LECTURE NOTES

On

Analog and Digital Communication
B.Tech, 6th Semester, EEE

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COURSE CONTENT

Analog and Digital Communication

B.Tech , 6^{th} Semester, EEE

Review of signals and system:-

Review of signals and systems, Frequency domain representation of signals, Principles of Amplitude Modulation Systems- DSB, SSB and VSB modulations. Angle Modulation, Representation of FM and PM signals, Spectral characteristics of angle modulated signals.

Review of probability and random process:-

Gaussian and white noise characteristics, Noise in amplitude modulation systems,Noise in Frequency modulation systems. Pre-emphasis and Deemphasis, Threshold effect in angle modulation.

Pulse modulation:-

Sampling process. Pulse Amplitude and Pulse code modulation (PCM), Differential pulse code modulation. Delta modulation, Noise considerations in PCM, Time Division multiplexing, Digital Multiplexers.

Elements of Detection Theory:-

Optimum detection of signals in noise, Coherent communication with waveforms- Probability of Error evaluations. Base band Pulse Transmission- Inter symbol Interference and Nyquist criterion. Pass band Digital Modulation schemes- Phase Shift Keying, Frequency Shift Keying, Quadrature Amplitude Modulation, Continuous Phase Modulation and Minimum Shift Keying.

Digital Modulation

Trade-offs. Optimum demodulation of digital signals over band-limited channels- Maximum likelihood sequence detection (Viterbi receiver). Equalization Techniques. Synchronization and Carrier Recovery for Digital modulation.

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B.Tech, 6th Semester, CSE

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Digital Learning Resources:

[1] CourseName:Analogcommunication Course Link: https://nptel.ac.in/noc/courses/noc19/SEM2/noc19-ee46

[2] Course Instructor: Prof. Goutam Das, IIT Kharagpur B. Tech (ECE/ ETE) Syllabus from Admission Batch 2018-19 5th Semester.

[3] Course Name: Modern Digital Communication Techniques Course Link: https://nptel.ac.in/courses/117/105/117105144/ Course Instructor: Prof. S.S. Das, IIT Kharagpur.

[4] Course Name: Communication Engineering Course Link: https://nptel.ac.in/courses/117/102/117102059/

CHAPTER-I

Review of signals and system

[Review of signals and systems, Frequency domain representation of signals, Principles of Amplitude Modulation Systems- DSB, SSB and VSB modulations. Angle Modulation, Representation of FM and PM signals, Spectral characteristics of angle modulated signals.]

1.1 Signal and System

Signal and System is a fundamental concept in electrical engineering and applied mathematics that deals with analyzing, processing, and understanding signals and the systems that manipulate them. It is widely used in communication, control systems, signal processing, and more.

1. Signals

A **signal** is a function that conveys information. It can be:

- **Continuous-Time (CT) Signals:** Defined for every value of time (e.g., speech, radio waves).
- **Discrete-Time (DT) Signals:** Defined at discrete time intervals (e.g., digital signals).

Types of Signals:

- **Deterministic vs. Random Signals:** Predictable vs. unpredictable signals.
- **Periodic vs. Aperiodic Signals:** Signals that repeat over time vs. those that do not.
- **Even vs. Odd Signals:** Symmetric vs. anti-symmetric signals.
- **Energy vs. Power Signals:** Based on finite or infinite energy.

2. Systems

A **system** processes an input signal to produce an output signal.

Types of Systems:

- **Linear vs. Nonlinear:** Follows superposition vs. does not.
- **Time-Invariant vs. Time-Variant:** System behavior is constant vs. changes over time.
- **Causal vs. Non-Causal:** Depends on past/present inputs vs. future inputs.
- **Stable vs. Unstable:** Produces bounded output vs. unbounded output.

3. System Representation

- **Impulse Response (h(t) or h[n]):** The system's response to a unit impulse signal.
- **Convolution:** Determines output for any input using impulse response.
- **Fourier Series and Fourier Transform:** Frequency domain representation of signals.
- **Laplace Transform & Z-Transform:** Used for system analysis and stability check.

4. Applications

- **Digital Signal Processing (DSP):** Audio/image processing, communications.
- **Control Systems:** Robotics, automation, feedback control.
- **Communication Systems:** Modulation, filtering, noise reduction.

1.2 Frequency Domain Representation of Signals

The **frequency domain representation** describes signals in terms of their frequency components instead of time. This is useful in analyzing and processing signals, especially in communication and control systems.

1. Why Frequency Domain?

- Many real-world signals are combinations of sinusoids of different frequencies.
- Frequency analysis simplifies operations like filtering and modulation.
- It helps in understanding how a system responds to different frequency components.

2. Fourier Series (For Periodic Signals)

A periodic signal $x(t)$ can be expressed as a sum of sinusoidal components:

$$x(t) = \sum_{n=-\infty}^{\infty} C_n e^{jn\omega_0 t}$$

where:

- C_n are the Fourier coefficients, representing the amplitude and phase of each frequency component.
- $\omega_0 = \frac{2\pi}{T}$ is the fundamental angular frequency.
- The signal is represented as a sum of harmonics.

3. Fourier Transform (For Aperiodic Signals)

For non-periodic signals, the **Fourier Transform** is used to convert a time-domain signal into the frequency domain:

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-j2\pi ft}dt$$

$$x(t) = \int_{-\infty}^{\infty} X(f)e^{j2\pi ft}df$$

where:

- $X(f)$ is the frequency spectrum of $x(t)$, representing its frequency content.
- The **Magnitude Spectrum** $|X(f)|$ shows how much of each frequency is present.
- The **Phase Spectrum** $\angle X(f)$ represents the phase shift of each frequency component.

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4. Properties of Fourier Transform

1. **Linearity:** $ax_1(t) + bx_2(t) \rightarrow aX_1(f) + bX_2(f)$
2. **Time Shifting:** $x(t - t_0) \rightarrow X(f)e^{-j2\pi ft_0}$
3. **Frequency Shifting:** $x(t)e^{j2\pi f_0 t} \rightarrow X(f - f_0)$
4. **Convolution Theorem:** $x_1(t) * x_2(t) \rightarrow X_1(f)X_2(f)$

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5. Laplace and Z-Transforms (Extensions of Fourier Transform)

- **Laplace Transform:** Used for system analysis in the **s-domain**.
- **Z-Transform:** Used for discrete-time signals in the **z-domain**.

6. Applications of Frequency Domain Analysis

- **Filtering:** Low-pass, high-pass, and band-pass filters.
- **Modulation:** Used in communication systems (AM, FM, PM).
- **Spectral Analysis:** Used in speech, audio, and biomedical signal processing.

1.3 Principles of Amplitude Modulation (AM) Systems

Amplitude Modulation (AM) is a method of transmitting information by varying the **amplitude** of a high-frequency carrier wave in proportion to a message signal while keeping the frequency and phase constant.

1. Basic Concept of AM

An **AM wave** is generated by modulating a high-frequency carrier $c(t)$ with a low-frequency message signal $m(t)$.

- **Carrier signal:**

$$c(t) = A_c \cos(2\pi f_c t)$$

 where A_c is the carrier amplitude and f_c is the carrier frequency.

- **Message signal (modulating signal):**

$$m(t) = A_m \cos(2\pi f_m t)$$

 where A_m is the message amplitude and f_m is the message frequency.

- **AM Signal:**

$$s(t) = [A_c + m(t)] \cos(2\pi f_c t)$$

 Expanding $m(t)$:

$$s(t) = A_c \cos(2\pi f_c t) + A_m \cos(2\pi f_m t) \cos(2\pi f_c t)$$

2. Modulation Index (Depth of Modulation)

The **modulation index** μ determines how much the carrier amplitude varies:

$$\mu = \frac{A_m}{A_c}$$

- $\mu < 1$: **Undermodulation** (Low-quality signal).
- $\mu = 1$: **100% Modulation** (Optimal).
- $\mu > 1$: **Overmodulation** (Distortion occurs).

3. Frequency Spectrum of AM Signal

The AM signal contains three main frequency components:

- **Carrier frequency** f_c
- **Upper Sideband (USB)** $f_c + f_m$
- **Lower Sideband (LSB)** $f_c - f_m$

Bandwidth of AM:

$$B = 2f_m$$

where f_m is the highest frequency of the message signal.

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4. Types of AM Systems

1. **Double Sideband with Carrier (DSB-AM):** Standard AM with both sidebands and carrier.
2. **Double Sideband Suppressed Carrier (DSB-SC):** Carrier is suppressed, reducing power wastage.
3. **Single Sideband (SSB-AM):** Only one sideband is transmitted, saving bandwidth.

5. Power in AM Signals

Total power in AM:

$$P_{total} = P_c \left(1 + \frac{\mu^2}{2}\right)$$

where $P_c = \frac{A_c^2}{2}$ is the carrier power.

- Carrier power is wasted in standard AM.
- Suppressing the carrier (DSB-SC) or using SSB improves efficiency.

Advantages:

✅ Simple transmitter and receiver design.
✅ Used in long-range communication (e.g., AM radio).

Disadvantages:

❌ Low power efficiency (carrier consumes unnecessary power).
❌ Susceptible to noise and interference.
❌ Requires more bandwidth than necessary (except SSB).

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7. Applications of AM Systems

- AM Radio Broadcasting (DSB-AM)
- Aviation Communication (SSB-AM)
- Television Video Transmission (VSB-AM)
- Two-way Radio Communication

1.4 Double Sideband (DSB) Amplitude Modulation

Double Sideband (DSB) is a type of amplitude modulation (AM) where both the **upper sideband (USB)** and **lower sideband (LSB)** are transmitted. It can be classified into two types:

1. **DSB with Carrier (DSB-AM)** – Standard AM with a transmitted carrier.
2. **DSB-Suppressed Carrier (DSB-SC)** – Carrier is suppressed to save power.

1. Mathematical Representation of DSB-SC

A message signal $m(t)$ modulates a high-frequency carrier $c(t)$.

- **Message Signal:**

$$m(t) = A_m \cos(2\pi f_m t)$$

- **Carrier Signal:**

$$c(t) = A_c \cos(2\pi f_c t)$$

- **DSB-SC Modulated Signal:**

$$s(t) = m(t)c(t) = A_m A_c \cos(2\pi f_m t) \cos(2\pi f_c t)$$

Using the trigonometric identity:

$$\cos A \cos B = \frac{1}{2}[\cos(A + B) + \cos(A - B)]$$

We get:

$$s(t) = \frac{A_m A_c}{2} [\cos(2\pi(f_c + f_m)t) + \cos(2\pi(f_c - f_m)t)]$$

2. Power in DSB-SC

Since the carrier is suppressed, the **total power** is concentrated in the sidebands.

$$P_{DSB-SC} = \frac{A_m^2 A_c^2}{4}$$

This makes DSB-SC more **power-efficient** than standard AM.

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3. Generation of DSB-SC

Balanced Modulator

- A nonlinear device that removes the carrier while allowing sidebands to pass.
- Uses **multiplication** of $m(t)$ and $c(t)$.

Ring Modulator

- Uses diodes and transformers to remove the carrier.
- Commonly used in analog signal processing.

4. Detection (Demodulation) of DSB-SC

Coherent Detection (Synchronous Demodulation)

- Requires a **synchronous local oscillator** at the receiver.
- Multiplies the received signal with the same carrier frequency.
- A low-pass filter extracts the original message.

$$s(t) \cdot \cos(2\pi f_c t) = \frac{A_m A_c}{2} \cos(2\pi f_m t) + \frac{A_m A_c}{2} \cos(2\pi(2f_c - f_m)t)$$

- The **low-pass filter** removes the high-frequency component, leaving $m(t)$.
- Requires **phase synchronization**, otherwise **distortion** occurs.

5. Advantages & Disadvantages of DSB-SC

✅ Advantages:

- More power-efficient than standard AM.
- Carrier does not waste power.
- Used in suppressed-carrier transmission like stereo FM.

❌ Disadvantages:

- Requires a **coherent receiver** (carrier synchronization needed).
- More complex demodulation than standard AM.

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6. Applications of DSB-SC

- **Broadcasting and Two-Way Communication**
- **Single Sideband (SSB) Generation** (SSB starts from DSB-SC)
- **Speech Processing and Telephony**

1.5 What is SSB

Single Sideband (SSB) is a type of **amplitude modulation (AM)** where only **one sideband** (either **upper** or **lower**) is transmitted, while the other and the carrier are suppressed.

Why SSB:--* Standard **AM** transmits **both** sidebands and the **carrier**, wasting bandwidth and power.

*__SSB__ saves bandwidth and improves power efficiency.

3. SSB Signal Generation

(A) Frequency Discrimination Method (Filtering)

- Generate **DSB-SC** and use a **bandpass filter** to remove one sideband.
- Used for **high-frequency** signals like HF radio.

(B) Phase Shift Method

- Uses **Hilbert Transform** to shift phase by 90°.
- Generates **SSB** without filtering.
- Used for **low-frequency** signals like audio processing.

5. Advantages & Disadvantages of SSB

✅ Advantages:

✔ **Bandwidth Efficiency** (Uses half the bandwidth of AM).

✔ **Power Efficiency** (No carrier power wasted).

✔ **Better Signal-to-Noise Ratio (SNR).**

❌ Disadvantages:

✖ **Complex Demodulation** (Requires phase synchronization).

✖ **Expensive Receivers.**

✖ **Distortion if the carrier is not properly reinserted.**

6. Applications of SSB

📡 **Long-Distance HF (High-Frequency) Communication** (Military, Aviation, Marine).

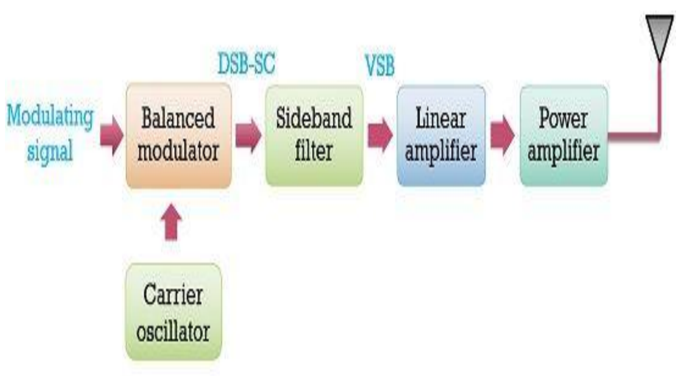
📻 **Amateur (Ham) Radio.**

🎙 **Speech Processing & Audio Transmission.**

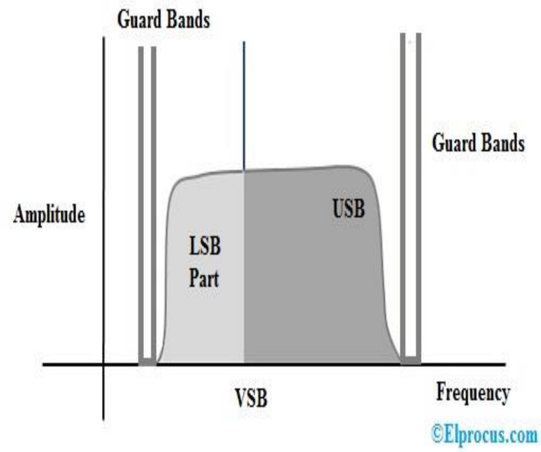
📞 **Telephony (e.g., Old Landline Systems).**

1.6 VSB Modulation

The term VSB stands for vestigial sideband; it is a kind of amplitude modulation technique, where a part of the signal named as a vestige and it is modulated with one sideband.
For the transmission, both the bands are not necessary, because it is a waste. But if a single band is transmitted, then the data will be lost. Therefore, this method has developed. A VSB signal is designed like the following image.



VSB signal generation and transmission



Advantages

1. The advantages of VSB modulation include the following.
2. The main benefit of this modulation is the decrease in BW. It is approximately efficient like SSB
3. High efficient
4. Designing of the filter is simple when high accuracy is not necessary.
5. Because of the transmitting allowance of a lower sideband part, the filter constraints will be relaxed.
6. It makes possible for the low-frequency components transmission as well as good phase characteristics without trouble.
7. For incomplete LSB suppression, practical filters are utilized.

Disadvantages

1. The disadvantages of VSB modulation include the following.
2. Bandwidth is higher when compared with the single-sided band (SSB).
3. Demodulation is difficult

VSB Modulation Applications

The applications of VSB modulation include the following.

1. VSB modulation is standard for the transmission of TV signals. Because the video signals require a large transmission BW using the techniques like DSB-FC otherwise DSF-SC.
2. This is a type amplitude modulation that is mainly used for the TV broadcast worldwide. In this broadcast, it is essential to broadcast the information of video and audio concurrently.

4. In the transmission of VSB, the higher sideband of video signal & picture carrier are broadcasted without any control. Where a vestige is a fraction of lower sideband and it is transmitted & the residual part is covered up
5. When the usage of BW is considered, then this is the most suitable and proficient technique.

1.7 Angle Modulation

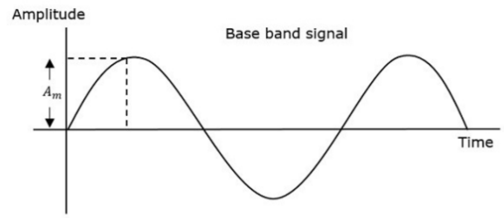
The other type of modulation in continuous-wave modulation is angle modulation. Angle Modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.
The standard equation of the angle modulated wave is

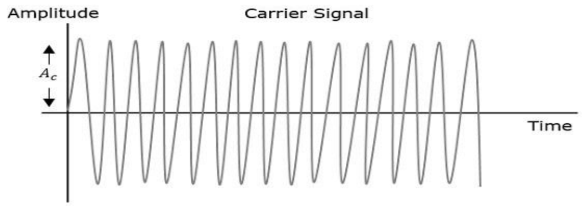
$$s(t) = A_c \cos\theta_i(t)$$

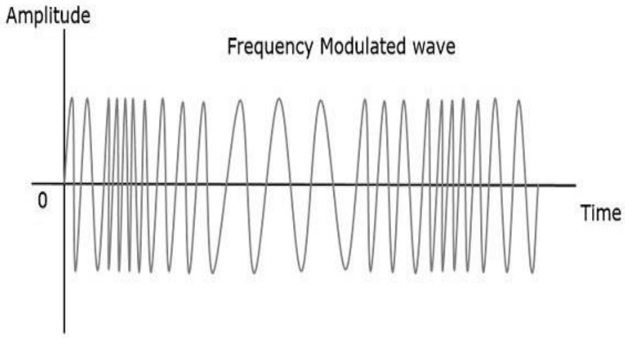
(Where, A_c is the amplitude of the modulated wave, $\theta_i(t)$ is the angle of the modulated wave)
Angle modulation is further divided into frequency modulation and phase modulation.

- **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the message signal.
- **Phase Modulation** is the process of varying the phase of the carrier signal linearly with the message signal.

Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.







Mathematical Representation

The equation for instantaneous phase ϕ_i in phase modulation is

$$\phi_i = k_p m(t)$$

Where,

- k_p is the phase sensitivity
- $m(t)$ is the message signal

The standard equation of angle modulated wave is

$$s(t) = A_c \cos(2\pi f_c t + \phi_i)$$

Substitute, ϕ_i value in the above equation.

$$s(t) = A_c \cos(2\pi f_c t + k_p m(t))$$

This is the **equation of PM wave**.

If the modulating signal, $m(t) = A_m \cos(2\pi f_m t)$, then the equation of PM wave will be

$$s(t) = A_c \cos(2\pi f_c t + \beta \cos(2\pi f_m t))$$

1.8 Difference between FM and PM

S.NO.	Frequency Modulation	Phase Modulation
1.	In Frequency Modulation amplitude and phase remain the same.	In Phase Modulation, the frequency and amplitude remain the same.
2.	Frequency Modulation is inversely proportional to modulating frequency.	Phase Modulation is proportional to modulating voltage.
3.	Associated with the change in frequency, there is some phase change.	Associated with the change in phase, there is some frequency change.
4.	It is possible to receive FM on a PM receiver.	It is possible to receive a PM on a FM receiver.
5.	Noise immunity is better to AM and PM.	Noise immunity is better than AM but worse than FM.
6.	Signal to noise ratio is better than in phase modulation.	Signal to noise ratio is poor than in frequency modulation.
7.	Frequency Modulation is widely used.	Phase Modulation is used in mobile system.

CHAPTER-II
Review of probability and random process

[Gaussian and white noise characteristics, Noise in amplitude modulation systems,Noise in Frequency modulation systems. Pre-emphasis and Deemphasis, Threshold effect in angle modulation.]

1.9 Gaussian noise

Gaussian noise means the probability density function of the noise has a Gaussian distribution, which basically defines the probability of the signal having a certain value. Whereas white noise simply means that the signal power is distributed equally over time.

1. Gaussian White Noise refers to a type of additive noise commonly found in electrical devices, characterized by a Gaussian distribution with all frequencies present. It is modeled with a zero mean and variance parameter that determines the strength of the noise in the input signal.
2. white noise is a random signal having equal intensity at different frequencies, giving it a constant power spectral density.[1] The term is used with this or similar meanings in many scientific and technical disciplines,
3. White refers to the idea that it has uniform power across the frequency band for the information system. It is an analogy to the color white which has uniform emissions at all frequencies in the spectrum. Gaussian because it has a normal distribution in the time domain with

an average time domain value of zero.

Noise in FM

Frequency Modulation is much more immune to noise than amplitude modulation and is significantly more immune than phase modulation. In order to establish the reason for this and to determine the extent of the improvement, it is necessary to examine the effect of noise on a carrier.

Effects of Noise on Carrier—Noise Triangle: A single Noise and Frequency Modulation will affect the output of a receiver only if it falls within its band pass. The carrier and noise voltages will mix, and if the difference is audible, it will naturally interfere with the reception of wanted signals. If such a single-noise voltage is considered vector ally, it is seen that the noise vector is superimposed on the carrier, rotating about it with a relative angular velocity ωn-ωc . This is shown in Figure 5-5. The maximum deviation in amplitude from the average value will be Vn, whereas the maximum phase deviation will be Φ = sin-1 (Vn/Vc).

Pre-emphasis and De-emphasis: The noise triangle showed that noise has a greater effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal-to-noise ratio. This boosting of the higher modulating frequencies, in accordance with a prearranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis. An example of a circuit used for each function is shown in Figure

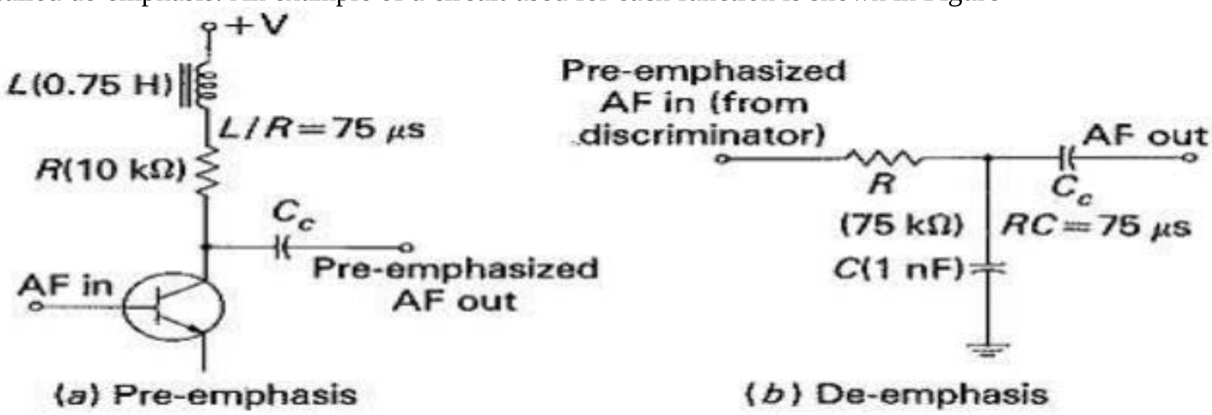


FIGURE 5-7 75-μs emphasis circuits.

Two modulating signals having the same initial amplitude, with one of them pre emphasized to twice this amplitude, whereas the other is unaffected (being at a much lower frequency). The receiver will naturally have to de-emphasize the first signal by a factor of 2, to ensure that both signals have the same amplitude in the output. of the receiver. Before demodulation, i.e., while susceptible to noise interference, the emphasized signal had twice the deviation it would have had without pre-emphasis and was thus more immune to noise. When this signal is de-emphasized, any noise sideband voltages are de-emphasized with it and therefore have correspondingly lower amplitude than they would have had without emphasis. Their effect on the output is reduced.

Threshold effect

In angle modulation (like FM), the threshold effect is a sudden and significant degradation in output signal-to-noise ratio (SNR) when the input signal-to-noise ratio (SNR) falls below a certain level, making signal detection difficult.

Threshold Effect in Angle Modulation

1. The noise analysis of angle demodulation schemes is based on the assumption that the SNR at the demodulator input is high. With this assumption, the signal and noise components at the demodulator output are additive.

2. Due to the nonlinear nature of the demodulation process, There is no reason that the additive signal and noise components at the input of the modulator result in additive signal and noise components at the output of the demodulator.
3. The high SNR assumption is not at all correct in general, The signal and noise processes at the output of the demodulator are completely mixed in a single process by a complicated nonlinear functional.

and Pulse code modulation (PCM), Differential pulse code modulation. Delta modulation, Noise considerations in PCM, Time Division multiplexing, Digital Multiplexers.

CHAPTER-3
Pulse modulation

[Sampling process. Pulse Amplitude and Pulse code modulation (PCM), Differential pulse code modulation. Delta modulation, Noise considerations in PCM, Time Division multi plexing, Digital Multiplexers.]

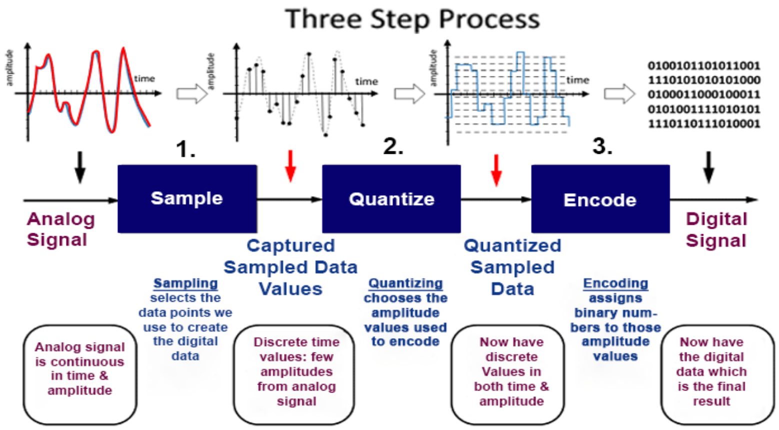
3.1 Pulse modulation:-

Sampling process:- The sampling process refers to the method of converting a continuous-time signal into a discrete-time representation by selecting samples at a specific frequency, known as the Nyquist sampling rate, to maintain the signal's information content.

Sampling Process in Digital Communication

The sampling process includes the following steps:

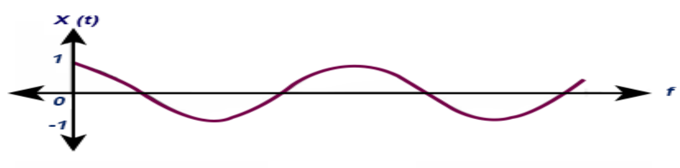
1. The continuous signal is taken as an input.
2. Sampling is performed to convert this signal into a digital representation.
3. In addition to sampling, quantization of a signal is performed.
4. After the above step, encoding of the signal is done.



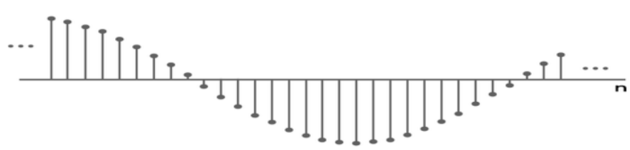
TYPE OF SIGNAL

1. Continuous Time Signals
2. Discrete Time Signals
3. Digital Signals

1. Continuous Time Signals:- CTS are those signals that are continuous in both time and amplitude. They are represented by functions that stay continuous over a range of time and values of amplitude.



2. Discrete Time Signals:- DTS are those that are continuous in amplitude but discrete in time meaning that they are signals with values at some specific instants of time. It is to be taken care of that these signals are discrete only in time whereas they can be either continuous or discrete in amplitude.



3. Digital Signals:- After sampling and quantization, the resulting signals are in digital format are hence called Digital Signals. These signals have both their time and amplitude in discrete format.

Sampling in Digital Communication:-

There are few important terminologies of Sampling in Digital Communication discussed below:
Sampling:-

- Sample
- Sampling Rate or Sampling Frequency
- Nyquist Rate

- Nyquist Interval
- Quantization

1. Sampling:- It is the process by which, we convert CTS (continuous time signal) into DTS (discrete time signal) by taking the signal values at some distinct points in time, meaning that this is used to take samples of analog signals at some points in time (regular or irregular).

2. Sample:- It can be defined as the numeric value of an analog signal at a specific time. It is just the signal's measured amplitude at a particular time and converting it to a digital representation.

3. Sampling Rate or Sampling Frequency:- It refers to the number of samples or data points taken per unit of time from an analog signal to convert it into a digital format. It is also known as sampling frequency. It is measured in Hertz (Hz).

The formula for sampling rate or sampling frequency is given by:

$$SamplingRate = \frac{1}{T_s} = f_s$$

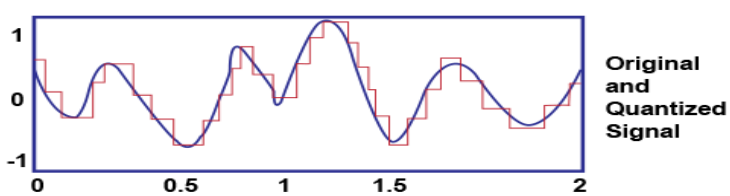
Ts=sampling/time, fs = sampling

4. Nyquist Rate:- The Nyquist interval, also known as the Nyquist period, is the time interval between consecutive samples in a digital signal or digital sampling system. It is the reciprocal of the Nyquist rate, which is the smallest sampling rate required to accurately capture an analog signal in digital form without information loss. Mathematically it can be represented as:

$$T = \frac{1}{NyquistRate}$$

T = Nyquist interval (sec)
Nyquist Rate is the sampling rate (Hz)

5.Quantization:- It is the process to represent a continuous-valued signal with a limited set of discrete values. In other words, it involves mapping a continuous signal's infinite range of potential values to a finite collection of discrete values.

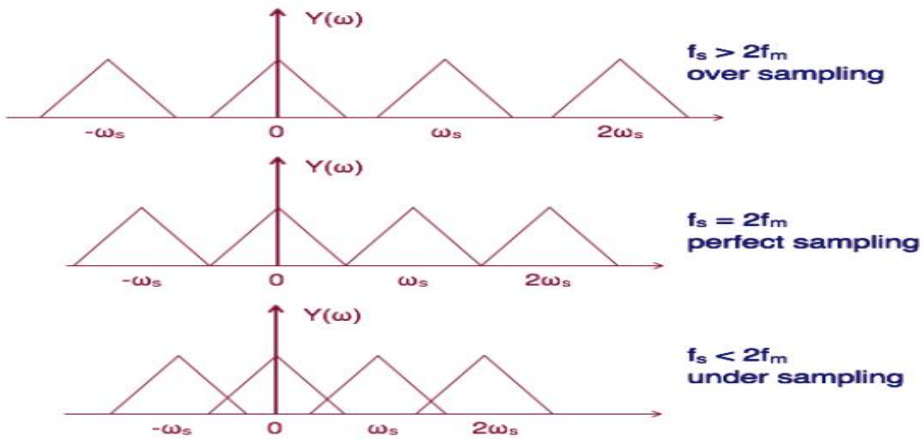


<u>Nyquist - Shannon Sampling Theorem</u>

The theorem states that for reconstructing a sampled signal accurately from the available samples, the sampling frequency should be at least twice as much as the highest frequency component of the signal.

expression: $2 \times f_{max} \leqslant f_s$

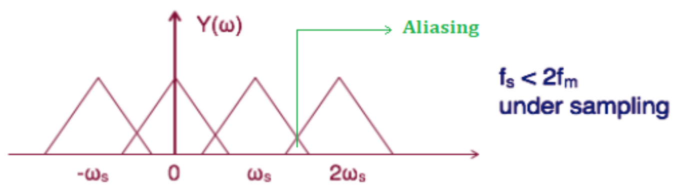
Where, fmax = maximum frequency component of the original signal
fs= sampling frequency



3.2 Oversampling & Undersampling

1. Undersampling

The spectra of X(ω) are overlapped in this scenario because the sampling rate is less than the Nyquist rate, making it impossible to extract the original signal from the sampled signal. Because the spectra overlap, some frequency components of the original signal will acquire a new frequency; this process is known as frequency aliasing.



2. Oversampling

Over-sampling is when more samples are taken that are necessary to capture the signal's frequency. It can be done to measure more accurately, enhancing SNR, providing more detailed information for further processing. It can be seen in the above figure-'Sampling cases'.

Methods of Sampling

1. Ideal Sampling 2. Natural Sampling 3. Flat-Top Sampling

Scope of Fourier Transform

It is well noticed that we seek the assistance of Fourier series and Fourier transforms in analyzing signals and proving theorems. This is because:

The Fourier Transform is used for a non-periodic signal which is helpful in analysis of the signal.

The Fourier transform is helps to observe signals in several domains and readily analyze them making it a strong mathematical tool.

Using this Fourier transform, any signal may be decomposed into the sum of sines and cosines.

Advantages of Sampling

It is an important step in converting analog signal to digital signal which allows efficient digital storage and signal processing

By taking samples at some specific interval time, it helps compressing the original signal, which in turn helps in efficient transmission

Digital signals can be easily processed using various algorithms

Disadvantages of Sampling

When sampling rate is not proper, it may lead to the problem of aliasing, resulting in distorted signals

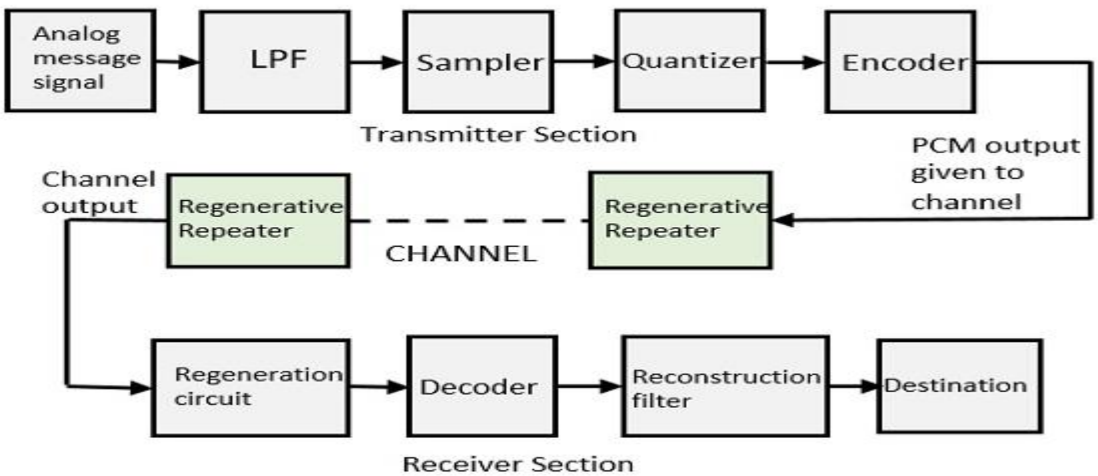
In the process of Analog to Digital conversion, the next step after sampling is quantization which may result in: loss of information

While conversion (Analog to Digital) sampling may introduce errors due to factors such as quantization noise, temperature variations, etc.

3.3 Pulse code modulation (PCM)

The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train. Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.



Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component W of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections (LPF, Sampler, and Quantizer) will act as an analog to digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

*Reconstruction Filte*r

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

3.4 Delta Modulation

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of a smaller value Δ, such a modulation is termed as delta modulation.

Features of Delta Modulation

Following are some of the features of delta modulation.

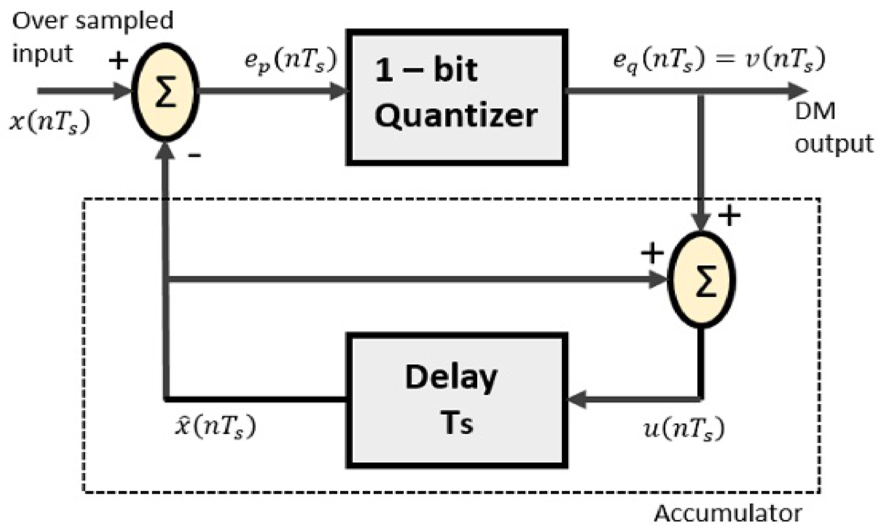
1. An over-sampled input is taken to make full use of the signal correlation.
2. The quantization design is simple.
3. The input sequence is much higher than the Nyquist rate.
4. The quality is moderate.
5. The design of the modulator and the demodulator is simple.
6. The stair-case approximation of output waveform.
7. The step-size is very small, i.e., Δ (delta).
8. The bit rate can be decided by the user.

This involves simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

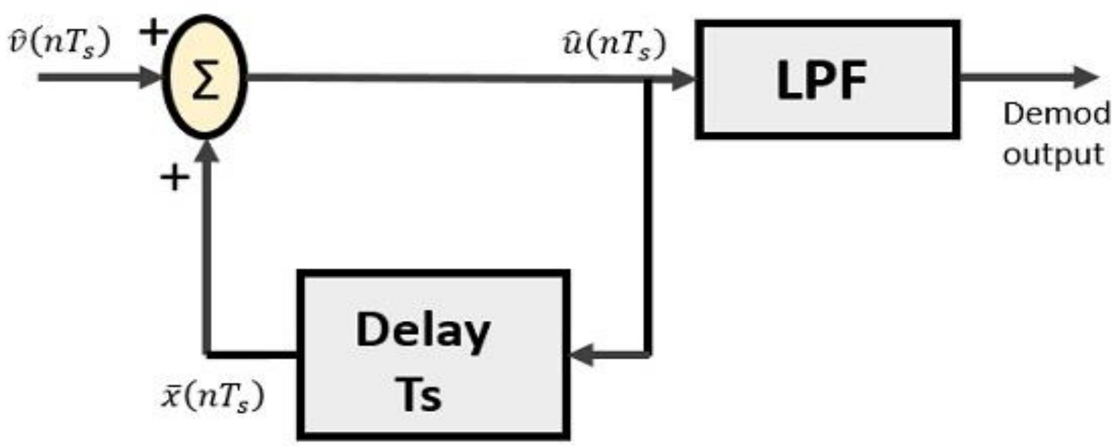
The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.



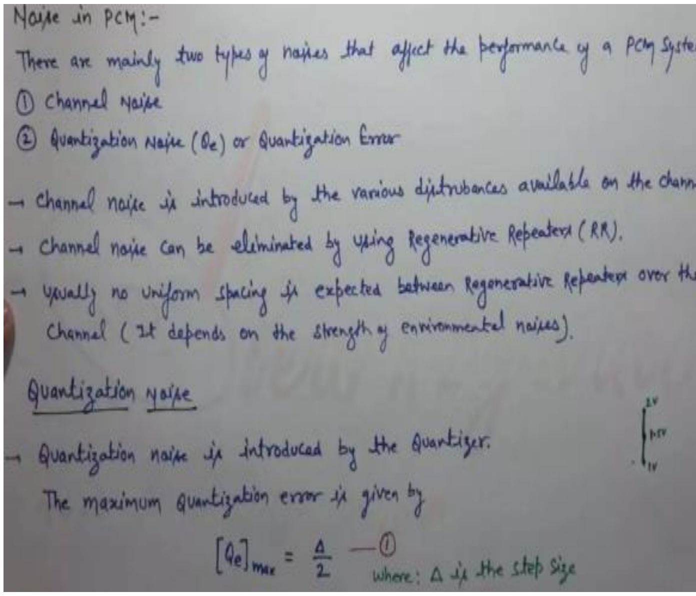
Delta Demodulator

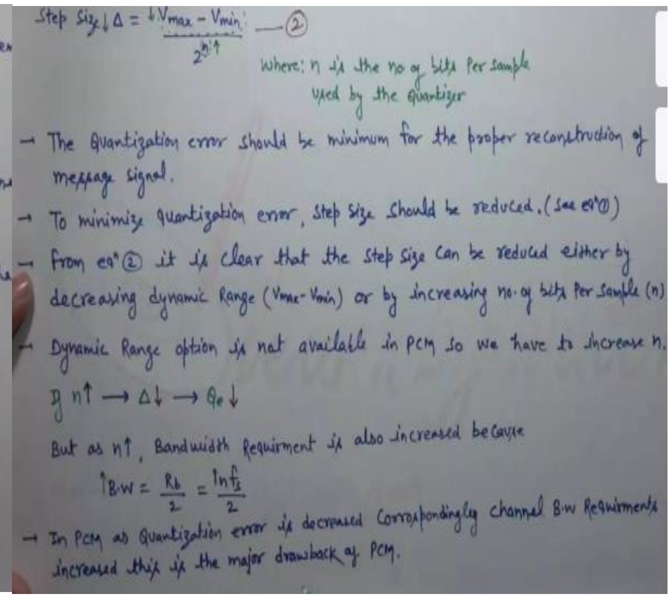
The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

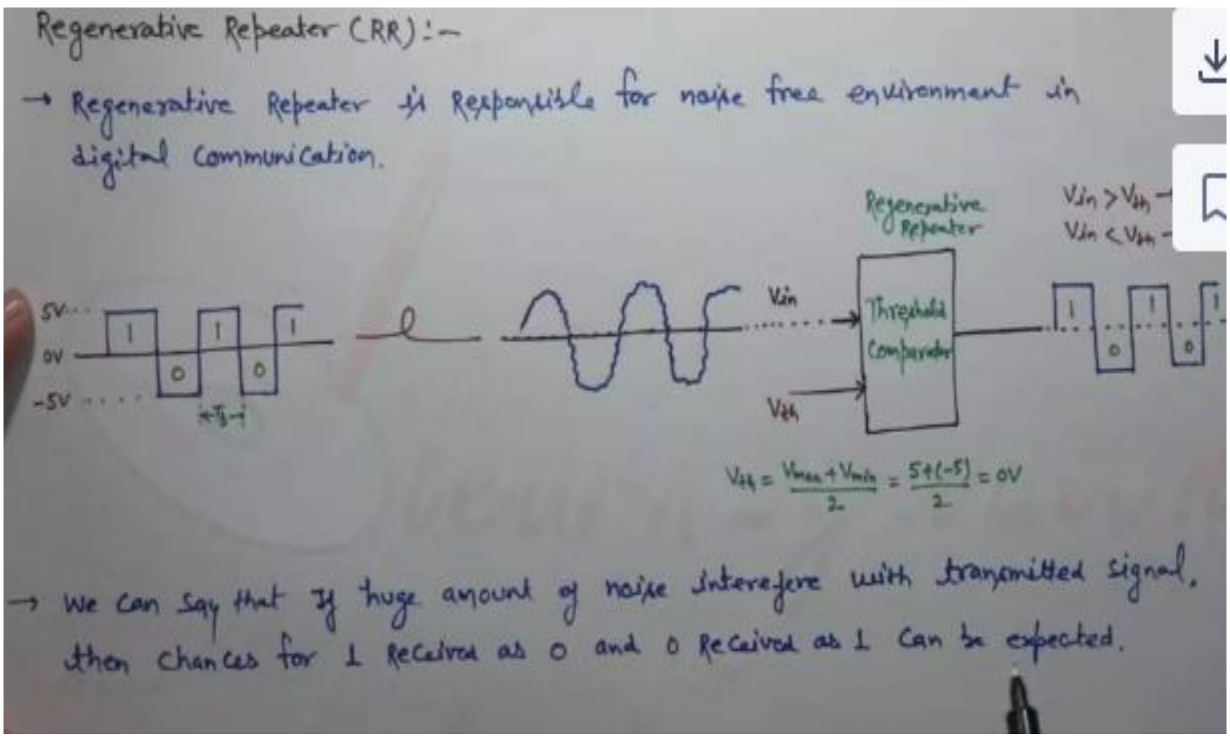
Following is the diagram for delta demodulator.



3.5 Noise in PCM







3.6 Time Division Multiplexing (TDM)

The Time Division Multiplexing (TDM) is a digital procedure. Here, each sender is given the entire possession of the whole bandwidth of the channel for a fixed duration of time. After this, the control is moved to the next sender, and the process continues on a round-robin basis.

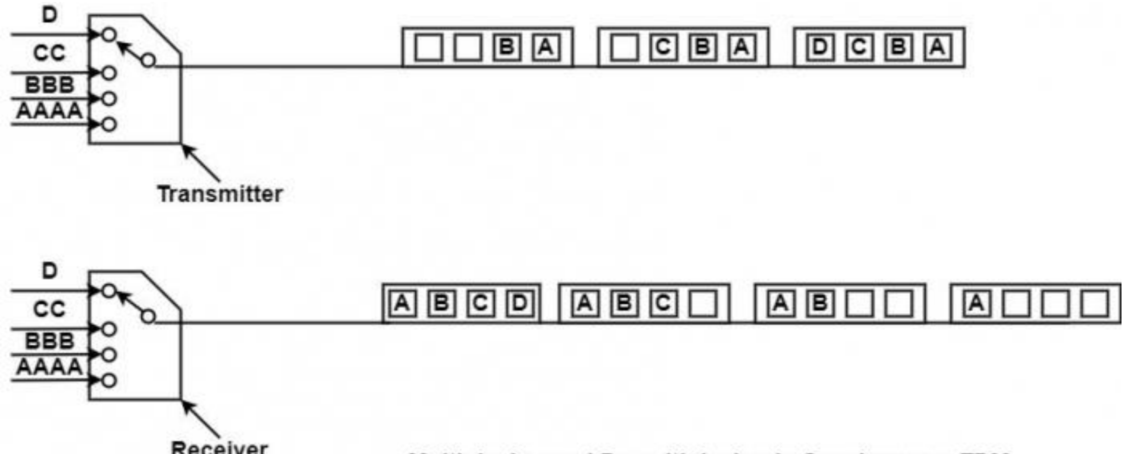
The following variants on TDM are as follows

1. **ATDM** (Asynchronous Time Division Multiplexing): Multiplexing in which the data is transmitted asynchronously.

2. **STDM** (Statistical Time Division Multiplexing): A multiplexing method that polls nodes and immediately skips any nodes with nothing to send.
3. **STM** (Synchronous Transfer Node): Designer for use in BISDN (broadband ISDN) and also supported in the SONET (Asynchronous Optical Network) architecture.

Types of Time Division Multiplexing:-

1. Synchronous TDM:-Synchronous TDM is known as synchronous and is essential because, each time slot is pre-assigned to a constant source. The time slots are sent irrespective of whether the sources have a few records to share or not.TDM devices can manage the source of various data rates. This is completed by authorising fewer slots per cycle to the passive input devices than the rapid device.



Multiplexing and Demultiplexing in Synchronous TDM

2. Statistical TDM:-

In the Synchronous Time Division Multiplexing (STDM), the multiplexer assigns an equal time slot to every device at all times, whether or not a device has anything to send. Time slot A, for instance, is authorised to device A alone and cannot be used by any other device.

Disadvantage:

1. In synchronous time-division multiplexing, an equal time slot is given to each sender to load its data on the channel.
2. Different senders load different volumes of data, and frames are usually empty.

3.7 Digital Multiplexers:-Multiplexer, also referred to as MUX, is a combination logic circuit that is designed to accept multiple input signals and transfer only one of them through the output line. In simple words, a multiplexer is a digital logic device that selects one-out-of-N (N = 2n) input data sources and transmits the selected data to a single output line.
The multiplexer is also called data selector as it selects one from several. The block diagram of a typical **2n:1 multiplexer** is shown

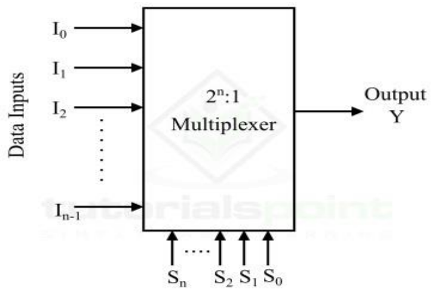


Figure 1 - Digital Multiplexer

4×1 Multiplexer

4×1 Multiplexer has four data inputs I3, I2, I1 & I0, two selection lines s1 & s0 and one output Y.One of these 4 inputs will be connected to the output based on the combination of inputs present at these two selection lines. Truth table of 4×1 Multiplexer is shown below.

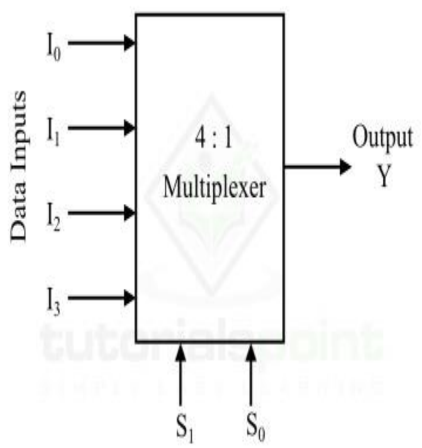
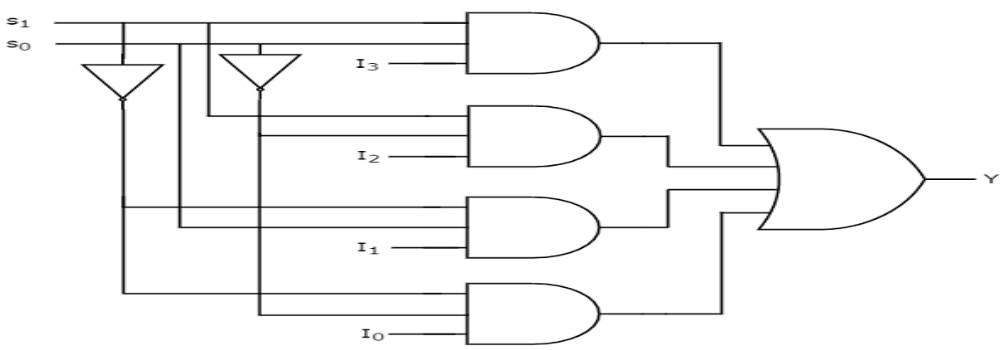


Figure 3 - 4:1 Multiplexer

Selection Lines		Output
S1	S0	Y
0	0	I0
0	1	I1
1	0	I2
1	1	I3

using Inverters, AND gates & OR gate



CHAPTER-4
Elements of Detection Theory

[Optimum detection of signals in noise, Coherent communication with waveforms- Probability of Error evaluations. Base band Pulse Transmission- Inter symbol Interference and Nyquist criterion. Pass band Digital Modulation schemes- Phase Shift Keying, Frequency Shift Keying, Quadrature Amplitude Modulation, Continuous Phase Modulation and Minimum Shift Keying.]

4.1 Optimum detection of signals in noise:-

(a) Receiver Noise Noise is the unwanted electromagnetic energy that interferes with the ability of the receiver to detect the wanted signal. It may enter the receiver through the antenna along with the desired signal or it may be generated within the receiver.

Noise power PN is expressed in terms of the temperature To of a matched resistor at the input of the receiver P.N = kToβ Watt

[where: k – Boltzmann's Constant (1.38×10-23 J/K), To – System Temperature (usually 290K), β – Receiver Noise Bandwidth (Hz)]

(b) Noise Probability Density Functions:-

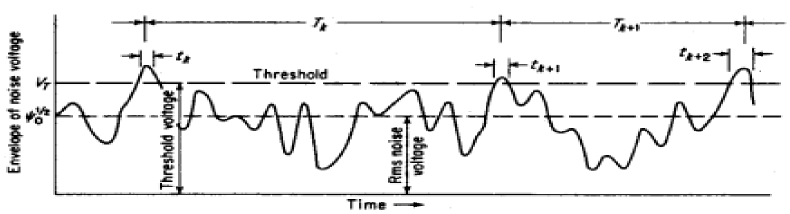
The noise entering the IF filter is assumed to be Gaussian (as it is thermal in nature) with a probability density function (PDF) given by

$$p(v) = \frac{1}{\sqrt{2\pi\psi_o}} \exp\frac{-v^2}{2\psi_o},$$

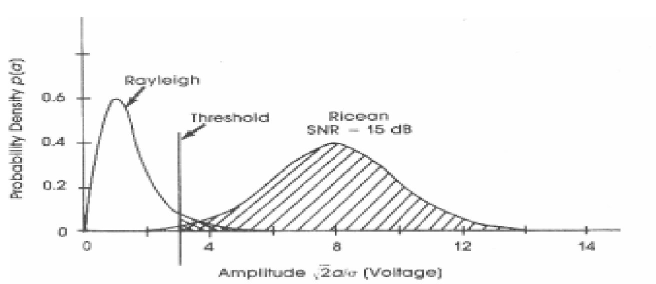
where $p(v)dv$ - probability of finding the noise voltage v between v and v+dv,

ψ_o - variance of the noise voltage.

(c) Probability of False Alarm:-



(d) Probability of Detection:-



Probability density functions of noise and signal-plus-noise with threshold shown for $P_{fa} = 10^{-2}$

Intersymbol Interference (ISI):

ISI is a form of distortion where one symbol interferes with the reception of subsequent symbols, leading to signal degradation.

It's caused by the channel's impulse response spreading the transmitted signal in time, making it difficult to distinguish between symbols.

This interference can lead to errors in the received data, especially at high data rates.

Nyquist criterion

The Nyquist criterion provides conditions for pulse shaping to eliminate ISI, ensuring that samples at the sampling instants are either 1 or 0. o achieve zero intersymbol interference (ISI), samples must have only one non-zero value at each sampling instant