**Muhammad Israr Akbar     ID 11396**

Q1. Select the correct answer of the given ones.

1) Interactive transmission of data independent of a time sharing system may be best suited to
   (b) **half-duplex lines**

2) The loss in the signal power as of an Electromagnetic signal is called
   (a) **Attenuation**

3) Early detection of packet losses improves _____________ acknowledgment performance.
   (c) **positive**

4) Additional signal introduced in the desired signal in producing hypes is called
   (d) **Dispersion**

5) Token is a 3 **byte Frame** that rotates around the ring.

6) Ring may have up to _____________ (802.5) or _____________ (IBM) nodes.

7) FDDI can support a maximum of ____**5000**___________ stations.

8) Error-correcting codes are __________________ enough to handle all errors.

9) ACK is a small ____________________ confirming reception of an earlier frame

10) Electronics are _________________ as compared to optics

**Q 2:- Distinguish between error correction and error detection. Explain any two error detection techniques with mathematical examples other than given in slides, search from internet?**

## Distinguish between error correction and error detection.

Digital bit plays a vital role in the processing and communication in which the signals are converted into digital bits. These digital bits are then processed and transmitted from one system to another system. These are data bits are grouped and transmitted which are called data stream. In this entire steam if any single bit changes during processing or communication, it leads a high error in the output. With increase in digital techniques the errors are also high due to noise and these effects the entire data steam if it is not corrected because these errors will be propagated if they are not detected and corrected.

The errors can occur in the systems or in the communication network and to avoid the errors in the communication network, the network should be able to transfer the data from one device to another device with highest accuracy. In the systems level the errors must be detected and corrected at the input acceptance check for reliable processing.

**Bit Error:** if a bit 1 is changed to 0 or 0 changed to 1 due to noise and it is called bit error.

**Two error detection techniques with mathematical examples:-**

## Types of Errors

- Single-Bit Error
- Burst Error

## Single-Bit Error

The only one bit of a given data unit is changed from 1 to 0 or from 0 to 1.

## Burst Error

The two or more bits are changed from 0 to 1 or from 1 to 0 is known as Burst Error.

The Burst Error is determined from the first corrupted bit to the last corrupted bit.

## Error Detection with Example:-

The most popular Error Detecting Techniques are:

- Longitudinal Redundancy Check (LRC)
- Hamming distance based checks

## Longitudinal Redundancy Check (LRC)

A longitudinal redundancy check (LRC) is an error-detection method for determining the correctness of transmitted and stored data.

LRC verifies the accuracy of stored and transmitted data using parity bits. It is a redundancy check applied to a parallel group of bit streams. The data to be transmitted is divided into transmission blocks into which additional check data is inserted.

This term is also known as a horizontal redundancy check.

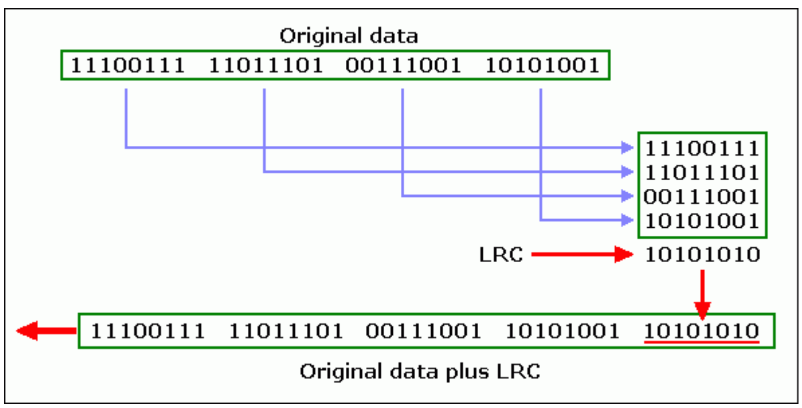
LRC fields consist of one byte containing an eight bit binary value. LRC values are calculated by transmitting devices, which append LRC to messages. The device at the receiving end recalculates the LRC on receipt of the message and compares the calculated value to the actual value received in the LRC field. If the values are equal, the transmission was successful; if the values are not equal, this indicates an error.

LRC is generated through the following steps:

i. Add all bytes in messages excluding the starting colon and the ending the carriage return line feed.
ii. Add this to the eight-bit field and discard the carries.
iii. Subtract the final field value from FF hex, producing one's complement.
iv. Add one, producing two's complement.

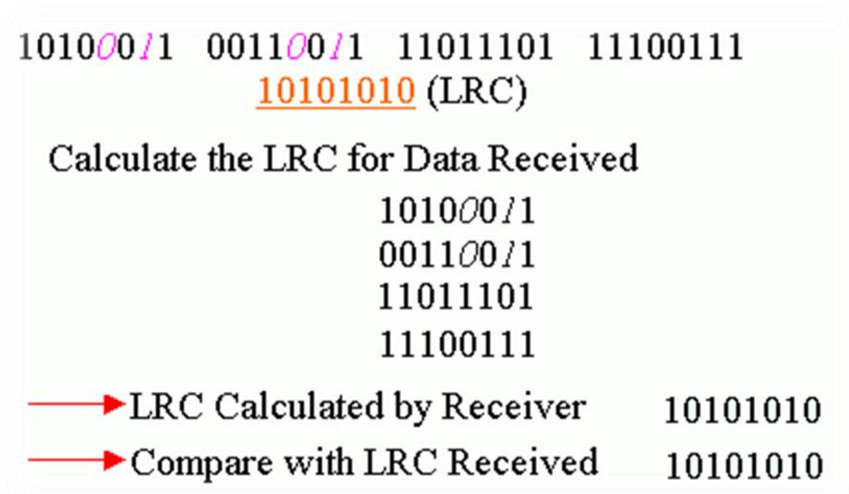
## LRC Code Generation

In this error detection method, a block of bits is organized in a table with rows and columns. Then the parity bit for each column is calculated and a new row of eight bits, which are the parity bits for the whole block, is created. After that the new calculated parity bits are attached to the original data and sends to the receiver.




LRC increases the likelihood of detecting burst error. An LRC of n bits can easily detects a burst error of n bits.

However, if two bits in one data unit are damaged and two bits in exactly the same positions in another data unit are also damaged, the LRC checker will not detect an error.



How LRC Fail to Detect the Burst Noise

Notice that although the 5th bit and the 7th bit for $1^{st}$ and $2^{nd}$ data unit have been changed but the LRC calculated by receiver is still the same as the LRC received. Thus the receiver checker cannot detect this burst error.

## Hamming distance based checks

Hamming distance is a metric for comparing two binary data strings. While comparing two binary strings of equal length, Hamming distance is the number of bit positions in which the two bits are different.

The Hamming distance between two strings, a and b is denoted as d(a,b).

It is used for error detection or error correction when data is transmitted over computer networks. It is also using in coding theory for comparing equal length data words.

## Calculation of Hamming Distance

In order to calculate the Hamming distance between two strings, and we perform their XOR operation, ($a \oplus b$), and then count the total number of 1s in the resultant string.

## Example 1:

Suppose there are two strings 1101 1001 and 1001 1101.

11011001 ⊕ 10011101 = 01000100. Since, this contains two 1s, the Hamming distance, d(11011001, 10011101) = 2.

## Minimum Hamming Distance

In a set of strings of equal lengths, the minimum Hamming distance is the smallest Hamming distance between all possible pairs of strings in that set.

## Example 2:

Suppose there are four strings 010, 011, 101 and 111.

010 ⊕ 011 = 001, d(010, 011) = 1.

010 ⊕ 101 = 111, d(010, 101) = 3.

010 ⊕ 111 = 101, d(010, 111) = 2.

011 ⊕ 101 = 110, d(011, 101) = 2.

011 ⊕ 111 = 100, d(011, 111) = 1.

101 ⊕ 111 = 010, d(011, 111) = 1.

Hence, the Minimum Hamming Distance, $d_{min}$ = 1.

## Q3: What is encoding? Write down different types of encoding. Explain characteristics of AM, FM and PM with mathematical equations?

Encoding is the process of converting data from one form to another during transmission it over some computer networks. While "encoding" can be used as a verb, it is often used as a noun, and refers to a specific type of encoded data. There are several types of encoding, including image encoding, audio and video encoding, and character encoding.  OR

The encoding of a message is the production of the message. It is a system of coded meanings, and in order to create that, the sender needs to understand how the world is comprehensible to the members of the audience.

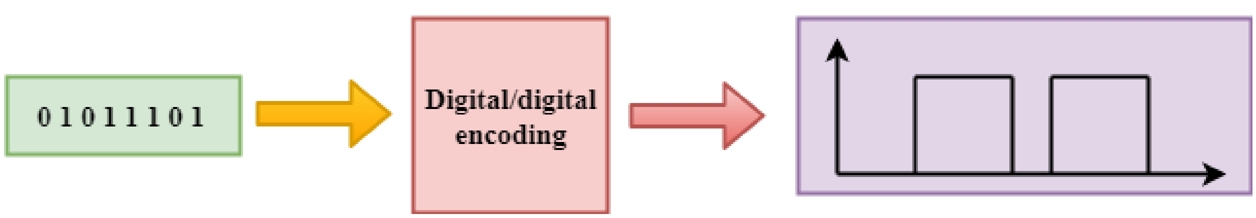
In the process of encoding, the sender (i.e. encoder) uses verbal (e.g. words, signs, images, video) and non-verbal (e.g. body language, hand gestures, face expressions) symbols for which he or she believes the receiver (that is, the decoder) will understand. The symbols can be words and numbers, images, face expressions, signals and/or actions. It is very important how a message will be encoded; it partially depends on the purpose of the message.

## Digital Transmission

Data can be represented either in analog or digital form. The computers used the digital form to store the information. Therefore, the data needs to be converted in digital form so that it can be used by a computer.

## DIGITAL-TO-DIGITAL CONVERSION

Digital-to-digital encoding is the representation of digital information by a digital signal. When binary 1s and 0s generated by the computer are translated into a sequence of voltage pulses that can be propagated over a wire, this process is known as digital-to-digital encoding.

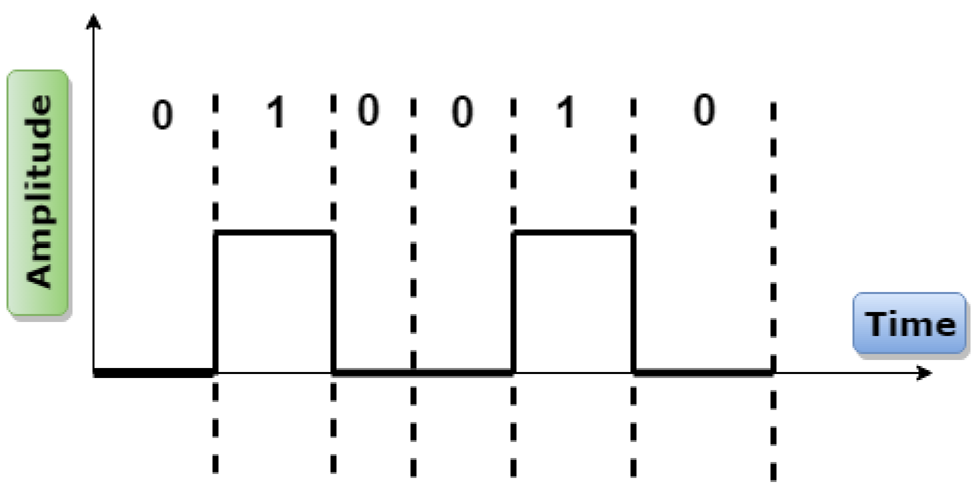



## Digital-to-digital encoding is divided into three categories:

- **Unipolar Encoding**
- **Polar Encoding**
- **Bipolar Encoding**

## Unipolar

- Digital transmission system sends the voltage pulses over the medium link such as wire or cable.
- In most types of encoding, one voltage level represents 0, and another voltage level represents 1.
- The polarity of each pulse determines whether it is positive or negative.
- This type of encoding is known as Unipolar encoding as it uses only one polarity.
- In Unipolar encoding, the polarity is assigned to the 1 binary state.
- In this, 1s are represented as a positive value and 0s are represented as a zero value.
- In Unipolar Encoding, '1' is considered as a high voltage and '0' is considered as a zero voltage.
- Unipolar encoding is simpler and inexpensive to implement.




## Polar

- Polar encoding is an encoding scheme that uses two voltage levels: one is positive, and another is negative.
- By using two voltage levels, an average voltage level is reduced, and the DC component problem of unipolar encoding scheme is alleviated.

## Bipolar

- Bipolar encoding scheme represents three voltage levels: positive, negative, and zero.
- In Bipolar encoding scheme, zero level represents binary 0, and binary 1 is represented by alternating positive and negative voltages.
- If the first 1 bit is represented by positive amplitude, then the second 1 bit is represented by negative voltage, third 1 bit is represented by the positive amplitude and so on. This alternation can also occur even when the 1bits are not consecutive.

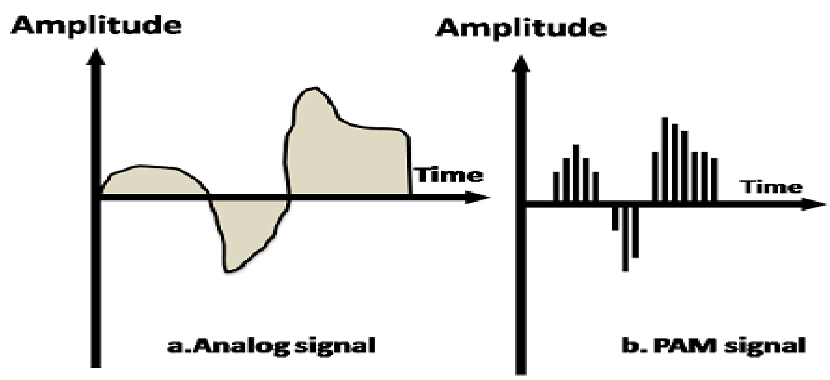
# ANALOG-TO-DIGITAL CONVERSION

- When an analog signal is digitalized, this is called an analog-to-digital conversion.
- Suppose human sends a voice in the form of an analog signal, we need to digitalize the analog signal which is less prone to noise. It requires a reduction in the number of values in an analog message so that they can be represented in the digital stream.
- In analog-to-digital conversion, the information contained in a continuous wave form is converted in digital pulses.

## Techniques for Analog-To-Digital Conversion

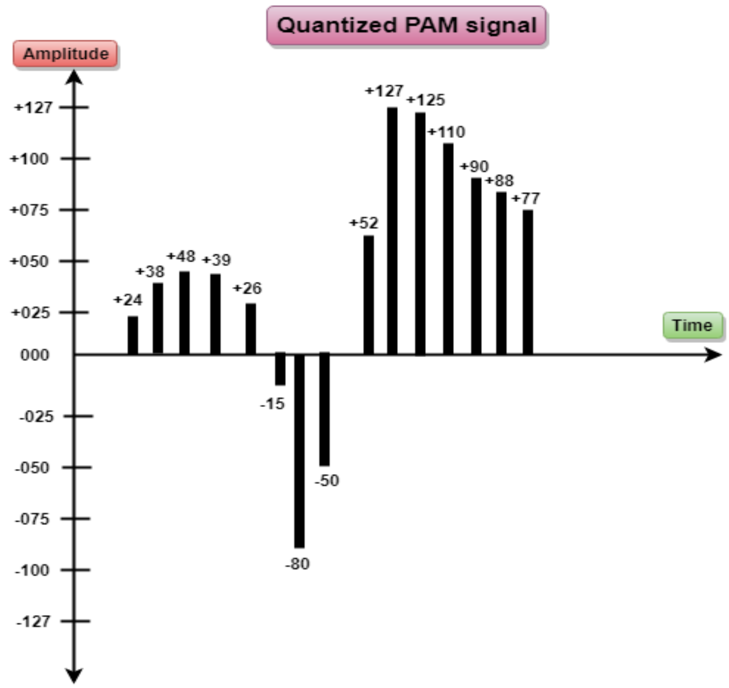
## PAM

- PAM stands for pulse amplitude modulation.
- PAM is a technique used in analog-to-digital conversion.
- PAM technique takes an analog signal, samples it, and generates a series of digital pulses based on the result of sampling where sampling means measuring the amplitude of a signal at equal intervals.
- PAM technique is not useful in data communication as it translates the original wave form into pulses, but these pulses are not digital. To make them digital, PAM technique is modified to PCM technique.




## PCM

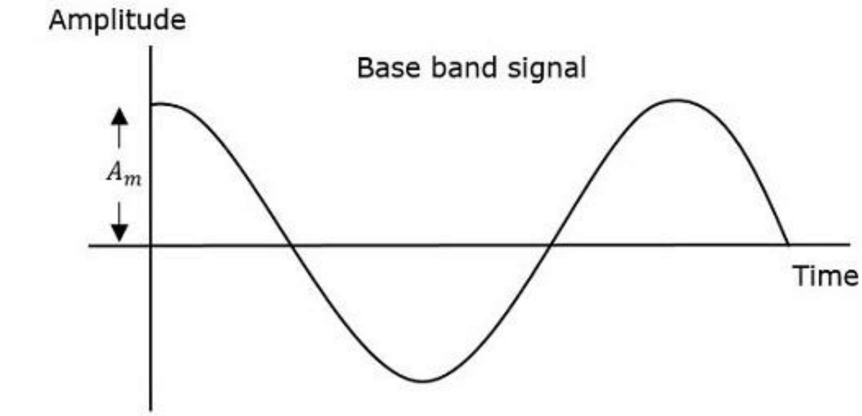
- PCM stands for **Pulse Code Modulation**.
- PCM technique is used to modify the pulses created by PAM to form a digital signal. To achieve this, PCM quantizes PAM pulses. Quantization is a process of assigning integral values in a specific range to sampled instances.
- PCM is made of four separate processes: PAM, quantization, binary encoding, and digital-to-digital encoding.

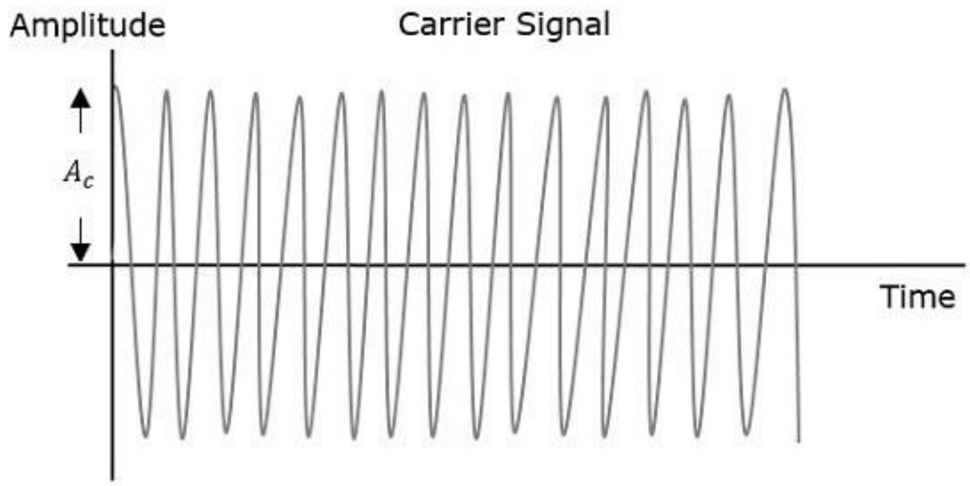


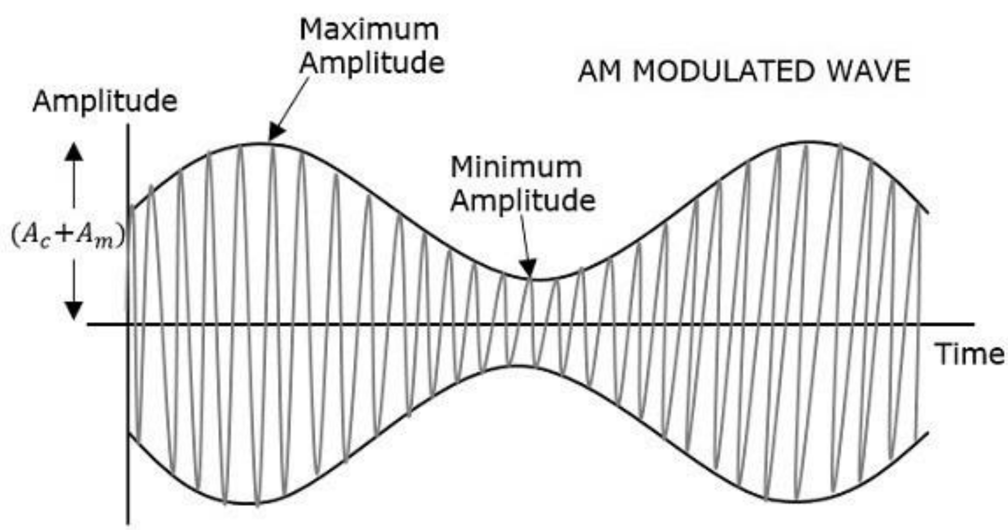

## Characteristics of Amplitude Modulation (AM)

A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.

According to the standard definition, "The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures.


Amplitude
Base band signal
$A_m$
Time


Amplitude
Carrier Signal
$A_c$
Time


Maximum
Amplitude
AM MODULATED WAVE
Amplitude
Minimum
Amplitude
$(A_c+A_m)$
Time

The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.

## Mathematical Expressions

Following are the mathematical expressions for these waves.

Time-domain Representation of the Waves

Let the modulating signal be,

$$m(t)=Amcos(2\pi fmt)m(t)=Amcos(2\pi fmt)$$

and the carrier signal be,

$$c(t)=Accos(2\pi fct)c(t)=Accos(2\pi fct)$$

Where,

AmAm and AcAc are the amplitude of the modulating signal and the carrier signal respectively.

fmfm and fcfc are the frequency of the modulating signal and the carrier signal respectively.

Then, the equation of Amplitude Modulated wave will be

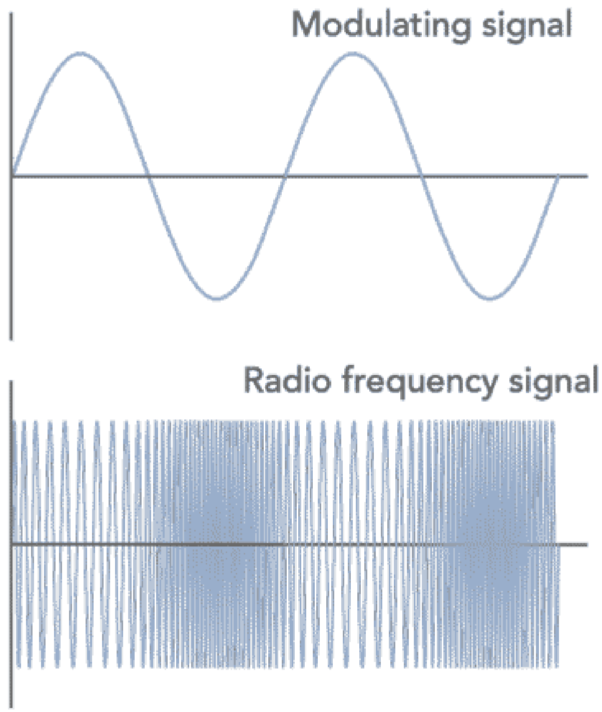
$s(t)=[Ac+Amcos(2\pi fmt)]cos(2\pi fct)$

### Characteristics of Frequency Modulation (FM)

While changing the amplitude of a radio signal is the most obvious method to modulate it, it is by no means the only way. It is also possible to change the frequency of a signal to give frequency modulation or FM. Frequency modulation is widely used on frequencies above 30 MHz, and it is particularly well known for its use for VHF FM broadcasting.

Although it may not be quite as straightforward as amplitude modulation, nevertheless frequency modulation, FM, offers some distinct advantages. It is able to provide near interference free reception, and it was for this reason that it was adopted for the VHF sound broadcasts. These transmissions could offer high fidelity audio, and for this reason, frequency modulation is far more popular than the older transmissions on the long, medium and short wave bands.

In addition to its widespread use for high quality audio broadcasts, FM is also used for a variety of two way radio communication systems. Whether for fixed or mobile radio communication systems, or for use in portable applications, FM is widely used at VHF and above.




When the audio signal is modulated onto the radio frequency carrier, the new radio frequency signal moves up and down in frequency. The amount by which the signal moves up and down is important. It is known as the deviation and is normally quoted as the number of kilohertz deviation. As an example the signal may have a deviation of plus and minus 3 kHz, i.e. ±3 kHz. In this case the carrier is made to move up and down by 3 kHz.

Broadcast stations in the VHF portion of the frequency spectrum between 88.5 and 108 MHz use large values of deviation, typically ±75 kHz. This is known as wide-band FM (WBFM). These signals are capable of supporting high quality transmissions, but occupy a large amount of bandwidth. Usually 200 kHz is allowed for each wide-band FM transmission. For communications purposes less bandwidth is used. Narrow band FM (NBFM) often uses deviation figures of around ±3 kHz.

It is narrow band FM that is typically used for two-way radio communication applications. Having a narrower band it is not able to provide the high quality of the wideband transmissions, but this is not needed for applications such as mobile radio communication.

## FM Equation

The basic FM equation is presented in Equation:

$$y(t)=A\sin(2\pi f_c t+I\sin(2\pi f_m t)),$$

where the parameters are defined as follows:

- fc= carrier frequency (Hz)
- fm= modulation frequency (Hz)
- I= modulation index

The Figure screen cast video continues the discussion by explaining the significance of each part of Equation, and demonstrates in a qualitative fashion how the different parameters of the equation influence the spectrum of the audio signal.

## Characteristics of Phase Modulation (FM)

We have all heard of AM radio and FM radio. But phase modulation seems to be in a different category—"**PM** radio" is by no means a common term. It turns out that phase modulation is more relevant in the context of digital RF. In a way, though, we can say that PM radio is as common as FM radio simply because there is little difference between phase modulation and frequency modulation. FM and PM are best considered as two closely related variants of *angle modulation*, where "angle" Phase modulation is similar to frequency modulation and is an important technique in digital communication systems.

## The Math

We saw in the previous page that frequency modulation is achieved by adding the integral of the baseband signal to the argument of a sine or cosine function (where the sine or cosine function represents the carrier

$$x_{FM}(t) = \sin(\omega_C t + \int_{-\infty}^{t} x_{BB}(t)dt)$$

You will recall, though, that we introduced frequency modulation by first discussing phase modulation: adding the baseband signal itself, rather than the integral of the baseband signal, causes the phase to vary according to the baseband value. Thus, phase modulation is actually a bit simpler than frequency modulation.

$$x_{PM}(t) = \sin(\omega_C t + x_{BB}(t))$$

As with frequency modulation, we can use the modulation index to make the phase variations more sensitive to the changes in the baseband value:

$$x_{PM}(t) = \sin(\omega_C t + m x_{BB}(t))$$

The similarity between phase modulation and frequency modulation becomes clear if we consider a single-frequency baseband signal. Let's say that xBB(t) = sin(ωBBt). The integral of sine is negative cosine (plus a constant, which we can ignore here)—in other words, the integral is simply a time-shifted version of the original signal. Thus, if we perform phase modulation and frequency modulation with this baseband signal, the only difference in the modulated waveforms will be the alignment between the baseband value and the variations in the carrier; the variations themselves are the same. This will be more clear in the next section, where we'll look at some time-domain plots.

It's important to keep in mind that we're dealing with instantaneous phase, just as frequency modulation is based on the concept of instantaneous frequency. The term "phase" is rather vague. One familiar meaning refers to the initial state of a sinusoid; for example, a "normal" sine wave begins with a value of zero and then increases toward its maximum value. A sine wave that begins at a different point in its cycle has a phase offset. We can also think of phase as a specific portion of a full waveform cycle; for example, at a phase of π/2, a sinusoid has completed one-fourth of its cycle.

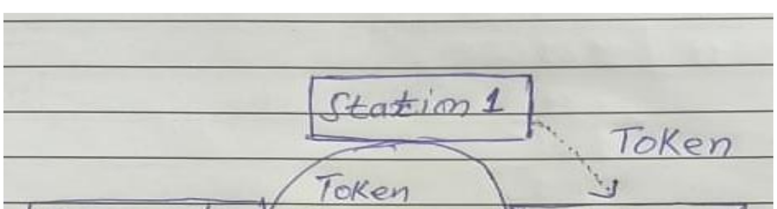
These interpretations of "phase" don't help us very much when we're dealing with a phase that continuously varies in response to a baseband waveform. Rather, we use the concept of *instantaneous* phase, i.e., the phase at a given moment, which corresponds to the value passed (at a given moment) to a trigonometric function. We can think of these continuous variations in instantaneous phase as "pushing" the carrier value farther from or closer to the preceding state of the waveform.

One more thing to keep in mind: Trig functions, including sine and cosine, operate on angles. Changing the argument of a trig function is equivalent to changing the angle, and this explains why both FM and PM are described as angle modulation.

**Q4: Compare Ethernet and Token Ring concept of data networking with diagrams. Which one is better in your opinion and why?**

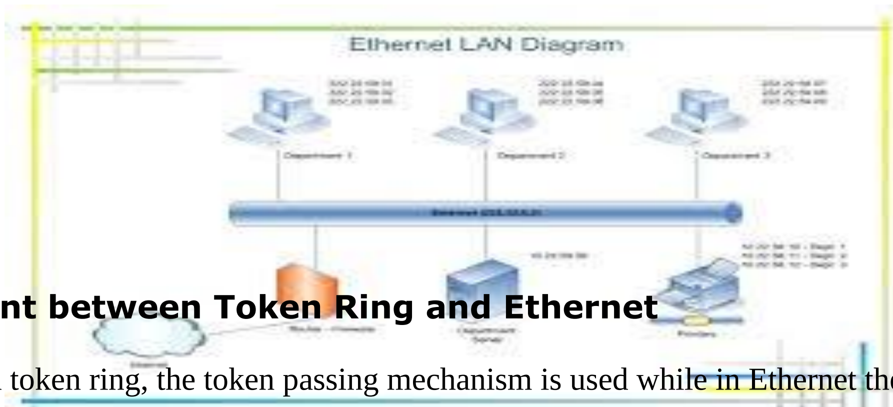
## 1. Token Ring:

Token ring is a data networking concept where tokens are passed over the network. This concepts was defined by IEEE 802.5 standard. The token ring network uses a station and a special frame called token. A station in this networking concept can transmit a data frame when it holds a token. When a successful transmission of data frame is made, then the tokens are issued. Token ring is a Star shaped topology and handles priority in which some nodes may give priority to the token. The below figure represents a token ring.



## 2. Ethernet:

Ethernet was defined IEEE 802.3 standard. The main mechanism behind Ethernet is the CSMA/CD mechanism. The CSMA/CD mechanism means that if many stations exists at the same time to communicate, all stations will be closed. O start again the station, a random resume can be made. The Ethernet network doesn’t employ any priorities. And it is less costly than token ring network.




## Different between Token Ring and Ethernet

- In token ring, the token passing mechanism is used while in Ethernet the CSMA/CD mechanism is used.
- Token ring is a deterministic network while Ethernet is non deterministic.
- Token ring is based on start topology while Ethernet uses bus topology.
- The token ring handles priority in which some nodes may give priority to the token while the Ethernet does not allow priority.
- Token ring costs more while Ethernet has low cost compare to token ring.
- Token ring uses telephone cable while Ethernet uses coaxial cable.

## Ethernet is better than token Ring. Justification:

Ethernet has much cheaper cost as compared to token ring which has higher cost. Ethernet networks are much faster than token ring. Token ring as comparatively old protocol than Ethernet so Ethernet can work efficiently with cheap cost and faster speed. Also, the cables used in Ethernet have higher security than that of Token ring.

**Q5. Explain the concept and review of Reliable Transmission with diagram (from a research paper of 2019 or 2020) and its functionality. The name and reference of paper should be given**?

Area: Wireless Sensor Network

Paper Title: New approach of multi-path reliable transmission for marginal wireless sensor network

Explanation:

In the application environment having dense distribution of marginal wireless sensor network (WSN), the data transmission process will generate a large number of conflicts, which will result in loss of transmission data and increase of transmission delay. The multi-path data transmission method can effectively solve the problem of large data loss and transmission delay caused by collisions. . A new approach of multi-path reliable transmission for application of marginal WSN (named RCBMRT) was proposed in this paper. The approach was built on the network topology after the hierarchical clustering is completed. The original data packet is divided into several sub-packets according to the routing information, and then the concurrent multi-path is used. The mode is transferred to the aggregation node of marginal WSN. Compared with the existing CB-RACO protocol, DE-MRT protocol and FRTSMC protocol, the data loss rate is reduced, the delay is reduced, the reliability of reliable data transmission is improved, and the life cycle of the network of marginal WSN is prolonged.

The paper introduces the mechanism of concurrent braided multipath reliable transmission routing protocol based on redundancy strategy in marginal WSN. For ease of research, the agreed marginal WSN network model has the following properties:

(1) With high density characteristics, isolated nodes will not appear under initial conditions.

(2) The BS is unique and the energy is not limited.

(3) The initial energy of the sensor node is, and the energy cannot be increased

This approach adopts redundancy mechanism to realize the reliability of data transmission in the marginal WSN, and uses concurrent woven multi-path technology to improve the transmission efficiency of data packets. It divides the data packets that the sensor node needs to transmit into several sub-packets with data redundancy, and then forwards the sub-packets to the aggregation node through multi-path by the intermediate nodes of marginal environment.

At the same time, they have done the relative experiments according to some more parameters like throughput and routing overhead to test the approach. Based on the experimental results, they found that the throughput of this approach has added about 20% than that of the best existed approach (such as FRTSMC protocol), but the routing overhead has only added about 3%.

Reference:

Zhang, D. G., Wu, H., Zhao, P. Z., Liu, X. H., Cui, Y. Y., Chen, L., & Zhang, T. (2020). New approach of multi-path reliable transmission for marginal wireless sensor network. Wireless Networks, 26(2), 1503-1517.