CAPITAL UNIVERSITY - KODERMA

COMMUNICATION SYSTEM AND CIRCUITS ASSIGNMENT

Name : Arshad Nazir

Electrical and Electronics Engineering

Signature:

Date :

PART – 1

1. What is pre envelope and complex envelope? BTL 1 Remembering

   An analytic signal is a complex signal created by taking a signal and then adding in quadrature its Hilbert Transform. It is also called the pre-envelope of the real signal. A new quantity based on the analytic signal, called the Complex Envelope. Complex Envelope is defined as
   $g + (t) = \tilde{g}(t) e\ j2\pi f t$
   $\tilde{g}(t)$ is the Complex Envelope.

2. Give the advantages of VSB-AM. BTL 1 Remembering ?

   Ans:

   AM vestigial sideband is a form of amplitude modulation in which the carrier and one complete sideband are transmitted, but only part of the second sideband is transmitted.

3. State heterodyning principle. BTL 1 Remembering

   Ans:

   Heterodyning is used **to shift one frequency range into another, new frequency range**, and is also involved in the processes of modulation and demodulation. The two input frequencies are combined in a nonlinear signal-processing device such as a vacuum tube, transistor, or diode, usually called a mixer.

4. Mention the advantages of modulating low frequency signal into high frequency signal. BTL 1 Remembering

   Ans:

   - Ease of transmission
   - Multiplexing
   - Reduced noise
   - Narrow bandwidth
   - Frequency assignment
   - Reduce the equipments limitations

5. List the types of AM modulators. BTL 1 Remembering

   Ans:

   - Amplitude modulation.
   - Angle Modulation
     - Frequency modulation
     - Phase modulation.

6. Define Coherent Detection. BTL 1 Remembering

Ans:

If the local oscillator signal used in receiver is exactly coherent or synchronized, in both frequency and phase, with the carrier used in the transmitter then it is called as coherent detection or synchronous demodulation.

7. Why do you need modulation in communication systems? BTL 3 Applying

   Ans:

   The signals within 20 Hz to 20 kHz frequency range can travel only a few distances. To send the message signal, the length of the antenna should be a quarter wavelength of the used frequency. Thus, modulation is required **to increase the frequency of the message signal and to enhance its strength to reach the receiver**.

8. Identify the differences between single sideband and vestigial sideband systems. BTL 2 Understanding?

   Ans:

   Two modulation schemes that are derivatives of DSB modulation are single-sideband (SSB) and **vestigial-sideband (VSB)** modulation. SSB modulation is produced by filtering out all of one sideband of a DSB signal, while VSB is produced by leaving a vestige of one sideband and all of the other sideband of a DSB signal.

9. Write about diagonal clipping and negative peak clipping. BTL 2 Understanding

   Ans:

   Diagonal Clipping :

   Distortion that occurs in an AM demodulator (usually associated with diode detection), where the capacitor discharge time constant is set too long for the detector to accurately follow fast changes in the AM signal envelope.

   Negative Peak Clipping :

   This distortion occurs **due to a fact that the modulation index on the output side of the detector is higher than that on its input side** . Hence, at higher depth of modulation of the transmitted signal, the overmodulation may takes place at the output of the detector.

10. Suggest a modulation scheme for broadcast video transmission. BTL 6 Creating

    Ans:

    During the tv transmission , Frequency modulation (FM) is used for audio transmission and amplitude modulation (AM) is used for picture transmission.

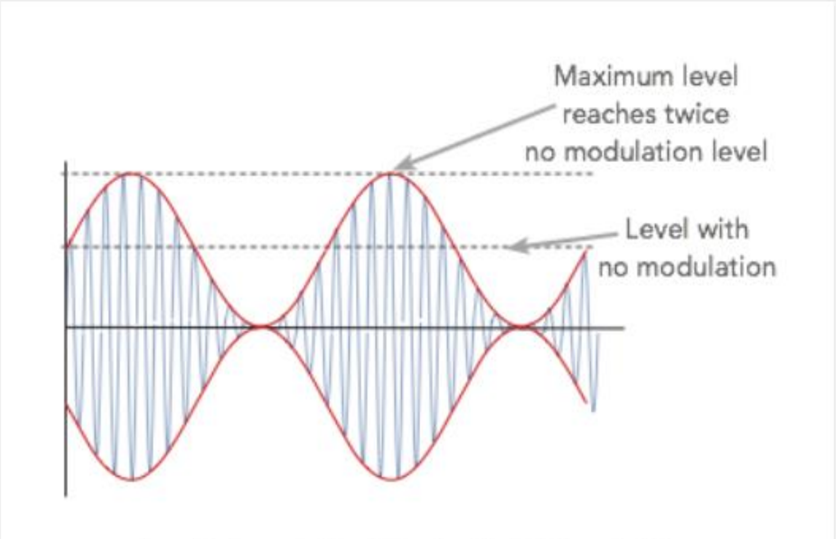
Answer: In Television transmission, FM is used for audio transmission and **AM** is used for video transmission.

11. Apply the concepts of sensitivity and selectivity in AM receiver. BTL 3 Applying

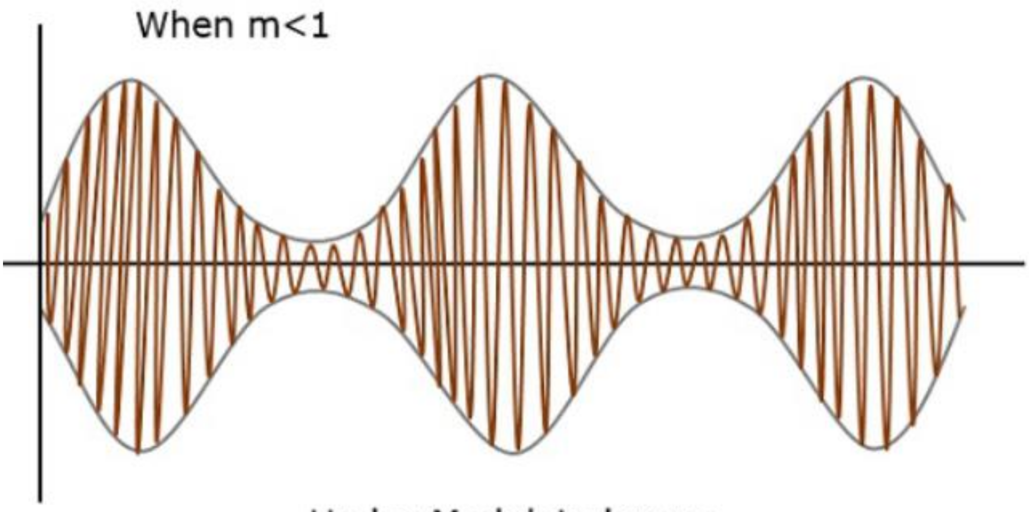
Ans:

Selectivity of radio receiver is **the ability of a receiver to accept the wanted signal and to reject the unwanted**. **signals**. Sensitivity of a radio receiver is its ability is its ability to amplify the desired weak

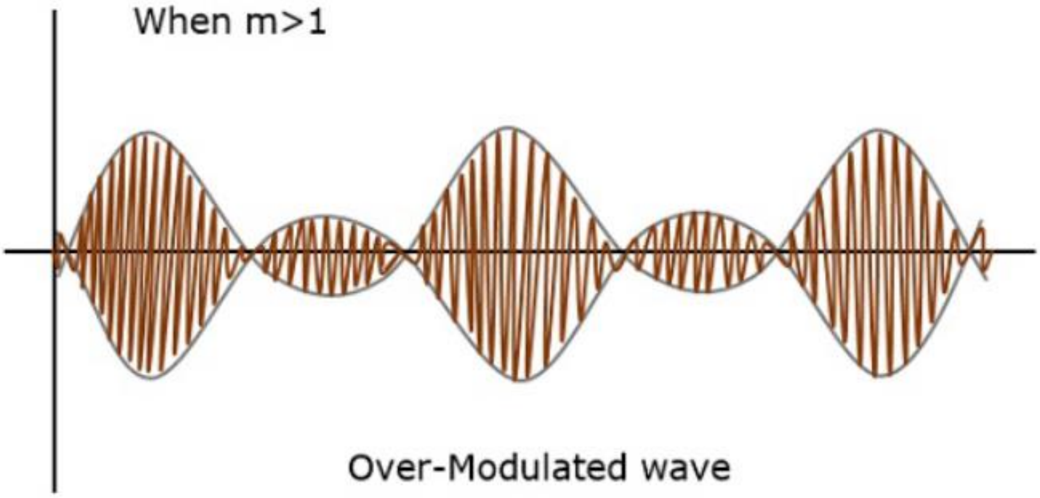
12. Draw the AM modulated wave for over, under & 100% modulation. BTL 2 Understanding



Amplitude modulated signal with 100% modulation



Under-Modulated wave



13. If incoming frequency is f1 and translated carrier frequency is f2, apply and find the local oscillator frequency. BTL 3 Applying

14. Compare AM with DSB-SC and SSB-SC. BTL 4 Analyzing?

    Ans:

    In the DSB-SC modulation, unlike in AM, **the wave carrier is not transmitted**; thus, much of the power is distributed between the side bands, which implies an increase of the cover in DSB-SC, compared to AM, for the same power use. DSB-SC transmission is a special case of double-sideband reduced carrier transmission.

    An AM signal consists of two redundant sideband signals that each contain the operator's voice, along with a so-called carrier signal between them. An SSB signal uses **only** one of the sidebands. An AM radio receiver mixes the carrier with the sidebands, and out pops the original audio signal.

PART IA

1. Describe the concepts of AM modulation and derive the equation of an AM wave. Draw the phasor diagram, spectrum and modulated AM wave for various degrees of modulation index.

   Ans:

   Amplitude modulation is considered to be a process in **which the wave signals are transmitted by modulating the amplitude of the signal**. The amplitude modulation is often called AM.

   Amplitude Modulation Equation Derivation

The mathematical representation of amplitude-modulated waves in the time domain is as follows.

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

$m(t) = A_c \cos(2\pi f_c t)$ (carrier signal)

$s(t) = [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$ (equation of amplitude modulated wave)

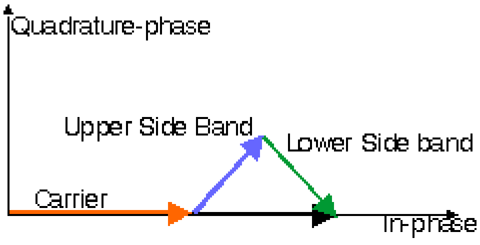
Where,

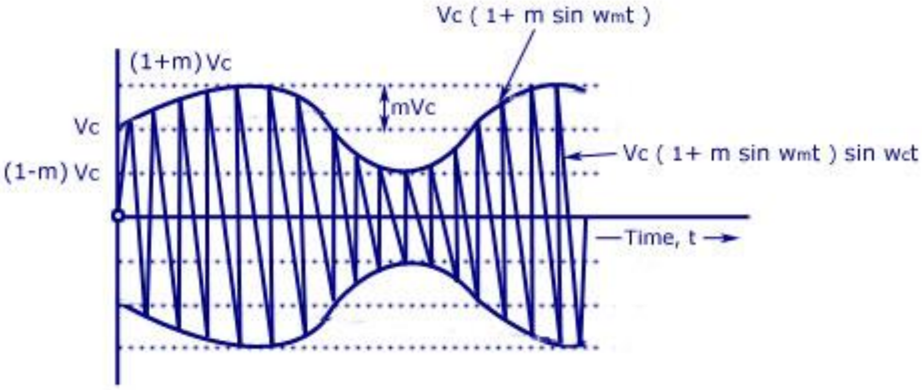
A_m: Amplitude of modulating signal

A_c: Amplitude of carrier signal

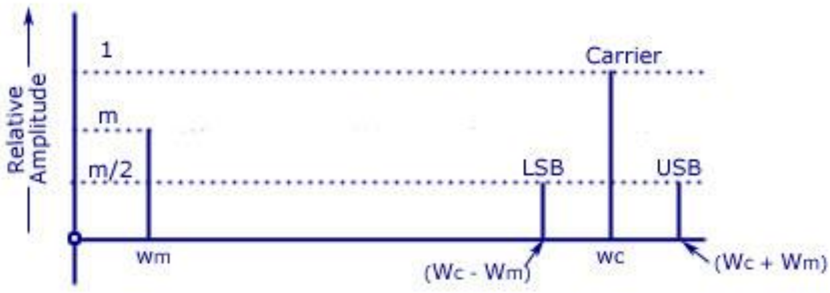
f_m: Frequency of modulating signal

f_c: Frequency of carrier signal





Amplitude Modulated Sinewave with m < 1



Frequency Spectrum of a Sinusoidally Modulated Wave

www.CircuitsToday.com

2. (i) Find the phasor representation, current relation and efficiency of AM.

Ans:

Phasor representation:

A phasor is simply a shorthand way of representing a signal that is sinusoidal in time. Though it may seem difficult at first, it makes the mathematics involved in tha analysis of systems with sinusoidal inputs much simpler. To start, we take a sinusoidal signal in time defined by magnitude, phase and frequency (A, θ and ω)

$f(t)=A\cos(\omega t+\theta)$

and represent it in phasor form as a complex number with a magnitude and phase (A, and θ). Note that frequency (ω) is not included, but is implicit in the concept of a phasor.

$F=Aej\theta=A(\cos(\theta)+j\sin(\theta))$

phasor in terms of its magnitude and phase.

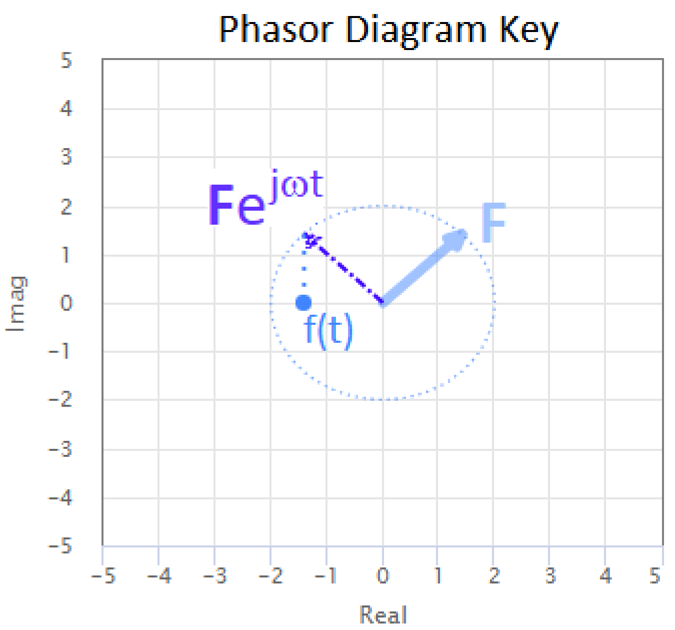
$F=A\angle\theta$

If we multiply the phasor, **F**, by ejωt, we are simply rotating **F** by an angle ωt (*i.e.*, the resulting vector has an angle of (ωt+θ).

$Fej\omega t=Aej\theta ej\omega t=Aej(\omega t+\theta)=A\angle(\omega t+\theta)$

We can recover the time domain function, f(t), by taking the real part of this rotating vector.

$f(t)=Re\{Fe^{j\omega t}\}=Re\{Ae^{j(\omega t+\theta)}\}=Re\{A(\cos(\omega t+\theta)+j\sin(\omega t+\theta))\}=A\cos(\omega t+\theta)$



Current relationship :

It has been shown that the carrier component of the modulated wave has the same amplitude as the unmodulated carrier. That is, the amplitude of the carrier is unchanged; energy is either added or subtracted. The modulated wave contains extra energy in the two sideband components.

Efficiency: The power efficiency of amplitude modulation is **very low**. ... This is called 100% modulation as it is the maximum amount that can be applied. If the level of modulation is increased beyond the 100% level, then distortion will arise because the envelope cannot fall below zero.

(ii) In a superheterodyne receiver the input AM signal has a centre frequency of 1425 KHz and bandwidth 5 KHz. The input is down converted to 455 KHz. Identify the image frequency

Ans:

3. (i) Write the working of low level and high level AM Transmitters with the help of a neat block diagram.

Ans:

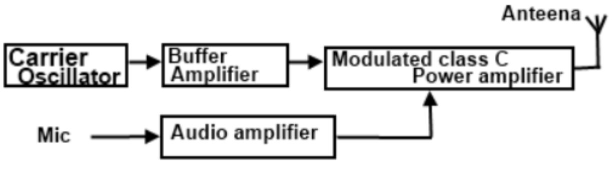


Figure (b): Block Diagram of Low Level AM Transmitter

The low-level AM transmitter shown in the figure (b) is similar to a high-level transmitter, except that the powers of the carrier and audio signals are not amplified. These two signals are directly applied to the modulated class C power amplifier.

Modulation takes place at the stage, and the power of the modulated signal is amplified to the required transmitting power level. The transmitting antenna then transmits the signal.

(ii) Obtain the types of AM modulators based on their placement in a transmitter circuit.

Ans:

Double-sideband suppressed-carrier transmission (DSB-SC):

Double-sideband suppressed-carrier transmission (DSB-SC) is transmission in which **frequencies produced** by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed.

single-sideband modulation (SSB):

In radio communications, single-sideband modulation (SSB) or single-sideband suppressed-carrier modulation (SSB-SC) is **a type of modulation used to transmit information, such as an audio signal**, by radio waves. A refinement of amplitude modulation, it uses transmitter power and bandwidth more efficiently.

Vestigial Sideband Modulation (VSB):

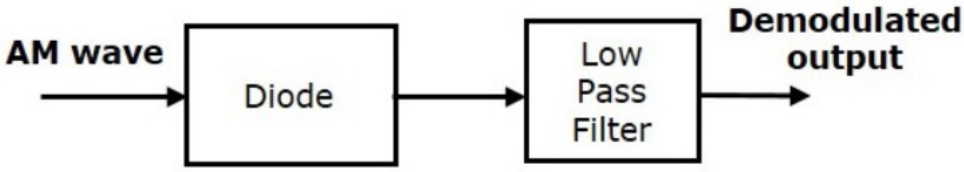
Vestigial Sideband Modulation (VSB) refers to **the process where the "vestige" part of a signal is modulated alongside one sideband**. A vestigial sideband is a form of amplitude modulation (AM) that encodes data in a signal by altering the amplitude of the carrier frequency.

4. (i) Demonstrate the concepts of envelope detection for demodulation of AM and explain its operation.

   Ans:

Envelope Detector

Envelope detector is used to detect (demodulate) high level AM wave. Following is the block diagram of the envelope detector.



This envelope detector consists of a diode and low pass filter. Here, the diode is the main detecting element. Hence, the envelope detector is also called as the **diode detector**. The low pass filter contains a parallel combination of the resistor and the capacitor.

The AM wave $s(t)$ is applied as an input to this detector.

We know the standard form of AM wave is

$$s(t) = A_c\left[1 + k_a m(t)\right]\cos(2\pi f_c t)$$

In the positive half cycle of AM wave, the diode conducts and the capacitor charges to the peak value of AM wave. When the value of AM wave is less than this value, the diode will be reverse biased. Thus, the capacitor will discharge through resistor **R** till the next positive half cycle of AM wave. When the value of AM wave is greater than the capacitor voltage, the diode conducts and the process will be repeated.

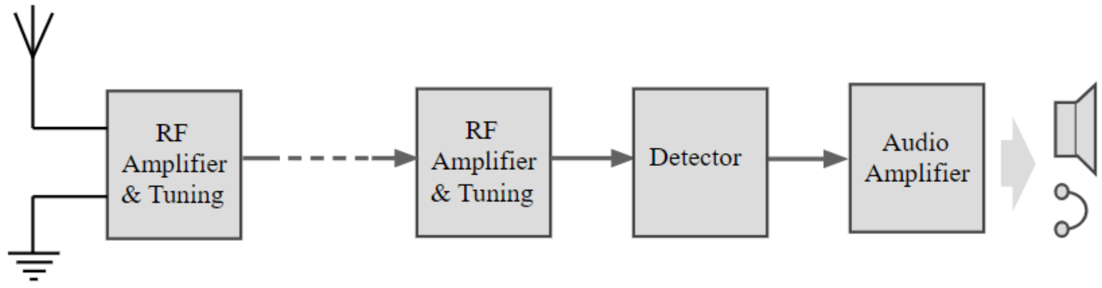
We should select the component values in such a way that the capacitor charges very quickly and discharges very slowly. As a result, we will get the capacitor voltage waveform same as that of the envelope of AM wave, which is almost similar to the modulating signal.

(ii)Illustrate non coherent tuned radio frequency receiver.

Ans:

Typically a TRF receiver would consist of three main sections:

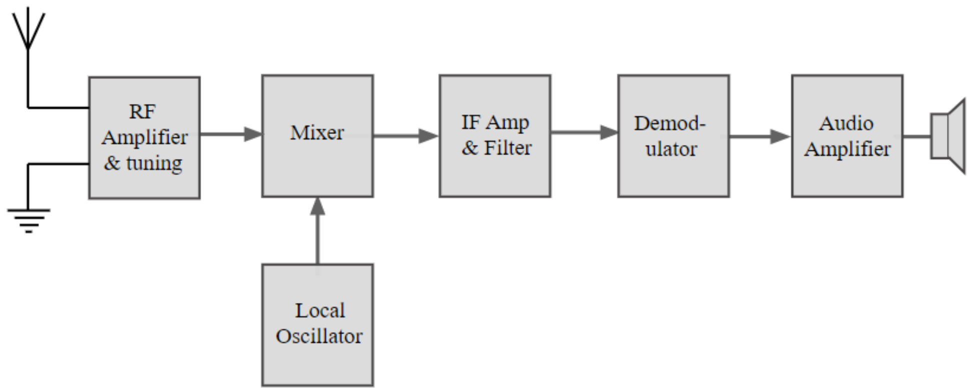
- ***Tuned radio frequency stages:*** This consisted of one of more amplifying and tuning stages. Early sets often had several stages, each proving some gain and selectivity.
- ***Signal detector:*** The detector enabled the audio from the amplitude modulation signal to be extracted. It used a form of detection called envelope detection and used a diode to rectify the signal.
- ***Audio amplifier:*** Audio stages to provide audio amplification were normally, but not always included.



Tuned radio frequency receiver, TRF, block diagram

5. Give main idea about super heterodyne receiver with neat block diagram and explain the various parameters

   Ans:



Block diagram of a basic superheterodyne receiver

The superhet radio receiver is used in many forms of radio broadcast reception, two way radio communications and the like.

The main types of circuit bock used in the RF circuit design for superhet receivers is given below.

RF tuning & amplification: This RF stage within the overall block diagram for the receiver provides initial tuning to remove the image signal. It also provides some amplification. There are many different approaches used within the RF circuit design for this block dependent its application.

The electronic circuit design presents some challenges. Low cost broadcast radios may have an amplifying mixer circuit that gives some RF amplification. HF radios may not want too much RF gain because some of the very strong signals received could overload later stages. The RF design may incorporate some amplification as well as RF attenuation to overcome this issue. Radios for VHF and above will tend to use more gain to have a noise figure that is sufficiently low to receive the signal. Noise is a particular problem for VHF / UHF radio communications systems.

If noise performance for the receiver is important, then this stage will be designed for optimum noise performance. This RF amplifier circuit block will also increase the signal level so that the noise introduced by later stages is at a lower level in comparison to the wanted signal.

All radios will need a sufficiently high level of image rejection, and this is provided by the RF tuning. High IF frequencies enable the RF tuning to be more effective as the difference between the wanted signal and image is increased.

Local oscillator: Like other areas of the RF circuit design, the local oscillator circuit block within the superhet radio can take a variety of forms.

Early receivers used free running local oscillators. There was a considerable degree of RF circuit design expertise used with these oscillators in high performance superhet radios to ensure the lowest possible drift. High Q coils, low drift circuit configurations, heat management (because heat causes drift), etc . .

Today most receivers use one or more of a variety of forms frequency synthesizer. The most common approach in the RF circuit design is to use a phase locked loop approach. Single and multi-loop synthesizers are used dependent upon the requirements, performance, cost and the like. Direct digital synthesizers are also being used increasingly.

Whatever form of synthesizer is used in the RF design, they provide much greater levels of stability and enable frequencies to be programmed digitally in a variety of ways, normally using some form of microcontroller or microprocessor system. They are more complicated than the older variable frequency oscillators, requiring many more electronic components, but providing a very much higher level of performance.
Mixer: The mixer can be one of the key elements within the overall RF design of the receiver. Ensuring that the mixer performance matches that of the rest of the radio is particularly important.

Both the local oscillator and incoming signal enter this block within the superheterodyne receiver. The wanted signal is converted to the intermediate frequency.

The actual implementation requires that the minimum number of spurious signals are generated. In some very low cost broadcast receivers, self oscillating mixers that provide RF amplification from a single transistor and a few other electronic components may be used, these do not offer high performance. For a high performance radio used for two way radio communications an the like, much better performance is required. To achieve this mixer circuits such as balanced mixers, double balanced mixers, and the like may be seen within the oerall electronic circuit design.

IF amplifier & filter: This superheterodyne receiver block provides the majority of gain and selectivity. Often comparatively little gain will be provided in the previous blocks of the RF circuit design of the radio. The IF stages are where the main gain is provided. Being fixed in frequency, it is much easier to achieve high levels of gain and overall performance.

Originally the IF stage might have included a number of different transistors, FETs or thermionic valves / vacuum tubes and other electronic components, but nowadays it is possible to obtain integrated circuits that contain a complete IF strip.

This circuit block of the radio also provides the adjacent channel selectivity. High performance filters like crystal filters may be used, although LC or ceramic filters may be used within domestic radios. The type of filter will depend upon the radio RF design and its application.

Also within a multi-conversion superhet, the IF may be on a number of different frequencies, typically the earlier stages are at higher frequencies to provide higher levels of image rejection, and later ones at lower frequencies to provide gain and adjacent channel selectivity.

Demodulator: The superheterodyne receiver block diagram only shows one demodulator, but in reality many radio RF designs may have one or more demodulators dependent upon the type of signals being receiver. Those radios used for professional radio communications applications and monitoring may need to be able to demodulate a variety of modulation schemes and waveforms and this may require a number of different demodulators that can be switched in as appropriate.

Even many broadcast radios will have AM and FM, but professional radios used for monitoring and two way radio communications may require a larger variety in some instances. Having a variety of demodulators will enable many different signal modes to be received and increase the capability of the radio.

Automatic Gain Control, AGC: An automatic gain control is incorporated into most superhet radio block diagrams. The function of this circuit block is to reduce the gain for strong signals so that the audio level is maintained for amplitude sensitive forms of modulation, and also to prevent overloading.

Although the basic concept is the same through all radio RF circuit designs, there are some variations in the implementation and the electronic circuit design required. Some of the key variations are the time constant of the AGC system. For AM and the like a relatively slow time constant is acceptable. For SSB, a shorter time constant is needed so that the envelope of the SSB signal is followed.

There are also variations in the way the AGC voltage is derived, and where it is applied. Often it is applied to the IF circuit blocks first and then to the RF circuit block. In this way the best signal to noise ratio is preserved. Generally the AGC is relatively easy to implement, having relatively few electronic components.

Audio amplifier: Once demodulated, the recovered audio is applied to an audio amplifier block to be amplified to the required level for loudspeakers or headphones. Alternatively the recovered modulation may be used for other applications whereupon it is processed in the required way by a specific circuit block.

In many ways, this circuit block within the superheterodyne radio is the most straightforward. For many applications, the audio amplifier will involved some straightforward electronic circuit design, especially if the audio is applied to simple headphones or a loudspeaker. For two way radio communication applications, the audio bandwidth may need to be limited to the "telecommunications" bandwidth of about 300 Hz to 3.3 kHz. Audio filters could be employed as well.

For applications requiring a higher quality output, more thought may need to be applied during the electronic circuit design to achieving high fidelity performance.

6. Identify the need for carrier suppression in AM system? Draw and explain the functioning of such system?

   Ans:

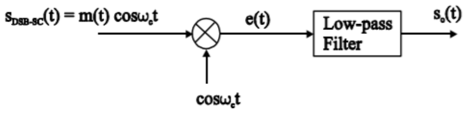
   Need for carrier suppression in AM System :

   Reduced-carrier transmission is an amplitude modulation (AM) transmission in which the carrier signal level is reduced to reduce wasted electrical power. Suppressed carriers are often used for single sideband (SSB) transmissions, such as for amateur radio on shortwave.

- DSB-SC is useful to ensure the discrete carrier signal is suppressed: $$s(t) = A_c m(t) \cos \omega_c t$$
- The voltage or current spectrum of DSB-SC will be

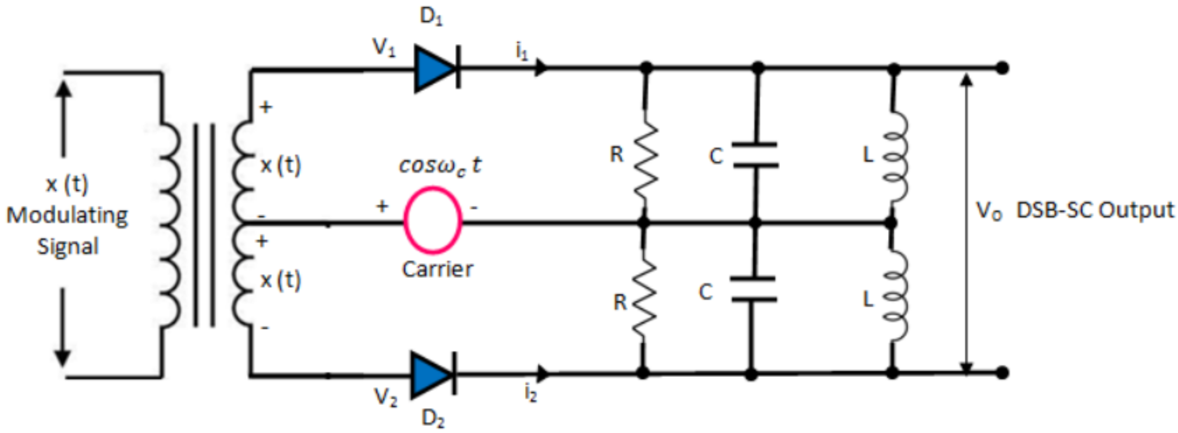
$$S(f) = \frac{A_c}{2}[M(f - f_c) + M(f + f_c)]$$

- Therefore no waste of power for discrete carrier component !
- What is the modulation efficiency? → 100 Percent!
  - Effic = <m(t) ^{2}>/ <m(t) ^{2}>
- Generating DSB-SC



7. Construct the balanced modulator circuit for the generation of DSBSC-AM and explain its operation

Ans:



The modulating signal x(t) is applied equally with 180^o phase reversal at the inputs of both the diodes through the input center tapped transformer .
The carrier is applied to the center tap of the secondary .
Hence, input voltage to D_1 is given by :

$$v_1 = cos\omega_c t + x(t) \quad \text{........(1)}$$

And the input voltage to D_2 is given by :

$$v_2 = cos\omega_c t - x(t) \quad \text{.........(2)}$$

The parallel RLC circuits on the output side form the band pass filters .

Analysis

The diode current i_1 and i_2 are given by :

$$i_1 = av_1 + bv_1{}^2$$

$$i_1 = a[x(t) + cos\omega_c t] + b[x(t) + cos\omega_c t]^2$$

$$i_1 = ax(t) + acos\omega_c t + bx^2(t) + 2bx(t)cos\omega_c t + bcos^2\omega_c t$$ ………………(3)

Similarly,

$$i_2 = av_2 + bv_2{}^2$$

$$i_2 = a[x(t) - cos\omega_c t] + b[x(t) - cos\omega_c t]^2$$

$$i_2 = av_2 + bv_2{}^2 = ax(t) - acos\omega_c t + bx^2(t) - 2bx(t)cos\omega_c t + bcos^2\omega_c t$$

…………(4)

The output voltage is given by :

$$v_o = i_1 R - i_2 R$$

Substituting the expression for i_1 and i_2 from equations (3) and (4), we get

$$v_o = R[2\,a\,x(t) + 4\,b\,x(t)\,cos\omega_c\,t]$$

Or,

$$v_o = \underbrace{2aRx(t)}_{\text{Modulating Signal}} + \underbrace{4bRx(t)}_{\text{DSB-SC Signal}}\,cos\omega_c t$$

Hence, the output voltage contains a modulating signal term and the DSB-SC signal. The modulating signal term is eliminated, and the second term is allowed to pass through to the output by the LC band pass filter section.

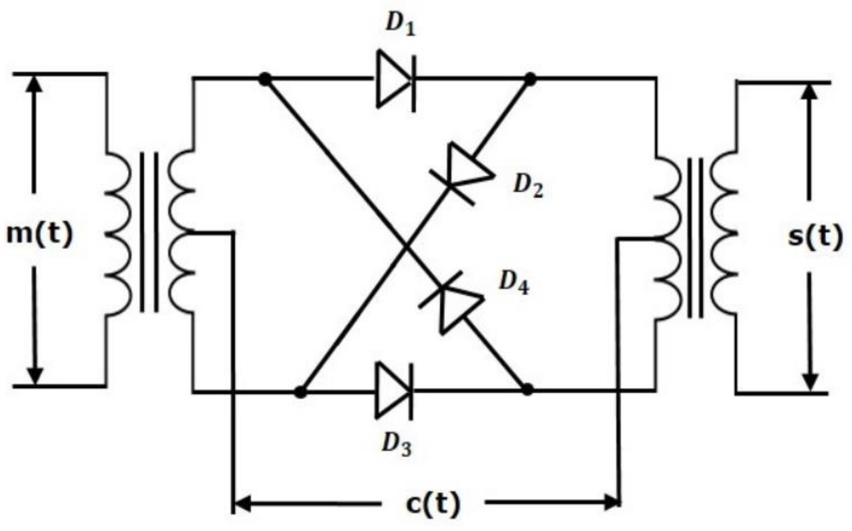
Therefore, final output = 4 b R x(t) cos ω_ct

= K x(t) cos ω_ct

Thus, the diode balanced modulator produces the DSB-SC signal at its output.

8. How do you examine ring modulator for the generation of DSB-SC signal?

Ans:



In this diagram, the four diodes D1D1,D2D2,D3D3 and D4D4 are connected in the ring structure. Hence, this modulator is called as the ring modulator. Two center tapped transformers are used in this diagram. The message signal m(t)m(t) is applied to the input transformer. Whereas, the carrier signals c(t)c(t) is applied between the two center tapped transformers.
For positive half cycle of the carrier signal, the diodes D1D1 and D3D3 are switched ON and the other two diodes D2D2 and D4D4 are switched OFF. In this case, the message signal is multiplied by +1.
For negative half cycle of the carrier signal, the diodes D2D2 and D4D4 are switched ON and the other two diodes D1D1 and D3D3 are switched OFF. In this case, the message signal is multiplied by -1. This results in 18001800 phase shift in the resulting DSBSC wave.
From the above analysis, we can say that the four diodes D1D1, D2D2, D3D3 and D4D4 are controlled by the carrier signal. If the carrier is a square wave, then the Fourier series representation of c(t)c(t) is represented as

$$c\,(t) = \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos[2\pi f_c t\,(2n-1)]$$

We will get DSBSC wave s(t)s(t), which is just the product of the carrier signal c(t)c(t) and the message signal m(t)m(t) i.e.,

$$s\,(t) = \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos[2\pi f_c t\,(2n-1)]m\,(t)$$

The above equation represents DSBSC wave, which is obtained at the output transformer of the ring modulator.
DSBSC modulators are also called as **product modulators** as they produce the output, which is the product of two input signals.

PART II

1. Define modulation index of frequency modulation and phase modulation ?

   Ans:

   The modulation index of FM is defined as the ratio of the frequency deviation of the carrier to the frequency of the modulating signal. **mf = Modulation Index of FM = Δf/fm**.

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.

2. Why frequency modulation is more preferred for voice transmission?

   Ans:

   In radio transmission, an advantage of frequency modulation is **that it has a larger signal-to-noise ratio and therefore rejects radio frequency interference better** than an equal power amplitude modulation (AM) signal. For this reason, most music is broadcast over FM radio.

3. List the advantages of AM and FM ?

   Ans:

	AM	FM
Stands For	AM stands for Amplitude Modulation	FM stands for Frequency Modulation
Origin	AM method of audio transmission was first successfully carried out in the mid 1870s.	FM radio was developed in the United states in the 1930s, mainly by Edwin Armstrong.
Modulating differences	In AM, a radio wave known as the "carrier" or "carrier wave" is modulated in amplitude by the signal that is to be transmitted. The frequency and phase remain the same.	In FM, a radio wave known as the "carrier" or "carrier wave" is modulated in frequency by the signal that is to be transmitted. The amplitude and phase remain the same.
Pros and cons	AM has poorer sound quality compared with FM, but is cheaper and can be transmitted over long distances. It has a lower bandwidth so it can have more stations available in any frequency range.	FM is less prone to interference than AM. However, FM signals are impacted by physical barriers. FM has better sound quality due to higher bandwidth.

4. What are the types of modulation?

   Ans:

   Modulation techniques are roughly divided into four types: **Analog modulation, Digital modulation, Pulse modulation , and Spread spectrum method**.

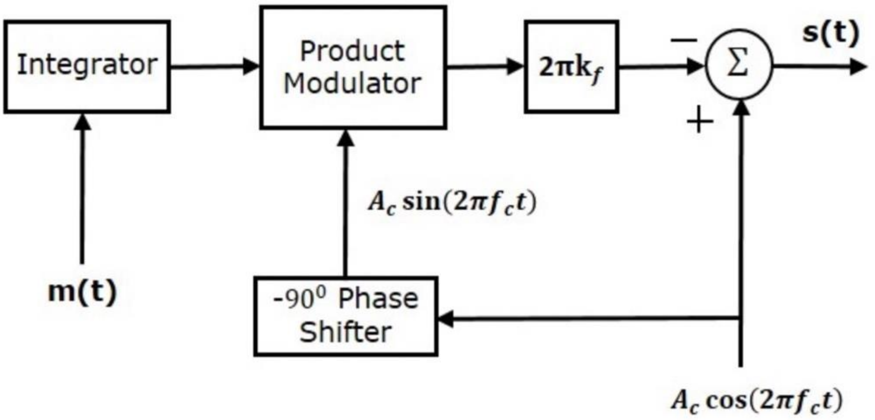
5. State the Carson’s rule to determine the bandwidth of FM.

Ans:

Carson's rule estimates the FM signal bandwidth as **BT = 2(75 + 15) = 180 kHz** which is six times the 30 kHz bandwidth that would be required for AM modulation.

6. Draw the block diagram of a method for generating a narrow band FM.

Ans:



7. Give the mathematical expression for FM and PM.?

Ans:

The basic FM equation is presented in Equation: y(t)=Asin(2πfct+Isin(2πfmt)), where the parameters are defined as follows: fc= carrier frequency (Hz)

The PM wave is obtained by varying the phase angle Φ of a carrier in proportion with the amplitude of the modulating voltage. If the carrier voltage is expressed as under: Then, the PM wave can be expressed as under: Here, **Φm = Maximum phase change corresponding to the maximum amplitude of the** modulating signal.

8. Compare WBFM and NBFM ?

Ans:

When spectrum efficiency is important, Narrowband FM (NBFM) is used but when better signal quality is required, Wideband FM (WBFM) is used at the expense of greater spectrum usage. The term WBFM is used in applications where the modulation index is equal to or larger than 1.

PART IIA

1. Obtain the expression for the single tone frequency modulated signal and hence prove that is the constant envelope modulation requiring infinite bandwidth?

   Ans:

   Let $m(t) = A_m \cos \omega_m t$. Then

$$s(t) = A_c \cos\left(\omega_c t + \frac{k_\omega A_m}{\omega_m} \sin \omega_m t\right)$$

   The *modulation* index is defined as

$$\beta = \frac{k_\omega A_m}{\omega_m} = \frac{\text{peak frequency deviation}}{\text{modulating frequency}}$$

   It can be shown that $s(t)$ has the series expansion

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[(\omega_c + n\omega_m)t] \qquad (12)$$

   where $J_n(\beta)$ is the n-th order Bessel function of the first kind. These functions can be computed by the series

$$J_n(x) = \sum_{m=0}^{\infty} (-1)^m \frac{\left(\frac{1}{2}x\right)^{n+2m}}{m!(n+m)!}$$

   Clearly, the spectrum of the FM signal is much more complex than that of the AM signal.

   - There are components at the infinite set of frequencies $\{\omega_c + n\omega_m; n = -\infty,\cdots,\infty\}$
   - The sinusoidal component at the carrier frequency has amplitude $J_0(\beta)$ and can actually become zero for some β.

2. (i) Show the mathematical expression for Wideband Frequency Modulation. Also compare and contrast its characteristics with Narrowband Frequency modulation.
   (ii) How do you obtain FM from PM and vice versa? Explain

3. What are the methods of FM generation and explain an indirect method to generate an FM signal?

   Ans:

   - Methods of FM Generation.

- Direct Methods.
- Indirect Methods.
- Direct-FM Varactor Diode Modulator.
- Crossby Direct FM Transmitter.
- PLL Direct FM Modulator.
- FM Reactance Modulator.
- Frequency-stabilized Reactance FM.

Indirect Method:

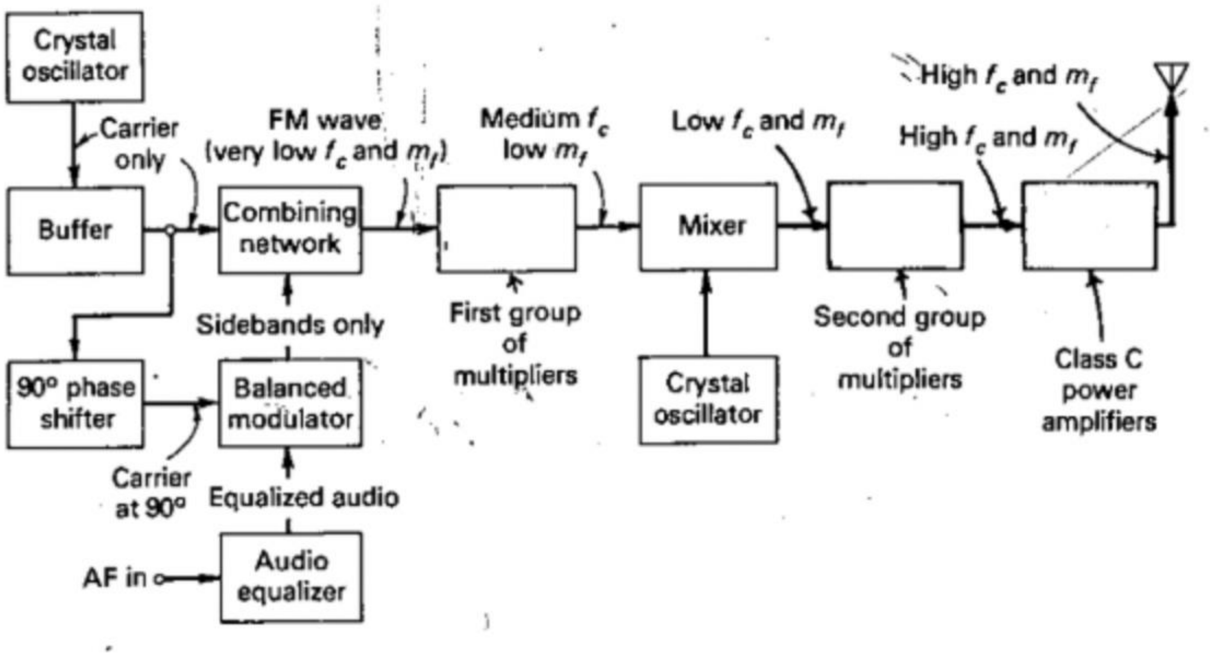


Figure: Armstrong method (Indirect method of FM generation)

Operation:

- The crystal oscillator generates the carrier at low frequency typically at 1MHz. This is applied to the combining network and a 90° phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies. The modulating signal is then applied to a balanced modulator.
- The balanced modulator produced two side bands such that their resultant is 90° phase shifted with respect to the unmodulated carrier.
- The unmodulated carrier and 90° phase shifted sidebands are added in the combining network.
- At the output of the combining network we get FM wave. This wave has a low carrier frequency fc and low value of the modulation index mf .
- The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a

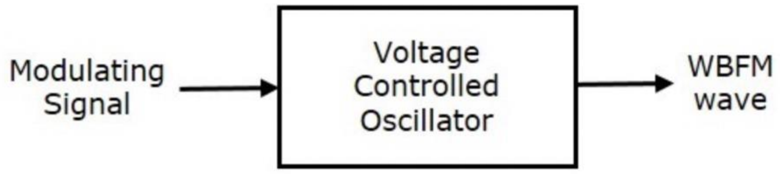
mixer and then the fc and mf both are raised to required high values using the second group of multipliers.

- The FM signal with high fc and high mf is then passed through a class C power amplifier to raise the power level of the FM signal.
- The Armstrong method uses the phase modulation to generate frequency modulation.

4. With a neat diagram, describe the concepts of FM transmitters with direct method of generation.

Ans:

In direct FM generation, the instantaneous frequency of the carrier is changed directly in proportion with the message signal. For this, a device called **voltage controlled oscillator (VCO)** is used. A VCO can be implemented by using a sinusoidal oscillator with a tuned circuit having a high value of Q.



Here, the modulating signal $m(t)$ is applied as an input of Voltage Controlled Oscillator (VCO). VCO produces an output, which is nothing but the WBFM.

$$f_i \; \alpha \; m(t)$$

$$\Rightarrow f_i = f_c + k_f m(t)$$

Where,

f_i is the instantaneous frequency of WBFM wave.

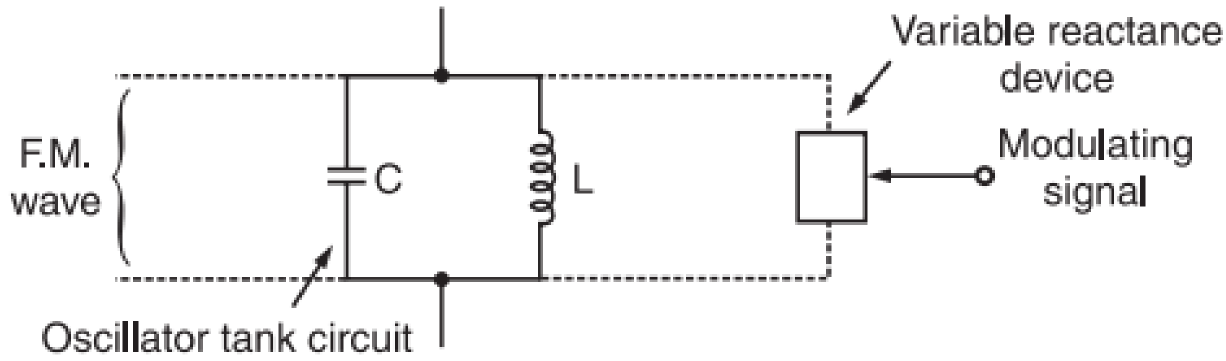
5. Describe how FM generation is achieved using Varactor and reactance modulators.

Ans:

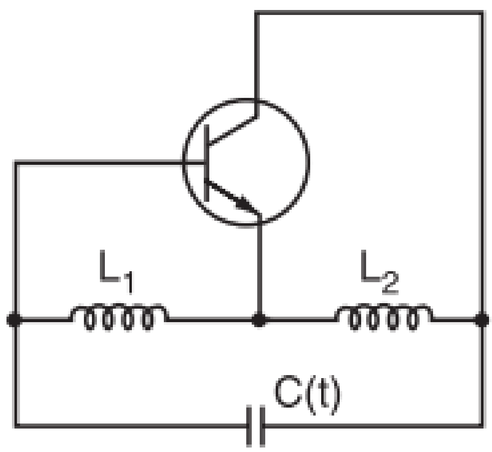
- **Reactance Modulator**

In direct FM generation, the instantaneous frequency of the carrier is changed directly in proportion with the message signal.
For this, a device called voltage controlled oscillator (VCO) is used.

A VCO can be implemented by using a sinusoidal oscillator with a tuned circuit having a high value of Q.
The frequency of this oscillator is changed by changing the reactive components involved in the tuned circuit. If L or C of a tuned circuit of an oscillator is changed in accordance with the amplitude of modulating signal then FM can be obtained across the tuned circuit as shown in figure 1 below.



A two or three terminal device is placed across the tuned circuit. The reactance of the device is varied proportional to modulating signal voltage. This will vary the frequency of the oscillator to produce FM. The devices used are FET, transistor or varactor diode.
An example of direct FM is shown in figure 1 which uses a Hartley oscillator along with a varactor diode.
The varactor diode is reverse biased. Its capacitance is dependent on the reverse voltage applied across it. This capacitance is shown by the capacitor C(t) in figure 2.



- **Varactor Diode Modulator**

The varactor diode FM modulator has been shown below in figure 3.

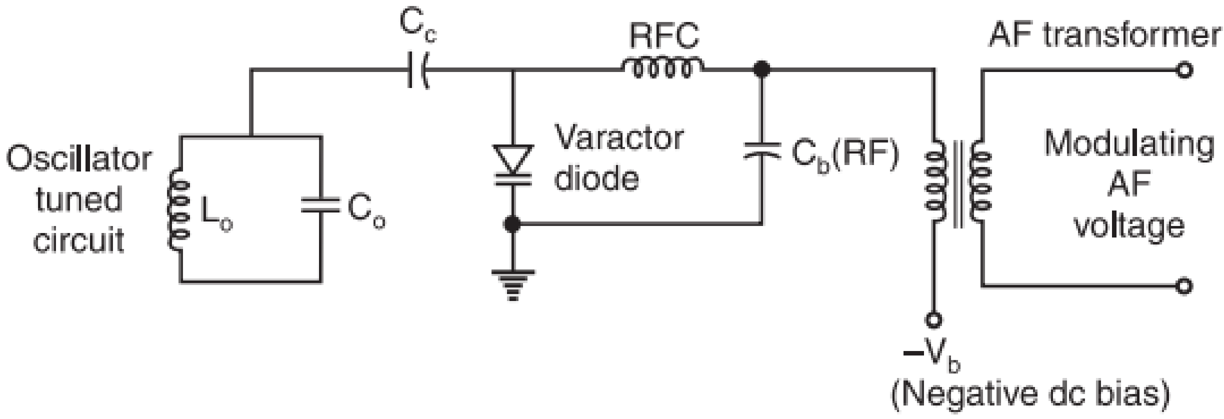


Fig.3 : Varactor Diode Modulator
A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied bias and The varactor diode must be reverse biased.

Working Operation:

The varactor diode is reverse biased by the negative dc source $-V_b$.
The modulating AF voltage appears in series with the negative supply voltage. Hence, the voltage applied across the varactor diode varies in proportion with the modulating voltage.
This will vary the junction capacitance of the varactor diode.
The varactor diode appears in parallel with the oscillator tuned circuit.
Hence the oscillator frequency will change with change in varactor dioide capacitance and FM wave is produced.

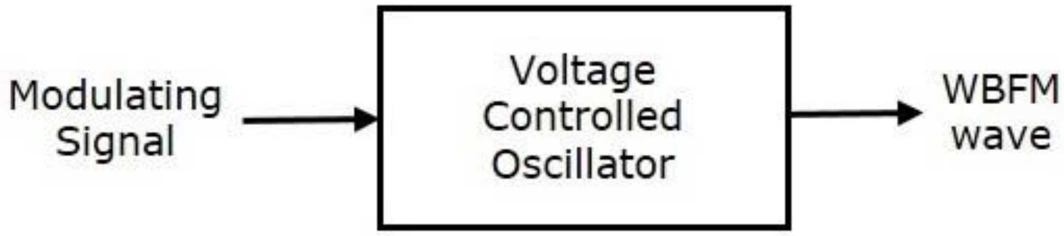
The RFC will connect the dc and modulating signal to the varactor diode but it offers a very high impedance at high oscillator frequency. Therefore, the oscillator circuit is isolated from the dc bias and modulating signal.

6. Explain the principle of indirect method of generating a wideband FM signal

Ans:

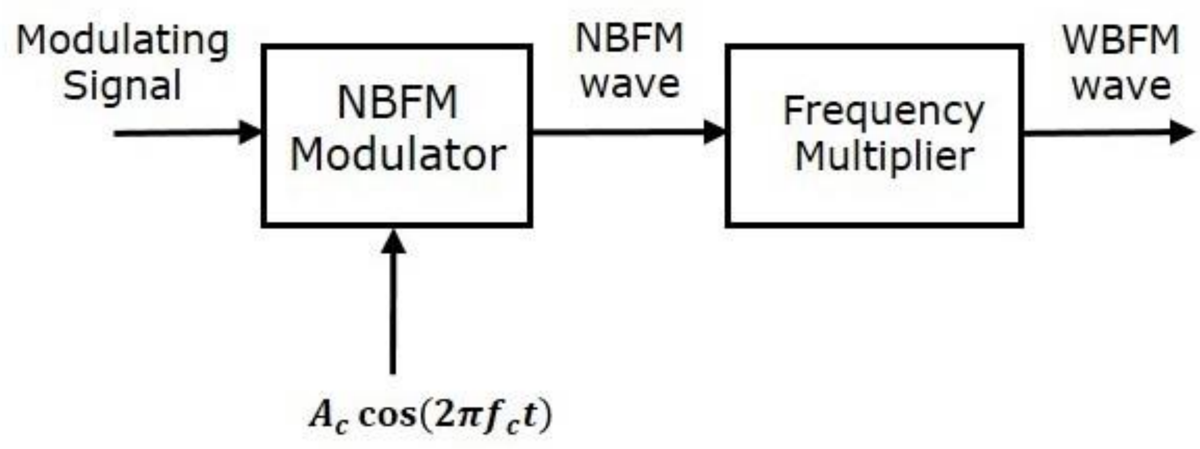
- Direct Method

This method is called as the Direct Method because we are generating a wide band FM wave directly. In this method, Voltage Controlled Oscillator (VCO) is used to generate WBFM. VCO produces an output signal, whose frequency is proportional to the input signal voltage. This is similar to the definition of FM wave. The block diagram of the generation of WBFM wave is shown in the following figure.



- Indirect Method

This method is called as Indirect Method because we are generating a wide band FM wave indirectly. This means, first we will generate NBFM wave and then with the help of frequency multipliers we will get WBFM wave. The block diagram of generation of WBFM wave is shown in the following figure.



7. With the phasor representations, demonstrate and explain the working of Foster Seeley discriminator

   Ans:

   The FOSTER-SEELEY DISCRIMINATOR is also known as the PHASE-SHIFT DISCRIMINATOR. It uses a double-tuned rf transformer to convert frequency variations in the received fm signal to amplitude variations. These amplitude variations are then rectified and filtered to provide a dc output voltage. This voltage varies in both amplitude and polarity as the input signal varies in frequency. A typical discriminator response curve is shown in figure 3-10. The output voltage is 0 when the input frequency is equal to the carrier frequency (f_r). When the input frequency rises above the center frequency, the output increases in the positive direction. When the input frequency drops below the center frequency, the output increases in the negative direction.

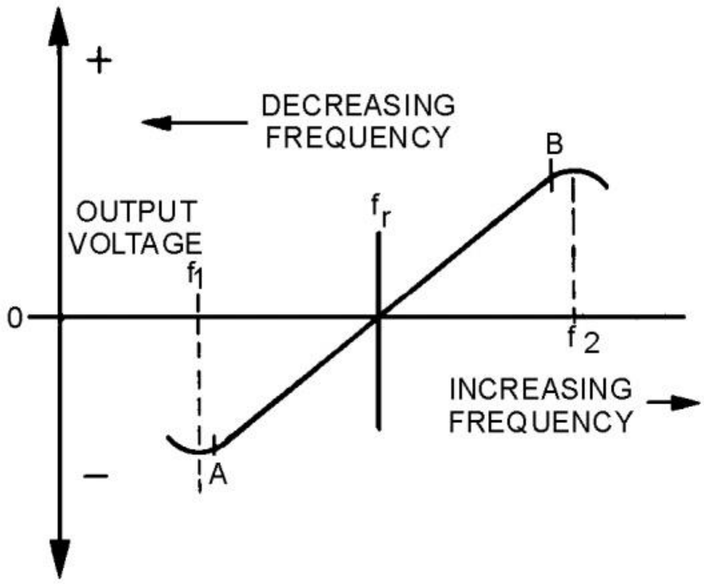


Figure 3-10.—Discriminator response curve.

The output of the Foster-Seeley discriminator is affected not only by the input frequency, but also to a certain extent by the input amplitude. Therefore, using limiter stages before the detector is necessary.

Circuit Operation of a Foster-Seeley Discriminator

View (A) of figure 3-11 shows a typical Foster-Seeley discriminator. The collector circuit of the preceding limiter/amplifier circuit (Q1) is shown. The limiter/amplifier circuit is a special amplifier circuit which limits the amplitude of the signal. This limiting keeps interfering noise low by removing excessive amplitude variations from signals. The collector circuit tank consists of C1 and L1. C2 and L2 form the secondary tank circuit. Both tank circuits are tuned to the center frequency of the incoming fm signal. Choke L3 is the dc return path for diode rectifiers CR1 and CR2. R1 and R2 are not always necessary but are usually used when the back (reverse bias) resistance of the two diodes is different. Resistors R3 and R4 are the load resistors and are bypassed by C3 and C4 to remove rf. C5 is the output coupling capacitor.

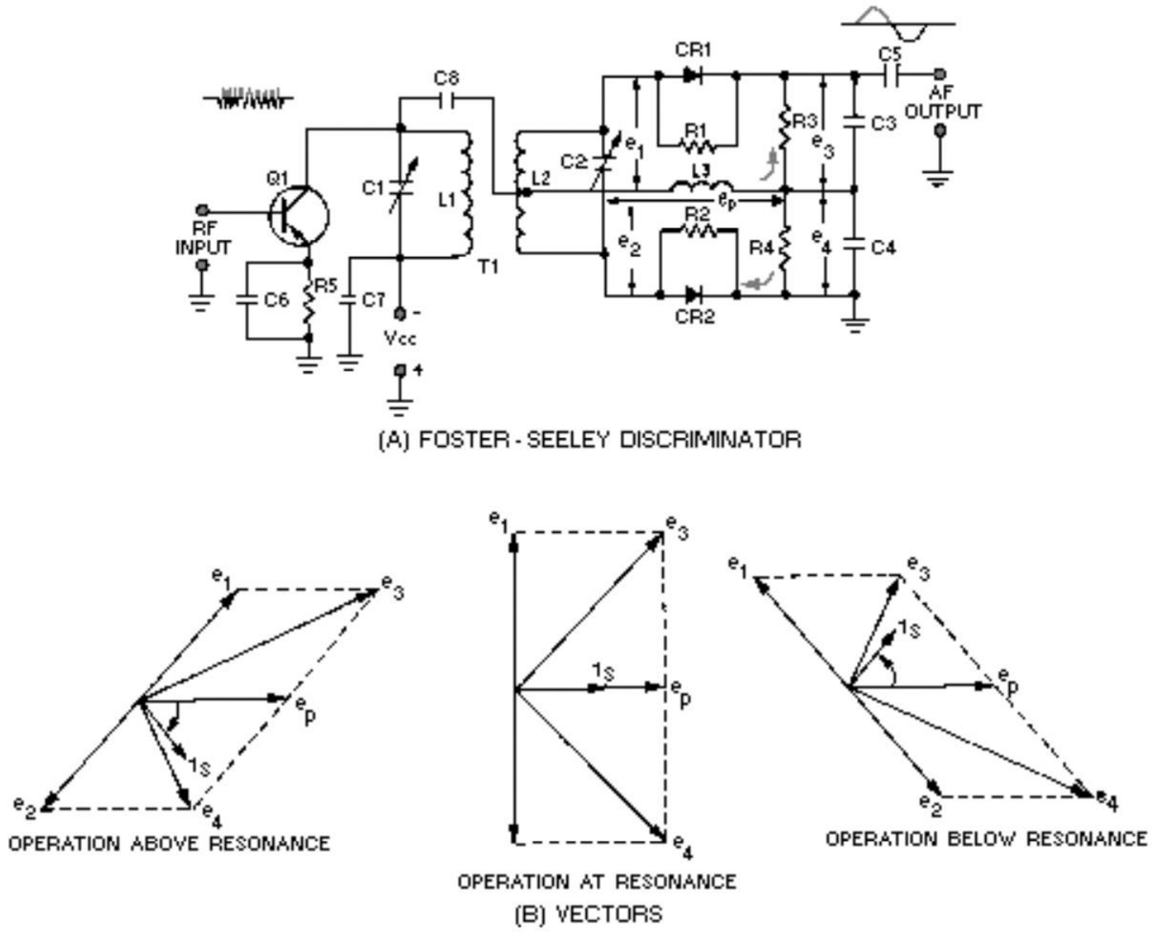


Figure 3-11.—Foster-Seeley discriminator. FOSTER-SEELEY DISCRIMINATOR.

PART III

1. State central limit theorem.

   Ans:

   The central limit theorem states that if you have a population with **mean μ and standard deviation σ** and take sufficiently large random samples from the population with replacement , then the distribution of the sample means will be approximately normally distributed.

2. Define random variable. Specify the sample space and the random variable for a coin tossing experiment.

Ans:

A sample space is a collection of all possible outcomes of a random experiment. A random variable is a function defined on a sample space. So A sample space for a coin toss is a set {H,T}. defines $Heads$= 0 and $Tails$=1, and says that X={0,1}X={0,1} is a random variable.

3. List the properties of the cumulative distributive function.

   Ans:

   1 The cumulative distribution function $F_X(x)$ is a non-decreasing function. This follows directly from the result we have just derived: For $a < b$, we have

   $$\Pr(a < X \le b) \ge 0 \implies F_X(b) - F_X(a) \ge 0 \implies F_X(a) \le F_X(b).$$

   2 As $x \to -\infty$, the value of $F_X(x)$ approaches 0 (or equals 0). That is, $\lim_{x\to-\infty} F_X(x) = 0$. This follows in part from the fact that $\Pr(\varnothing) = 0$.

   3 As $x \to \infty$, the value of $F_X(x)$ approaches 1 (or equals 1). That is, $\lim_{x\to\infty} F_X(x) = 1$. This follows in part from the fact that $\Pr(\mathcal{E}) = 1$.

4. Describe mean, autocorrelation and covariance of a random process.

   Ans:

   Mean : The mean of a random process is **the average of all realizations of that process**. In. order to find this average, we must look at a random signal over a range of time (possible. values) and determine our average from this set of values.

   Autocorrelation : Basically the autocorrelation function **defines how much a signal is similar to a time-shifted version of itself**. A random process X(t) is called a second order process if $E[X^2(t)] < \infty$ for each $t \in T$.

   Covariance : covariance is **a measure of the joint variability of two random variables**. The sign of the covariance therefore shows the tendency in the linear relationship between the variables.

5. What are the properties of an autocorrelation function?

   Ans:

   Properties of Auto-Correlation Function R(Z):

   (i) The mean square value of a random process can be obtained from the auto-correlation function R(Z).

   (ii) R(Z) is even function Z.

   (iii) R(Z) is maximum at Z = 0 e.e. $|R(Z)| \le R(0)$. In other words, this means the maximum value of R(Z) is attained at Z = 0.

(iv) If R(Z) is the auto-correlation of a stationary random process {x(t)} with no periodic components and with non-zeros means then

6. Outline Ergodic processes and Gaussian processes.

   Ans:

   Ergodic processes : a stochastic process is said to be ergodic if its statistical properties can be deduced from a single, sufficiently long, random sample of the process. Conversely, a process that is not ergodic is a process that changes erratically at an inconsistent rate.

   Gaussian processes : a Gaussian process is a stochastic process (a collection of random variables indexed by time or space), such that every finite collection of those random variables has a multivariate normal distribution, i.e. every finite linear combination of them is normally distributed.

7. Express the autocorrelation function and power spectral density of white noise.

   Ans:

   The spectral density of white noise is Uniform and the autocorrelation function of White noise is the Delta function. Explanation: The power spectral density is basically the Fourier transform of the autocorrelation function of the power signal, i.e. **S x (f) = F .**

8. Point out the Rayleigh and Rician probability density functions.

   Ans:

   The Rayleigh distribution is **a distribution of continuous probability density function**. It is named after the English Lord Rayleigh. This distribution is widely used for the following: Communications - to model multiple paths of densely scattered signals while reaching a receiver.

9. Distinguish between random variable and random process.

   Ans:

   A random variable is a variable which can take **different** values and the values that it takes depends on some probability distribution rather than a deterministic rule. A random process is a process which can be in a number of different states and the transition from one state to another is random.

10. Give the conditions to be satisfied for wide sense stationary.

    Ans:

    A random process is called weak-sense stationary or wide-sense stationary (WSS) if **its mean function and its correlation function do not change by shifts in time**.

PART IIIA

1. Explain the following terms: Random variable, Gaussian process and Central limit theorem

   Ans:

   Random variable : A random variable is a **variable whose value is unknown or a function that assigns values to each of an experiment's outcomes**. A random variable can be either discrete (having specific values) or continuous (any value in a continuous range).

   Gaussian process : a Gaussian process is **a stochastic process (a collection of random variables indexed by time or space)**, such that every finite collection of those random variables has a multivariate normal distribution, i.e. every finite linear combination of them is normally distributed.

   Central limit theorem : In probability theory, the central limit theorem (CLT) states that **the distribution of a sample variable approximates a normal distribution** (i.e., a "bell curve") as the sample size becomes larger, assuming that all samples are identical in size, and regardless of the population's actual distribution shape.

2. (i) For ergodic process show that mean of the time average is equal to ensemble mean.

   (ii) Differentiate the strict-sense stationary with that of wide sense stationary process.

3. (i) Analyse the following terms mean, correlation, covariance and ergodicity.

   Ans:

   Mean : The mean of a random process is **the average of all realizations of that process**. In. order to find this average, we must look at a random signal over a range of time (possible. values) and determine our average from this set of values.

Correlation : Correlation is a **statistical measure that expresses the extent to which two variables are linearly related** (meaning they change together at a constant rate). It's a common tool for describing simple relationships without making a statement about cause and effect.

Covariance : Covariance is a measure of the joint variability of two random variables. If the greater values of one variable mainly correspond with the greater values of the other variable, and the same holds for the lesser values, the covariance is positive.

Ergodicity : ergodicity expresses the idea that a point of a moving system, either a dynamical system or a stochastic process, will eventually visit all parts of the space that the system moves in, in a uniform and random sense.

(ii) Explain the properties of the auto correlation function.

Ans:

Properties of Auto-Correlation Function R(Z):

(i) The mean square value of a random process can be obtained from the auto-correlation function R(Z).

(ii) R(Z) is even function Z.

(iii) R(Z) is maximum at Z = 0 e.e. $|R(Z)| \leq R(0)$. In other words, this means the maximum value of R(Z) is attained at Z = 0.

(iv) If R(Z) is the auto-correlation of a stationary random process {x(t)} with no periodic components and with non-zeros means then

4. (i) Demonstrate the advantages of Gaussian Modelling of a random process.

Ans:

Gaussian Processes (GP) are a generic supervised learning method designed to solve regression and probabilistic classification problems.

The advantages of Gaussian processes are:

- The prediction interpolates the observations (at least for regular kernels).
- The prediction is probabilistic (Gaussian) so that one can compute empirical confidence intervals and decide based on those if one should refit (online fitting, adaptive fitting) the prediction in some region of interest.
- Versatile: different kernels can be specified. Common kernels are provided, but it is also possible to specify custom kernels.

(ii)Describe about stationary processes and its classifications.

Ans:

We can classify random processes based on many different criteria. Intuitively, a random process **{X(t),t∈J} is stationary** if its statistical properties do not change by time. For example, for a stationary process, X(t) and X(t+Δ) have the same probability distributions.

- **Strict stationarity** means that the joint distribution of any moments of any degree (e.g. expected values, variances, third order and higher moments) within the process is **never** dependent on time. This definition is in practice too strict to be used for any real-life model.
- **First-order stationarity** series have means that never changes with time. Any other statistics (like variance) *can* change.
- **Second-order stationarity** (also called weak stationarity) time series have a constant mean, variance and an autocovariance that doesn't change with time. Other statistics in the system are free to change over time. This constrained version of strict stationarity is very common.
- **Trend-stationary models** fluctuate around a deterministic trend (the series mean). These deterministic trends can be linear or quadratic, but the amplitude (height of one oscillation) of the fluctuations neither increases nor decreases across the series.
- **Difference-stationary models** are models that need one or more differencing to become stationary

5. Generalize the equation for finding the probability density function of a one to one differential function of a given random variable.

Ans:

The Probability Density Function(PDF) defines the probability function representing the density of a continuous random variable lying between a specific range of values. In other words, the probability density function produces the likelihood of values of the continuous random variable. Sometimes it is also called a probability distribution function or just a probability function. However, this function is stated in many other sources as the function over a broad set of values. Often it is referred to as cumulative distribution function or sometimes as probability mass function(PMF). However, the actual truth is PDF (probability density function) is defined for continuous random variables, whereas PMF (probability mass function) is defined for discrete random variables.

In the case of a continuous random variable, the probability taken by X on some given value x is always 0. In this case, if we find P(X = x), it does not work. Instead of this, we must calculate the probability of X lying in an interval (a, b). Now, we have to figure it for P(a< X< b), and we can calculate this using the formula of PDF. The Probability density function formula is given as,

$$\mathbf{P(a < X < b) = \int_a^b f(x)\,dx}$$

Or

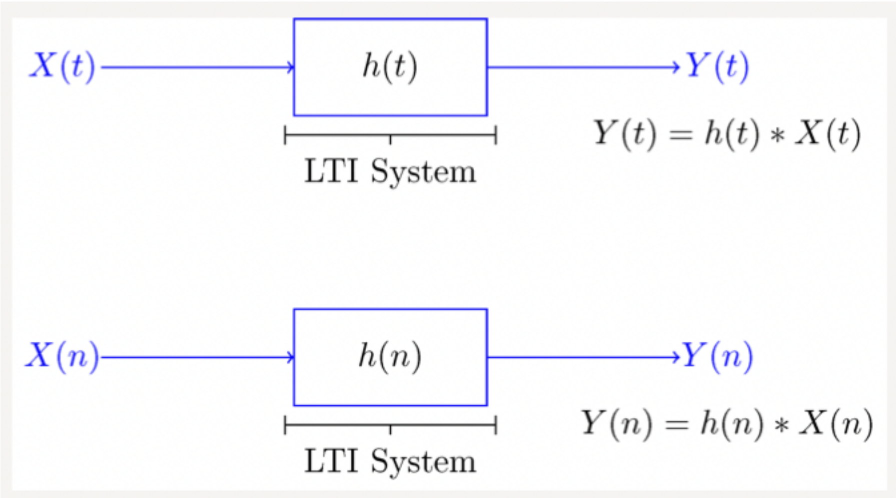
$$\mathbf{P(a \le X \le b) = \int_a^b f(x)\,dx}$$

This is because, when X is continuous, we can ignore the endpoints of intervals while finding probabilities of continuous random variables. That means, for any constants a and b,

$P(a \le X \le b) = P(a < X \le b) = P(a \le X < b) = P(a < X < b)$.

6. (i) Write about Transmission of random process through a Linear Time Invariant (LTI) filter.

Ans:



LTI Systems with Random Inputs:

Consider an LTI system with impulse response $h(t)$. Let $X(t)$ be a WSS random process. If $X(t)$ is the input of the system, then the output, $Y(t)$, is also a random process. More specifically, we can write

$$\begin{aligned} Y(t) &= h(t) * X(t) \\ &= \int_{-\infty}^{\infty} h(\alpha)X(t-\alpha)\, d\alpha. \end{aligned}$$

Here, our goal is to show that $X(t)$ and $Y(t)$ are jointly WSS processes. Let's first start by calculating the mean function of $Y(t)$, $\mu_Y(t)$. We have

$$\begin{aligned} \mu_Y(t) = E[Y(t)] &= E\left[\int_{-\infty}^{\infty} h(\alpha)X(t-\alpha)\, d\alpha\right] \\ &= \int_{-\infty}^{\infty} h(\alpha)E[X(t-\alpha)]\, d\alpha \\ &= \int_{-\infty}^{\infty} h(\alpha)\mu_X\, d\alpha \\ &= \mu_X \int_{-\infty}^{\infty} h(\alpha)\, d\alpha. \end{aligned}$$

We note that $\mu_Y(t)$ is not a function of t, so we can write

$$\mu_Y(t) = \mu_Y = \mu_X \int_{-\infty}^{\infty} h(\alpha)\, d\alpha.$$

Let's next find the cross-correlation function, $R_{XY}(t_1, t_2)$. We have

$$\begin{aligned} R_{XY}(t_1,t_2) = E[X(t_1)Y(t_2)] &= E\left[X(t_1)\int_{-\infty}^{\infty} h(\alpha)X(t_2-\alpha)\, d\alpha\right] \\ &= E\left[\int_{-\infty}^{\infty} h(\alpha)X(t_1)X(t_2-\alpha)\, d\alpha\right] \\ &= \int_{-\infty}^{\infty} h(\alpha)E[X(t_1)X(t_2-\alpha)]\, d\alpha \\ &= \int_{-\infty}^{\infty} h(\alpha)R_X(t_1, t_2-\alpha)\, d\alpha \\ &= \int_{-\infty}^{\infty} h(\alpha)R_X(t_1-t_2+\alpha)\, d\alpha \qquad \text{(since } X(t) \text{ is WSS).} \end{aligned}$$

We note that $R_{XY}(t_1, t_2)$ is only a function of $\tau = t_1 - t_2$, so we may write

$$\begin{aligned} R_{XY}(\tau) &= \int_{-\infty}^{\infty} h(\alpha)R_X(\tau+\alpha)\, d\alpha \\ &= h(\tau) * R_X(-\tau) = h(-\tau) * R_X(\tau). \end{aligned}$$

Similarly, you can show that

$$R_Y(\tau) = h(\tau) * h(-\tau) * R_X(\tau).$$

(ii) Find the autocorrelation of a sequence x(t) =A cos(2πfc(t+θ)) where A and fc are constant and θ is afc(t+θ)) where A and fc are constant and θ is aθ)) where A and fc are constant and θ is a)) where A and fc are constant and θ)) where A and fc are constant and θ is a is a random variable that is uniformly distributed over the interval [-πfc(t+θ)) where A and fc are constant and θ is a πfc(t+θ)) where A and fc are constant and θ is a].

7. (i) Define autocorrelation. Discuss the properties of autocorrelation function.

Ans:

Autocorrelation represents **the degree of similarity between a given time series and a lagged version of itself over successive time intervals**. Autocorrelation measures the relationship between a variable's current value and its past values.

Properties of Auto-Correlation Function R(Z):

(i) The mean square value of a random process can be obtained from the auto-correlation function R(Z).

(ii) R(Z) is even function Z.

(iii) R(Z) is maximum at Z = 0 e.e. $|R(Z)| \leq R(0)$. In other words, this means the maximum value of R(Z) is attained at Z = 0.

(iv) If R(Z) is the auto-correlation of a stationary random process {x(t)} with no periodic components and with non-zeros means then

(ii) Differentiate between random variable and random process

Ans:

A random variable is a variable which can take different values and the values that it takes depends on some **probability distribution** rather than a deterministic rule. A random process is a process which can be in a number of different states and the transition from one state to another is random.

PART IV

1. Describe white noise? Give its characteristics.

Ans:

In signal processing, white noise is a random signal having equal intensity at different frequencies, giving it a constant power spectral density. White noise draws its name from white light, although light that appears white generally does not have a flat power spectral density over the visible band. White noise has zero mean, constant variance, and is uncorrelated in time. As its name suggests, white noise has a power spectrum which is uniformly spread across all allowable frequencies.

2. Define noise figure and noise equivalent temperature.

   Ans:

   Noise figure (NF) measures of degradation of the signal-to-noise ratio (SNR), caused by components, such as amplifiers, in an RF signal chain. Noise temperature is a representation of noise in terms of the temperature required to produce an equivalent amount of Johnson-Nyquist Noise.

3. A Receiver is connected to an antenna of resistance of 50Ω has an equivalent noise resistance of 30Ω. Formulate the receiver noise figure.

   Ans:

   $F = (1 + (R_{eq} / R_a)) = (1 + (30 / 50)) = 1.6$

4. Give the expression for the thermal noise voltage across a resistor. Also define thermal noise.

   Ans:

   THE CHARACTERISTICS OF THERMAL NOISE. $\mathbf{b = k\,T_0}$, where k is Boltzmann's constant ($k = 1{,}38.10^{-23}$ J/°K) and T_0 is the absolute temperature of the dipole in degrees Kelvin. Generally spectral density for a dipole equals b = kT, with T representing the noise temperature of the dipole.

5. Formulate the narrow-band noise m(t) at the IF filter output in terms of its in-phase and quadrature components.
6. Discuss the need for pre-emphasis and de-emphasis

   Ans: Pre-emphasis refers to boosting the relative amplitudes of the modulating voltage for higher audio frequencies from 2 to approximately 15 KHz.

   De-emphasis means attenuating those frequencies by the amount by which they are boosted

   The main purpose is to improve the signal-to-noise ratio for FM reception. Pre-emphasis works by boosting the high-frequency portion of the signal. This compensates for the high-frequency loss in the cable. De-emphasis works by cutting the low-frequency portion of the signal. This may be coupled with an increased transmit voltage.

7. State threshold effect in AM receiver.

   Ans:

When the value of the input signal to noise ratio falls below a particular value (called Threshold value), a **rapid fall of the output signal to noise ratio** is observed by the receiver. The threshold effect occurs due to the presence of large noise in the modulated signal.

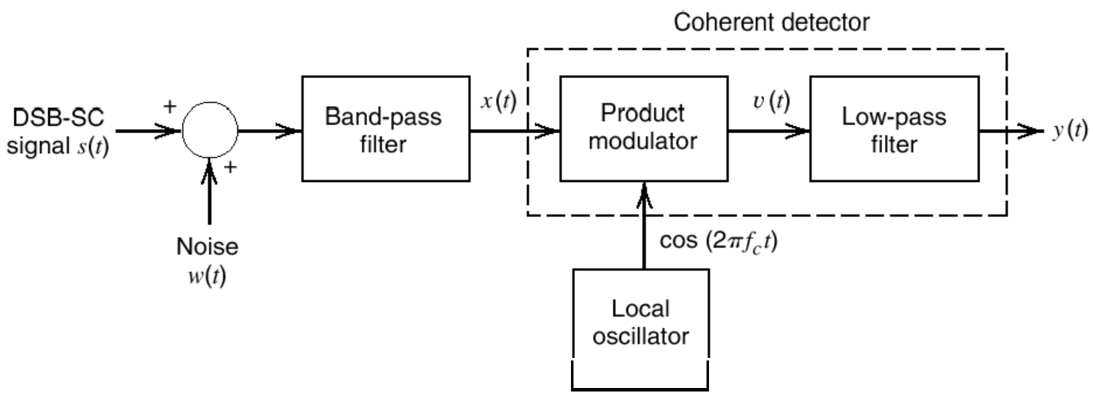
8. What is FM threshold effect?

Ans:

FM threshold effect: In an FM receiver, **the effect produced when the desired-signal gain begins to limit the desired signal, and thus noise limiting (suppression)**. FM threshold effect occurs at (and above) the point at which the FM signal-to-noise improvement is measured.

9. Distinguish the noise performance of DSBSC receiver using coherent detection with AM receiver using envelope detection.

Ans:



10. Illustrate the characteristics of white noise.

Ans:

White noise is **random noise that has a flat spectral density** that is, the noise has the same amplitude, or intensity, throughout the audible frequency range (20 to 20,000 hertz). For example, some people use white noise machines as sleep aids to drown out annoying noises in the environment

11. Classify the methods are to improve FM threshold reduction?

Ans:

Threshold reduction in FM receivers may be achieved by using **an FM demodulator with negative feedback** (commonly referred to as an FMFB demodulator), or by using a phase-locked loop demodulator.

12. Obtain the equation for transfer function of de-emphasis circuit.
13. Name what is capture effect? What do you understand by ‘capture effect’ in FM?

    Ans:

    The capture effect is defined as **the complete suppression of the weaker signal at the receiver's limiter (if present) where the weaker signal is not amplified, but attenuated**. The measurement of how well a receiver rejects a second signal on the same frequency is called its capture ratio.

14. Discuss threshold effect with respect to noise?

    Ans:

    The threshold effect is first noticed when **the input signal - to - noise ratio reaches the vicinity of unity**. Where this exists, the amplitude of the signal is smaller than that of the noise to the point that it becomes difficult if not impossible to detect the presence of signal.

15. Explain noise equivalent bandwidth.

    Ans:

    Equivalent noise bandwidth is **the bandwidth of a perfect rectangular filter that allows the same amount of power to pass as the cumulative bandwidth of the channel selective filters**. You can also refer to ENBW as effective noise bandwidth or noise bandwidth.

16. Calculate the noise voltages for the two resistors 20KΩ & 50KΩ in series at 3000K for a bandwidth of 100KHz.

    Ans:

    Noise voltage $V_n = \sqrt{(4R\ KTB)}$
    Where, $K = 1.381 \times 10^{-23}$ J/K, joules per Kelvin, the Boltzmann constant
    B is the bandwidth at which the power P_n is delivered.
    T noise temperature
    R is the resistance
    Noise voltage by resistors when connected in series is
    $V_n = \sqrt{\{4(R_1 + R_2)\ KTB\}}$
    $= \sqrt{\{4(20 * 10^3 + 50 * 10^3) * 1.381 \times 10^{-23} * 3000 * 100 * 10^3\}}$
    $= 340.59 * 10^{-3}$

17. Point out the characteristic of shot noise.

    Ans:

    Shot noise is **a form of noise that** is encountered in RF and other electronic circuits as a result of the granular nature of current. Shot noise is a form of noise that arises because of the discrete nature of the charges carried by charge carriers, electrons or holes.

18. Evaluate the thermal noise voltage generated at 2900 K for a bandwidth of 100KHz.

Ans:

Temperature, T = 2900 K
Bandwidth, B = 100KHz.
Assuming R = 1Ω

Noise voltage is given as:
$VN=\sqrt{(4RkTB)}$
$VN=\sqrt{(4RkTB)} =\sqrt{(4\times1\times1.38\times10^{-23}\times2900\times100\times10^{3})}= 0.12655253\mu V$

19. DC current of 2 mA flows through the semiconductor junction. Consider the effective noise bandwidth of 1 kHz and Infer the shot noise component.
20. Determine thermal noise voltage across the simple parallel RC circuit shown with R =1kΩ and C=1μF at T = 27oC.

PART IVA

1. Write a short note on

(i) Shot noise with its power spectral density

Ans:

Shot noise is **a form of noise that** is encountered in RF and other electronic circuits as a result of the granular nature of current. Shot noise is a form of noise that arises because of the discrete nature of the charges carried by charge carriers, electrons or holes.

Its spectral density is **proportional to the average current**, I, and is characterized by a white noise spectrum up to a certain cut-off frequency, which is related to the time taken for an electron to travel through the conductor. In contrast to thermal noise, shot noise cannot be eliminated by lowering the temperature.

(ii) Thermal noise with PSD

Ans:

Thermal noise in an ideal resistor is approximately **white**, meaning that the power spectral density is nearly constant throughout the frequency spectrum (however see the section below on extremely high frequencies). When limited to a finite bandwidth, thermal noise has a nearly Gaussian amplitude distribution.

2. Describe in detail various sources of noise.

   Ans:

- **Natural Noise**

Natural noise gets generated due to either natural phenomenon or atmospheric actions like solar flares, radiation in space, electronic storms etc.

It is further classified into atmospheric and extraterrestrial noise.

Atmospheric Noise

The atmospheric actions produce false or spurious signals that get added with the original signal thereby causing interference in the information signal. These spurious signals propagate in the same manner as the original signal.

Hence the receiver at the other end collects both message as well as spurious signals.

Extraterrestrial Noise

This type of noise is generated by either the sun or the outer space. This type of noise is classified into two categories:

Solar Noise: Solar noise is generated by the sun. As Sun is a large body with extremely high temperature thus it emits or releases high electrical energy in noise form over a broad frequency range.
However, the intensity of the produced noise signal changes timely. This is so because the temperature change of the sun follows 11 years of the life cycle. Hence large electrical disturbances occur after the period of every 11 years. While at other years the noise level is comparatively low.

Cosmic Noise: This noise originates from the stars present in the outer space. As distant stars are also very high-temperature bodies and are also termed as the sun. The noise generated from the star is similar to that generated from the sun. Cosmic noise is also known as **black body noise**.

Not only the stars but the galaxies and other virtual point sources like **quasars**and **pulsars** in the outer space produces cosmic noise.

- ### Man-made noise

This type of extrinsic noise is also known as industrial noise. These are basically the electrical noise that gets produced by the wear and tear of the circuit being used. The source of man-made noise is electric motors, high current circuits, florescent lights switch gears etc.

When these machines operate, arc discharge takes place and this discharge generates noise signals in the communication system.

The frequency spectrum of man-made noise lies between **1 MHz** to **600 MHz**.
2 ***Internal Noise*** is the fundamental noise that gets generated by the electronic equipment involved in the system itself. They are called so because these are nothing but an integral part of the system.
Proper designing of the communication system can reduce or overcome noise due to internal sources.

Internal Noise is classified as follows:

Thermal Noise

As we already know that an electrical signal is transmitted through a channel by the help of conductors. So, the electrons present in the conductors move randomly.

The random motion of the electrons is the reason for the thermal energy received by the conductor. However, these free electrons are non-uniformly distributed within the conductor.

Due to this a possibility also exist that at one end the number of free electrons will be comparatively higher than at the other end.

This non-uniform distribution of electrons provides the average voltage to be zero, however, the average power is not zero in this case.

So, this non zero power is nothing but the noise. And as it is the outcome of thermal action. Hence also known as thermal noise power. Thermal noise is sometimes referred as **Johnson noise** or **white noise**.
Shot Noise

Shot noise in a communication channel is the result of random variation in the appearance of electrons and holes at the output side of the device. These random movements are the result of discontinuities in the device which is being used by the system.

The shot noise generates sound like several lead shots are striking over a metal plate or tube.

It also occurs in pn junction diodes, as though movement of carriers within the diode is due to the action of an external potential. But, sometimes their random movement generates shot noise.

Thus we can say non-linearity or discontinuity in the system generates shot noise.

Partition Noise

Here the name itself is indicating the cause for generation of this type of noise.

As it gets generated when the system is composed of multiple paths, and during the flow, the current gets divided in these paths. These are nothing but the result of random variation in the divisions. Due to this reason some devices offer low partition noise while some offers, high.

Flicker Noise

It is also known as low-frequency noise and it occurs because of the variation in the carrier density. Due to this variation or fluctuation, the conductivity of the material gets varied.

So, when a direct current is allowed to flow through the conductor then fluctuating voltage drop across in the conductor results in flicker noise voltage.

It is to be noteworthy here that, **the mean square of flicker noise voltage is directly proportional to the square of the current** flowing through the device.
Transit Time Noise

It is also known as high-frequency noise. It arises when the charge carriers require comparatively more time to travel from one end to another within the conductor. This effect is called **transit time effect**.
For low-frequency applications, this effect is avoidable but for high-frequency applications the effect is unavoidable. Due to this transit time effect, random noise gets generated inside the device and is known as transit time noise.

3. What is coherent detector? Derive an expression for SNR at input (SNRc) and output of (SNRo) of a coherent detector.

4. Express and derive the output SNR for FM reception. Also obtain the figure of merit.

5. (i) Point out the significance of pre-emphasis and de-emphasis in communication system.

   Ans:

   - Pre-emphasis works by boosting the high-frequency portion of the signal. This compensates for the high-frequency loss in the cable.
   - De-emphasis works by cutting the low-frequency portion of the signal. This may be coupled with an increased transmit voltage.
   - Pre-emphasis and de-emphasis provide essentially the same function, which is to provide a flat frequency curve on the receiver side. In actual implementation, de-emphasis can be technically simpler, so it is more often seen between the two.
   - While pre/de-emphasis helps to create a more stable signal, it can also create issues if the system applies too much of either. For instance, if you surpass the

optimal amount, you can end up with too little low-frequency and too much high-frequency.

(ii) Write in detail about FM threshold effect.

Ans:

In an FM receiver, the effect produced when the desired-signal gain begins to limit the desired signal, and thus noise limiting (suppression). Note: FM threshold effect occurs at (and above) the point at which the FM signal-to-noise improvement is measured. The output signal to noise ratio of FM receiver is valid only if the carrier to noise ratio is measured at the discriminator input is high compared to unity. It is observed that as the input noise is increased so that the carrier to noise ratio decreased, the FM receiver breaks. At first individual clicks are heard in the receiver output and as the carrier to noise ratio decreases still further, the clicks rapidly merge in to a crackling or sputtering sound. Near the break point eqn8.50 begins to fail predicting values of output SNR larger than the actual ones. This phenomenon is known as the threshold effect.The threshold effect is defined as the minimum carrier to noise ratio that gives the output SNR not less than the value predicted by the usual signal to noise formula assuming a small noise power. For a qualitative discussion of the FM threshold effect, Consider, when there is no signal present, so that the carrier is unmodulated. Then the composite signal at the frequency discriminator input is

$$x(t) = [A_c + n_I(t)] \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t \text{-------------------}(1)$$

Where nI(t) and nQ(t) are inphase and quadrature component of the narrow band noise n(t) with respect to carrier wave Accos2 fct. The phasor diagram of fig8.17 below shows the phase relations b/n the various components of x(t) in eqn (1).This effect is shown in fig below, this calculation is based on the following two assumptions:

1. The output signal is taken as the receiver output measured in the absence of noise. The average output signal poweris calculated for a sinusoidal modulation that produces a frequency deviation Iequal to 1/2 of the IF filter bandwidth B, The carrier is thus enabled to swing back and forth across the entire IF band.

2. The average output noise power is calculated when there is no signal present, i.e.,the carrier is unmodulated, with no restriction placed on the value of the carrier to noise ratio.

Assumptions:

- Single-tone modulation, ie: m(t) = Am cos(2Δ fmt)
- The message bandwidth W = fm;
- For the AM system, μ = 1;
- For the FM system, β = 5 (which is what is used in commercial FM transmission, with Δf = 75 kHz, and W = 15 kHz).

6. Formulate the figure of merit for AM system using envelope detector

Ans:

Figure of merit of AM receiver is

$$(SNR)_{C,AM} = \frac{Average\ Power\ of\ AM\ Wave}{Average\ Power\ of\ noise\ in\ message\ bandwidth}$$

$$\Rightarrow (SNR)_{C,AM} = \frac{A_c^{\,2}\left(1 + k_a^{\,2}\right)P}{2WN_0}$$

$$(SNR)_{O,AM} = \frac{Average\ Power\ of\ demodulated\ signal}{Average\ Power\ of\ noise\ at\ Output}$$

$$\Rightarrow (SNR)_{O,AM} = \frac{A_c^{\,2}k_a^{\,2}P}{2WN_0}$$

$$F = \frac{(SNR)_{O,AM}}{(SNR)_{C,AM}}$$

$$\Rightarrow F = \left(\frac{A_c^2 k_a^2 P}{2WN_0}\right) / \left(\frac{A_c{}^2 \left(1 + k_a{}^2\right) P}{2WN_0}\right)$$

$$\Rightarrow F = \frac{K_a{}^2 P}{1 + K_a{}^2 P}$$

Therefore, the Figure of merit of AM receiver is less than one.

7. (i) Discuss in detail about the narrowband noise and analyse the properties of in-phase and quadrature components of narrow band noise.

   Ans:

   A random process X(t) is bandpass or narrowband random process if its power spectral density SX(f) is nonzero only in a small neighbourhood of some high frequency fc Deterministic signals: defined by its Fourier transform Random processes: defined by its power spectral density.
   In-Phase & Quadrature Sinusoidal Components

$$\sin(A + B) = \sin(A)\cos(B) + \cos(A)\sin(B)$$

From the trig identity, we have

$$\begin{aligned} x(t) &\stackrel{\Delta}{=} A\sin(\omega t + \phi) = A\sin(\phi + \omega t) \\ &= [A\sin(\phi)]\cos(\omega t) + [A\cos(\phi)]\sin(\omega t) \\ &\stackrel{\Delta}{=} A_1\cos(\omega t) + A_2\sin(\omega t). \end{aligned}$$

From this we may conclude that every sinusoid can be expressed as the sum of a sine function phase zero) and a cosine function (phase 2). If the sine part is called the ``in-phase'' component, the cosine part can be called the ``phase-quadrature'' component. In general, ``phase quadrature'' means ``90 degrees out of phase,'' i.e., a relative phase shift of ± 2. It is also the case that every sum of an in-phase and quadrature component can be expressed as a single sinusoid at some amplitude and phase. The proof is obtained by working the previous derivation backwards.Figure illustrates in-phase and quadrature components overlaid. Note that they only differ by a relative 00 degree phase shift.

(ii) Explain narrowband noise analyser and noise synthesizer

Narrowband Noise Representation:
In most communication systems, we are often dealing with **band-pass filtering of signals**. Wideband noise will be shaped into bandlimited noise. If the bandwidth of the bandlimited noise is relatively small compared to the carrier frequency, we refer to this as narrowband noise.

8. Evaluate the effective noise temperature of a cascade amplifier from Friss formula

   Ans:

   Noise figure (NF) is the increase in noise power of a device from the input to the output that is greater than the signal gain. In effect, it is the amount of decrease of the signal-to-noise ratio. Like gain, noise figure can be expressed either as a ratio or in decibels (dB). Note: NF is not a power level, so it never has units of dBm.

PART V

1. List the advantages and disadvantage of digital communication system.

   Ans:

   - The benefits of digital communication over analog are listed as follows -
   - In digital signals, the impact of noise interference, distortion is less.
   - It facilitates video conferencing that saves time, money, and effort. We can perform video conferencing with someone or a group of people without any traveling. In video conferencing, we can see the facial expressions, which are helpful in reading the reaction of people.
   - It is easy to implement, less expensive.
   - It is used in military applications.
   - The correction and detection of errors are easy in digital communication, as there is a use of channel coding.
   - As compared to analog signals, it is easy to save and retrieve digital signals.
   - In digital signals, the configuring process is easy as compared to analog signals.
   - There is a common encoding technique in most digital circuits, so for a number of processes, similar devices can be used.

- The probability of cross-talk is very less in digital communication.
- The implementation of hardware is more flexible in digital communication.
- In digital communication, to avoid signal jamming, the spread spectrum technique is used.
- It also facilitates us with audio conferencing by which we can talk to someone or a group of people in another location without traveling. Thus, it saves time, effort, and money.
- To maintain the secrecy of information, the signal processing functions like compression and encryption are employed in digital circuits.
- Digital communication is cheaper and simpler compared to analog signals because of the advancement of IC technologies.

The limitations of digital communication are listed as follows -

- There is high power consumption in digital communication.
- There is a requirement for synchronization in the case of synchronous modulation.
- There is a sampling error.
- The most common limitation of digital communication is that it requires more transmission bandwidth. It is due to the higher data rate because of analog to digital conversion.
- Digital communication requires analog to digital conversion at a high rate.
- There can be a possibility of miscommunication if a user doesn't understand something.

2. Define Band pass sampling.

   Ans:

   In signal processing, undersampling or bandpass sampling is a technique where **one samples a bandpass-filtered signal at a sample rate below its Nyquist rate** (twice the upper cutoff frequency), but is still able to reconstruct the signal.

3. Mention the definition of FDM.

   Ans:

   In telecommunications, **frequency-division multiplexing**(FDM) is a technique by which the total bandwidth available in a communication medium is divided into a series of non-overlapping frequency bands, each of which is used to carry a separate signal.

4. Distinguish natural and flat top sampling.

   Ans:

   The difference between natural sampling and flat top sampling is that: **In natural sampling the analog input is multiplied by a train of uniformly spaced,**

rectangular pulses. While in flat top sampling the top of the samples are flat, this means they have a constant amplitude.

5. Interpret the use of pre-filtering done before sampling.

   Ans:

   Pre-filtering is done before sampling **to remove ambiguity between the continuous signals and discrete signals**. Without such filtering there can be multiple continuous signals which can generate the same samples.

6. What is meant by aliasing?

   Ans:

   In signal processing and related disciplines, aliasing is **an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled**. Aliasing can occur in signals sampled in time, for instance digital audio, or the stroboscopic effect, and is referred to as temporal aliasing.

7. How would you show your understanding of the components required for signal reconstruction?

   Ans:

8. Write about non uniform quantization.

   Ans:

   The **type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic**, is termed as a Non-uniform Quantization. The quantization levels in this type are odd in number. Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

9. Illustrate the two fold effects of quantization process.

   Ans:

   The quantization Process has a two-fold effect: 1. the peak-to-peak range of the input sample values is subdivided into a finite set of decision levels or decision thresholds that are aligned with the risers of the staircase, and 2.

10. Illustrate the difference between uniform and non-uniform quantization.

    Ans:

The key difference between uniform and nonuniform quantization is that **uniform quantization has equal step sizes** while, in nonuniform quantization, the step sizes are not equal. Quantization is one step in the digitization process.

11. Construct the Nyquist sampling Theorem with equation.

Ans:

The Nyquist Sampling Theorem states that: A bandlimited continuous-time **signal can** be sampled and perfectly reconstructed from its samples if the waveform is sampled over twice as fast as it's highest frequency component.

Nyquist limit: the highest frequency component that can be accurately represented: $f\max < f_s/2.$

Nyquist frequency: sampling rate required to accurately represent up to

$f\max$:

$f_s > 2f\max.$

12. A Sinusoidal signal is transmitted using PCM scheme. The target output SNR should be greater than 13 dB. Can you identify the minimum number of representation levels and minimum number of bits required to represent each sample to achieve the above performance.

Ans:

13. A pulse code modulation system uses a uniform quantizer followed by a 6 bit encoder. The bit rate of the system is 50 Mbps. Determine the message bandwidth of the system.

Ans:

It has been given that : Bit rate r = 50 Mbps and N = 7

We know that it rate r = N f_s

There for f_s = 50 Mbps / 7

= 7.14 MHz

There for Max Band Width , BW = f_s /2 = 3.57MHz

14. Outline the input-output characteristic of a compressor and expander.

    Ans:

    The uniform quantizer will quantize the compressed signal, and the expander can extend and invert the compression mechanism to recreate the original signal. To replicate the signal at the receiver, the expander has complementary properties to the compressor so that the **compressor input equals the expander output**.

15. Point out the μ-law of compression

    Ans:

    μ-Law companding is a compression process. It explores the principle that **the higher amplitudes of analog signals are compressed before ADC while expanded after digital-to-analog conversion (DAC)**. As studied in the linear quantizer, the quantization error is uniformly distributed.

16. State in your own words the definition for PPM and PWM.

    Ans:

    PWM stands for Pulse Width Modulation whereas PPM stands for Pulse Position Modulation. PWM and PPM are the two radio receiver protocols that are used for transferring data between the RX and the flight controllers.

17. Express the Quantization noise of a PCM system

    Ans:

    Quantization Noise: It is a type of quantization error, which usually occurs in analog audio signal, **while quantizing it to digital**. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called Quantization Noise.

18. Formulate the concept of PAM and PCM.

    Ans:

    PAM is basically a sampling technique which converts an analog signal into a discrete signal(signal that is continuous in amplitude but discrete in time) where as PCM is an analog-to-digital conversion technique which is **sampling +quantization**.

19. Examine the concept of TDM.

Ans:

Time-division multiplexing (TDM) is a method of transmitting and receiving independent signals over a common signal path by **means of synchronized switches at each end of the transmission line** so that each signal appears on the line only a fraction of time in an alternating pattern.

20. Summarize the advantages and disadvantages of FDM

Ans:

Advantages of FDM:

- It does not need synchronization between its transmitter as well as receiver
- FDM is simpler and easy demodulation
- Less expensive
- FDM system does not need synchronization but TDM needs synchronization. It is an advantage of FDM over TDM
- FDM provides more latency than TDM
- Using FDM system multimedia data can be transferred with very high efficiency and low noise and distortion
- FDM system has high reliability
- It is used for analogue signals
- In this system due to slow narrowband fading, only one channel gets affected
- A large number of the signal can be transmitted simultaneously

Disadvantages of FDM:

- It is suffering the problem of cross talk
- FDM is only used only when a few low-speed channels are desired
- Intermodulation distortion takes place
- The circuitry for FDM is complex than TDM
- FDM requires more hardware than TDM
- FDM system extremely expensive
- FDM provides less throughput
- FDM has not dynamic coordination
- The full bandwidth of the channel cannot be used on the FDM system
- The communication must have very large bandwidth
- A large number of modulator and filter required
- FDM channel can get affected wideband fading
- FDM system needs a carrier wave or carrier signal but TDM does not need carrier signal

PART VA

1. (i) Explain the following terms with respect to sampling Aliasing, Signal Reconstruction and Aperture effect distortion

   Ans:

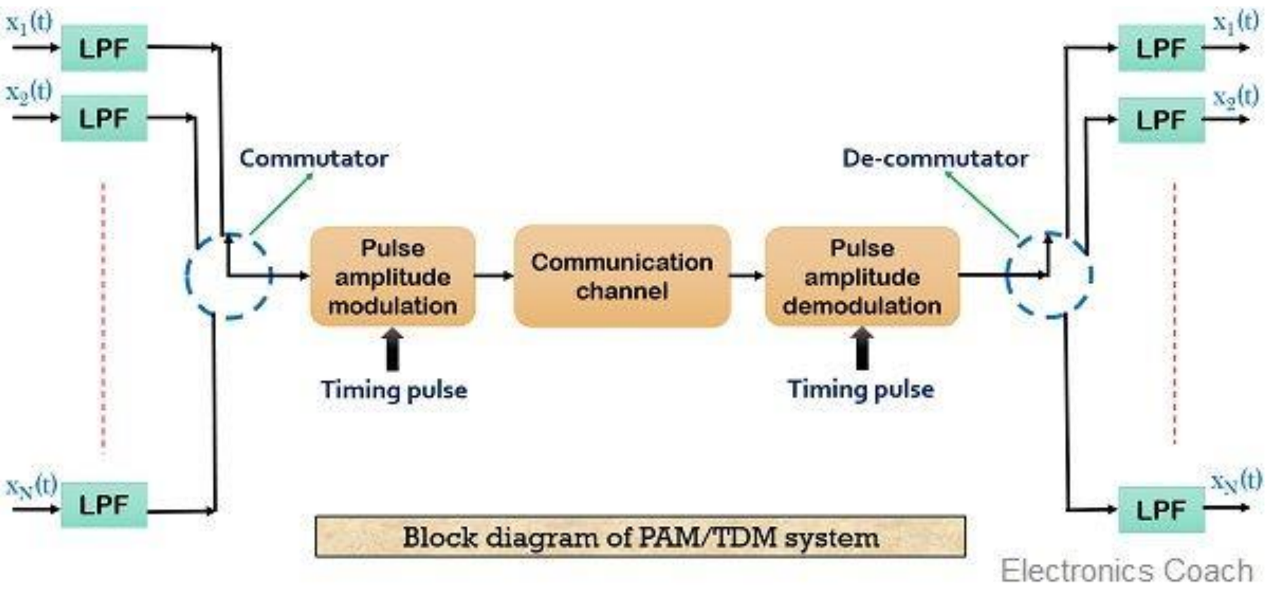
   Aliasing : Aliasing is **an effect of the sampling that causes different signals to become indistinguishable**. Due to aliasing, the signal reconstructed from samples may become different than the original continuous signal. This can drastically deteriorate the performance if proper care is not taken.

   Signal Reconstruction : In signal processing, reconstruction usually means **the determination of an original continuous signal from a sequence of equally spaced samples**. This article takes a generalized abstract mathematical approach to signal sampling and reconstruction.

   Aperture effect distortion : The sampled signal in the flat top sampling consists of attenuated high frequency components and this effect is known as Aperture effect. Aperture effect can be improved by selecting value of pulse **width τ** to be very small and by using equalizer circuit.

   (ii) Outline Time Division multiplexing system for N- number of channels

   A multiplexing technique by which multiple data signals can be transmitted over a common communication channel in different time slots is known as **Time Division Multiplexing (TDM)**.



Let f_m be the maximum signal frequency and f_s is the sampling frequency then

$$f_s \geq 2f_m$$

Thus, the time duration in between successive sample is given as, $T_s = \frac{1}{f_s}$

Rewriting in terms of f_m $T_s \leq \frac{1}{2f_m}$

Now, as we have considered that there are N input channels, then one sample is collected from each of the N samples.

Hence, each interval will provide us with N samples and the spacing between the two is given as

$$\frac{T_s}{N}$$

We know pulse frequency is basically the number of pulses per second and is given by

$$\text{Pulse frequency} = \frac{1}{Spacing\ between\ 2\ samples}$$

$$= \frac{1}{\frac{T_s}{N}}$$

$$= \frac{N}{T_s}$$

$$= \frac{N}{\frac{1}{f_s}}$$

$$= Nf_s$$

For a TDM signal pulse per second is the signalling rate denoted as ‘r’.

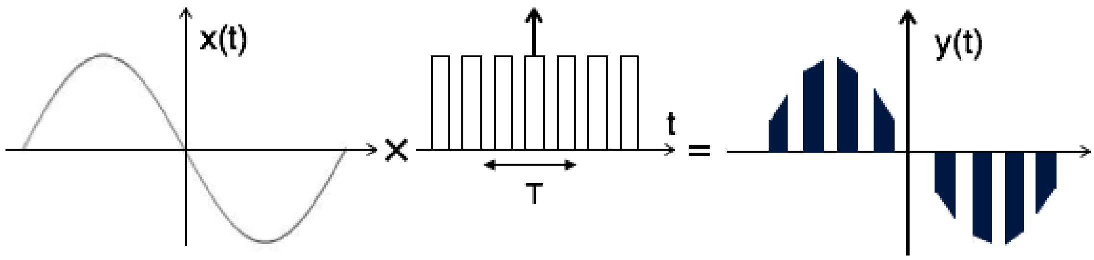
Thus, $r = N\,f_s$

2. Illustrate the following sampling procedures with proper details for Natural Sampling and Flat top Sampling

Ans:

Natural Sampling

Natural sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period T. i.e. you multiply input signal x(t) to pulse train $\Sigma_{n=-\infty}^{\infty} P(t-nT)$ as shown below



The output of sampler is

$$y(t) = x(t) \times \text{pulse train}$$

$$= x(t) \times p(t)$$

$$= x(t) \times \Sigma_{n=-\infty}^{\infty} P(t-nT) \ldots \ \ldots (1)$$

The exponential Fourier series representation of p(t) can be given as

$$p(t) = \Sigma_{n=-\infty}^{\infty} F_n e^{jn\omega_s t} \ \ldots \ \ldots (2)$$

$$= \Sigma_{n=-\infty}^{\infty} F_n e^{j2\pi n f_s t}$$

Where $F_n = \frac{1}{T} \int_{\frac{-T}{2}}^{\frac{T}{2}} p(t) e^{-jn\omega_s t} dt$

$$= \frac{1}{TP}(n\omega_s)$$

Substitute F_n value in equation 2

$$\therefore p(t) = \Sigma_{n=-\infty}^{\infty} \frac{1}{T} P(n\omega_s) e^{jn\omega_s t}$$

$$= \frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s) e^{jn\omega_s t}$$

Substitute p(t) in equation 1

$$y(t) = x(t) \times p(t)$$

$$= x(t) \times \frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s)\, e^{jn\omega_s t}$$

$$y(t) = \frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s)\, x(t)\, e^{jn\omega_s t}$$

To get the spectrum of sampled signal, consider the Fourier transform on both sides.

$$F.T\,[y(t)] = F.T[\frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s)\, x(t)\, e^{jn\omega_s t}]$$

$$= \frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s)\, F.T\,[x(t)\, e^{jn\omega_s t}]$$

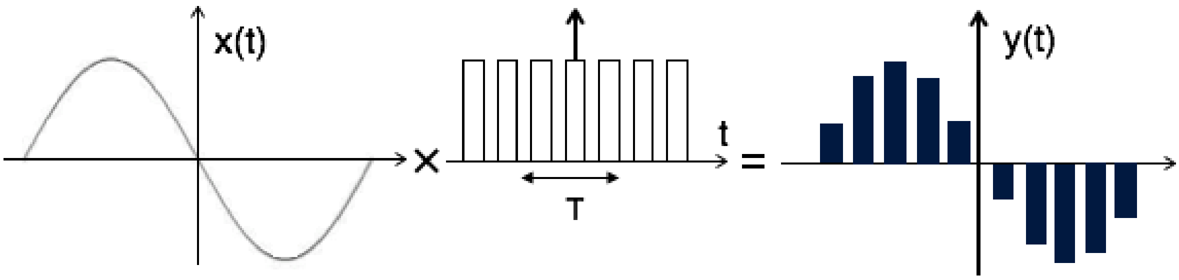
According to frequency shifting property

$$F.T\,[x(t)\, e^{jn\omega_s t}] = X[\omega - n\omega_s]$$

$$\therefore\; Y[\omega] = \frac{1}{T} \Sigma_{n=-\infty}^{\infty} P(n\omega_s)\, X[\omega - n\omega_s]$$

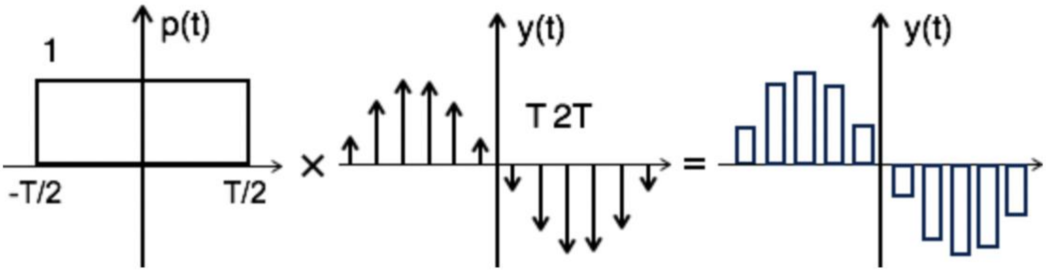
Flat Top Sampling

During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the pulse is in the form of flat top. Here, the top of the samples are flat i.e. they have constant amplitude. Hence, it is called as flat top sampling or practical sampling. Flat top sampling makes use of sample and hold circuit.



Theoretically, the sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal say $y_\delta(t)$ as shown in the diagram:

i.e. $y(t) = p(t) \times y_\delta(t) \ldots \ \ldots (1)$



To get the sampled spectrum, consider Fourier transform on both sides for equation 1

$$Y[\omega] = F.T\,[P(t) \times y_\delta(t)]$$

By the knowledge of convolution property,

$$Y[\omega] = P(\omega)\, Y_\delta(\omega)$$

Here $P(\omega) = TSa(\frac{\omega T}{2}) = 2 \sin \omega T/\omega$

3. Deduce the concept of Low pass sampling, Aliasing and Signal Reconstruction

Ans:

Low Pass Signal Sampling

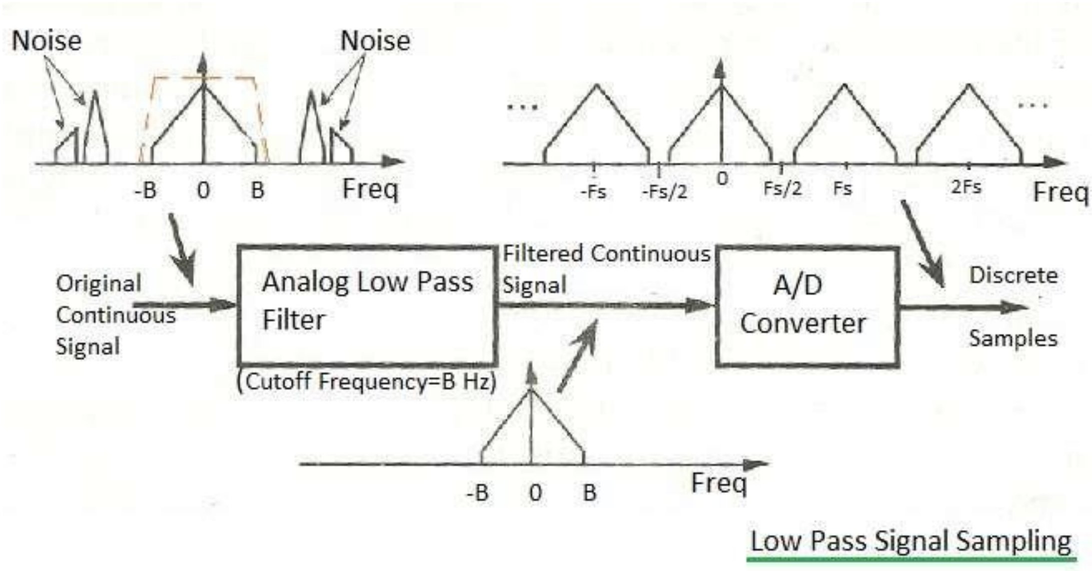


Figure depicts that **input spectrum to be sampled is symmetrical about zero Hz**. The spectral amplitude is zero above +B and below -B Hz. If the signal is sampled at Fs samples/second than we see spectral replications along with the original input spectrum.

Aliasing : Aliasing is **an effect of the sampling that causes different signals to become indistinguishable**. Due to aliasing, the signal reconstructed from samples may become different than the original continuous signal. This can drastically deteriorate the performance if proper care is not taken.

Signal Reconstruction : In signal processing, reconstruction usually means **the determination of an original continuous signal from a sequence of equally spaced samples**. This article takes a generalized abstract mathematical approach to signal sampling and reconstruction.

4. Compare the concept of Uniform and Non Uniform Quantisation with necessary illustrations.

   Ans:

   There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

   The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

5. Propose the ideas about Uniform Quantization and its types

   Ans:

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

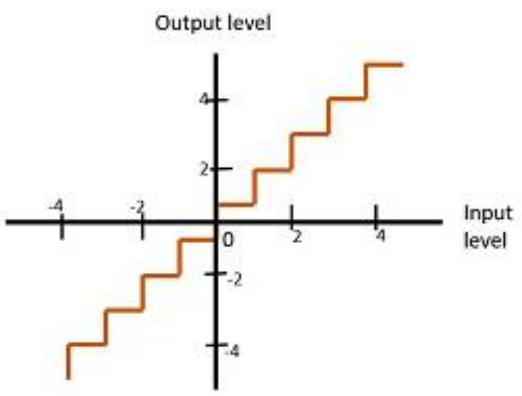


Fig 1 : Mid-Rise type Uniform Quantization

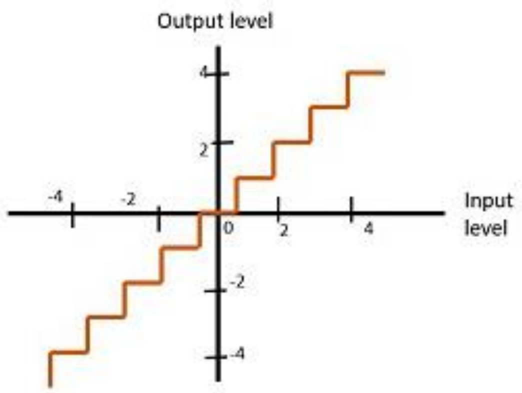


Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

6. Illustrate and describe the types of Quantizer. Describe the mid tread and midrise type characteristics of uniform quantizer with suitable diagram.

   Ans:

   There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

   The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

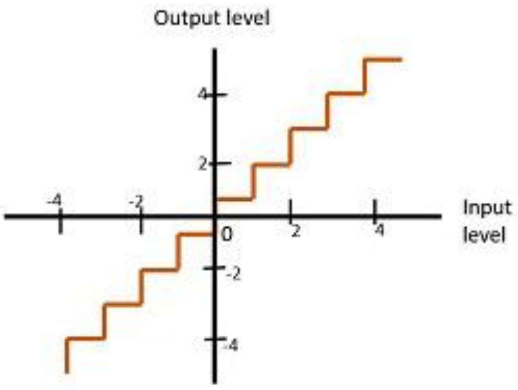


Fig 1 : Mid-Rise type Uniform Quantization

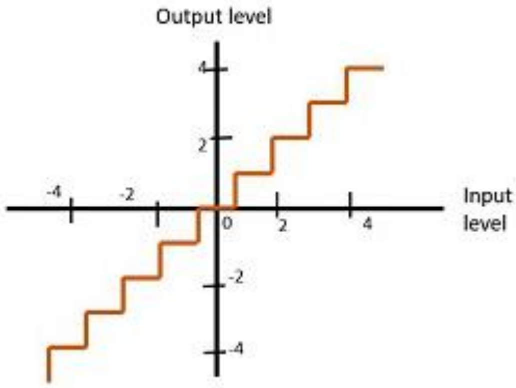


Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

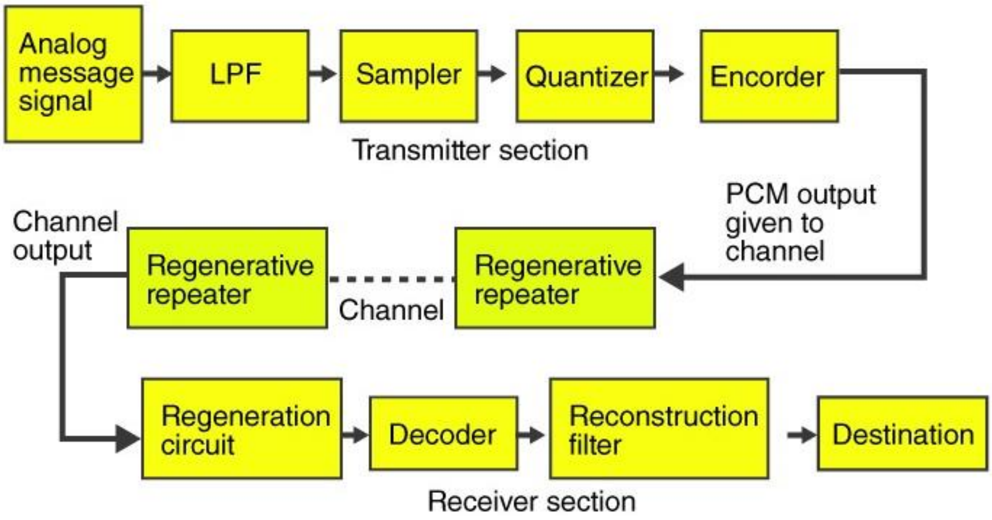
7. Distinguish various Pulse Modulation Techniques with necessary diagrams.

   Ans:

   Pulse Code Modulation (PCM)

   This type of modulation is different from all modulations learnt so far. It is clear from the block diagram given at the top that it is a type of digital modulation. That is the signals here are sampled and sent in pulse form. A common feature among other techniques is that pulse code modulation also uses sampling technique. In this case, instead of sending a pulse train which is capable of continuously varying parameters, this type of generator produces a series of numbers or digits. Each digit in it represents the appropriate length of the sample at a particular instant.

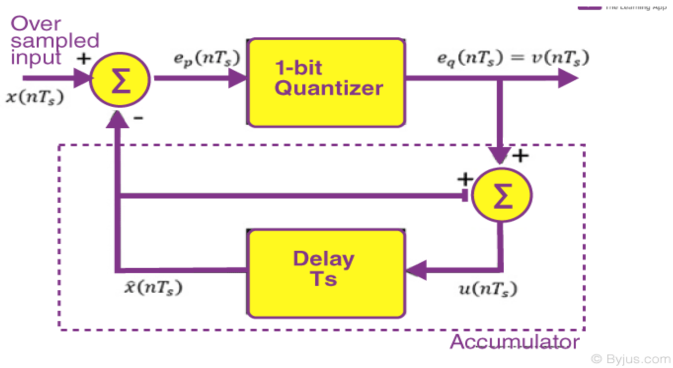
   Basic block diagram of its realisation is given below:



Delta Modulation

In this type of modulation, the sampling rate is very high. Here, stepsize after quantisation is of the smaller value. In this method, the quantisation design is very simple. Here the bit rate can be designed by the user.

Block diagram of a delta modulator is given below:



8. Determine the purpose of Non Uniform Quantization and mention the Laws for implementing the same

Ans:

Nonuniform quantization is achieved by first distorting the original signal with a logarithmic *compression* characteristic, and then using a uniform quantizer. For small magnitude signals the compression characteristic has a much steeper slope than large magnitude signals. Thus, a given signal change at small magnitudes will carry the uniform quantizer through more steps than the same change at large magnitudes. The compression characteristic effectively changes the distribution of the input signal magnitude. By compression, the low amplitudes are scaled up while the high amplitudes are scaled down. After compression, the distorted signal is used as input to the uniform quantizer. Thus, we achieve nonuniform quantization. There are two compression algorithms commonly used, the μ-law and the *A*-law [6, 10].

We use a device called *expander* at the receiver with characteristics complementary to the compression. It is used so that the overall transmission is not distorted. The whole process (compression and expansion) is called *companding* [7]. The μ - and *A*-law compression characteristics as given below are used.

- **μ-Law** compression characteristic used in North America is given as:

υ_{in} and υ_{out} are the normalized input and output voltages, respectively

μ = a positive constant

μ = 0 represents uniform quantization, μ = 255 is used in North America

- ***A*-Law** compression characteristic used in Europe and most of the rest of the world is given as:

where *A* is the positive constant. *A* = 87.6 is the standard value used in Europe. The case of uniform quantization corresponds to *A* = 1.

9. (i) Describe PCM system with neat block diagram?

Ans:

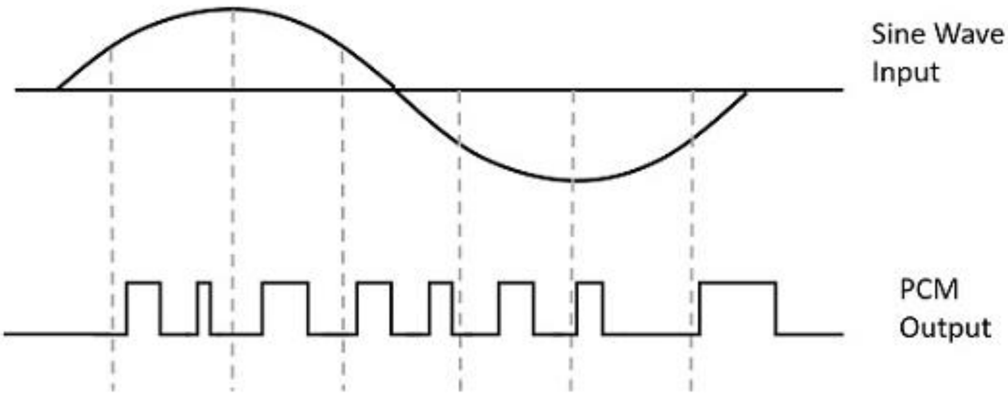
Pulse Code Modulation :

Modulation is the process of varying one or more parameters of a carrier signal in accordance with the instantaneous values of the message signal.

The message signal is the signal which is being transmitted for communication and the carrier signal is a high frequency signal which has no data, but is used for long distance transmission.

There are many modulation techniques, which are classified according to the type of modulation employed. Of them all, the digital modulation technique used is Pulse Code Modulation PCM.

A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., 1sand 0s. The output of a PCM will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.



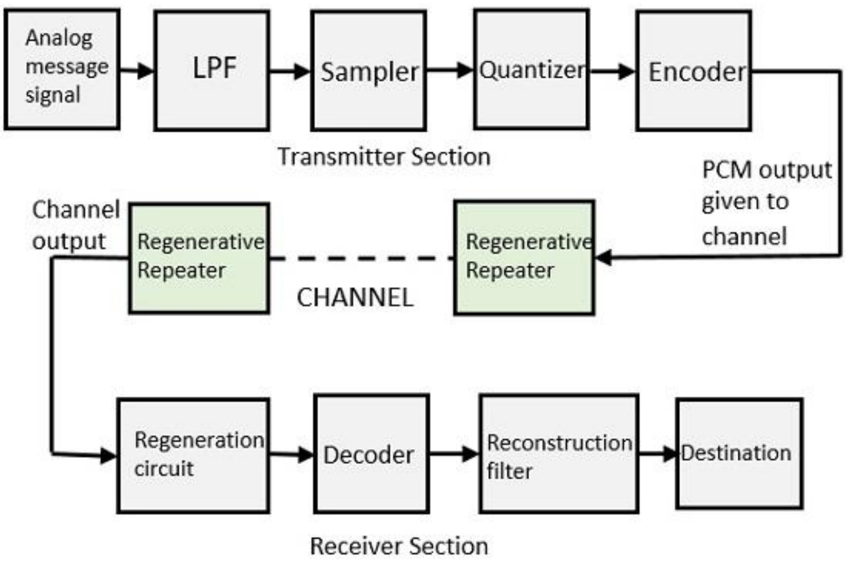
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of 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,andQuantizerLPF,Sampler,andQuantizer 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 Filter

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.

(ii) What is TDM and mention its applications. Explain the difference between analog TDM and digital TDM

Ans:

Time Division multiplexing (TDM)

TDM phone technology is based on electrical circuits that are physically switched on the public switched telephone network (PTSN). ... These systems have been in place for a long time – before IP server technology and the Internet was invented.

The main application of TDM(Time Division Multiplexing) is **to transmit or receive different data and signal using a common single transmission line**. TDM is used to transmit and receive a huge amount of independent data over a single line.