20/8/24

ASSIGNMENT 2

Unit-2 Physical Layer

Q.1 Illustrate analog and digital signal.

→ Analog Signal

An analog signal is a continuous signal that varies over time and can take on any value within a given range. It represents information in a form that is analogous to the actual physical phenomenon, such as sound waves, temperature or light intensity.

They are typically represented as smooth, continuous waveforms like sine waves. The amplitude of the wave can take any value within a given range.

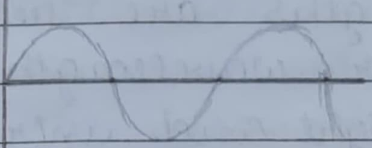
eg: The signal from a microphone picking up a sound wave.

→ Digital Signal.

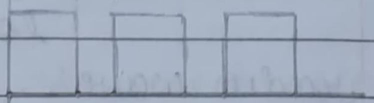
A digital signal is a discrete signal that represents information in binary form, using a series of 0s and 1s. It only takes on specific values, typically two levels, representing on(1) and off(0).

They are represented by square waves, where the signal jumps between distinct levels without intermediate values.

eg: The data sent from a computer to a digital device like a USB drive.



Analog signal



Digital signal

Q.2 Define the following terms:

i) Frequency.

→ Frequency refers to the number of oscillations or cycles that a wave completes in one second. It is measured in Hertz (Hz). For example, a wave that completes 1000 cycles in one second has a frequency of 1000 Hz (1 KHz).

It determines the rate at which data can be transmitted. Higher frequencies can carry more data. It is used in frequency modulation (FM) for radio transmission

ii) Wavelength.

→ Wavelength is the physical distance between two consecutive points in phase on a wave, such as the distance between two consecutive crests or troughs. It is usually measured in meters (m). For example, for a wave travelling at a speed of 300,000,000 meters per second (speed of light in vacuum) and a frequency of 1,000,000 Hz (1MHz), the wavelength would be 300 meters.

longer wavelengths are used for radio waves, while shorter wavelengths are used for visible light and infrared signals.

iii) Bandwidth.
-v . Bandwidth refers to the range of frequencies over which a signal can be transmitted or the amount of data that can be transmitted over a communication channel in a given time period. It is typically measured in Hertz (Hz) for analog signals or bits per second (bps) for digital signals. For example, if a communication channel can transmit signals between 1 MHz and 3 MHz, its bandwidth is 2 MHz. For digital communication, if a channel can transmit 1 Gbps, that is its bandwidth.

It determines the maximum data rate that can be transmitted. wider bandwidths allow for higher data rates. It is also used to measure the capacity of a communication channel.

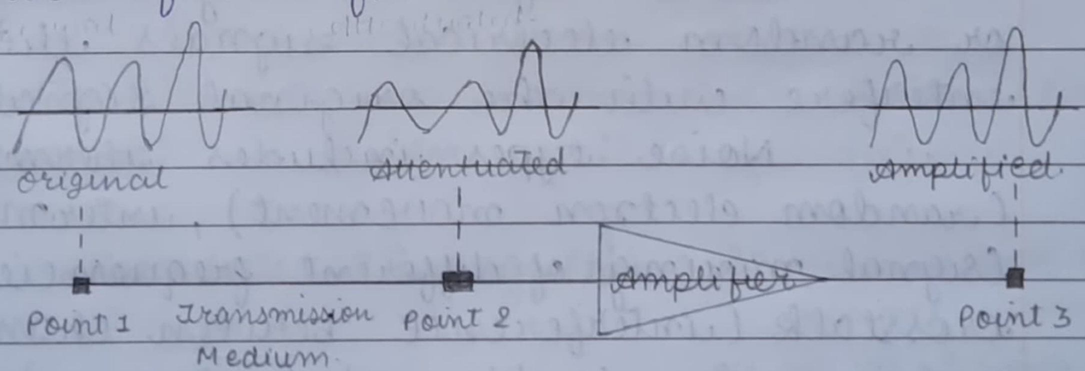
Q.3 Describe causes of transmission impairment.
-4 Transmission impairment refers to any degradation of the signal as it travels through a transmission medium, which can lead to errors or loss of data.

1. Attenuation.

Attenuation is the reduction in the strength or amplitude of a signal as it propagates through a medium.

causes: As the signal travels over a long distance, it loses energy and becomes weaker. Different ~~communication~~ transmission media, such as copper cables, fiber optics or air, have different levels of resistance or absorption that cause the signal to weaken.

effects: A weakened signal may become indistinguishable from noise, leading to loss of information.

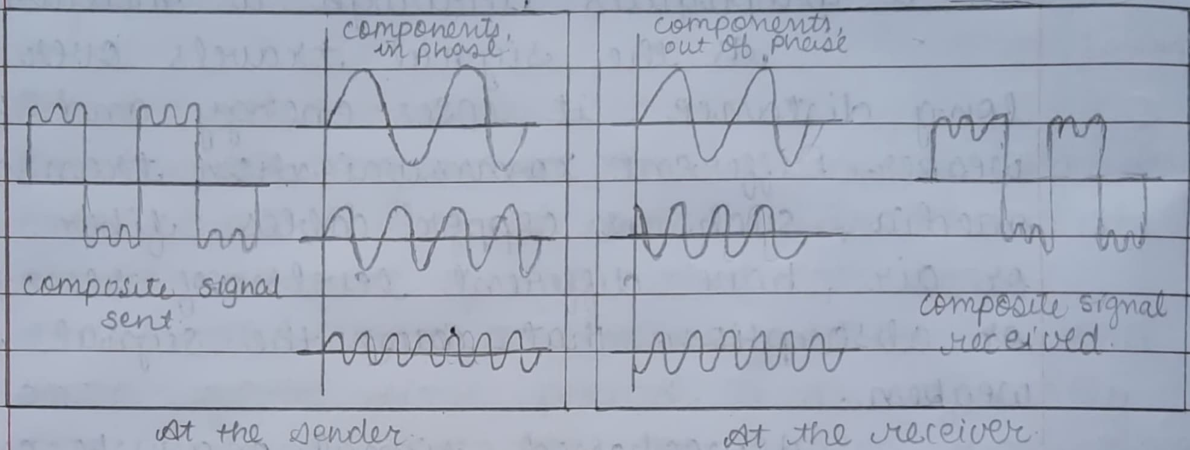


2. Distortion.

Distortion occurs when the shape or form of the signal is altered as it travels through the medium.

causes: Different frequencies travel at different speeds, causing phase shifts and signal distortion. Multiple paths cause signal distortion in wireless communiccation.

effects: Distortion can cause the signal to become unclear, making it ~~unclear~~ difficult for the receiver to interpret the original information accurately.

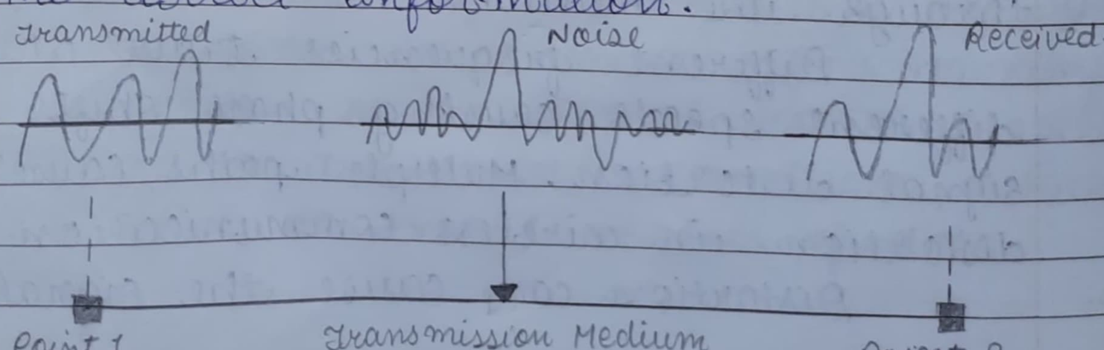


3. Noise

Noise refers to any unwanted or random electrical signals that interfere with the original signal.

causes: Noise types includes thermal (random electron movement), intermodulation (signal mixing of different frequencies), crosstalk (interference between channels) and impulse (sudden disturbances).

effects: Noise can cause errors in the transmitted data by altering the original signal, making it difficult to recover the correct information.



Q.4 Describe following:

i) Nyquist's theorem.

→ Nyquist's theorem, also known as sampling theorem, states that to accurately reconstruct a continuous signal from its samples, the sampling rate must be at least twice the signal's highest frequency. This minimum rate is called the Nyquist rate.

This prevents aliasing, where different signals become indistinguishable. This theorem is essential for digital signal processing, audio/video digitization, and telecommunications.

The maximum data rate (in bits per second) over a band-limited channel is given by:

$$\text{Data rate} = 2B \log_2(M)$$

; B = bandwidth of the channel in Hz

M = no. of discrete levels used to represent each symbol

If a signal is sampled at a rate lower than Nyquist rate, aliasing occurs. For example, if a signal has a maximum frequency of 5 KHZ, it should be sampled at a rate of at least 10 KHZ to avoid aliasing and to accurately reconstruct the original signal.

ii) Shannon's theorem.

—› Shannon's theorem, also known as the Shannon - Hartley theorem, defines the maximum data rate (capacity) that can be transmitted over a noisy communication channel depends on bandwidth and signal-to-noise ratio.

The theorem indicates that increasing the bandwidth or improving the signal-to-noise ratio allows for a higher data transmission rate. It is crucial for communication system design and analysis.

The maximum achievable information rate over a noisy channel is given by:

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

; B = bandwidth of the channel in Hz

S = Signal Power

N = Noise Power.

For example, a noisy phone can transmit more data if you improve its quality and reduce noise.

Q.5. Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with four signal levels. Calculate the Bit Rate.

—› Given,

Bandwidth, B = 3000 Hz

No. of signal levels, M = 4.

ans So, to calculate the bit rate for a noiseless channel, we can use the Nyquist formula:

$$\text{Bit Rate} = 2 \times B \times \log_2(M)$$

$$\therefore \text{Bit Rate} = 2 \times 3000 \times \log_2(4)$$
$$= 2 \times 3000 \times 2$$
$$= 12000 \text{ bits per second (bps)}$$

⇒ The bitrate for the noiseless channel is 12,000 bps (12 Kbps).

Q.6 Consider a telephone line with a bandwidth of 3000. The signal-to-noise ratio is usually 3162. What is the capacity of channel?

→ Given,

Bandwidth, $B = 3000$ Hz

Signal-to-Noise ratio, $S/N = 3162$.

ans So, to calculate the capacity of a channel in the presence of noise, we use Shannon's-Hartley Theorem:

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

$$\therefore C = 3000 \times \log_2(1 + 3162)$$
$$= 3000 \times \log_2(3163) \approx \log_{10}(3163) / \log_{10}(2)$$
$$= 3000 \times 11.627$$
$$= 34881 \text{ bps (bits per second)}.$$

⇒ The capacity of the telephone line is approximately 34,920 bps (34.92 Kbps)

Q.7 Define the following terms :

i) Throughput

→ Throughput is the actual rate at which data is successfully transmitted between two points in a network. It measures how much data is transferred from one point to another within a specified period and is usually expressed in bits per second (bps) [also in kbps, Mbps, Gbps]. The actual achieved data rate, which can be lower due to factors like network congestion, errors and protocol overhead.

For Example, if a network has a bandwidth of 100 Mbps but due to congestion and other factors, only 60 Mbps is being achieved, the throughput is 60 Mbps.

ii) Latency.

→ Latency is the time delay experience in a system, particularly the time it takes for a data packet to travel from the source to the destination. Latency is typically measured in milliseconds (ms) or seconds (s). It also includes propogation, transmission, processing and queuing delay. which affects the latency.

For example, in a satellite communication system, the latency might be high (e.g, 500 ms) due to the long distance the signal must travel.

iii) Bandwidth - Delay Product.

→ The Bandwidth - Delay Product (BDP) is a metric that represents a measure of the maximum amount of data that can be in transit ["in flight"] within a network at any given time. It is measured in bits.

∴ BDP = Bandwidth × Round-Trip Time (RTT)

- Bandwidth measured in bps.
- RTT is the time it takes for a signal to travel from the sender to the receiver and back, measured in seconds.

A higher BDP indicates that the network can handle more data in transit, which is important for optimizing the throughput of long-distance, high-bandwidth connections.

For example, if a network has a bandwidth of 10 Mbps and an RTT of 100 ms, the BDP would be 125 KB (Kilo Bytes).

Q.8 What is digital to digital conversion? Enlist various digital to digital conversion techniques.

→ Digital-to-Digital Conversion (D2D) refers to the process of converting digital data (binary bits) into a digital signal that can be transmitted over a communication channel (medium). This involves representing the digital data (1s and 0s) in a format suitable for transmission. The primary goal is to ensure the signal ~~to~~ is transmitted with minimal errors, while also making efficient use of the available bandwidth.

→ Various D2D techniques :~

i) Line Coding.

It converts binary data into a digital signal by mapping the binary sequence to voltage levels, ensuring effective transmission over communication media. Different line coding schemes are used to achieve various objectives like minimizing bandwidth, ensuring synchronization and reducing errors.

ii) Block Coding.

Here, the block of bits are mapped to blocks of symbols, often ~~#~~ with redundancy to ensure error detection. Block coding is typically used in ~~conjuction~~ conjunction with line coding to improve the reliability of data transmission. The key aspects of block coding are error detection, error correction and redundancy (extra bits ~~of~~ are added to the original data to provide redundancy).

For example, 4B/5B encoding, where every 4-bit block of data is mapped to a 5-bit code, adding redundancy to help detect errors and maintain synchronization.

iii) Scrambling

Scrambling is used to prevent long sequences of zero's or ones, which can cause synchronization problems. It modifies the data to ensure a more balanced signal. Unlike block coding, scrambling does not add extra bits to the signal; instead, it reorders the data to reduce the likelihood of ~~synchro~~ synchronization.

For example, B8ZS (Bipolar 8-Zero Substitution) and HDB3 (High-Density Bipolar 3) are commonly used scrambling techniques in digital communication systems.

Q.9 Explain line coding briefly. Enlist different line coding schemes.

→ Line coding is a technique used to convert digital data into a digital signal for transmission, ensuring compatibility with the medium and receiver while maintaining efficient, error-free communication.

→ Different line coding Schemes :-

1. Unipolar : uses a single polarity (usually positive) for representing binary 1, while binary 0 is representing by zero voltage.
eg: Unipolar NRZ (Non-Return-to-zero).

2. Polar : uses two voltage levels - positive and negative - to represent binary 1 and 0.
eg: NRZ-L (Non-Return-to-zero-level), RZ (Return-to-zero).

3. Bipolar : Alternates between positive and negative voltages to represent binary 1, while binary 0 is represented by zero voltage.
eg: AMI (Alternate Mark Inversion), B8ZS (Bipolar with 8-zero Substitution).

4. Multilevel : uses more than two voltage levels to represent multiple bits per symbol.
eg: 2B1Q (Two Binary, One Quaternary), ~~MLT~~ MLT-3 (Multilevel Transmission - 3 levels)

5. Multitransition : ~~Incoorp~~ Incorporates multiple transitions within a single bit interval to represent data, often used to ensure

eg: Manchester Encoding, Differential Manchester encoding.

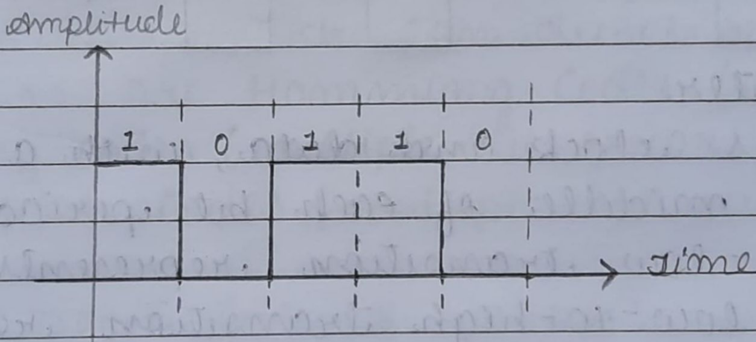
Q.10 Describe the following line coding schemes:

i) unipolar

→ Same definition as in Question 9

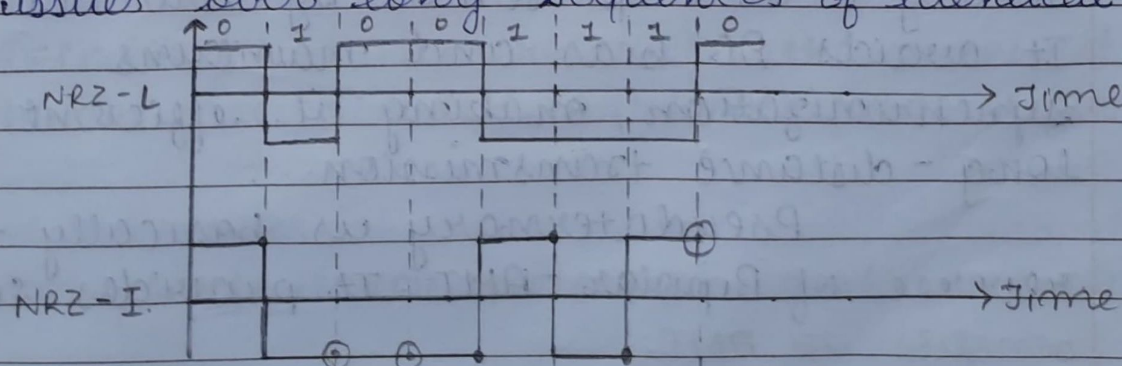
It is simple to implement and have low power consumption.

The problem here is that it uses DC component and it is susceptible to noise

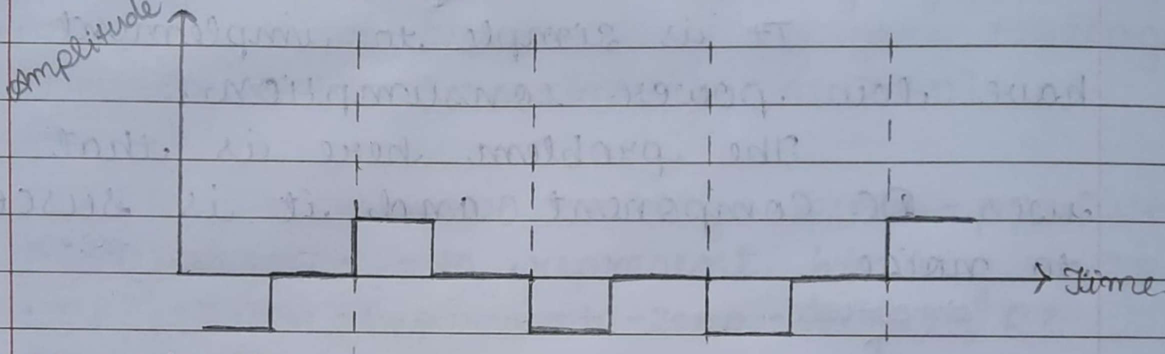


ii) Polar - NRZ (Non-Return-to-Zero)

→ Uses two voltage levels - positive for binary 1 and negative for binary 0. The signal does not return to zero between bits, leading to more efficient use of bandwidth but potential synchronization issues over long sequences of identical bits.



iii) Polar - RZ (Return-to-zero)
→ Uses positive and negative voltages for binary 1 and 0, but the signal returns to zero halfway through each bit period. This improves synchronization but requires more bandwidth than NRZ. I



iv) Manchester
→ Combines clock and data, with a transition in the middle of each bit period. A high-to-low transition represents binary 0 and a low-to-high transition represents a binary 1. It ensures good synchronization but requires more bandwidth.

v) Bipolar-AMI and Pseudoternary.
→ Bipolar-AMI (Alternate Mark Inversion) represents binary 1 by alternating between positive and negative voltages, while binary 0 is represented by zero voltage. It avoids DC bias and maintains synchronization, making it efficient for long-distance transmission.

Pseudoternary is basically the reverse of Bipolar-AMI. It provides similar benefits to AMI.

Q.11 Describe block coding digital to digital conversion technique.

→ Block coding is a technique that adds redundancy to a block of data to improve error detection and correction capabilities. By introducing extra bits into the data stream, block coding can increase the robustness of the transmitted information. The key features are error detection and correction, synchronization and efficiency.

The Common Block Coding techniques which can detect and correct errors are Hamming Codes (single-bit errors), Golay codes (multiple-bit errors) and Reed-Soloman Codes (multiple consecutive bits in error).

→ How Block Coding Works :-

(i) Data Division : The input data is divided into blocks of a fixed size (eg: 4 bits).

(ii) Encoding : Each block is encoded into a larger block containing additional bits (eg: 4bit block encoded into 8-bit block).

(iii) Transmission : The encoded blocks are transmitted over the communication channel.

(iv) Decoding : At the receiver, the received blocks are decoded to recover the original data.

• Advantages :-

→ Enhanced error detection and correction capabilities.

- Improved synchronization due to controlled signal transitions.
- Increased reliability of data transmission.

- Disadvantages :-
- Increased bandwidth usage due to the added redundancy.
- Complexity in encoding and decoding processes.

Q.12 What is analog to digtal conversion? Enlist various ~~anlog~~ analog to digital conversion techniques.

→ Analog-to-Digital Conversion (ADC) is the process of converting a continuous analog signal (sound, temperature) into a digital signal that can be processed by digital systems (computers, communication devices). This conversion involves key steps:

i) Sampling : Measuring the amplitude of the analog signal at regular intervals (sampling rate)

ii) Quantization : Converting each sampled amplitude to the nearest value within a finite set of discrete levels.

iii) Encoding : Converting the quantized values into a digital format, typically binary.

→ Various ADC techniques :-

1. Pulse Amplitude Modulation (PAM)
2. Pulse Code Modulation (PCM)
3. Delta Modulation. (DM).

Q.13 Explain following analog to digital conversion techniques:

i) Pulse Amplitude Modulation.

→ PAM is a technique where the amplitude of each pulse in a sequence of pulses is proportional to the amplitude of the analog signal at the same time of sampling. It is an ADC method where an analog signal is sampled and each sample is represented by a pulse whose amplitude corresponds to the sample value. ~~If~~ It is often used as an intermediate step in other conversion techniques like PCM.

PAM is simple to implement and works well for low-frequency signals, making it suitable for straightforward applications.

However, it is prone to noise interference and often requires extra modulation and demodulation stages to maintain signal quality.

ii) Pulse Code Modulation.

→ PCM is a digital representation of an analog signal where the amplitude of the analog signal is sampled, quantized and encoded into a binary form. PCM is a more common method for digital audio and data transmission compared to PAM.

PCM offers hight noise immunity and is efficient for a wide range of signals, making it ideal for various digital communication applications.

However, it requires more bandwidth than PAM, which can be a limitation in bandwidth - constrained environments.

iii) Delta Modulation (DM).

→ DM is a method where the difference between the current ~~can~~ sample and the previous sample is encoded into a single bit. It is a simpler and more efficient technique compared to PCM for representing analog signals with a lower bit rate.

DM is simple to implement and requires low bandwidth, making it efficient for basic applications.

However, it is susceptible to slope overload distortion and has a limited dynamic range, which can affect signal quality in more complex scenarios.

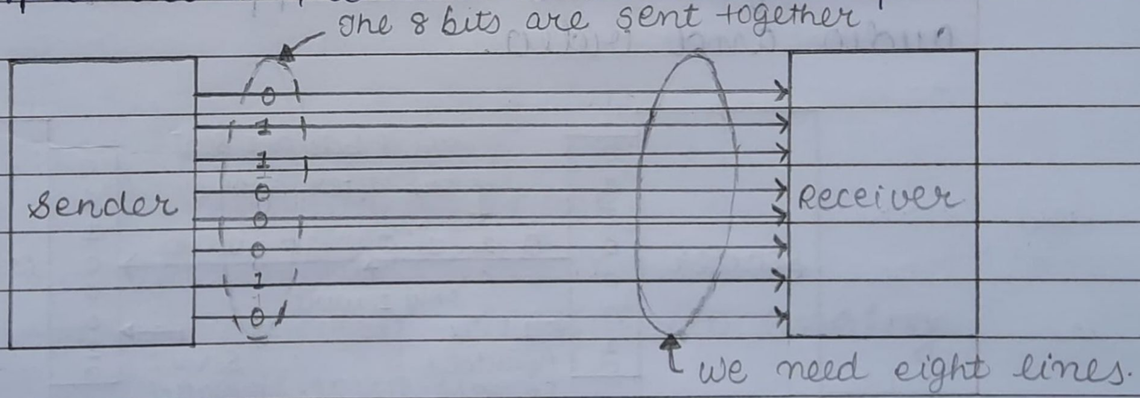
Q.14 Describe different transmission modes in digital transmission.

→ Digital transmission involves the transmission of discrete data, typically represented as a sequence of 0s and 1s. There are two primary transmission modes :

1. Parallel Transmission

Multiple bits of data are sent simultaneously over multiple channels or wires. Each bit of the data is transmitted over a separate wire, allowing for high data transfer rates.

It provides high speed and simplicity for short distances but faces performance issues over long distances and requires multiple wires, which can be costly.



2. Serial Transmission

Bits are sent one after another over a single channel or wire. This method is more suitable for long-distance communication.

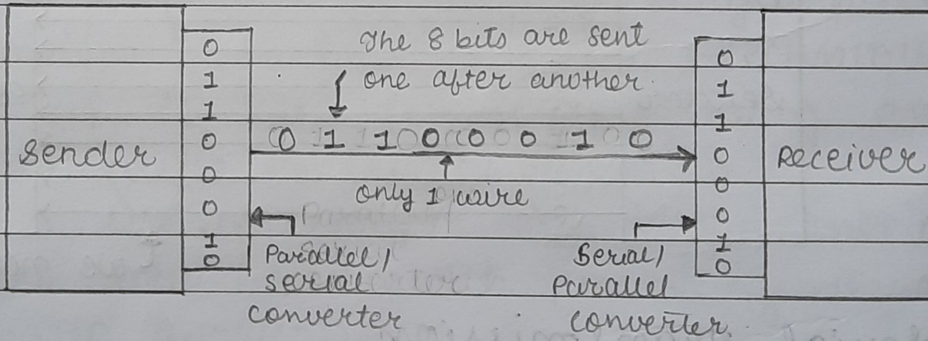
It reduces wiring complexity and is better for long distances but generally offers lower data transfer rates, though ~~advandanced~~ advanced technologies can improve speed.

There are types of serial transmission, which are :

(i) Asynchronous Transmission : sends data one byte at a time with start and stop bits, using timing for synchronization.

(ii) Synchronous Transmission : sends data in blocks with shared clock signals for continuous flow without start and stop bits.

(iii) Isochronous Transmission : Tranmits data at a constant rate with precise timing, ideal for real-time applications like audio and video.



Q.15 What is digital to analog conversion ? Enlist various digital to analog conversion techniques.

→ Digital-to-analog conversion (DAC) is the process of converting digital data (binary or coded) into an analog signal that can be used by analog devices. This is essential for applications where digital systems need to interface with analog systems, such as audio playback, video display and control systems. It involves steps like:

i) Digital Input: Receives a binary number representing the discrete values.
ii) Conversion: Transforms this binary number into a corresponding analog ~~to~~ voltage or current.
iii) Output Filtering: uses a low-pass filter to smooth the stepped signal ~~to~~ into a continuous analog output.

→ Various DAC techniques used ~~for~~ are :-

1. Amplitude Shift Keying (ASK)
2. Frequency Shift Keying (FSK)
3. Phase Shift Keying (~~$~~PSK)

Q.16 Explain following digital to analog conversion techniques:
i) ASK.
→ The amplitude of a carrier signal is varied according to the digital data signal. Each digital ~~signal~~ bit is represented by a different amplitude level of the carrier wave. Typically, one amplitude level ~~cc~~ represents a binary "1", and another represents a binary "0".
ASK is easy to implement and suitable for low frequencies (easy-to-use

modulation scheme) but prone to noise and less efficient than other methods.

ii) FSK.

→ FSK involves varying the frequency of the carrier signal based on the digital data. Each binary value is represented by a different frequency of the carrier wave. For instance, one frequency might represent a binary 1, and a different frequency represents a binary 0.

FSK is more resistant to noise and works well in noisy environments, but it requires a wider bandwidth compared to ASK.

iii) PSK.

→ The phase of the carrier signal is altered according to the digital data. Each binary value changes the phase of the carrier wave. For example, a phase shift of 0 degrees could represent a binary 1 and a phase shift of 180 degrees could represent a binary 0.

PSK is bandwidth-efficient with good noise immunity, but it is more complex to implement than ASK or FSK.

Q.17 What is analog to analog conversion? Enlist various analog to analog conversion technique.

→ Analog-to-Analog Conversion (AAC) is the process of transforming an analog signal into another form of analog signal. Unlike ~~D&D~~ DAC; which involves translating discrete digital values into continuous analog signals, AAC involves manipulating or processing continuous analog signals without converting them into a digital form. This conversion is often necessary to ensure compatibility between different electronic devices or systems.

→ Various AAC techniques used :-

1. Amplitude Modulation (AM)
2. Frequency Modulation (FM)
3. Phase Modulation (PM)

Q.18 Explain following analog to analog conversion techniques:

i) AM.

→ The amplitude of the carrier signal is varied in proportion to the amplitude of the input analog signal. Here, the frequency and phase of the carrier remain constant. The modulating signal contains the information to be transmitted.

AM is simple to implement and works well for low-frequency signals, but it is more susceptible to noise and less efficient compared to FM or PM.

For example, in radio broadcasting, AM is used where the sound signal modulates the amplitude of a high-frequency carrier wave.

ii) FM.

→ FM varies the frequency of the carrier signal is varied according to the amplitude of the input analog signal (modulating signal). The amplitude and phase remains constant but frequency changes to represent the information signal.

FM offers better noise resistance and audio quality with a wider bandwidth, but it requires more complex circuitry and higher bandwidth than AM.

For example, FM is commonly used in radio broadcasting where the audio signal modulates the frequency of the carrier wave, providing better noise immunity than AM.

iii) PM.

→ The Phase of the carrier signal is varied according to the amplitude of the modulating signal. Unlike FM, which changes frequency, PM directly alters the phase of the carrier signal and the

amplitude & frequency remains unchanged.
PM is bandwidth - efficient and provides good noise immunity but it is more complex to implement compared to AM or FM.
For example, PM is used in some digital modulation schemes like PSK and can be found in application like GPS and digital communication systems.

Q.19 What is multiplexing? Enlist various multiplexing techniques.

→ Multiplexing is a technique used in communications to combine multiple signals ~~to~~ into a single signal over a shared medium. This process allows multiple data streams to be transmitted simultaneously without interference, maximizing the efficiency of the communication channel. It optimizes the use of available bandwidth and reduces the number of channels needed.

→ Various Multiplexing techniques are :-

1. Frequency-Division Multiplexing (FDM)
2. Wavelength-Division Multiplexing (WDM)
3. Synchronous Time-Division Multiplexing (STDM)
4. Statistical Time-Division Multiplexing (Statistical TDM)

Q.20 Explain the following multiplexing techniques :

i) FDM.

→ FDM allocates different frequency bands to multiple signals, allowing them to be transmitted simultaneously over a single channel without interference. Each signal operates in its own unique frequency range.

FDM is simple to implement and efficient for low-frequency signals but is susceptible to interference and has a limited number of channels.

For example, traditional radio and television broadcasting use FDM to transmit multiple channels over the same frequency spectrum.

ii) WDM

→ WDM is similar to FDM but operates in the optical domain. It transmits multiple signals simultaneously over a single optical fiber by ~~assigning~~ assigning each signal a different wavelength (color) of light.

WDM offers high bandwidth, low loss and supports long-distance transmission, though it requires specialized optical equipment.

For example, used in fiber-optic

communications to increase the capacity of a single fiber.

iii) Synchronous TDM.

-4 Synchronous TDM divides the channel into fixed time slots, with each signal assigned a specific time slot in a repeating sequence. All signals share the channel in a synchronized manner.

STDM is efficient, deterministic and easy to synchronize but needs precise timing and can be inefficient with unused slots.

For example, T1 and E1 telecommunication system use STDM.

iv) Asynchronous TDM. (Statistical TDM)

-4 Statistical TDM dynamically allocates time slots to signals based on demands rather than fixed slots. This method optimizes bandwidth by only assigning slots to active signals.

ATDM handles bursty traffic efficiently and supports varying data rates but is complex to implement and requires buffering and schedulling.

For example, used in modern digital communication systems where data traffic is variable, such as in packet-switched networks

~~~~ x ~~~~ x ~~~~
~~~~