Data Communication Unit III

Digital Transmission

Digital transmission refers to the process of sending digital signals (binary or other discrete-level pulses) between multiple points in a communication system. This is achieved using physical mediums such as wires, coaxial cables, or optical fibers.

Advantages of Digital Transmission

1. **Noise Immunity:**
    - Digital signals are less prone to interference from noise compared to analog signals, ensuring more reliable communication.
2. **Multiplexing:**
    - Digital signals are better suited for techniques like multiplexing, allowing multiple signals to share a single transmission medium efficiently.
3. **Signal Regeneration:**
    - Digital systems regenerate signals rather than amplifying them, which prevents noise from accumulating along the transmission path.
4. **Ease of Processing:**
    - Digital signals are easier to measure, process, and evaluate using digital technologies.

Disadvantages of Digital Transmission

1. **Bandwidth Requirements:**
    - Transmitting digitally encoded analog signals often requires more bandwidth than transmitting the original analog signal.
2. **Conversion Overhead:**
    - Analog-to-digital conversion before transmission and digital-to-analog conversion after transmission adds complexity and cost.
3. **Synchronization:**
    - Precise time synchronization between transmitter and receiver clocks is essential for proper signal interpretation.
4. **Compatibility Issues:**
    - Digital systems are often incompatible with older analog systems.

Multiplexing is a technique used in digital communication to combine multiple signals into one signal over a shared medium. The primary goal of multiplexing is to increase the efficiency of the communication channel, allowing multiple transmissions to occur simultaneously without interference. Common types of multiplexing include:

1. **Time Division Multiplexing (TDM)**: This method divides the available bandwidth of the communication channel into time slots, each of which is allocated to a different signal. For example, if there are 4 data sources, each is assigned a time slot in a round-robin manner to send its data.
2. **Frequency Division Multiplexing (FDM)**: This method divides the available bandwidth into several frequency bands. Each signal is modulated onto its own frequency band and transmitted simultaneously.

What is Pulse Modulation?

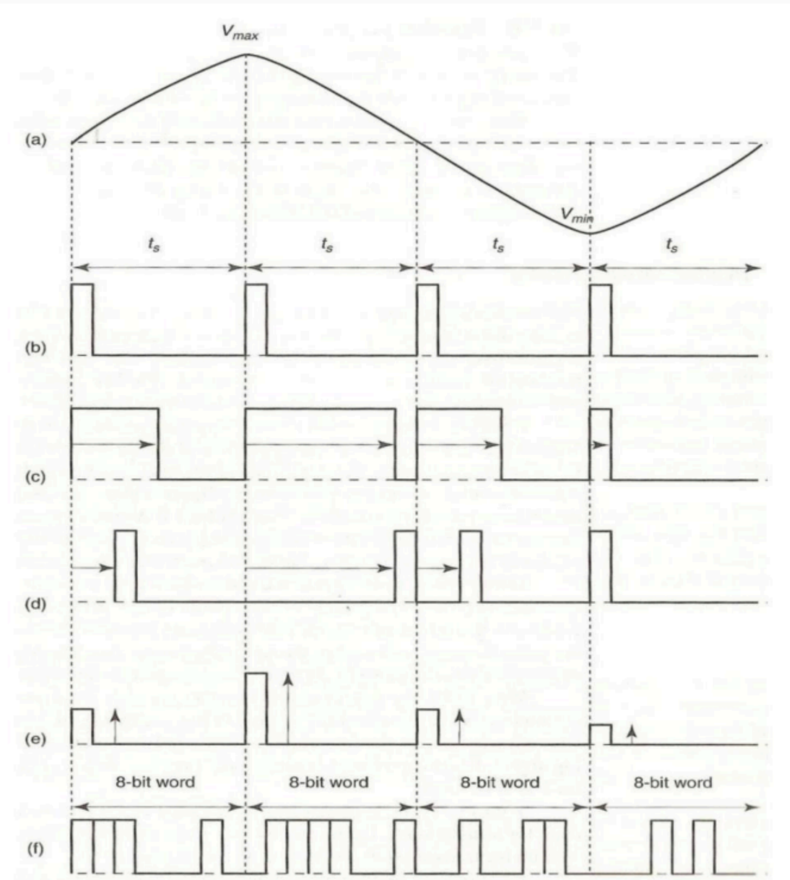
Pulse modulation is a method used to send information (like voice, music, or data) by converting it into a series of pulses (on/off signals). Instead of transmitting the original continuous analog signal, it **samples** the signal and encodes it into discrete pulses.

Types of Pulse Modulation

1. **Pulse Width Modulation (PWM):**
    - The width of a constant-amplitude pulse varies according to the amplitude of the analog signal at the time of sampling.
    - **Applications:**
        - Voltage regulators.
        - Class-D audio amplifiers.
2. **Pulse Position Modulation (PPM):**
    - The position of a pulse within a time slot changes based on the analog signal's amplitude.
    - **Applications:**
        - Optical fiber communications.
        - Radio-controlled (RC) vehicles like aircraft and cars.
3. **Pulse Amplitude Modulation (PAM):**
    - The amplitude of pulses varies in proportion to the analog signal.
    - Resembles the original analog signal more closely than PWM or PPM.
    - **Applications:**
        - Telephone modems.
        - Ethernet communication.
4. **Pulse Code Modulation (PCM):**
    - Converts an analog signal into binary code by sampling and quantization.

○ Widely used in digital communication systems, including telephony and audio recording.

Applications

- **Telecommunication Networks:** Enables efficient voice and data communication.
- **Audio and Video Processing:** Utilizes PCM and PWM techniques for high-fidelity audio and video signals.
- **Optical Communication:** Employs PPM for minimal multipath fading.
- 

Digital transmission, combined with pulse modulation techniques, forms the backbone of modern communication systems, balancing performance with complexity based on application needs.

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Pulse Code Modulation (PCM): Simplified Explanation

Pulse Code Modulation (PCM) is a method of converting analog signals (like voice or music) into a digital format that computers or other digital devices can process and store. It is widely used in audio recording, telephony, and communication systems.

Step-by-Step Breakdown

1. Analog Signal

- An **analog signal** is a continuous wave, such as sound from a microphone or a guitar.
- It changes smoothly over time and can have any value within a range.

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2. Why Digital Representation?

- Computers work with **digital data**—a series of numbers (0s and 1s).
- PCM is used to convert the analog wave into a digital format while preserving the signal's key information.

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3. PCM Process

Step 1: Sampling

- The analog signal is measured at regular intervals of time.
- These measurements are called **samples**.
- **Sampling Rate:** The number of times the signal is measured per second (e.g., 44.1 kHz for CDs).

Step 2: Quantization

- Each sample is assigned a numerical value that represents its amplitude (height of the wave) at that moment.
- The amplitude is rounded to the nearest available value. This process is called **quantization**.

Step 3: Encoding

- The numerical values are converted into binary format (0s and 1s) for digital storage and processing.

Key Features of PCM

1. **Accuracy:** The higher the sampling rate and the more precise the quantization, the better the digital representation of the original signal.
2. **Digital Format:** The signal becomes a series of numbers (binary code), making it easy to process, store, and transmit.
3. **Noise Resistance:** Once converted to digital, the signal is less affected by noise and interference.

Example

Imagine recording a voice:

1. The microphone captures the analog sound wave.
2. PCM samples the sound wave 44,100 times per second (common for audio CDs).
3. Each sample is converted to a binary number representing the wave's amplitude.

For example:

- Analog wave: 🎵 (smooth wave)
- After sampling: 📈 (discrete points on the wave).
- After quantization and encoding: 100101, 110011, etc.

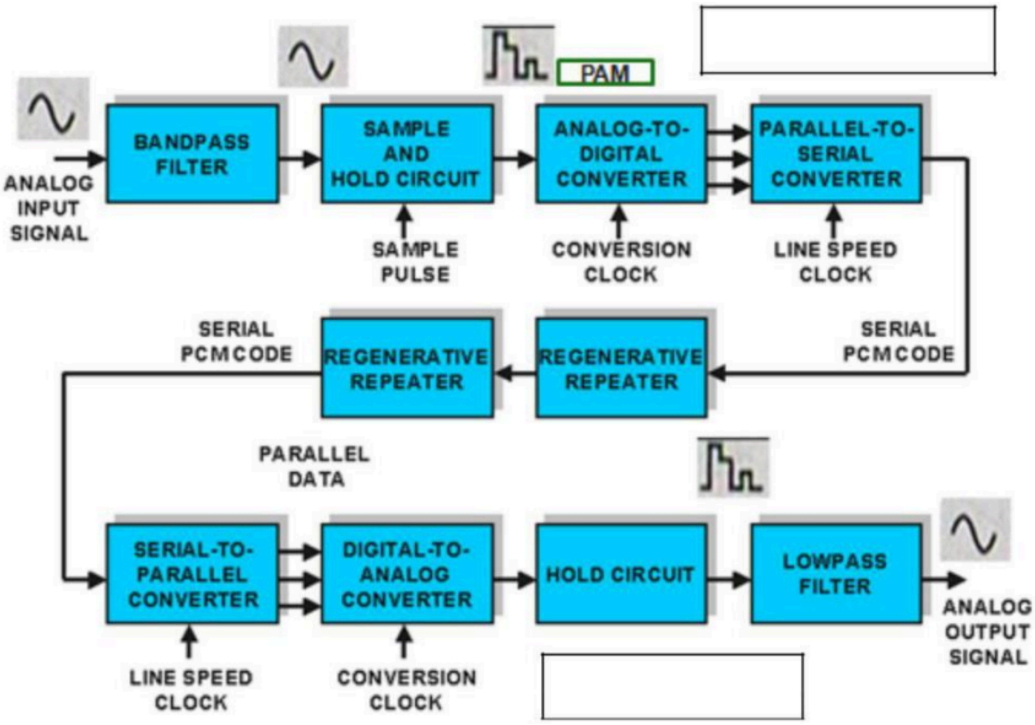
Applications of PCM

- **Telephony:** PCM is used in landlines to convert voice into digital data for transmission.
- **Audio CDs:** Audio is stored in PCM format.
- **Video Systems:** PCM handles audio in many video formats.

PCM is like taking snapshots of an analog signal at specific intervals, turning those snapshots into numbers, and then storing or transmitting the data in digital form!

(or)

Working of PCM:



The process of converting an analog signal into PCM involves several stages:

1. **Bandpass Filtering**: The analog signal is first passed through a bandpass filter that limits its frequency range. For example, in telephony, this range is typically **300 Hz to 3,000 Hz**, which corresponds to the frequency range of human speech.
2. **Sampling**: The analog signal is periodically sampled at discrete time intervals. The amplitude of the signal at each sample point is captured, typically using **natural sampling** or **flat-top sampling**.
3. **Quantization**: Each sampled signal is then quantized, which means it is mapped to the nearest value in a finite set of discrete levels. This process introduces **quantization error**, which is the difference between the actual signal value and the quantized value.
4. **Analog-to-Digital Conversion (ADC)**: The quantized values are converted into a digital binary code, often in parallel form.
5. **Parallel-to-Serial Conversion**: The parallel PCM codes are converted into a serial binary stream, which is transmitted over the communication channel.
6. **Transmission and Repeaters**: The serial binary data is transmitted through the channel. Repeaters are placed at regular intervals to regenerate the signal, compensating for attenuation and noise.
7. **Receiver Side**: At the receiver, the process is reversed:
    - **Serial-to-Parallel Conversion**: The serial data stream is converted back into parallel PCM codes.
    - **Digital-to-Analog Conversion (DAC)**: The parallel PCM codes are converted back into a multilevel pulse amplitude modulation (PAM) signal.
    - **Hold Circuit/Low Pass Filter**: The PAM signals are smoothed out and converted back to the original analog form.

An **integrated circuit (IC)** that performs both the encoding (PCM) and decoding (reverse PCM) functions is called a **codec** (coder/decoder).

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Dynamic Range It is the ratio of the largest possible magnitude to the smallest possible magnitude (other than 0V) that can be decoded by the digital-to-analog converter in the receiver. Mathematically,

$$DR = \frac{V_{max}}{V_{min}}$$

, where, DR is dynamic range and V_{min} is the quantum value (resolution) and V_{max} is the maximum voltage magnitude that can be discerned by the receivers DACs. Dynamic range can be expressed in decibels as,

$$DR_{(dB)} = 20 \log \frac{V_{max}}{V_{min}}$$ or $$DR_{(dB)} = 20 \log (2^n - 1)$$,where n is the number of PCM bits.

Comparison: Linear PCM vs. Nonlinear PCM

Feature	Linear PCM (LPCM)	Nonlinear PCM (NLPCM)
Quantization Levels	Evenly spaced across the range of the signal.	Unevenly spaced; smaller steps for low-amplitude signals.
Signal Representation	Accurately represents signals with uniform dynamic range.	Better representation of signals with a wide dynamic range.
Compression	No compression; all amplitudes are equally represented.	Companding reduces dynamic range and improves efficiency.
Bit Rate	Requires higher bit rate for high-quality representation.	More efficient bit rate usage for signals with large dynamic range.
Applications	High-fidelity audio, CDs, DVDs, digital storage.	Telephony, speech transmission, audio compression.
Complexity	Simpler to implement.	More complex due to companding and non-uniform quantization.
Advantages	Simple, accurate representation.	Better for signals with wide dynamic range; reduces quantization error in low-amplitude signals.
Disadvantages	Inefficient for signals with large dynamic range.	Not suitable for all types of signals, adds processing overhead.

Signal Voltage –To-Quantization Noise Voltage Ratio

The maximum quantization noise is half the resolution. Therefore, the worst possible signal voltage-to-quantization noise voltage ratio (SQR) occurs when the input signal is at its minimum amplitude. Mathematically, the worst-case voltage SQR is 2.

For linear PCM codes, the signal power-to-quantizing noise power ratio is determined by the formula:

$$SQR_{(dB)} = 10 \log \frac{v^2/R}{(q^2/12)/R}$$

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Time Division Multiplexing (TDM)

Time Division Multiplexing (TDM) is a method of transmitting multiple signals over a single communication channel by allocating a specific time slot to each signal in a sequential manner. Rather than combining the signals in frequency space (as in Frequency Division Multiplexing, FDM), TDM divides the time on the channel into small time intervals, or **time slots**, each dedicated to a different signal.

In TDM, each signal is transmitted in rapid succession, one after the other, during its allocated time slot. This method makes it possible to share the same physical medium (like a wire or radio spectrum) among multiple users or data streams without interference.

Key Concepts in TDM:

- **Multiplexing**: Combines multiple lower-speed channels into one higher-speed channel.
- **Time Slots**: Each user or signal gets a fixed, periodic time slot in the transmission cycle.
- **Data Rate**: The combined data rate of all signals is the sum of the individual rates.

Example:

Consider three signals, S1, S2, and S3, each transmitting at a rate of 1000 bps. In TDM, these signals are transmitted in sequence, with each signal occupying a fixed time slot. The combined signal will be transmitted at a rate of 3000 bps, with each signal getting one-third of the total bandwidth in a time-division manner.

Frame Synchronization in PCM-TDM

Frame synchronization refers to the process of aligning the start of each data frame so that the receiver knows where the beginning and end of the frame are. In systems like Pulse Code Modulation (PCM) used in Time Division Multiplexing (TDM), synchronization is crucial to ensure that each signal is correctly mapped to its corresponding time slot during transmission.

In a **PCM-TDM** system, digital data from multiple sources are multiplexed into a single higher-speed stream. Each frame of the multiplexed signal typically contains one sample from each of the channels in the system. Frame synchronization ensures that the receiver correctly identifies the beginning of each frame and thus knows which bits belong to which time slot and to which channel.

How Frame Synchronization is Achieved in a PCM-TDM System:

1. **Frame Structure**:
    - A frame consists of a sequence of time slots, each corresponding to a signal or user. In the case of PCM, each time slot carries one sample from each of the analog channels being digitized.
    - A **frame delimiter** or **synchronization bit pattern** is added at the beginning or end of each frame to mark the start or end of the frame. This pattern is recognizable by the receiver to align the data stream correctly.
2. **Synchronization Word/Pattern**:
    - The transmitter and receiver are preconfigured with a special word or bit pattern, called the **synchronization word**, at the beginning or end of each frame. This pattern is not part of the regular data and serves solely to help the receiver detect the start of a new frame.
    - This synchronization word could be a known sequence of bits, such as `101010`, which is easy to distinguish from regular data, ensuring the receiver can quickly identify the frame boundary.
3. **Frame Periodicity**:
    - TDM frames occur at regular intervals, and the receiver is expected to expect a new frame at a predictable rate. The synchronization word or pattern helps the receiver stay in sync with this periodic frame structure.
4. **Use of Special Bits**:
    - In some systems, a few extra bits may be inserted into the data stream at regular intervals. These bits carry synchronization information and are ignored by the receiver's data processing unit but are used to maintain accurate frame timing.
5. **Clock Recovery**:
    - TDM systems rely on **clock synchronization** between the transmitter and receiver to ensure data is correctly timed. The receiver extracts the clock from the incoming data stream using the synchronization pattern. If the timing deviates, the receiver can adjust its clock to maintain alignment with the transmitted data.

Summary of Frame Synchronization in PCM-TDM:

- **Purpose**: Frame synchronization ensures that the receiver knows the start of each frame, so it can correctly extract the data assigned to each time slot.

- **Achieved through**: Insertion of a synchronization word or pattern that marks the boundary between frames, allowing the receiver to align its timing with the transmitted data.
- **Clock Recovery**: Ensures that both the transmitter and receiver operate on synchronized clocks, allowing for correct data transmission without loss or misalignment.

In conclusion, frame synchronization is essential in PCM-TDM systems to properly identify the boundaries of each frame, ensuring the correct mapping of data from multiple sources into the multiplexed signal.

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Frame Synchronization

Frame synchronization ensures data is sent and received in the right sequence in Time Division Multiplexing (TDM). In TDM, frames are made up of individual time slots for each user, and synchronization identifies both the frame and its time slots.

Methods of Frame Synchronization

1. **Added-Digit Framing:**
   - A special framing bit is added to every frame.
   - Synchronization time depends on the number of bits and the frame period.
2. **Robbed-Digit Framing:**
   - The least important bit of every nth frame is replaced with a framing bit.
   - Maintains synchronization without interrupting transmission but introduces small data errors.
3. **Added-Channel Framing:**
   - Instead of adding bits, entire groups or words are added for synchronization.
   - Efficiency depends on the number of bits per frame and synchronizing word length.
4. **Statistical Framing:**
   - Uses patterns in the data (like a high chance of a "1" in specific bits) for synchronization without adding or robbing digits.
5. **Unique-Line Code Framing:**
   - Makes the framing bit stand out (e.g., higher amplitude or longer duration).
   - Synchronization is fast but requires extra processing.

Frequency Division Multiplexing (FDM)

FDM assigns **different frequency ranges** to multiple signals so they can share the same transmission medium simultaneously.

How FDM Works:

- Each signal is modulated to a unique frequency band.
- **Guard bands** (unused frequency gaps) prevent interference.
- Common in **radio, TV, and cable systems**.

Advantages:

1. Easy to add new users.
2. Supports full-duplex communication.
3. Less noise impact.

Disadvantages:

1. High initial cost (cables and equipment).
2. Issues with one user can affect others.
3. Requires precise frequency management.

Wavelength Division Multiplexing (WDM)

WDM uses **light waves** of different wavelengths (colors) to transmit multiple signals through a single fiber optic cable. It's like FDM but for optical signals.

How WDM Works:

- Each signal is modulated onto a unique wavelength of light.
- Light signals are combined and transmitted through the fiber, then separated at the receiver.

Advantages:

1. Increases capacity using a single fiber.
2. Easy to add/remove channels.
3. Optical components are reliable and cost-effective.

Disadvantages:

1. Signals must not interfere (wavelengths need proper spacing).
2. All signals need consistent strength.
3. Works only along predefined fiber routes.

Dense Wavelength Division Multiplexing (DWDM)

DWDM packs more wavelengths closely together, increasing the data capacity of a single fiber.

Advantages:

1. High capacity with low cost.
2. Compatible with multiple protocols (e.g., Ethernet, ATM).
3. Scalable and uses existing fibers effectively.

Disadvantages:

1. Issues like dispersion and attenuation can limit performance.
2. Four-wave mixing (non-linear interference) reduces capacity.

Synchronous Optical Network (SONET)

SONET is a **high-speed optical fiber transmission system** designed for data communication. It works like TDM but is optimized for optical fibers.

Key Features of SONET:

1. Speeds start at **51.84 Mbps** (STS-1) and can go up to several Gbps (e.g., OC-48 at 2.48 Gbps).
2. Can easily add/remove individual data streams (e.g., DS-1 signals) without disrupting the whole frame.

Applications:

1. High-speed internet backbone.
2. Supports **B-ISDN** (broadband networks).
3. Forms the base for **ATM (Asynchronous Transfer Mode)** and other high-speed optical networks.