DATA AND SIGNALS

1.WHAT IS ANALOG SIGNAL AND DIGITAL SIGNAL?

- **ANALOG SIGNAL:** An analog signal has infinitely many levels of intensity over a period of time. As the wave moves from value A to value B, it passes through and includes an infinite number of values along its path. The term analog data refers to information that is continuous.

For example, an analog clock that has hour, minute, and second hands gives information in a continuous form; the movements of the hands are continuous. Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal.

- **DIGITAL SIGNAL:** A digital signal, on the other hand, can have only a limited number of defined values. Although each value can be any number, it is often as simple as 1 and O.

For example digital clock that reports the hours and the minutes will change suddenly from 8:05 to 8:06. Digital data take on discrete values. For example, data are stored in computer memory in the form of Os and 1s.

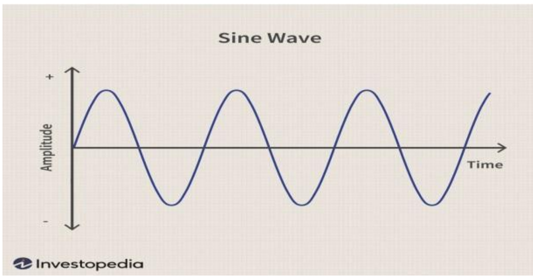
- **PERIODIC AND NON PERIODIC ANALOG**: A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a cycle.

A nonperiodic signal changes without exhibiting a pattern or cycle that repeats over time.

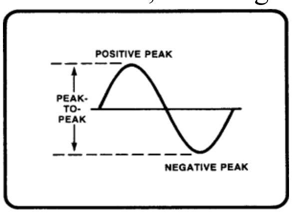
2.PERIODIC ANALOG SIGNALS: Periodic analog signals can be classified as simple or composite. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

Sine wave: The sine wave is the most fundamental form of a periodic analog signal. When we visualize it as a simple oscillating curve, its change over the course of a cycle is smooth and consistent, a continuous, rolling flow.

A sine wave can be represented by three parameters: the peak amplitude, the frequency, and the phase. These three parameters fully describe a sine wave.



Peak amplitude: The peak amplitude of a signal is the absolute value of its highest intensity, proportional to the energy it carries. For electric signals, peak amplitude is normally measured in volts.
Period and Frequency: Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle



Wavelength :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

Time and Frequency Domains: The time-domain plot shows changes in signal amplitude with respect to time. . A frequency-domain plot is concerned with only the peak value and the frequency.

Composite Signals: . A composite signal is made of many simple sine waves. For example, the power company sends a single sine wave with a frequency of 60 Hz to distribute electric energy to houses and businesses.

Bandwidth: The range of frequencies contained in a composite signal is its bandwidth. The bandwidth is normally a difference between two numbers. For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 - 1000, or 4000.

3. DIGITAL SIGNALS: A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.

Bit Rate: Most digital signals are nonperiodic, and thus period and frequency are not appropriate characteristics. Another *term-bit rate is* used to describe digital signals. The bit rate is the number of bits sent in 1s, expressed in bits per second (bps).
Bit Length: The bit length is the distance one bit occupies on the transmission 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. Attenuation is measured in terms of Decibels.

The decibel (dB) measures the relative strengths of two signals or one signal at two different points. Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.

$$dB=10\log_{10} P2/P1$$

Variables *PI* and *P2* are the powers of a signal at points 1 and 2, respectively.

.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 through a medium and, therefore, its own delay in arriving at the final destination.

Latency(Delay)**:** 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, bg queuing time and processing delay. Latency =propagation time +transmission time +queuing time + processing delay

Bandwidth The range of frequencies contained in a composite signal is its bandwidth. The bandwidth is normally a difference between two numbers. For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 - 1000, or 4000.

Bandwidth in Hertz We have discussed this concept. Bandwidth in hertz is the range of frequencies contained in a composite signal or the range of frequencies a channel can pass. For example, we can say the bandwidth of a subscriber telephone line is 4 kHz.

Bandwidth in Bits per Seconds The term bandwidth can also refer to the number of bits per second that a channel, a link, or even a network can transmit. For example, one can say the bandwidth of a Fast Ethernet network (or the links in this network) is a maximum of 100 Mbps. This means that this network can send 100 Mbps. Relationship There is an explicit relationship between the bandwidth in hertz and bandwidth in bits per seconds. Basically, an increase in bandwidth in hertz means an increase in bandwidth in bits per second.

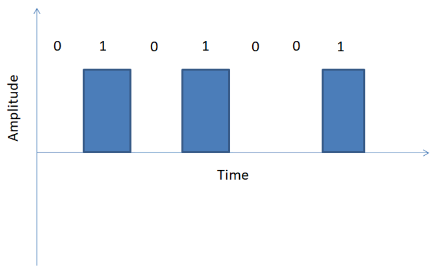
Throughput: The throughput is a measure of how fast we can actually send data through a network. Although, at first glance, bandwidth in bits per second and throughput seem the same, they are different. A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B. In other words, the bandwidth is a potential measurement of a link; the throughput is an actual measurement of how fast we can send data. For example, we may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps. This means that we cannot send more than 200 kbps through this link. Imagine a highway designed to transmit 1000 cars per minute from one point to another.

Digital to Digital Conversion :

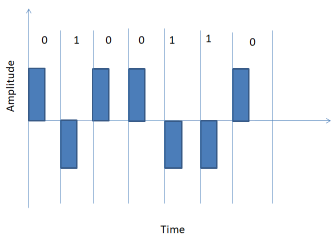
Digital-to-digital conversion in networking refers to the process of transforming digital data from one format or encoding to another for efficient transmission or storage. It is primarily used to prepare data for transmission across networks or to decode received data for interpretation by devices. Below are the key aspects of digital-to-digital conversion

1. Line Coding

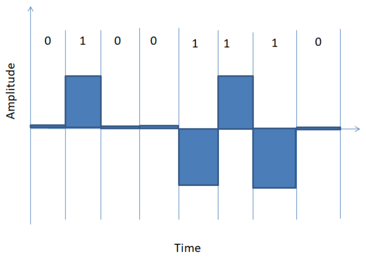
- **Definition:** Line coding involves converting digital data into a digital signal for transmission across a communication medium (e.g., cables, fiber optics).
- **Examples of Line Coding Schemes:**
    - **Unipolar Encoding:** Uses a single voltage level for binary data, e.g., 0 (low voltage) and 1 (high voltage).



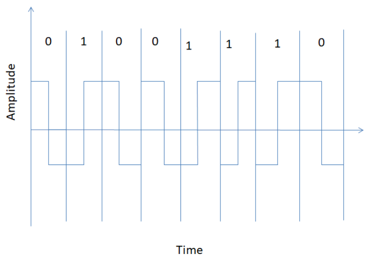
    - **Polar Encoding:** Uses two voltage levels (positive and negative) to represent binary data. Examples:
        - **NRZ (Non-Return to Zero):** Voltage remains constant during the bit interval (e.g., NRZ-L and NRZ-I).
        - **RZ (Return to Zero):** The signal returns to zero halfway through the bit interval.



    - **Bipolar Encoding:** Alternates between positive and negative voltages for successive 1s (e.g., AMI: Alternate Mark Inversion).



- **Manchester Encoding:** Combines clock and data, with transitions at the middle of the bit interval (1: high-to-low, 0: low-to-high).



ANALOG-TO-DIGITAL CONVERSION

- The techniques described above convert digital data to digital signals.
- Sometimes, however, we have an analog signal such as one created by a microphone or camera.
- We must keep in mind that a digital signal is superior to an analog signal.
- The tendency today is to change an analog signal to digital data.
- This change is achieved by Pulse Code Modulation.
- After the digital data are created (digitization), we can use technique described above i.e. Line coding and Block Coding to convert the digital data to a digital signal.

PULSE CODE MODULATION (PCM)

- The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
- Pulse-code modulation (PCM) is a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals of duration.
- This process is called sampling.
- Every sample is converted to a series of symbols in a digital code, which is usually a binary code.

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

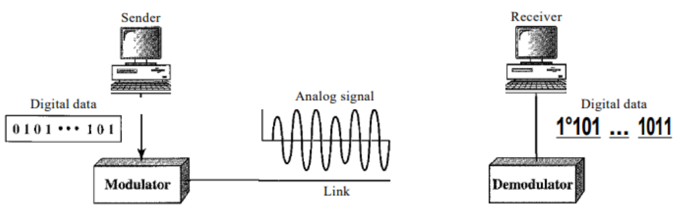
Delta Modulation (DM) :

PCM is a very complex technique. Other techniques have been developed to reduce the complexity of PCM. The simplest is delta modulation. PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample. Figure 4.28 shows the process. Note that there are no code words here; bits are sent one after another.

4.DIGITAL-TO-ANALOG CONVERSION: Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data. Figure 5.1 shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.

A sine wave is defined by three characteristics: amplitude, frequency, and phase. When we vary anyone of these characteristics, we create a different version of that wave. So, by changing one characteristic of a simple electric signal, we can use it to represent digital data. Any of the three characteristics can be altered in this way, giving us at least three mechanisms for modulating digital data into an analog signal: amplitude shift keying (ASK), frequency shift keying (FSK) , and phase shift keying (PSK).

Figure 5.1 *Digital-to-analog conversion*



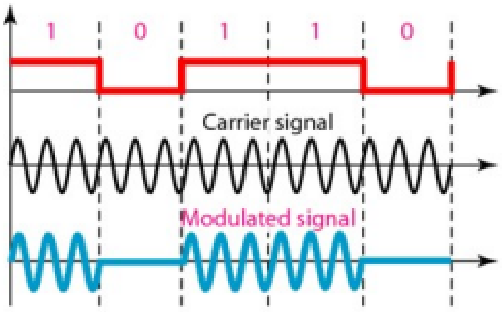
Aspects of Digital-to-Analog Conversion: Before we discuss specific methods of digital-to-analog modulation, two basic issues must be reviewed: bit and baud rates and the carrier signal.Data Element Versus Signal Element Data element as the smallest piece of information to be exchanged, the bit. We also defined a signal element as the smallest unit of a signal that is constant.

Data Rate Versus Signal Rate: We can define the data rate (bit rate) and the signal rate (baud rate). The relationship between them is S= N/r baud where N is the data rate (bps) and r is the number of data elements carried in one signal element. The value of r in analog transmission is r =log2 L, where L is the type of signal element, not the level.

Carrier Signal: In analog transmission, the sending device produces a high-frequency signal that acts as a base for the information signal. This base signal is called the carrier signal or carrier frequency. The receiving device is tuned to the frequency of the carrier signal that it expects from the sender.Digital information then changes the carrier signal by modifying one or more of its characteristics (amplitude, frequency, or phase). This kind of modification is called modulation (shift keying).

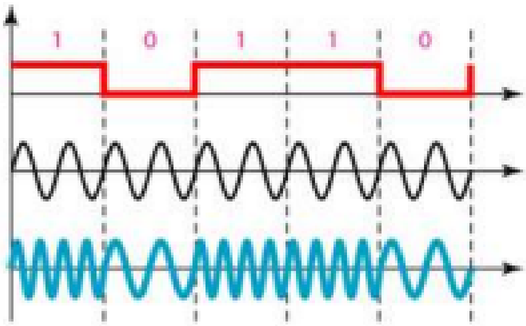
1. **Amplitude Shift Keying (ASK):** In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes. Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels. This is referred to as binary amplitude shift keying or on-off keying (OOK). The peak amplitude

of one signal level is 0; the other is the same as the amplitude of the carrier frequency.

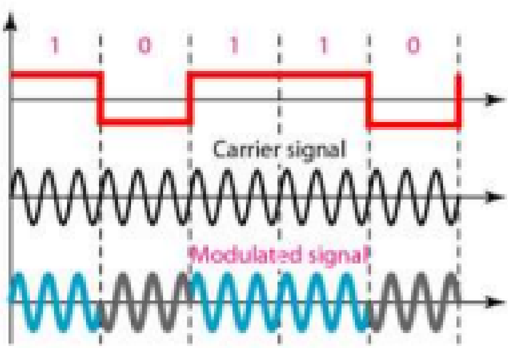


2. Frequency Shift Keying (FSK): In frequency shift keying, the frequency of the carrier signal is varied to represent data. The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements.

One way to think about binary FSK (or BFSK) is to consider two carrier frequencies., we have selected two carrier frequencies,f1 andf2. We use the first carrier if the data element is 0; we use the second if the data element is 1. However, note that this is n unrealistic example used only for demonstration purposes. Normally the carrier frequencies are very high, and the difference between them is very small.



2. **Phase Shift Keying (PSK) :**In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes. Today, PSK is more common than ASK or FSK. The simplest PSK is binary PSK, in which we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°.

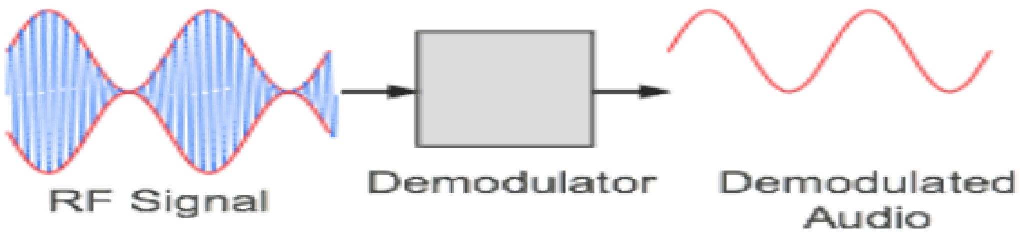


5.Analog to analog conversion:

A. AM (Amplitude demodulation) :

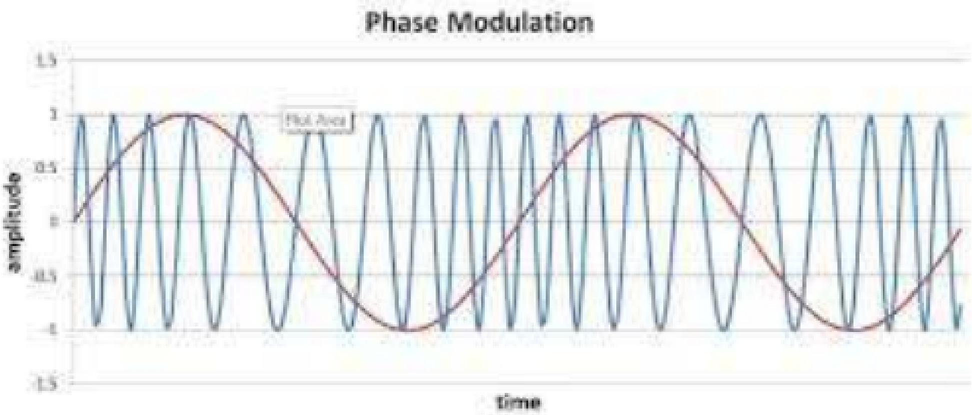
In order that a radio signal can carry audio or other information for broadcasting or for two way radio communication, it must be modulated or changed in some way. Although there are a number of ways in which a radio signal may be modulated, one of the easiest easiest is to change its amplitude in line with variations of the sound.

modulation, PM is sometimes used for analogue transmission, but it has become the basis for modulation schemes used for



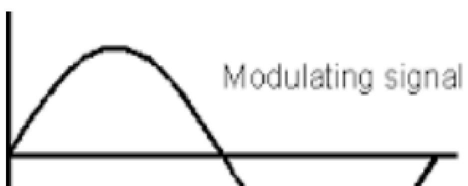
B. PM (Phase modulation) :

Phase carrying data. Phase shift keying, PSK is widely used for data communication.Phase modulation is also the basis of a form of modulation known as quadrature amplitude modulation, modulation, where both phase and amplitude amplitude are varied to provide additional capabilities.



C. FM(Frequency modulation) :

As with any form of modulation, it is necessary to be able to successfully demodulate it and recover the original signal. The FM



demodulator may be called a variety of names including FM demodulator, FM detector or an FM discriminator.

There are a number of different types of FM demodulator, but all of them enable the • There are a number of different types of FM demodulator, but all of them enable the frequency variations of the incoming signal to be converted into amplitude variations on the output. These are typically fed into an audio amplifier, or possibly a digital interface if data is being passed over the system.

6.Analog-to-Digital Conversion (ADC)

Analog-to-digital conversion is the process of converting an analog signal, which is a continuous wave, into a digital signal, which is a discrete sequence of numbers. This conversion is essential for many electronic devices, as digital signals are easier to process, store, and transmit.

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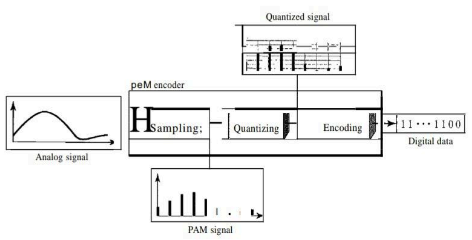
Figure 4.21 *Components ofPCM encoder*



1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

The Process of ADC

Sampling: The analog signal is sampled at regular intervals. The sampling rate must be at least twice the highest frequency component of the analog signal to avoid aliasing.

Quantization: The sampled values are rounded to the nearest discrete level. The number of levels depends on the resolution of the ADC, which is typically measured in bits.

Encoding: The quantized values are encoded into a binary code, which is a sequence of 0s and 1s.

Delta Modulation (DM) :

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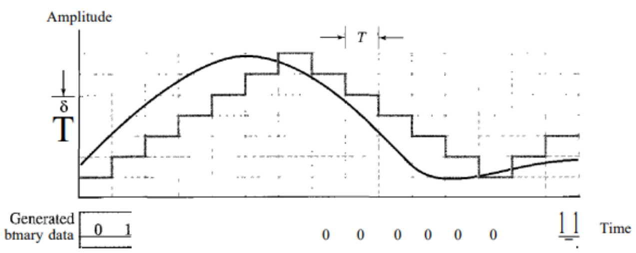
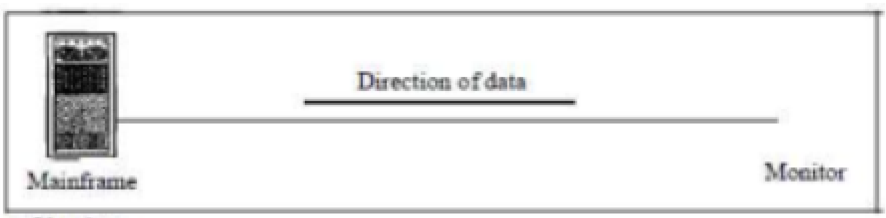
Figure 4.28 *The process ofdelta modulation*



7.DATA TRANSMISSION MODES:

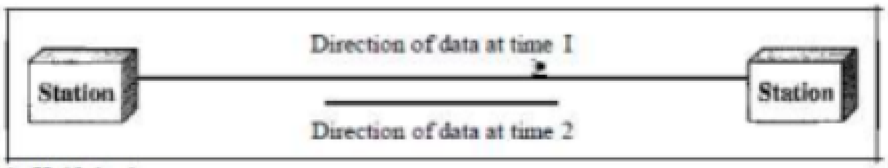
Communication between two devices can be simplex, half-duplex, or full-duplex.

Simplex: In simplex mode, the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit; the other can only receive. Keyboards and traditional monitors are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output. The simplex mode can use the entire capacity of the channel to send data in one direction.



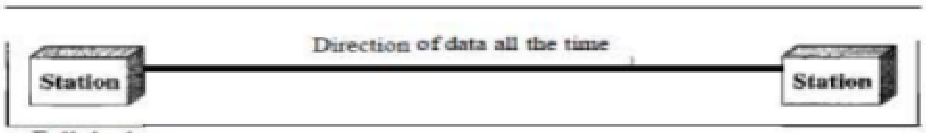
a. Simplex

Half-Duplex: In half-duplex mode, each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa. In a half-duplex transmission , the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time. Walkie-talkies and CB (citizens band) radios are both half-duplex systems. The half- duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of channel can be utilized for each direction..



b. Half-duplex

Full-Duplex: In full-duplex both stations can transmit and receive simultaneously. The full-duplex mode is like a two way street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link: with signals going in the other direction. One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time. The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.



c. Full-duplex

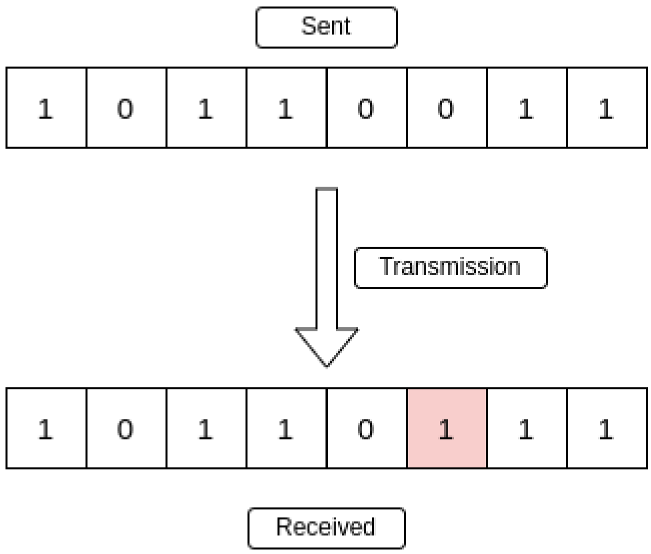
8. Error detection and correction

- Error is a condition when the receiver's information does not match the sender's. Digital signals suffer from noise during transmission that can introduce errors in the binary bits traveling from sender to receiver. That means a 0 bit may change to 1 or a 1 bit may change to 0.
- Data (Implemented either at the Data link layer or Transport Layer of the OSI Model) may get scrambled by noise or get corrupted whenever a message is transmitted. To prevent such errors, error-detection codes are added as extra data to digital messages. This helps in detecting any errors that may have occurred during message transmission.

Types of Errors

1. Single-Bit Error

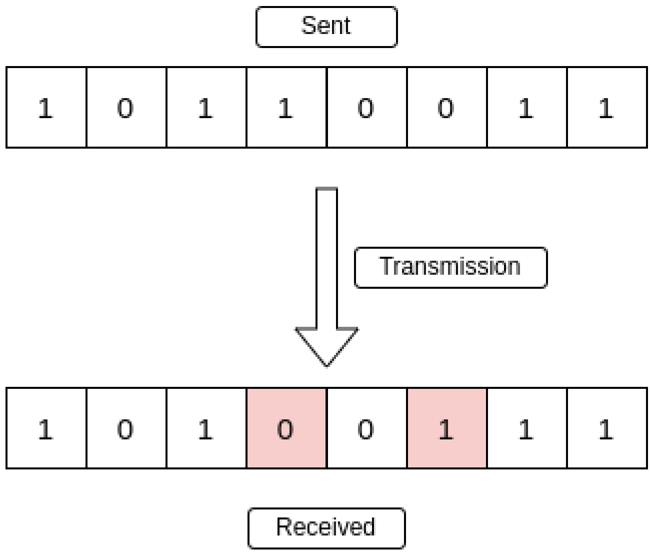
A single-bit error refers to a type of data transmission error that occurs when one bit (i.e., a single binary digit) of a transmitted data unit is altered during transmission, resulting in an incorrect or corrupted data unit.



Single-Bit Error

2. Multiple-Bit Error

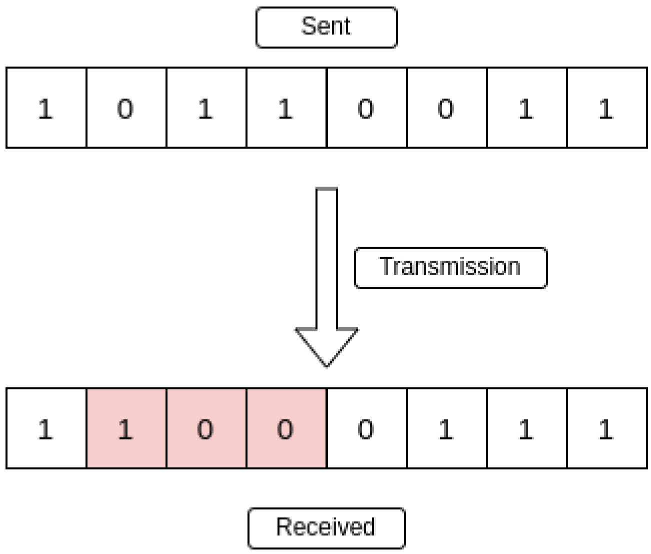
A multiple-bit error is an error type that arises when more than one bit in a data transmission is affected. Although multiple-bit errors are relatively rare when compared to single-bit errors, they can still occur, particularly in high-noise or high-interference digital environments.



Multiple-Bit Error

3. Burst Error

When several consecutive bits are flipped mistakenly in digital transmission, it creates a burst error. This error causes a sequence of consecutive incorrect values.



Burst Error

Error Detection Methods

To detect errors, a common technique is to introduce redundancy bits that provide additional information. Various techniques for error detection include:

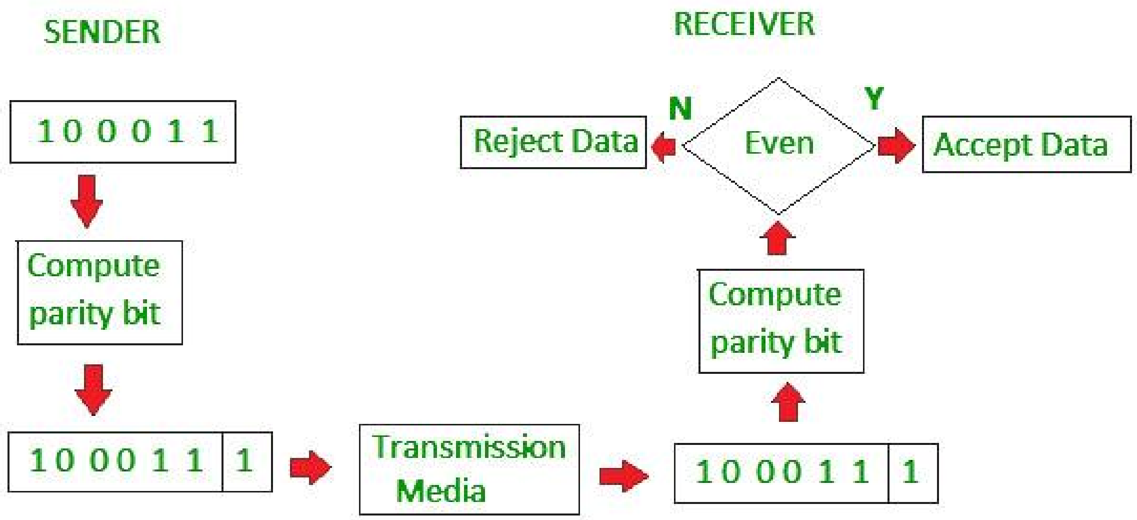
- Simple Parity Check
- Two-Dimensional Parity Check
- Checksum
- Cyclic Redundancy Check (CRC)

Simple Parity Check

Simple-bit parity is a simple error detection method that involves adding an extra bit to a data transmission. It works as:

- 1 is added to the block if it contains an odd number of 1's, and
- 0 is added if it contains an even number of 1's

This scheme makes the total number of 1's even, that is why it is called even parity checking.



Advantages of Simple Parity Check

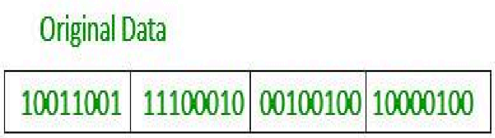
- Simple parity check can detect all single bit error.
- Simple parity check can detect an odd number of errors.
- **Fast Error Detection**: The process of calculating and checking the parity bit is quick, which allows for rapid error detection without significant delay in data processing or communication.
- **Single-Bit Error Detection**: It can effectively detect single-bit errors within a data unit, providing a basic level of error detection for relatively low-error environments.

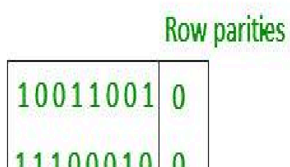
Disadvantages of Simple Parity Check

- Single Parity check is not able to detect even no. of bit error.
- **For example,** the Data to be transmitted is **101010**. Codeword transmitted to the receiver is 1010101 (we have used even parity).
  Let's assume that during transmission, two of the bits of code word flipped to 1111101.
  On receiving the code word, the receiver finds the no. of ones to be even and hence **no error,** which is a wrong assumption.

Two-Dimensional Parity Check

Two-dimensional Parity check bits are calculated for each row, which is equivalent to a simple parity check bit. Parity check bits are also calculated for all columns, then both are sent along with the data. At the receiving end, these are compared





with the parity bits calculated on the received data.

Advantages of Two-Dimensional Parity Check

- Two-Dimensional Parity Check can detect and correct all single bit error.
- Two-Dimensional Parity Check can detect two or three bit error that occur any where in the matrix.

Disadvantages of Two-Dimensional Parity Check

- Two-Dimensional Parity Check can not correct two or three bit error. It can only detect two or three bit error.
- If we have a error in the parity bit then this scheme will not work.

Checksum

Checksum error detection is a method used to identify errors in transmitted data. The process involves dividing the data into equally sized segments and using a [1's complement] to calculate the sum of these segments. The calculated sum is then sent along with the data to the receiver. At the receiver's end, the same process is repeated and if all zeroes are obtained in the sum, it means that the data is correct.

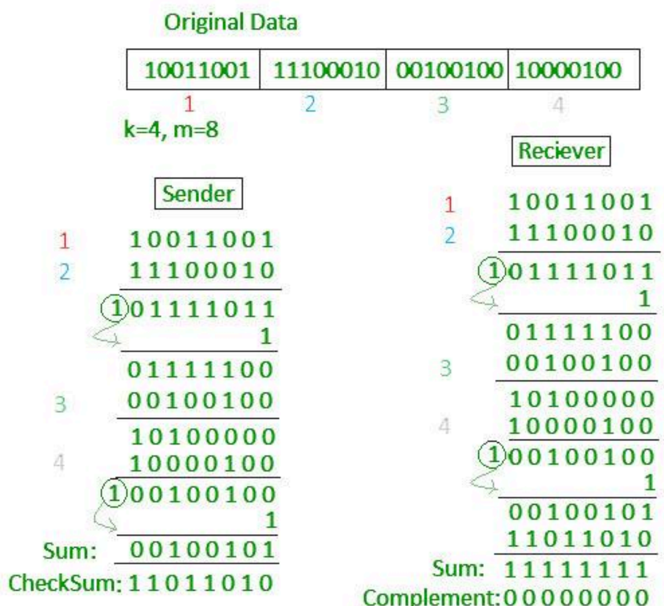
Checksum – Operation at Sender's Side

- Firstly, the data is divided into k segments each of m bits.
- On the sender's end, the segments are added using 1's complement arithmetic to get the sum. The sum is complemented to get the checksum.
- The checksum segment is sent along with the data segments.

Checksum – Operation at Receiver's Side

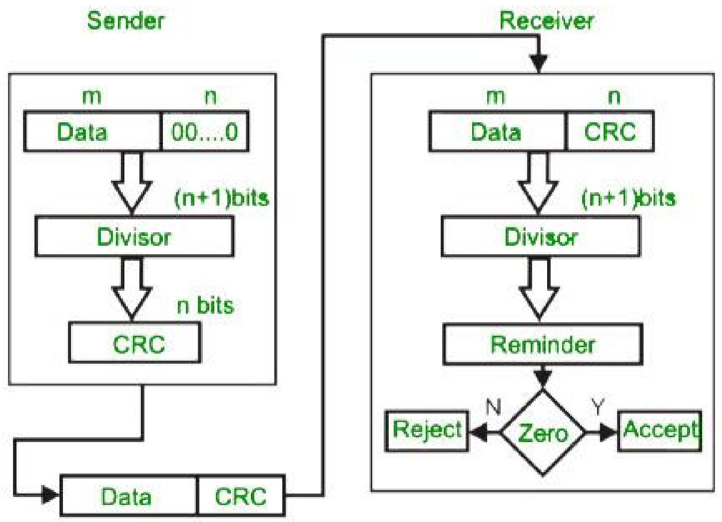
At the receiver's end, all received segments are added using 1's complement arithmetic to get the sum. The sum is complemented.

If the result is zero, the received data is accepted; otherwise discarded.



Cyclic Redundancy Check (CRC)

- Unlike the checksum scheme, which is based on addition, CRC is based on binary division.
- In CRC, a sequence of redundant bits, called cyclic redundancy check bits, are appended to the end of the data unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number.
- At the destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit is assumed to be correct and is therefore accepted.
- A remainder indicates that the data unit has been damaged in transit and therefore must be rejected.



CRC Working

We have given dataword of length n and divisor of length k.

Step 1: Append (k-1) zero's to the original message

Step 2: Perform modulo 2 division

Step 3: Remainder of division = CRC

Step 4: Code word = Data with append k-1 zero's + CRC

Note:

- CRC must be k-1 bits
- Length of Code word = n+k-1 bits

<u>Error correction :</u>

Hamming Code

Hamming code is an error-correcting code used to ensure data accuracy during transmission or storage. Hamming code detects and corrects the errors that can occur when the data is moved or stored from the sender to the receiver. This simple and effective method helps improve the reliability of communication systems and digital storage. It adds extra bits to the original data, allowing the system to detect and correct single-bit errors. It is a technique developed by Richard Hamming in the 1950s.

Algorithm of Hamming Code

Hamming Code is simply the use of extra parity bits to allow the identification of an error.

Step 1: Write the bit positions starting from 1 in binary form (1, 10, 11, 100, etc).

Step 2: All the bit positions that are a power of 2 are marked as parity bits (1, 2, 4, 8, etc).

Step 3: All the other bit positions are marked as data bits.

Step 4: Each data bit is included in a unique set of parity bits, as determined its bit position in binary form:

- **a.** Parity bit 1 covers all the bits positions whose binary representation includes a 1 in the least significant position (1, 3, 5, 7, 9, 11, etc).
- **b.** Parity bit 2 covers all the bits positions whose binary representation includes a 1 in the second position from the least significant bit (2, 3, 6, 7, 10, 11, etc).
- **c.** Parity bit 4 covers all the bits positions whose binary representation includes a 1 in the third position from the least significant bit (4–7, 12–15, 20–23, etc).
- **d.** Parity bit 8 covers all the bits positions whose binary representation includes a 1 in the fourth position from the least significant bit bits (8–15, 24–31, 40–47, etc).
- **e.** In general, each parity bit covers all bits where the bitwise AND of the parity position and the bit position is non-zero.