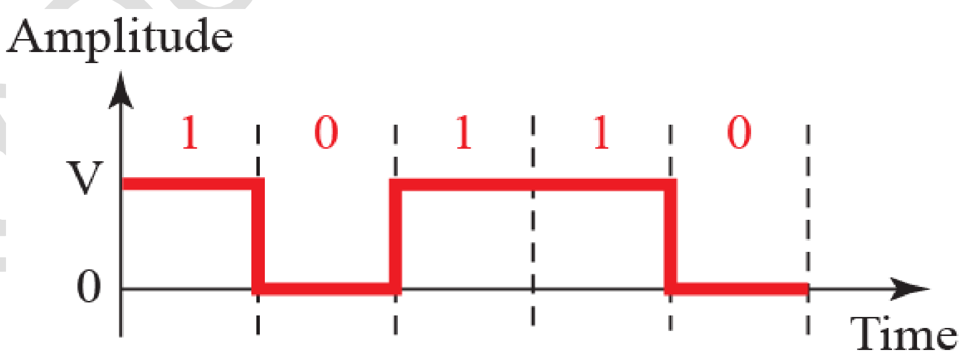
For unipolar line coding:

- The positive voltage (+V) is represented by binary 1.
- Binary 0 is zero voltage.

Simple unipolar line coding is easy to implement and decode. Even with several zeros in the data, the constant positive signal reduces signal power efficiency and transmits power consistently.

Unipolar line coding is employed in short-distance communication systems since it is easy to build and decode and power efficiency is less important.

In Unipolar NRZ, a binary 1 has a constant voltage level throughout the bit period, while a binary 0 has no voltage. Unipolar NRZ retains voltage signal polarity for all binary values, unlike other NRZ methods. However, it only uses positive voltage for binary 1 and zero voltage for binary 0.



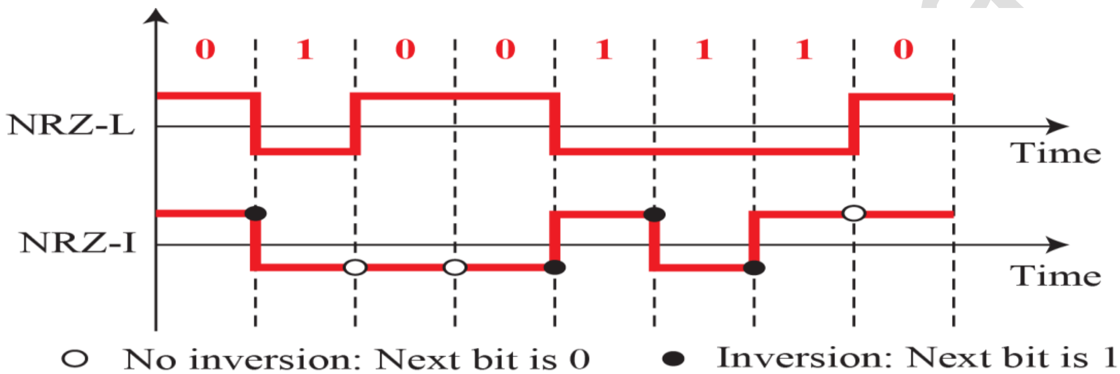
Lecture # 58

Polar-NRZ Line Coding:

Binary 1 has a constant voltage level, positive or negative, during the bit period.

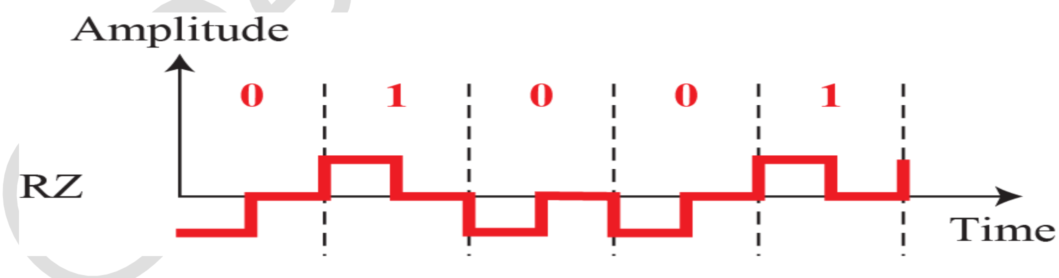
The voltage level of 0 in binary is opposite (negative or positive) throughout the bit period.

In Polar NRZ, the voltage level for a binary 1 remains constant during the bit period, while the voltage level for a binary 0 remains constant but reversed. The signal is not reset to zero each bit period in this technique. Instead, it keeps the voltage constant throughout each bit.



The Polar RZ Line Coding Scheme:

In binary, 1 is represented by a positive voltage level in the first half of the bit period and zero value in the second half.
The binary number 0 is represented by a negative voltage level for half the bit period and zero for the other half.
Polar RZ signals always revert to zero voltage (or baseline) midway through each bit period, independent of bit value. This improves synchronisation and reduces issues caused by long sequences of identical bits, a common concern in non-return-to-zero methods.



Polar NRZ and Polar RZ were separate polar line coding methods. In Polar NRZ, the voltage level remains constant during the bit period, whether it's 1 or 0. However, Polar RZ voltage returns to zero halfway through each bit cycle. Consider communication requirements like bandwidth efficiency and synchronisation while choosing between these systems.

Lecture # 59

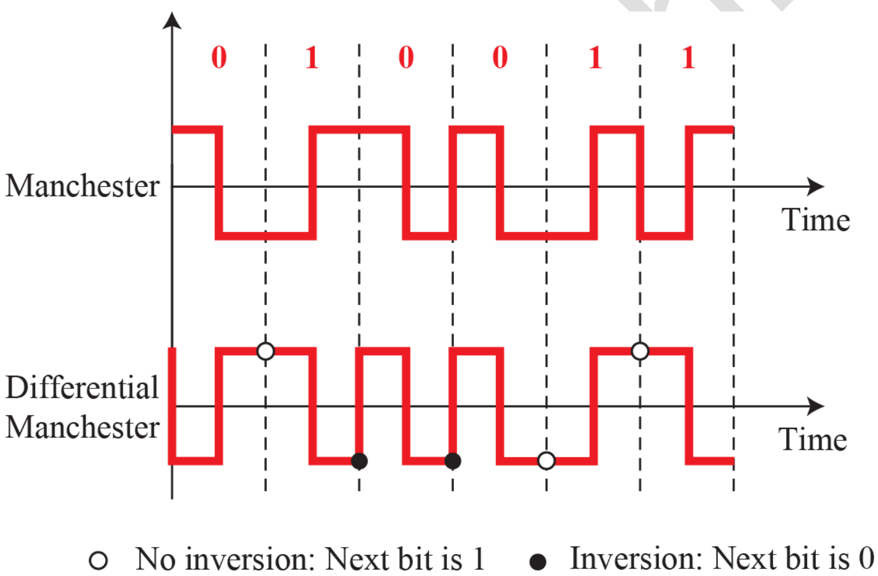
Polar Biphase, also known as Manchester encoding, is a line coding scheme commonly employed in digital communication. Within this scheme, every bit period is split into two equal intervals. In each bit period, the polarity of the voltage signal switches back and forth, allowing for regular transitions that help the sender and receiver stay in sync.

When considering Polar Biphase:

Binary 1 is indicated by a change from a positive voltage level to a negative voltage level (or vice versa) during the bit period. This transition is commonly known as a 'Manchester transition' or 'Manchester violation.'

In binary, a 0 is represented when there is no transition within the bit period, which can happen at the beginning or end of the period.

Polar Biphase encoding possesses the convenient feature of being self-clocking. This means that the receiver can extract the clock signal directly from the received data stream, thanks to the guaranteed transition in the middle of each bit period. This property is highly advantageous in preventing synchronisation errors, making it a popular choice for Ethernet LANs and other digital communication systems that require precise clock recovery.



Lecture # 60

AMI/Pseudoternary Bipolar Schemes:

AMI uses zero voltage for binary 0 and positive and negative voltage for binary 1. AMI's 1s alternate polarity is unique. Code for "1101" is "+0-+". AMI is used in T1 and E1 lines for data transmission and synchronization.

Digital communication systems encode data using Pseudoternary line coding. Balanced line codes ensure equal positive and negative voltage levels. Telecoms and networks utilize this coding strategy to send binary data over a channel. It reduces errors and ensures data transmission and is dependable and efficient.

Binary 1 has a continuous zero voltage in Pseudoternary, while binary 0 alternates between positive and negative. AMI has alternating polarities, but Pseudoternary has no voltage change for binary 1s. This encoding method is used in older telecommunication systems to maintain a balanced distribution of positive and negative voltage levels for long-distance transmission.

Communication applications often use bipolar systems like AMI and Pseudoternary. These systems encode binary data with positive, negative, and zero voltage.

Lecture # 61

Block coding converts one amount of bits into more. Often used in digital signal encoding. In mB/nB encoding,'m' represents the number of bits in the original data block and 'n' represents the encoded block, which is usually larger.

Reason for Block Coding:

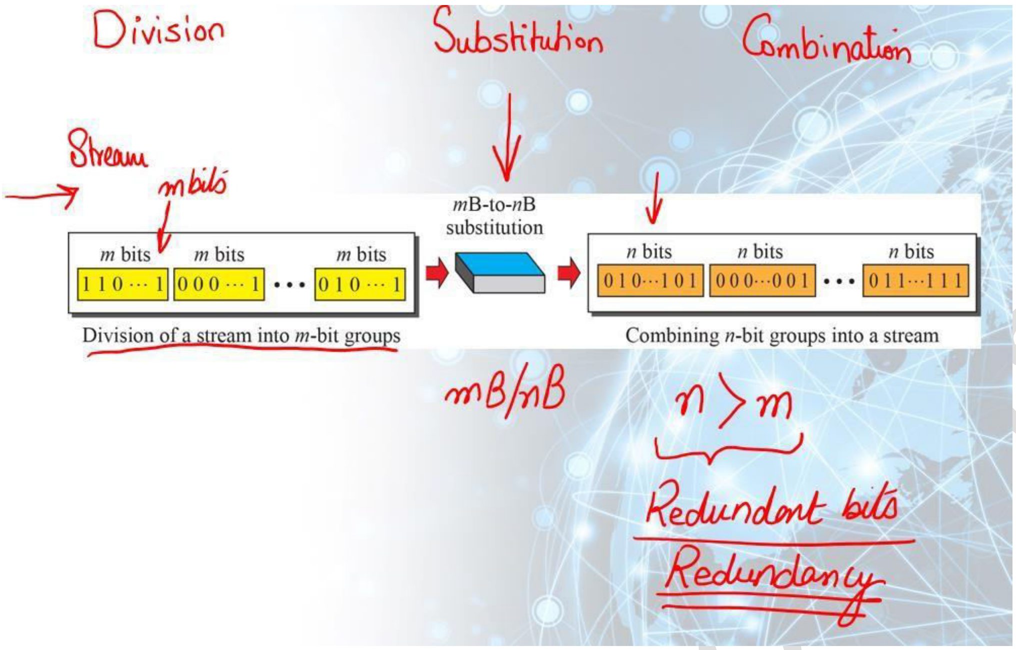
Block coding adds signal redundancy, improving signal reliability. Redundancy adds data. Additional components aid receiver synchronization, error identification, and correction.

Synchronization requires transmitter and receiver alignment at the start and end of each data block. This minimizes errors and streamlines data flow. Synchronization is needed to decode the signal and recover the data.

Benefits of Block Coding:

Block coding includes redundancy to improve data transmission reliability. Redundancy detects and fixes transmission problems to maintain data integrity.

Efficient Synchronization: Redundant bits help sender-receiver synchronization. Accurate signal decoding requires correct synchronization.



Lecture # 62

Using block coding 4B/5B with NRZ-I line coding

In digital communication systems, the utilisation of 4B/5B block coding alongside NRZ-I line coding is a widely employed technique. This approach is particularly prevalent in high-speed networks and interfaces like Gigabit Ethernet and Fibre Channel. Now, let's analyse the meaning behind this combination:

In **4B/5B block coding**, groups of 4 bits are assigned to distinct 5-bit code words. There are additional benefits to using this encoding method compared to regular NRZ encoding. It helps maintain a balance between 1s and 0s and also offers error detection capabilities. This scheme enables a wider variety of patterns compared to simple NRZ encoding, which enhances the reliability of data transmission.

NRZ-I Line Coding: NRZ-I, also known as Non-Return-to-Zero Inverted, is a line coding technique that utilises transitions to represent binary 1s and the absence of transitions to represent binary 0s. NRZ-I operates in a self-clocking manner, eliminating the need for an external clock signal to ensure synchronisation. On the contrary, it utilises the transitions within the data itself to ensure synchronisation between the sender and receiver.

4B/5B mapping codes
(**FYI:** Table on next page.)

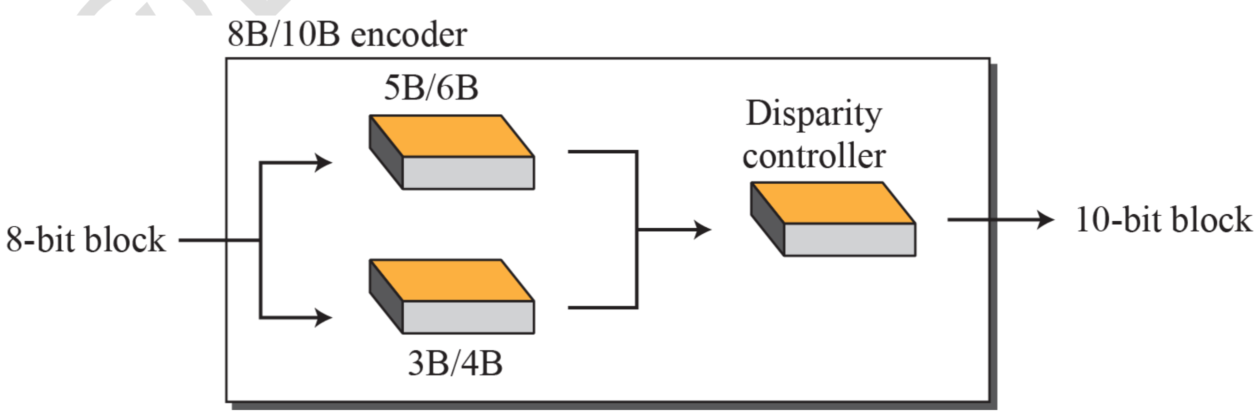
Data Sequence	Encoded Sequence	Control Sequence	Encoded Sequence
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

Lecture # 63

Block coding converts one amount of bits into more. Often used in digital signal encoding. In mB/nB encoding,'m' represents the number of bits in the original data block and 'n' represents the encoded block, which is usually larger.

8B/10B block encoding is a technique used in data transmission to ensure reliable and efficient communication.

8B/10B is a block encoding scheme commonly utilised in high-speed digital communication to optimise data transmission and maintain synchronisation and error detection. In this scheme, 8 bits of data are assigned to distinct 10-bit code words. This mapping ensures a balanced distribution of 1s and 0s, which facilitates synchronisation and enhances error-detection capabilities.



Lecture # 64

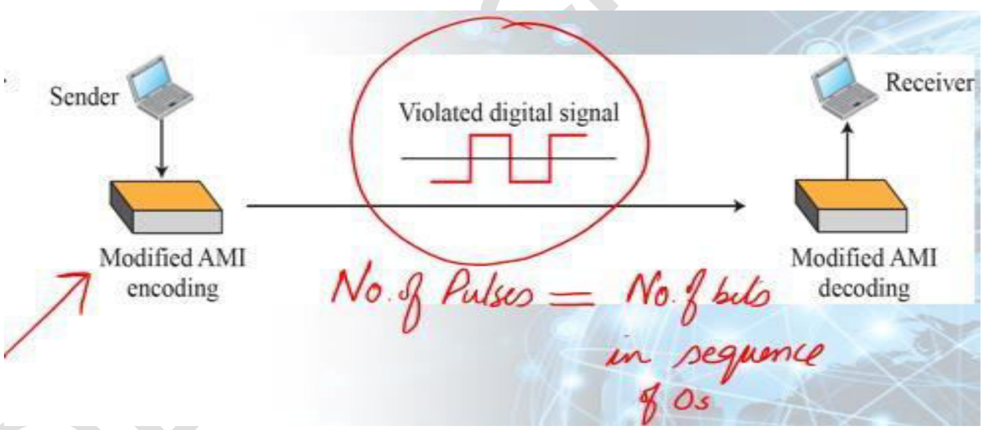
Scrambling

Scrambling is a technique used in data communication to improve the reliability of data transmission and to ensure that the data can be accurately recovered at the receiving end.

- **B8ZS and HDB3** are methods of scrambling data to make sure it's sent in a way that prevents synchronization and data integrity issues, especially over long-distance communication lines.
- **Biphase schemes**, such as Manchester encoding, are often used for LAN (Local Area Network) communication due to their advantages in terms of synchronization and low error rates.
- **Biphase schemes** suitable for LAN but not for Long Distance
- **Block Coding + NRZ-I** solves synch issue but has DC component.
- **Bipolar AMI** has a narrow bandwidth (no DC Component) but synch issue (long series of 0s)
- The system needs to insert the required pulses based on the defined scrambling rules.

AMI is used with scrambling.

- In AMI encoding, a 0-bit is represented alternately by a positive voltage (mark) and a negative voltage (space), while a 1-bit is represented by no voltage change (zero).
- This alternate marking of 0s ensures that there are frequent voltage transitions, which is important for clock recovery and synchronization.



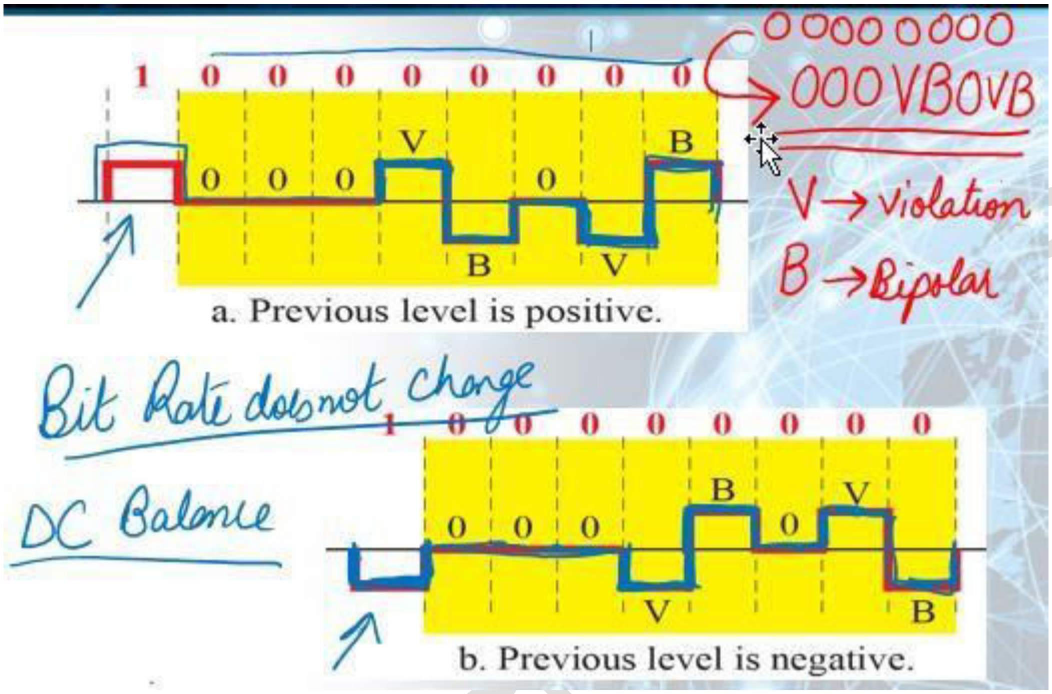
Types of Scrambling Techniques
Two common scrambling techniques are B8ZS and HDB3

Bipolar with 8-Zero Substitution (B8ZS): B8ZS is a specific method of scrambling. When you have a long string of consecutive zeros in your data, B8ZS replaces these with a special code to keep the data balanced and maintain timing. This substitution helps ensure that there are enough transitions in the data to keep everything synchronized during transmission.

High-density bipolar 3-zero (HDB3): also focuses on keeping the data balanced and ensuring that there are enough transitions. When there are four consecutive zeros in the data, HDB3 replaces them with a unique pattern to maintain the balance of ones and zeros in the data, which is crucial for reliable communication over long distances.

Lecture # 65

Two cases of B8ZS scrambling technique



- In this technique, eight consecutive zero-level voltages are replaced by the sequence **000VB0VB**.
- **The V in the sequence denotes violation:** this is a nonzero voltage that breaks an AMI rule of encoding (opposite polarity from the previous).
- **The B in the sequence denotes bipolar**, which means a nonzero level voltage in accordance with the AMI rule.

Types of Scrambling Techniques

Two common scrambling techniques are B8ZS and HDB3

Bipolar with 8-Zero Substitution (B8ZS): B8ZS is a specific method of scrambling. When you have a long string of consecutive zeros in your data, B8ZS replaces these with a special code to keep the data balanced and maintain timing. This substitution helps ensure that there are enough transitions in the data to keep everything synchronized during transmission.

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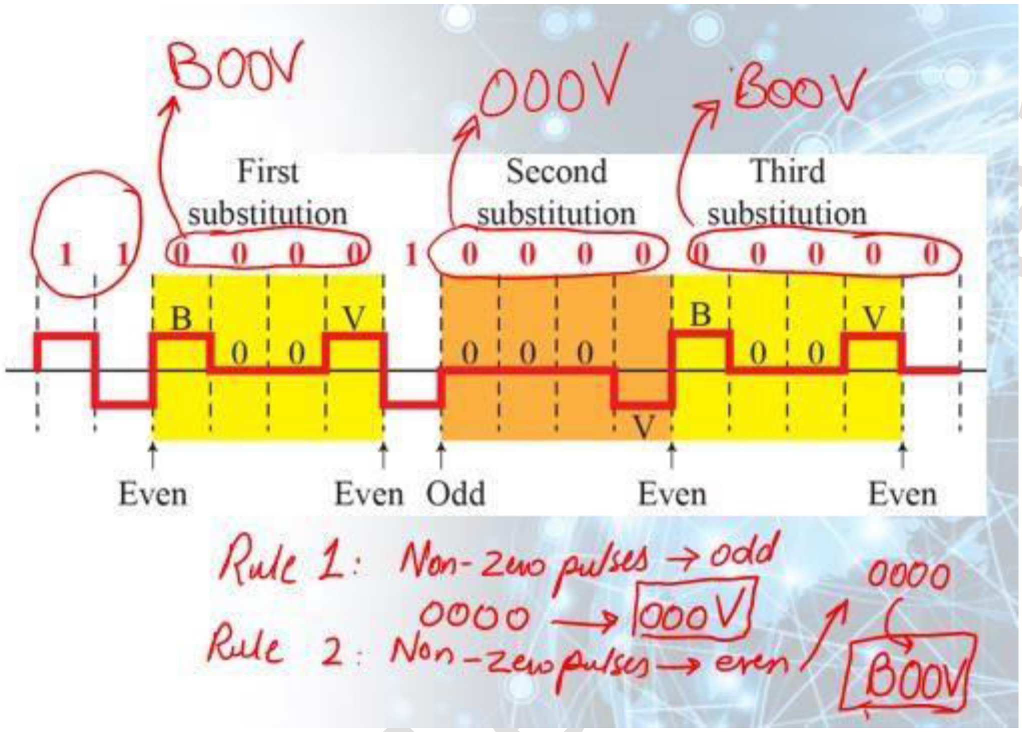
Lecture # 66

Different situations in HDB3 scrambling technique.

In HDB3 four consecutive zero-level voltages are replaced with a sequence of 000V or B00V.



- If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be 000V, which makes the total number of nonzero pulses even.
- If the number of nonzero pulses after the last substitution is even, the substitution pattern will be B00V, which makes the total number of nonzero pulses even.
- The V in the sequence denotes violation; this is a nonzero voltage that breaks an AMI rule of encoding.
- The B in the sequence denotes bipolar, which means a nonzero level voltage in accordance with the AMI rule.

Lecture # 67

Analog-to-digital Conversion

- **Analog Data to Digital Data:** Analog data to digital data conversion involves sampling the analog signal, quantizing the sampled values into discrete levels, encoding these values into binary form, and then using digital systems to process, store, or transmit the digital representation of the original analog information.
- **Process of Digitization:** The process of digitization involves converting analog information into digital data, making it easier to store, process, and transmit.

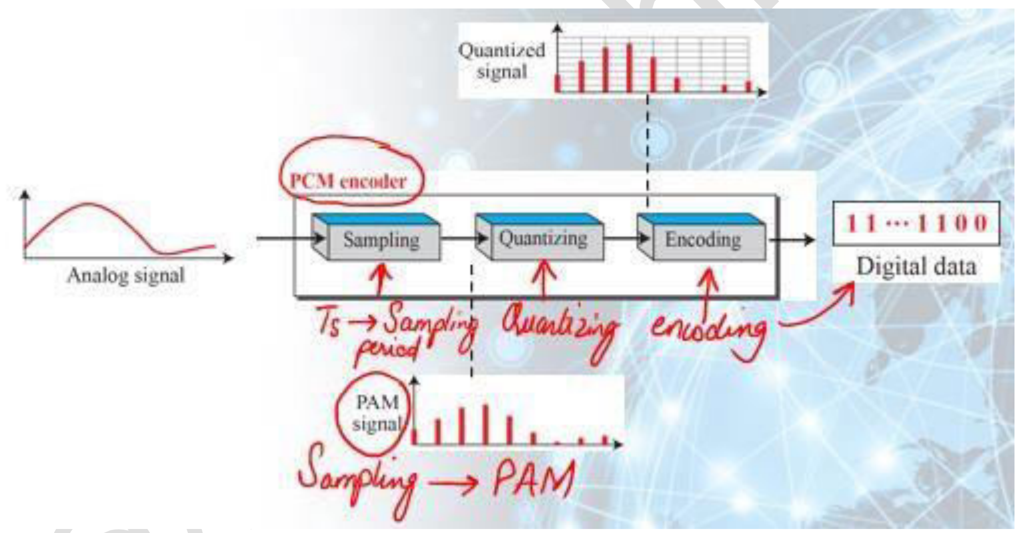
Pulse Code Modulation (PCM)

Pulse Code Modulation (PCM) is a widely used method for digitally representing analog signals, such as audio or video, in a format that can be easily processed and transmitted by digital systems.

Delta Modulation (DM)

Delta Modulation (DM) is a simple method used to digitally encode analog signals, typically for voice or low-quality audio transmission.

Components of PCM encoder



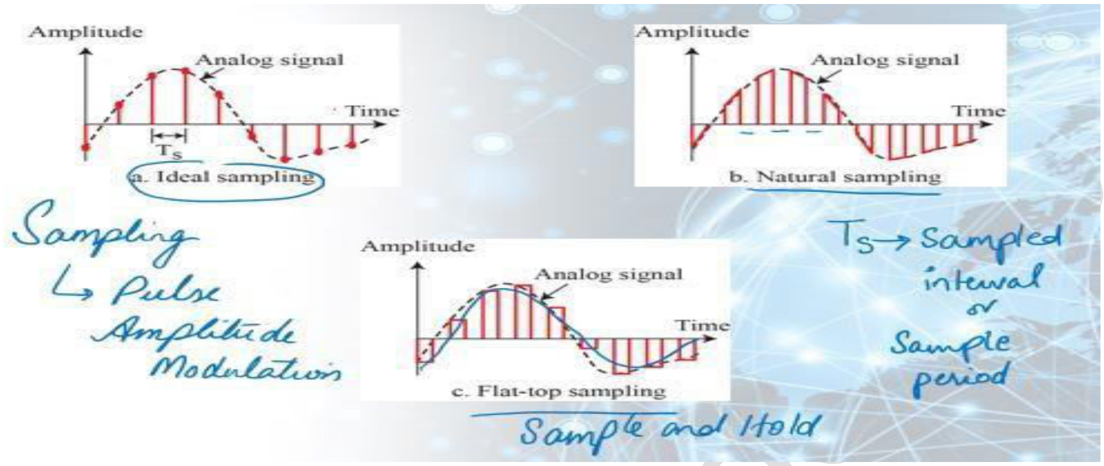
Pulse Code Modulation (PCM)

Pulse Code Modulation (PCM) is a widely used method for digitally representing analog signals, such as audio or video, in a format that can be easily processed and transmitted by digital systems.

- **Sampling:** The first step in PCM is to take samples of the analog signal at regular intervals.
- **Quantization:** Each of the sampled values is then quantized, which means assigning a numerical value to represent the amplitude of the sample.
- **Encoding:** The quantized values are encoded into a digital format, typically using binary code (0s and 1s)

Lecture # 68

Three different sampling methods for PCM



Ideal Sampling:

- Often referred to as Nyquist sampling, is a theoretical concept in signal processing.
- According to the Nyquist-Shannon theorem, to accurately represent an analog signal in digital form, you should sample it at a rate that is at least twice the signal's highest frequency (the Nyquist rate).

Natural Sampling:

- Also known as impulse or instantaneous sampling, is a practical approach to capturing analog signals.
- In natural sampling, the analog signal is sampled at specific points in time when an impulse or a brief sampling pulse occurs.

Flat-Top Sampling:

- Flat-top sampling is a modification of natural sampling that helps reduce the effects of quantization noise in analog-to-digital conversion.
- In flat-top sampling, the sampling pulse has a finite duration (flat top) rather than being instantaneous.

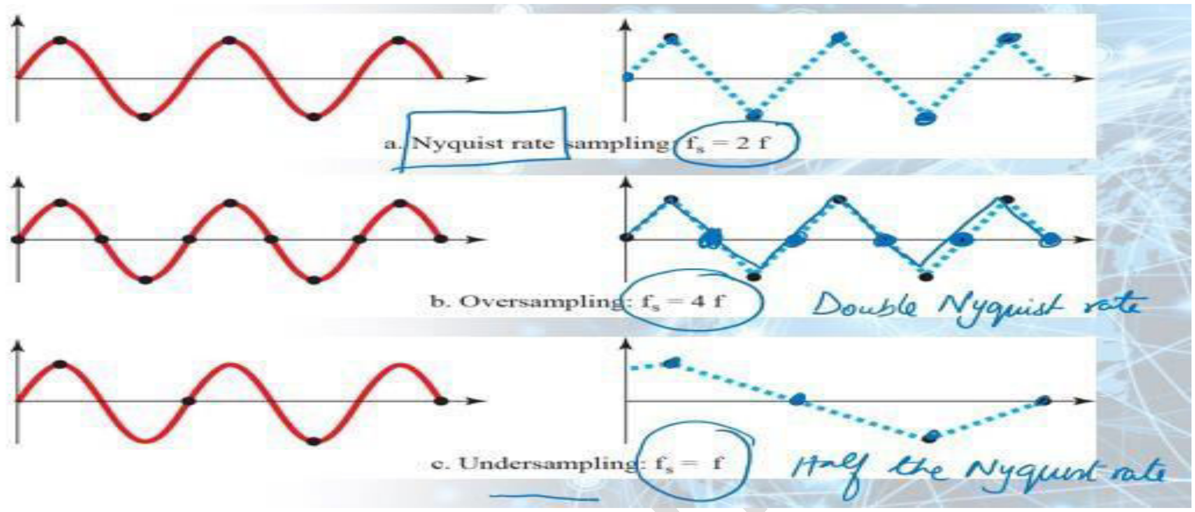
Nyquist Sampling Rate
Nyquist's theorem states that to accurately capture an analog signal, the sampling rate should be at least twice the signal's maximum frequency (Nyquist rate).

$$\boldsymbol{Nyquist} \text{ à } \boldsymbol{fs = 2fh}$$

Sampling Period (Ts):
The analog signal is sampled every Ts s, where **Ts is the sample interval or period.**

Sampling rate or sampling frequency:
The inverse of the sampling interval is called the sampling rate or sampling frequency and denoted by fs, where fs = 1/Ts.

Nyquist Sampling Rate



Sampling sine wave at three sampling rates:

f_s = 4f (2 times the Nyquist rate)

- Oversampling involves taking more samples of a signal than the minimum required by the Nyquist-Shannon theorem.
- By oversampling, you collect more data points per unit of time, which can improve the accuracy and reliability of the captured information.

f_s = 2f (Nyquist rate)

- Nyquist's theorem states that to accurately capture an analog signal, the sampling rate should be at least twice the signal's maximum frequency (Nyquist rate).

fs = f (one-half the Nyquist rate)

- Under sampling, on the other hand, involves sampling a signal at a rate lower than the Nyquist rate.
- Under sampling intentionally captures fewer data points to reduce data volume or handle multiple signals within a limited bandwidth.

Lecture # 69

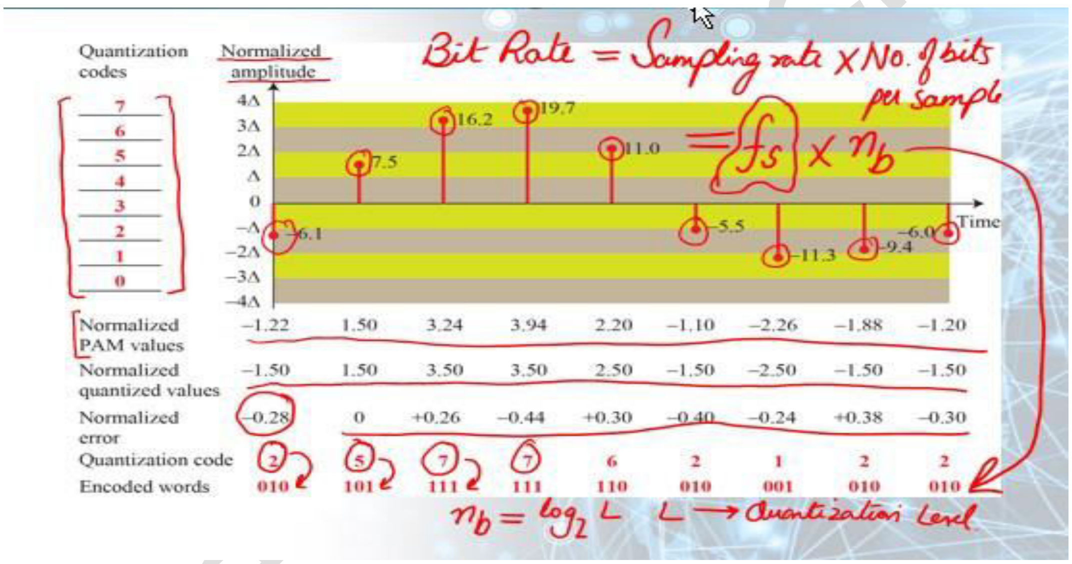
Quantization & encoding of a sampled signal.

- **Sampling** → Series of pulses with amplitude values between min and max signal amplitude
- Infinite set with non-integral values not suitable for encoding
- We quantize the sampling output into certain levels based on range of amplitudes and how much accuracy is needed.

Quantization

After sampling, each of the sampled values is then quantized, which means assigning a numerical value to represent the amplitude of the sample.

Quantization & encoding of a sampled signal.



- After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an **n_b-bit code word. In the above Figure,** A quantization code of 2 is encoded as 010; 5 is encoded as 101 and so on.
- The number of **bits for each sample** is determined from the **number of quantization levels**.
- If the number of **quantization levels is L**, the number of bits is **n_b = log2 L.**
- In the above example L is 8 and n_b is therefore 3.
- Bit rate = sampling rate x number of bits per sample = fs x n_b

Example:

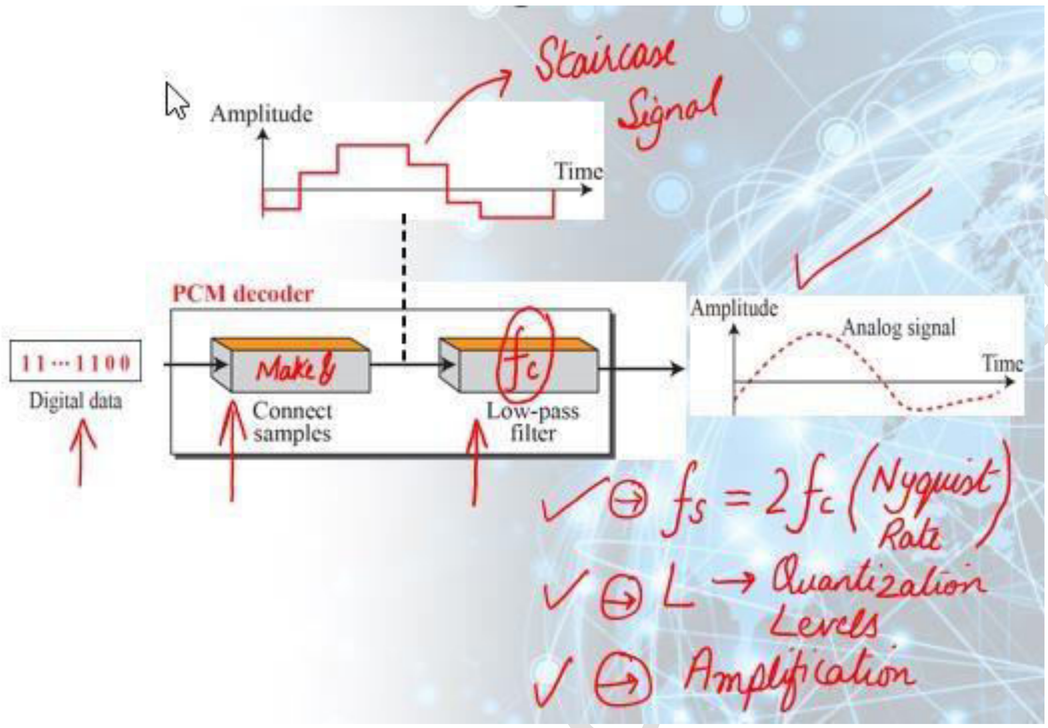
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows

Sampling rate = 4000 × 2 = 8000 samples/s

Bit rate = 8000 × 8 = 64,000 bps = 64 kbps

Lecture # 70

Original Signal Recovery- PCM Decoder



- The recovery of the original signal requires the **PCM decoder**.
- The decoder first uses **circuitry** to convert the code words into a pulse that holds the amplitude until the next pulse.
- After the **staircase signal** is completed, it is passed through a **low-pass filter** to smooth the staircase signal into an **analog signal**.
- The filter has the same **cutoff frequency** as the original signal at the sender.
- If the signal has been sampled at (or greater than) the **Nyquist sampling rate** and if there are enough **quantization levels**, the original signal will be recreated.
- The maximum and minimum values of the original signal can be achieved by using **amplification**.

Lecture # 71

Delta Modulation (DM)

- **PCM** is an extraordinarily complex technique.
- **Delta modulation** is a simpler technique.
- **PCM** finds the value of the signal amplitude for each sample; **DM** finds the change from the previous sample.
- No code words in **delta modulation**; bits are sent one after another.

Modulator:
The modulator is used at the sender site to create a stream of bits from an analog signal.

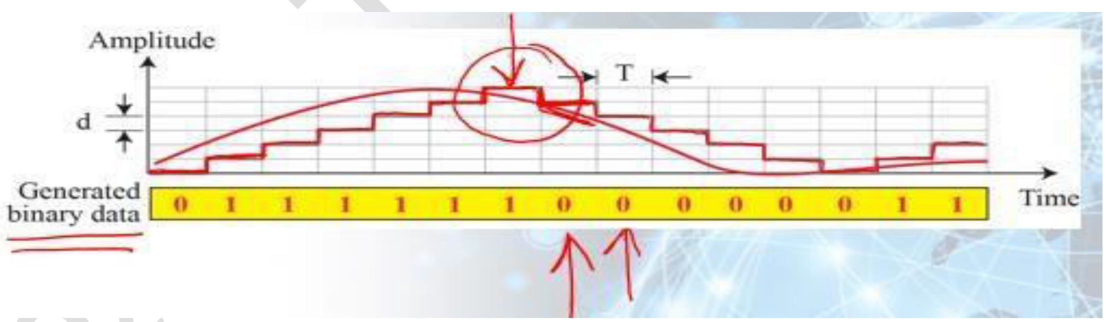
Delta δ:

- The process records the small positive or negative changes, called delta δ.
- If the delta is positive, the process records a 1
- If it is negative, the process records a 0.

Staircase:

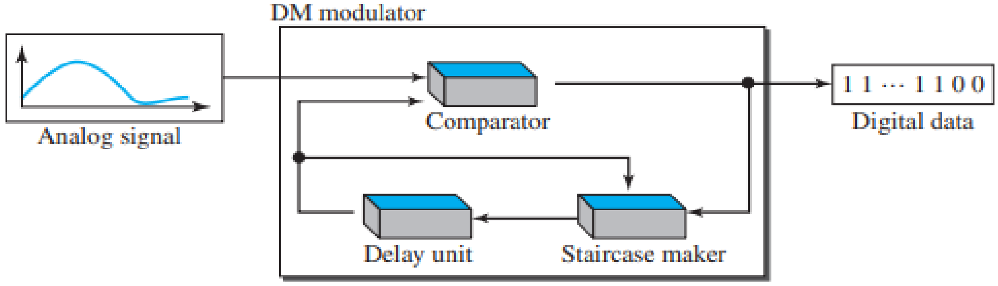
- The process also needs a base against which the analog signal is compared.
- The modulator builds a second signal that resembles a staircase.
- Finding the change is then reduced to comparing the input signal with the gradually made staircase signal.

The process of delta modulation



The process of delta modulation

- The **modulator**, at each sampling interval, compares the value of the analog signal with the last value of the **staircase signal**. If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is 0.
- The output of the **comparator**, however, also makes the **staircase** itself. If the next bit is 1, the staircase maker moves the last point of the staircase signal δ up; if the next bit is 0, it moves it δ down.
- Note that we need a **delay unit** to hold the staircase function for a period between two comparisons.

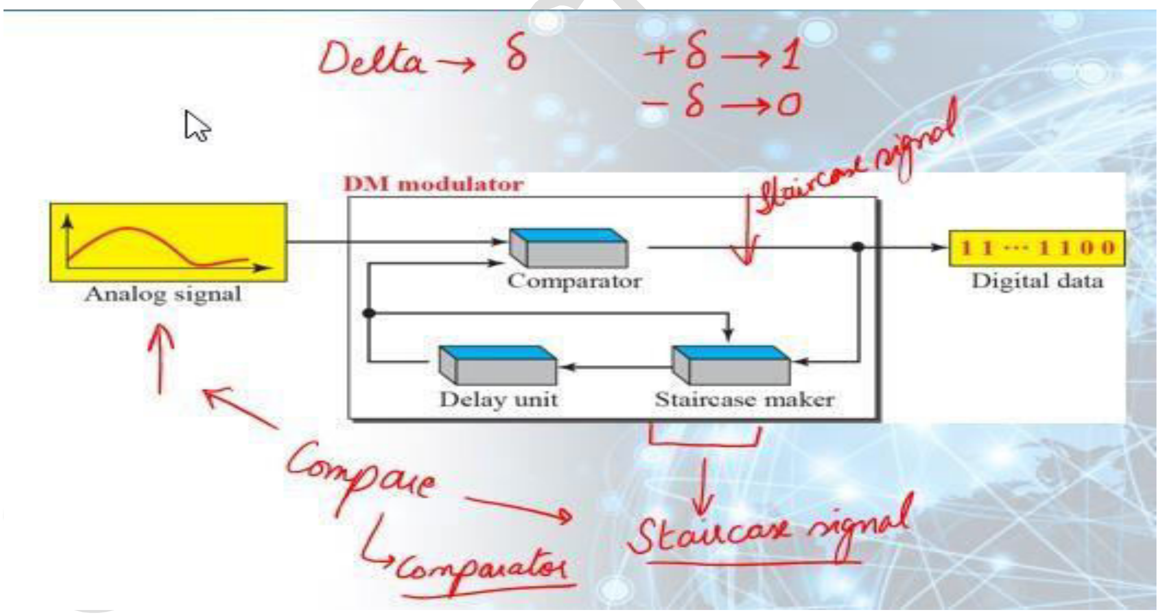


Lecture # 72

Delta Modulation (DM)

- **Delta modulation** is a simpler technique.
- **DM** finds the change from the previous sample.
- No code words in **delta modulation**; bits are sent one after another.

Delta Modulation Components

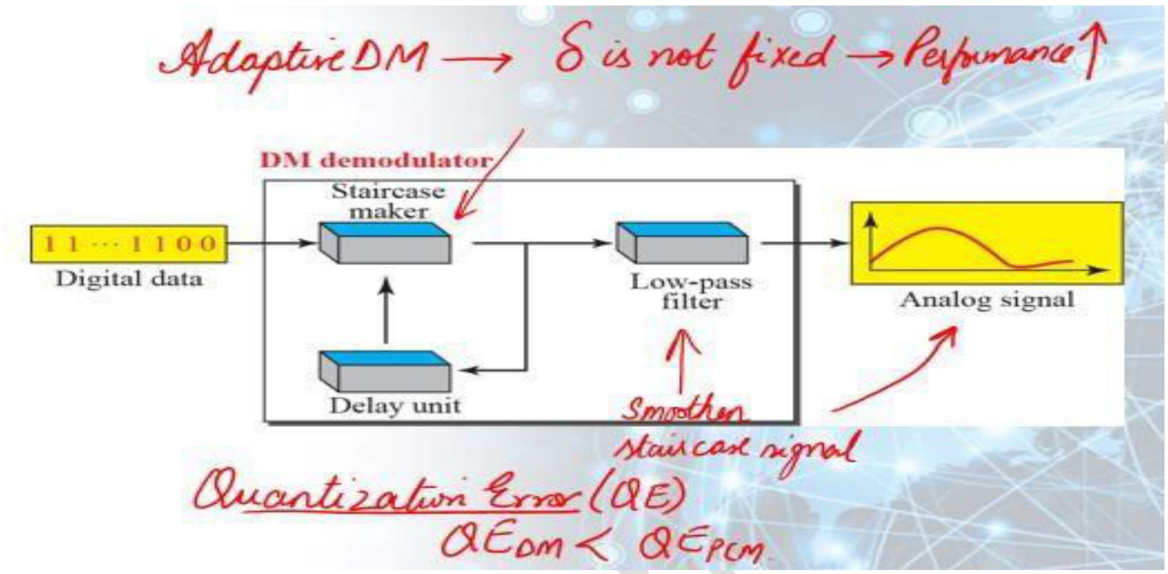


Delta Modulation Components

- The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal.
- If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is 0.
- The output of the comparator, however, also makes the staircase itself.

- If the next bit is 1, the staircase maker moves the last point of the staircase signal δ up.
- If the next bit is 0, it moves it δ down.
- We need a delay unit to hold the staircase function for a period between two comparisons.

Delta De modulation Components



Delta De modulation Components

- The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal.
- The created analog signal, however, needs to pass through a low-pass filter for smoothing.

Adaptive DM

- A better performance can be achieved if the value of δ is not fixed.
- In adaptive delta modulation, the value of δ changes according to the amplitude of the analog signal.

Lecture # 73

Wiring: when we are considering the wiring is the data stream.

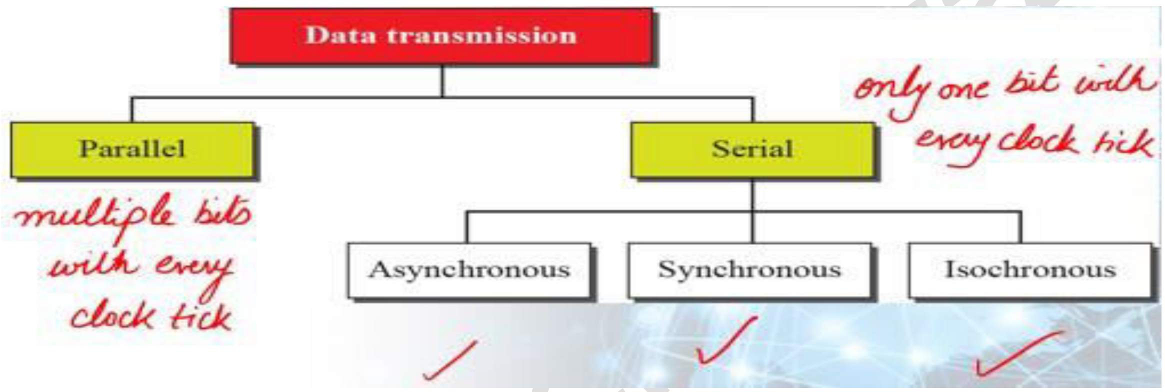
Data Stream: Do we send 1 bit at a time; or do we group bits into larger groups and, if so, how?

Transmission Modes: Parallel or Serial Transmission

In **parallel mode**, multiple bits are sent with each clock tick.

In **serial mode**, 1 bit is sent with each clock tick.

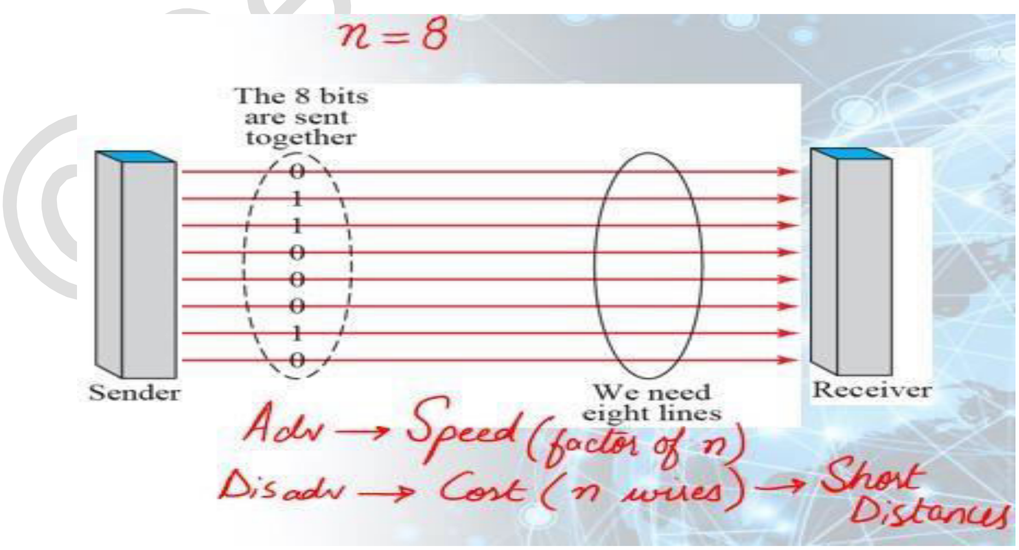
Data transmission modes:



Parallel Transmission

- Binary data (1s ad 0s) organized in groups of 'n' bits.
- We send 'n' bits at a time instead of just one.
- 'n' wires required to send 'n' bits at one time.

Parallel Transmission



- Computers produce and consume data in groups of bits. By grouping, we can send data n bits at a time instead of 1. This is called parallel transmission.

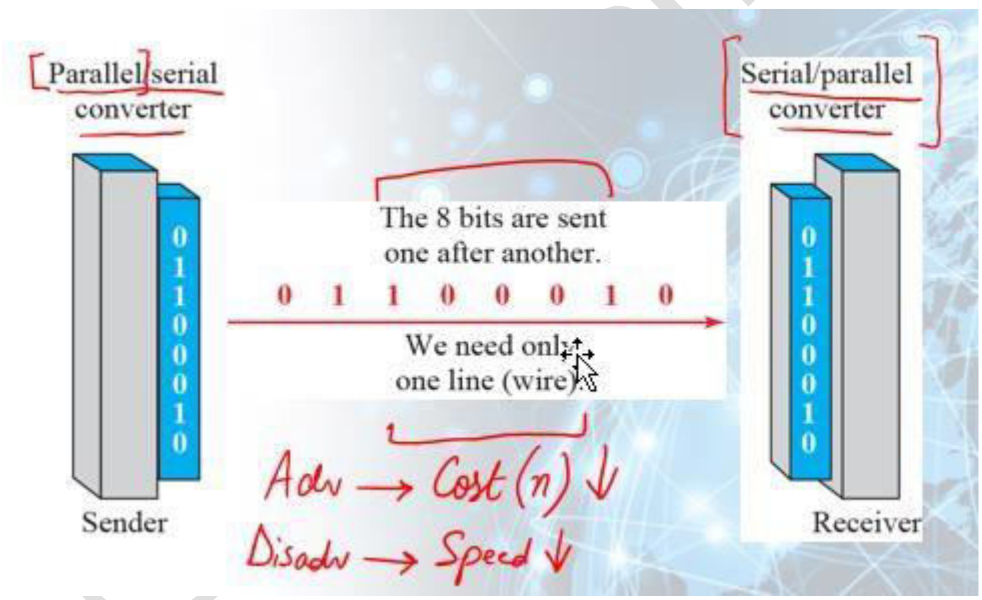
- In parallel transmission, we use n wires to send n bits at one time. That way each bit has its own wire, and all n bits of one group can be transmitted with each clock tick from one device to another.
- In the above figure, we can see that how parallel transmission works for n = 8.
- Typically, the eight wires are bundled in a cable with a connector at each end.

Advantage:
Speed: Parallel transmission can increase the transfer speed by a factor of n over serial transmission.

Disadvantage:
Cost: Parallel transmission requires n communication lines (wires in the example) just to transmit the data stream. It is expensive so parallel transmission is usually limited to short distances.

Lecture # 74

Serial Transmission



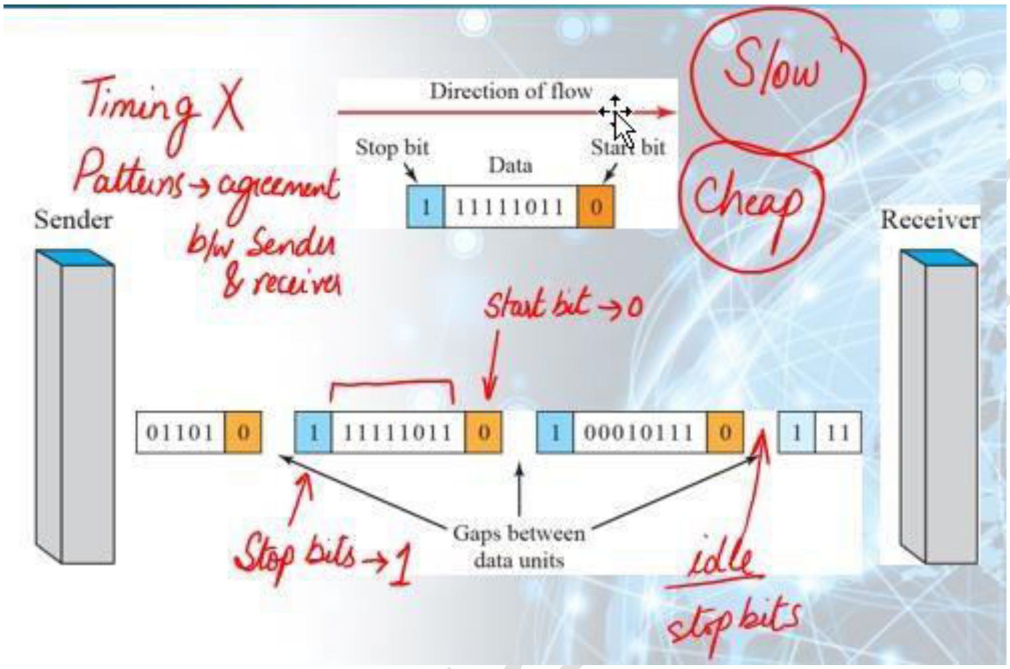
In serial transmission one bit follows another, so we need only one communication channel rather than n to transmit data between two communicating devices. It also reduces the cost of transmission over parallel by roughly a factor of n.

Since communication within devices is parallel, conversion devices are needed at the interface between the sender and the line that is performing parallel-to-serial conversion and between the line and the receiver which is conducting serial-to-parallel conversion.

Serial transmission occurs in one of three ways: **asynchronous, synchronous, and isochronous**.

Asynchronous Transmission

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.



Asynchronous transmission is so named because the timing of a signal is unimportant. Information is received and translated by agreed upon patterns. If those patterns are followed, the receiving device can retrieve the information without regard to the rhythm in which it is sent.

Start Bit: An extra bit is added to the beginning of each byte to alert the receiver to the arrival of a new group. This bit, usually a 0, is called the **start bit**.

Stop Bit: To let the receiver know that the byte is finished, 1 or more additional bits are appended to the end of the byte. These bits, usually 1s, are called stop bits.

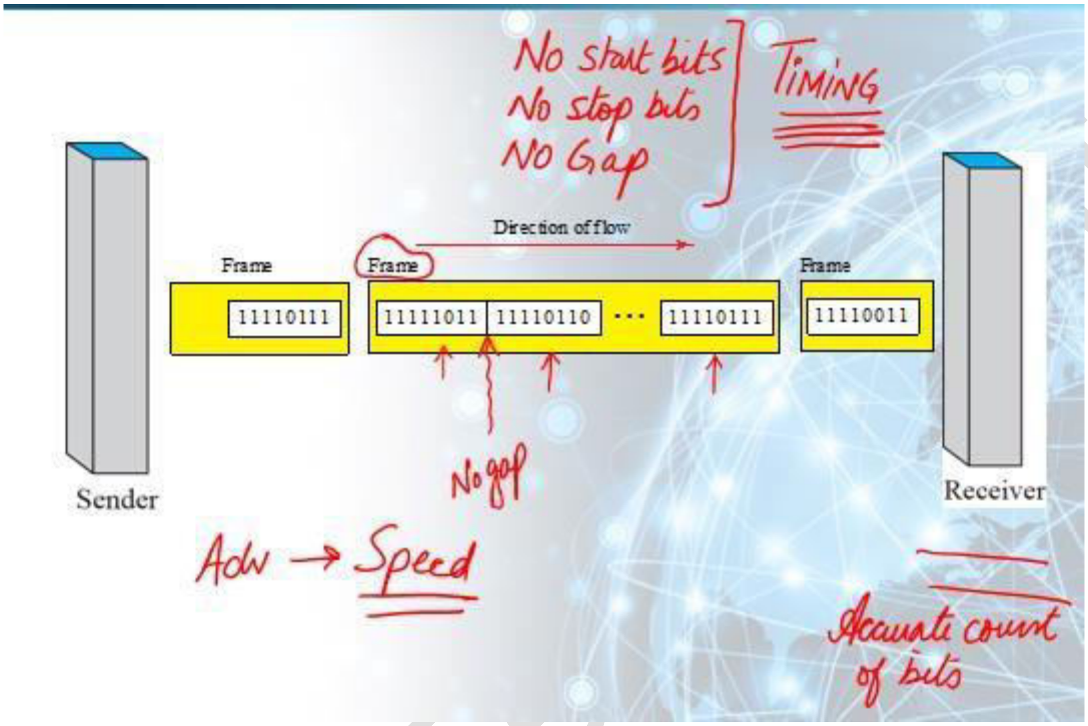
By this method, each byte is increased in size to at least 10 bits, of which 8 bits is information and 2 bits or more are signals to the receiver.

Idle channel: In addition, the transmission of each byte may then be followed by a gap of varying duration. This gap can be represented either by an idle channel or by a stream of additional stop bits.

Lecture # 75

Synchronous Transmission

In synchronous transmission, we send bits one after another without start or stop bits or gaps.



- In the above figure it is showing divisions between bytes, in reality, those divisions do not exist.
- The sender puts its data onto the line as one long string.
- When a sender wants to transmit data in separate bursts, they need to insert a specific sequence of 0s and 1s, signifying idle periods, in between these segments.
- The receiver counts the bits as they arrive and groups them into 8-bit units.
- In the absence of these idle intervals, as well as start and stop bits, there is no inherent mechanism to assist the receiving device in adjusting its bit synchronization during the data transmission process.
- As a result, precise timing becomes essential, as the accuracy of the received information relies entirely on the receiving device's ability to maintain an accurate bit count as the data is received.

Isochronous Transmission

- Used to send data in a predictable and timely manner.
- Data is sent at a fixed and constant rate, maintaining a precise timing schedule.
- This method is commonly used for real-time applications, such as audio and video streaming, where it is crucial to deliver data at a consistent pace to ensure smooth playback or communication.

Lecture # 76

FYI: There's no text in the reading section @VULMS for Lecture # 76. You can go through the lecture.

Lecture # 77

Before we discuss specific methods of digital-to-analog modulation, two basic issues must be reviewed:

Bit and Baud rates and The Carrier Signal:

The Baud rate refers to the total number of signal units transmitted in one second.

The Bit rate refers to the total Bits transmitted in one unit time.

In telecommunications, a carrier wave, carrier signal, or just carrier, is a waveform (usually sinusoidal) that is modulated (modified) with an information-bearing signal (called the message signal or modulation signal) for the purpose of conveying information.

In Analog Transmission of Digital Data, Baud Rate is less than or equal to the Bit Rate

Data Element vs. Signal Element

Data Rate vs. Signal Rate

Bandwidth Required ∝ Signal Rate (except FSK)

Carrier Signal

Lecture # 78

Example:

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate?

$r = 4 \quad S = 1000$

$N = ?$

$S = \frac{N}{r} \Rightarrow N = S \times r$

$N = 1000 \times 4$

$= 4000 bps$

Example:

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

$N = 8000 \text{ bps}$

$S = 1000 \text{ baud}$

$L = ?$

$r = ?$

$S = \frac{N}{r} \Rightarrow r = \frac{N \text{ (bits)}}{S \text{ (bauds)}}$

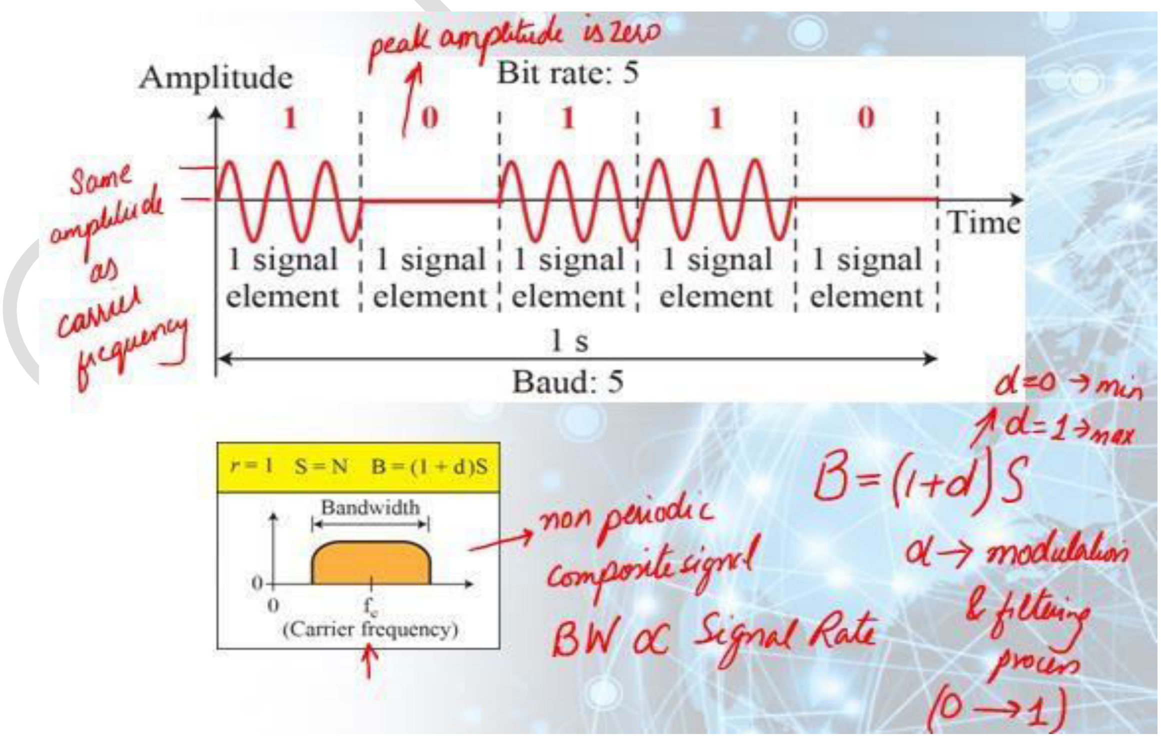
$\longrightarrow r = 8 \text{ bits/baud}$

$r = \log_2 L$

$\longrightarrow L = 2^r = 2^8 = 256$

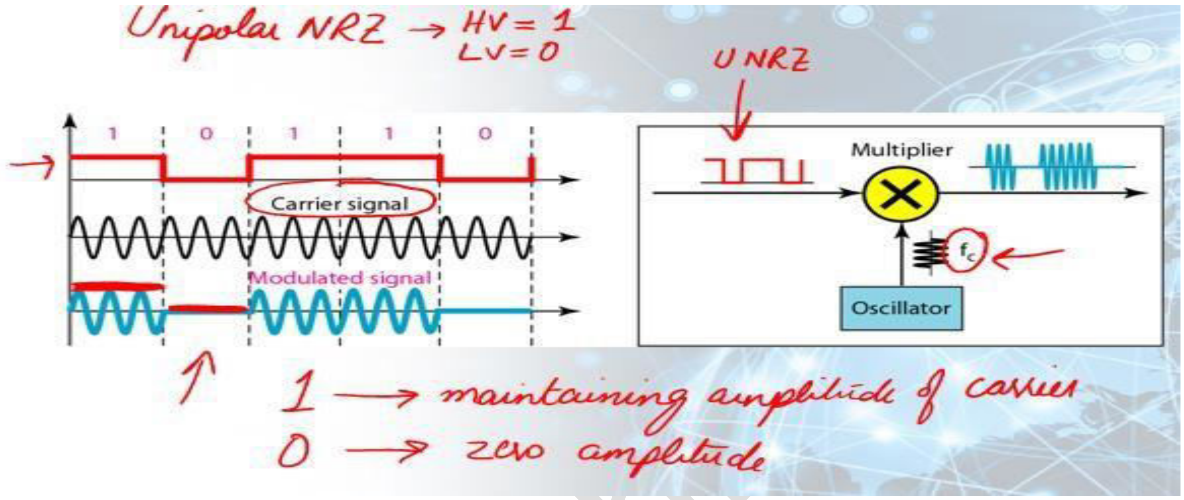
Lecture # 79

- The amplitude of the carrier signal is varied to create signal elements
- Both frequency and phase remain constant while the amplitude changes
- Binary ASK or On-Off Keying (OOK)



Lecture # 80

The amplitude of the carrier signal is varied to create signal elements.
Both frequency and phase remain constant while the amplitude changes
Binary ASK or On-Off Keying (OOK)
Implementation of Binary ASK or On-Off Keying (OOK)



(FYI: Numerical Problem) Example

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with d = 1?

$$f_c = 250\,kHz$$

$$B = (1+d)\,S$$

$$B = (1+1)\,S$$

$$B = 2S$$

$r = 1$

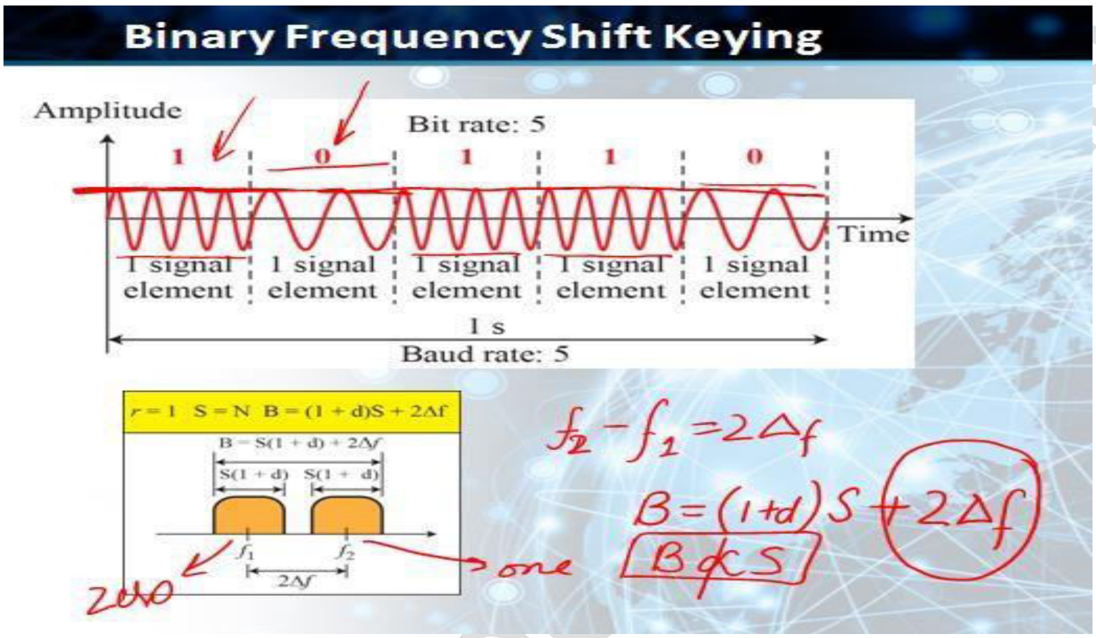
$$B = 2\left(\frac{N}{r}\right)$$

$$B = 2N \quad (r = 1)$$

$$N = \frac{B}{2} = \frac{100\,kHz}{2} = 50\,kbps$$

Lecture # 81

- The frequency of the carrier signal is varied to represent data.
- The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes.
- Both peak amplitude and phase remain constant



Lecture # 82

The frequency of the carrier signal varies to represent data.
Both peak amplitude and phase remain constant.

(**FYI:** Numerical problem) Example:

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with d = 1?

$$B = (1+d) \times S + 2\Delta f$$

$$\Delta f = 50 kHz$$

$$B = 2S + 50$$

$$2S + 50 = 100$$

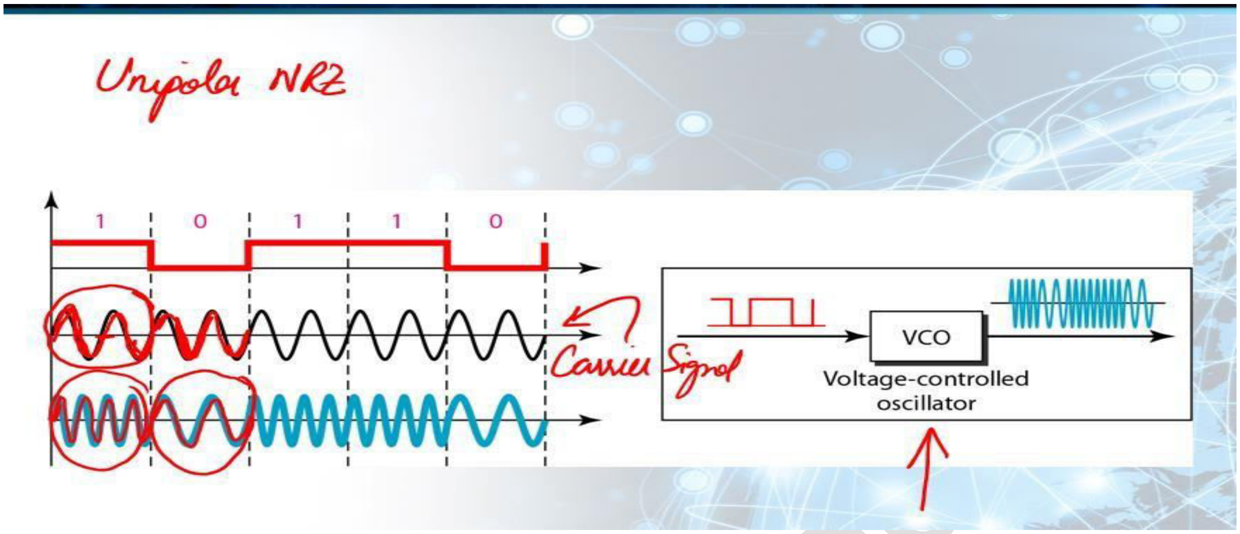
$$2S = 50 kHz$$

$$S = 25 \text{ kbaud}$$

$$N = S$$

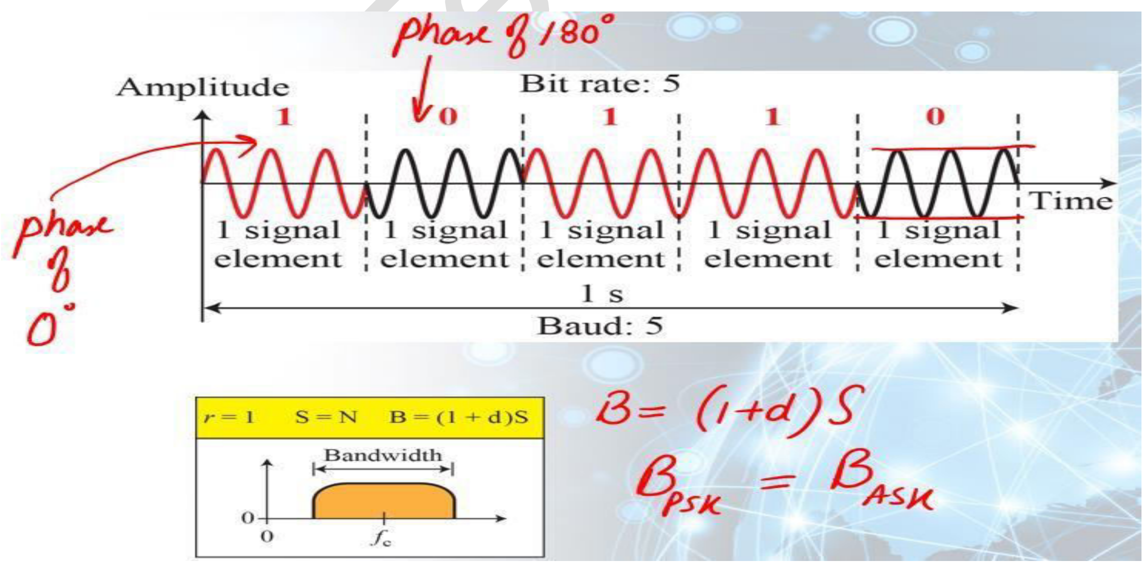
$$N = 25 \text{ kbps}$$

Implementation of BFSK:



Lecture # 83

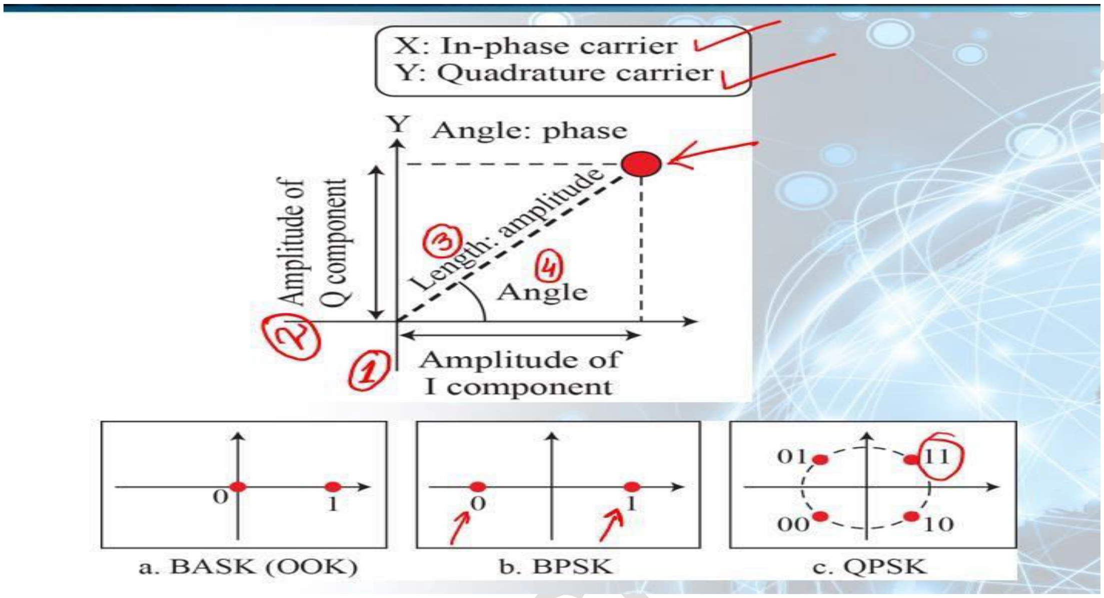
The phase of the carrier is varied to represent two or more different signal elements.
Both peak amplitude and frequency remain constant.
PSK is relatively more common than ASK or FSK.
Quadrature Phase Shift Keying (QPSK) is a form of Phase Shift Keying in which two bits are modulated at once, selecting one of four possible carrier phase shifts (0, 90, 180, or 270 degrees).
Binary Phase Shift Keying



Implementation of Binary Phase Shift Keying

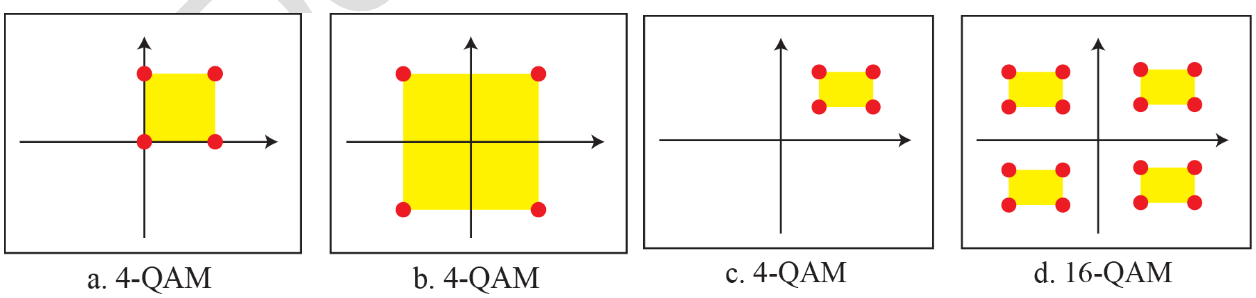
Lecture # 84

- **Helps us define the phase and amplitude of a signal element when we are using two carriers (one in phase and other in quadrature)**
- **Signal element is represented as a dot.**



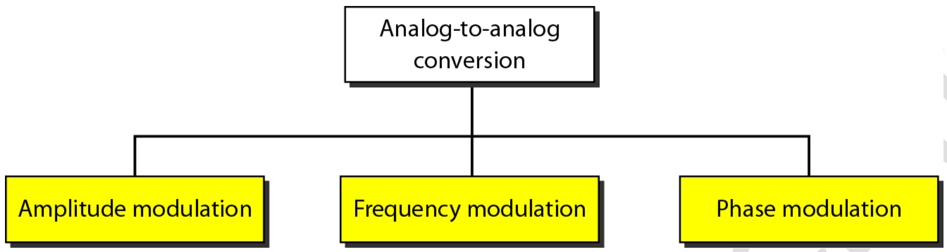
Lecture # 85

PSK is limited by the ability of the equipment to distinguish small differences in phase which limits its potential bit rate.
We have been altering only one of the three characteristics of a sine wave at a time; but what if we alter two?
Why not combine ASK and PSK?
Constellation diagrams for some QAMs.

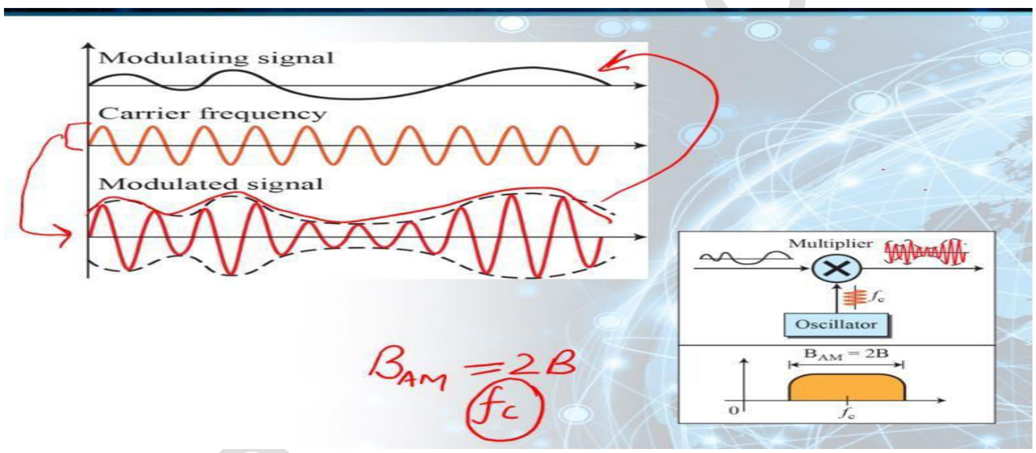


Lecture # 86

Representation of Analog information by an Analog signal
Amplitude Modulation (AM)
Frequency Modulation (FM)
Phase Modulation (PM)
Types of Analog-to-Analog Modulation



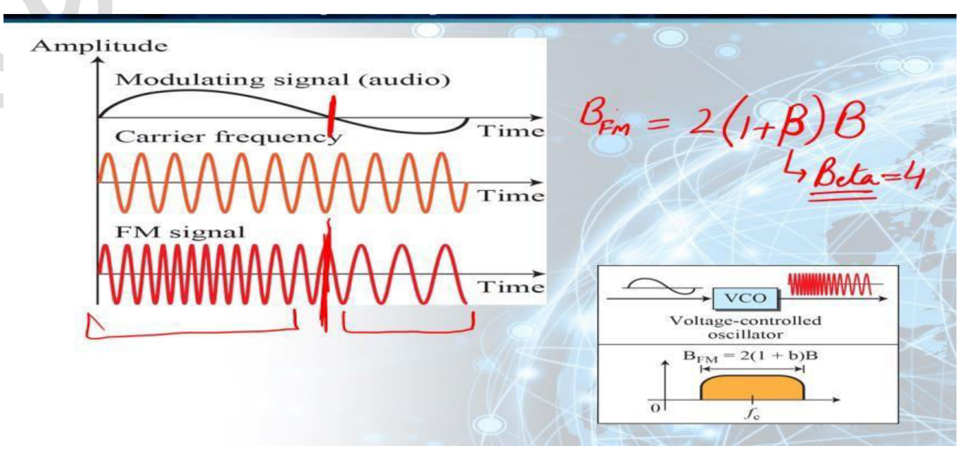
Amplitude modulation



Lecture # 87

FYI: This reading section contains the same stuff except the below heading and pic.

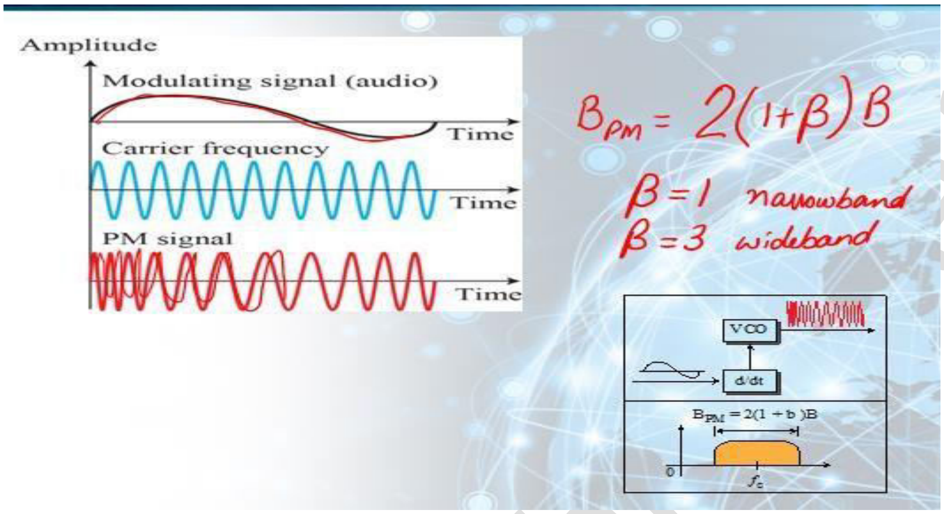
Frequency Modulation (FM)



Lecture # 88

FYI: This reading section contains the same stuff except the below heading and pic.

Phase Modulation (PM)



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So I'm releasing this first part of "First Test Edition" on 18th November 2023. Inshallah, I'll release it remaining as soon as the reading section will be open for students to read. I may also release its Mid-term file and Final-term file separately.
If you find any error or have anything to recommend me about it, then please feel free to contact me at given platforms in the "**Better-to-Know**" section.

Regards!!