Engineering Notebook

VOLUME 1

Subject Title Analog Communication

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UNIT No 1

TOPIC: - AMPLTUDE MODULATION

Q1. Draw the block diagram of a general communication system and explain briefly the function of each stage

Answer:

In communication systems, we often need to design and analyse systems in which many independent message can be transmitted simultaneously through the same channel. It is possible with a technique called frequency multiplexing, in which each message is translated in frequency to occupy a different range of spectrum. This involves an auxiliary signal called carrier which determines the amount of frequency translation. It requires either the amplitude, frequency or phase of the carrier be instantaneously varied as according to the instantaneous value of the message signal. The resulting signal then is called a modulated signal. When the amplitude of the carrier is changed as according to the instantaneous value of the message/baseband signal, it results in Amplitude Modulation. The systems implanting such modulation are called as Amplitude modulation systems.

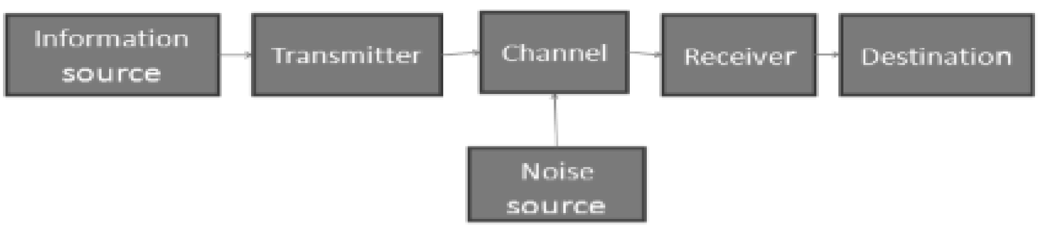


Figure 1: Block diagram of a communication system

The block diagram of a communication system will have five blocks, including the information source, transmitter, channel, receiver and destination blocks.

1. **Information source**:-

- The objective of any communication system is to convey information from one point to the other. The information comes from the information source, which originates it
- Information is a very generic word signifying at the abstract level anything intended for communication, which may include some thoughts, news, feeling, visual scene, and so on.
- The information source converts this information into physical quantity.
- The physical manifestation of the information is termed as message signal

2. **Transmitter**:-

- The objective of the transmitter block is to collect the incoming message signal and modify it in a suitable fashion (if needed), such that, it can be transmitted via the chosen channel to the receiving point.
- Channel is a physical medium which connects the transmitter block with the receiver block.
- The functionality of the transmitter block is mainly decided by the type or nature of the channel chosen for communication.

3. **Channel** :-

- Channel is the physical medium which connects the transmitter with that of the receiver.
- The physical medium includes copper wire, coaxial cable, fibre optic cable, wave guide and free space or atmosphere.
- The choice of a particular channel depends on the feasibility and also the purpose of the communication system.

4. **Receiver**:-

- The receiver block receives the incoming modified version of the message signal from the channel and processes it to recreate the original (non-electrical) form of the message signal.

- There are a great variety of receivers in communication systems, depending on the processing required to recreate the original message signal and also final presentation of the message to the destination.

5. **Destination**:-

- The destination is the final block in the communication system which receives the message signal and processes it to comprehend the information present in it.
- Usually, humans will be the destination block.

Q.2 Discuss the need of modulation and derive an expression for the sinusoidal carrier voltage which has been amplitude modulated by another sinusoidal modulating voltage.

Answer:

Baseband signals are incompatible for direct transmission over the medium so, modulation is used to convey (baseband) signals from one place to another

1) To reduce the size of antenna.

2) Efficient transmission

3) Reduced noise and interference.

4) Allows frequency translation:
   - Frequency Multiplexing
   - Avoids mixing of signals
   - Narrow banding

In amplitude modulation, the amplitude of a carrier signal is varied by the modulating voltage, whose frequency is invariably lower than that of the carrier. In practice, the carrier may be high frequency (HF) while the modulation is audio. Finally; AM is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating voltage.

Let the carrier voltage and the modulating voltage, *vc* and *vm*, respectively, be represented by

$$vm = Vm \sin \omega mt \qquad (1)$$

$$vc = Vc \sin \omega ct \qquad (2)$$

The phase angle has been ignored in both expressions since it is unchanged by the amplitude modulation process. Its inclusion here would merely complicate the proceedings, without affecting the result. From the definition of AM, you can see that the (maximum) amplitude Vc of the unmodulated carrier will have to be made proportional *to* the instantaneous modulating voltage $Vm \sin \omega mt$ when the carrier is amplitude modulated.

When a carrier is amplitude modulated, the proportionality constant is rnade equal to unity, and the instantaneous modulating voltage variations are superimposed onto the carrier amplitude. Thus when there is temporarily no modulation, the amplitude of the carrier is equal to its unmodulated value. When modulation is present, the amplitude of the carrier is varied by its instantaneous value. The situation is illustrated in , which shows how the maximum amplitude of the amplitude modulated voltage is rnade to vary with changes in the modulating voltage. Figure also shows that something unusual (distortion) will occur if *Vm* is greater than *Vc*. The fact that the ratio $\frac{Vm}{Vc}$ often occurs; leads to the definition of the *modulation index* given by

$$m=\frac{Vm}{Vc} \qquad (3)$$

$$A=Vc + Vm$$

$$=Vc+Vm \sin \omega mt=Vc +mVc \sin \omega mt$$

$$A=Vc(1+m \sin \omega mt) \qquad (4)$$

The instantaneous voltage of the resulting amplitude modulated wave is

$$V_{AM} =A\sin \theta=A\sin \omega ct= Vc\ (1+m \sin \omega mt) \sin \omega ct \qquad (5)$$

Q. 3 A broadcast transmitter radiates 4.72KW when the modulation percentage is 60.Calculate the total power when the modulation has been reduced to 40 percent.

Answer:

Given, P_t=4.72

m_a=0.6

$$Pt = Pc(1+\frac{ma^2}{2})$$
$$Pc = \frac{Pt}{1.18}$$
$$Pc = \frac{4.72}{1.18}$$
$$Pc = 4kw$$

Pt when %m_a=40

m_a=0.4

$$Pt = Pc(1+\frac{ma^2}{2})$$
$$Pt = 4(1+\frac{0.4^2}{2})$$
$$Pt = 4.32kw$$

Q.4 An audio frequency signal 10sin(2*3.14*500t) is used to amplitude modulate a carrier of 50 sin(5*3.14*10^5t). Calculate (1) Modulation index (2) Side band frequencies (3) BW required (4) Total power delivered to the load of 600Ω.

Answer:

Now comparing equation to genral form we get

$$V_C = 50$$
$$V_M = 10$$
$$\omega_C = 5\pi \times 10^5$$
$$\omega_M = 2\pi \times 500$$ Now,

$$Ma = \frac{V_M}{V_C} = \frac{10}{50} = 0.2$$
$$Fc = \frac{\omega_C}{2\pi} = \frac{5\pi \times 10^5}{2\pi} 250khz$$
$$Fm = \frac{\omega_m}{2\pi} = \frac{2\pi \times 500}{2\pi} = 500khz$$

$$F_{USB} = F_C + F_m = 250khz + 500hz = 250.5khz$$
$$F_{LSB} + F_C + F_M = 250khz - 500hz = 249.5khz$$

$$Bw_{req} = 2F_m = 2 \times 500 = 1000hz$$

Carrier power= $Pc = \frac{v^2}{R} = \frac{50^2}{600} = 4.1667w$

$$P_{AM} = Pc(1+\frac{ma^2}{2})$$
$$= 4.16667 \times (1+\frac{0.2^2}{2})$$
$$= 4.25w$$

Q.5 Calculate the percentage power saving when the carrier and one of the sidebands are suppressed in an AM wave modulated to a depth of i. 100 % ii. 50 %.

Answer:

ma=1

$$P_{SSB} = \frac{ma^2}{4} \times Pc = 0.25Pc$$
$$P_{AM} = Pc(1+\frac{ma^2}{2}) = 1.5Pc$$

% Power saved

$$\frac{P_{Am} - P_{SSB}}{P_{Am}} \times 100$$
$$= \frac{1.5 - 0.25}{1.5} \times 100$$
$$= 83.33\%$$

Now,

m_a=0.5

$$P_{SSB} = \frac{ma^2}{4} \times Pc$$

$$= \frac{1}{16} \times Pc$$

$$P_{AM} = Pc(1 + \frac{ma^2}{4})$$

$$= \frac{9}{8} \times Pc$$

% Power Saving

Now,

$$\frac{P_{AM} - P_{SSB}}{P_{AM}} \times 100$$

$$\frac{\frac{9}{8} - \frac{1}{16}}{\frac{9}{8}} \times 100$$

$$= 94.44\%$$

Q.6 A 400 W carrier is amplitude modulated to a depth of 100%.Calculate the total power in case of AM wave and DSBSC techniques. How much power saving (in W) is achieved for DSBSC? If the depth of modulation is changed in 75% then how much power (inW) is required for transmitting the DSBSC wave? Compare the powers required for DSBSC in both the cases and comment on the reason for change in the power levels.

Answer:

Pc= 400

m=1

Total power in SSB

$$P_{SSB} = Pc(\frac{m^2}{4}) =$$
$$= 100w$$

Power saving compared to AM

$$P_{AM} - P_{SSB} = 500w$$

Now,

Pc=400

m=0.75

$$P_{SSB} = Pc(\frac{m^2}{4})$$
$$= 400 \times (\frac{0.75^2}{4})$$
$$= 56.25w$$

For m=1, total power required is more and less for m<1

Q. 7 A certain transmitter radiates 9 kW with the carrier unmodulated and 10.12 kW when the carrier is sinusoidally modulated. Calculate modulation index. If another sine wave is simultaneously transmitted with modulation index 0.4, determine the total radiated power.
Answer:

$$\frac{m2}{2} = \frac{Pt}{Pc} - 1 = \frac{10.125}{9} - 1 = 1.125 - 1 = 0.125$$

m2=0.125*2=0.250

m=0.50

$$m_t = \sqrt{m1^2 + m2^2} = \sqrt{0.5^2 + 0.4^2} = 0.64$$

$$Pt = Pc\ (1+ \frac{mt2}{2})$$

Pt =10.84kW

Q.8 The antenna current of an AM broadcast transmitter, modulated to a depth of 40% by an audio sine wave is 11 A. It increases to 12 A as a result of simultaneous modulation by another audio sine wave. What is the modulation index due to this second wave?

Answer:

$$Ic = \frac{It}{\sqrt{1+\frac{m^2}{2}}} = \frac{11}{\sqrt{1+\frac{0.4^2}{2}}} = 10.58A$$

$$mt = \sqrt{2[(\frac{It}{Ic})^2 - 1]}$$

$$= \sqrt{2 \times 0.286} \quad = 0.757$$

$$m2 = \sqrt{mt^2 + m1^2} = \sqrt{0.757^2 + 0.4^2}$$

$$= 0.643$$

Q.9 Draw & explain the block diagram of an SSB transmitter using the filter system.

Answer: The filter method of SSB generation produces double sideband suppressed carrier signals (using a balanced modulator), one of which is then filtered to leave USB or LSB. It uses two filters that have different pass band centre frequencies for USB and LSB respectively. The resultant SSB signal is then mixed (heterodyned) to shift its frequency higher.
Limitations:

1) This method can be used with practical filters only if the baseband signal is restricted at its lower edge due to which the upper and lower sidebands do not overlap with each other. Hence it is used for speech signal communication where lowest spectral component is 70 Hz and it may be taken as 300 Hz without affecting the intelligibility of the speech signal.

2) The design of band-pass filter becomes quite difficult if the carrier frequency is quite higher than the bandwidth of the baseband signal.

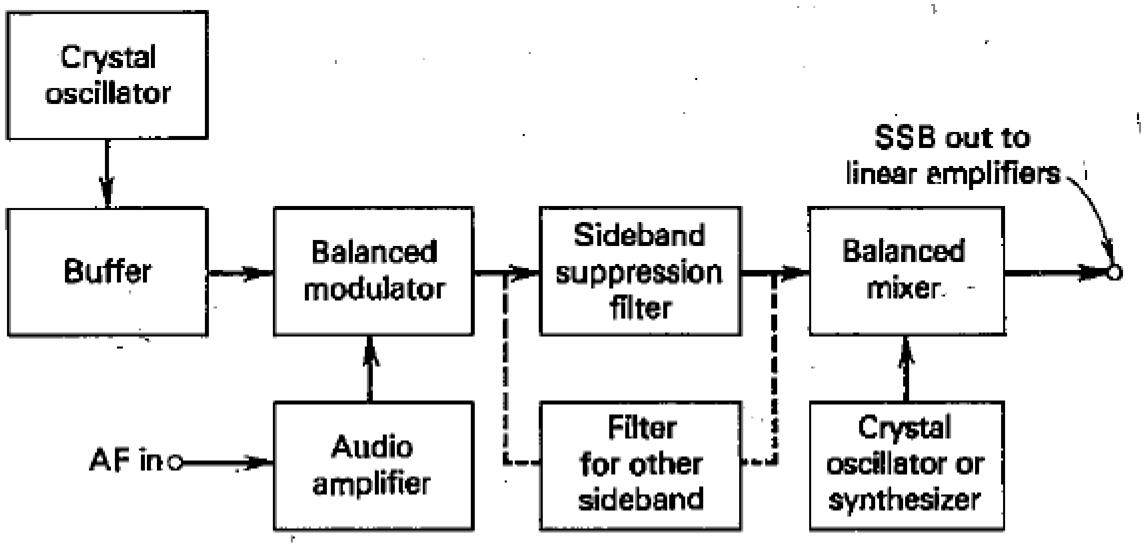


Figure 2:-Filter Method of SSB Transmission

Q.10 Give the principle of square law modulator. Draw the basic circuit of square law diode modulator.

Answer:
Square law diode modulation makes use of non-linear current-voltage characteristics of diode. This method is suited for low voltage levels as the current-voltage characteristic of diode is highly nonlinear in the low voltage region. So the diode is biased to operate in this non-linear region for this application. A DC battery *Vc* is connected across the diode to get such a operating point on the characteristic. When the carrier and modulating signal are applied at the input of diode, different frequency terms appear at the output of the diode. These when applied across a tuned circuit tuned to carrier frequency and a narrow bandwidth just to allow the two pass-bands, the output has the carrier and the sidebands only which is essentially the DSBSC AM signal.

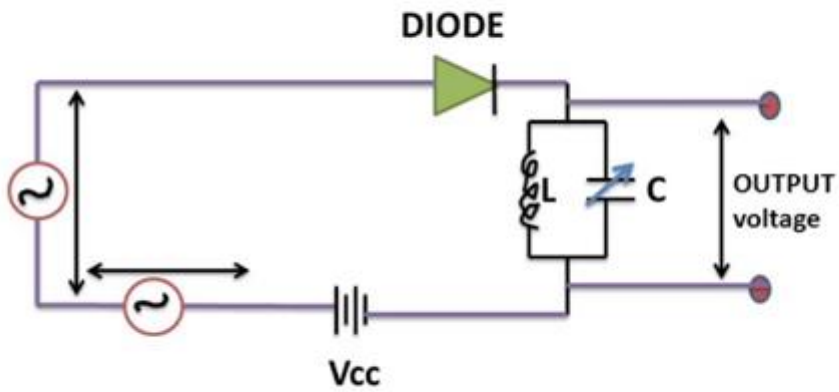


Figure 3:-Square Law Diode Modulator

UNIT No 2

TOPIC:-ANGLE MODULATION

Q.1 What is frequency modulation? Also draw necessary waveforms.

Answer:

A major problem in AM is its susceptibility to noise superimposed on the modulated carrier signal. To improve on this, the first frequency modulation (FM) radio communication system was developed in 1936, which is much more immune to noise than its AM counterpart. Unlike the AM, FM is difficult to treat mathematically due to the complexity of the sideband behavior resulting from the modulation process.

To generate a frequency modulated signal, the frequency of the radio carrier is changed in line with the amplitude of the incoming audio signal. The modulating signal changes the freq. fc of the carrier signal. While changing the amplitude of a radio signal is the most obvious method to modulate it, it is by no means the only way. It is also possible to change the frequency of a signal to give frequency modulation or FM. Frequency modulation is widely used on frequencies above 30 MHz, and it is particularly well known for its use for VHF FM broadcasting.

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

The frequency modulation index is the equivalent of the modulation index for AM , but obviously related to FM. In view of the differences between the two forms of modulation, the FM modulation index is measured in a different way. The FM modulation index is equal to the ratio of the frequency deviation to the modulating frequency.

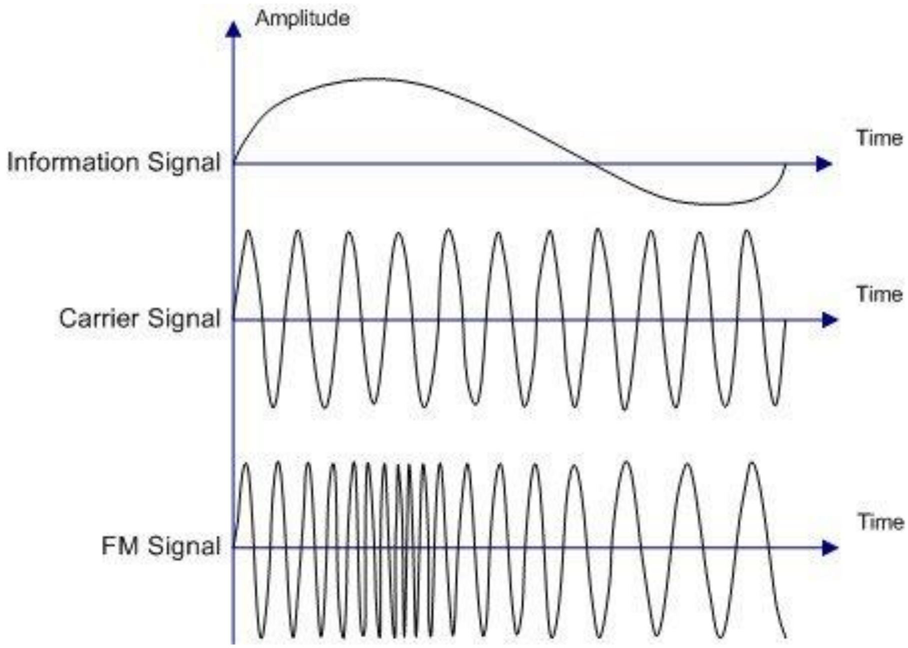


Figure 1:-FM Waveform

Q.2 Explain the frequency spectrum in FM.

Answer:

The FM wave equation is

$$V_{FM} = Vc \sin(\omega ct + mf \sin \omega mt) \qquad (1)$$

When a comparable stage was reached with AM theory it was possible to tell at a glance what frequencies were present in the modulated wave. Unfortunately, the situation is far more complex, mathematically speaking, for FM. Since Equation (1) is the sine of a sine, the only solution involves the use of *Besse/ functions.* Using these, it may then be shown that Equation(1) may be expanded to yield

$$v = A\{J_o(mf)\sin \omega ct$$
$$+ J_1(mf)\,[\sin(\omega c + \omega m)t - \sin(\omega c - \omega m)t]$$
$$+ J_2(mf)\,[\sin(\omega c + 2\omega m)t + \sin(\omega c - 2\omega m)t]$$
$$+ J_3(mf)\,[\sin(\omega c + 3\omega m)t - \sin(\omega c - 3\omega m)t]$$
$$+ J_4(mf)\,[\sin(\omega c + 4\,\omega m)t + \sin(\omega c - 4\,\omega m)t] \cdot \cdot \cdot\}$$

It can be shown that the output consists of a carrier and an apparently infinite number of pairs of sidebands. each preceded by *J* coefficients. These are Bessel functions.

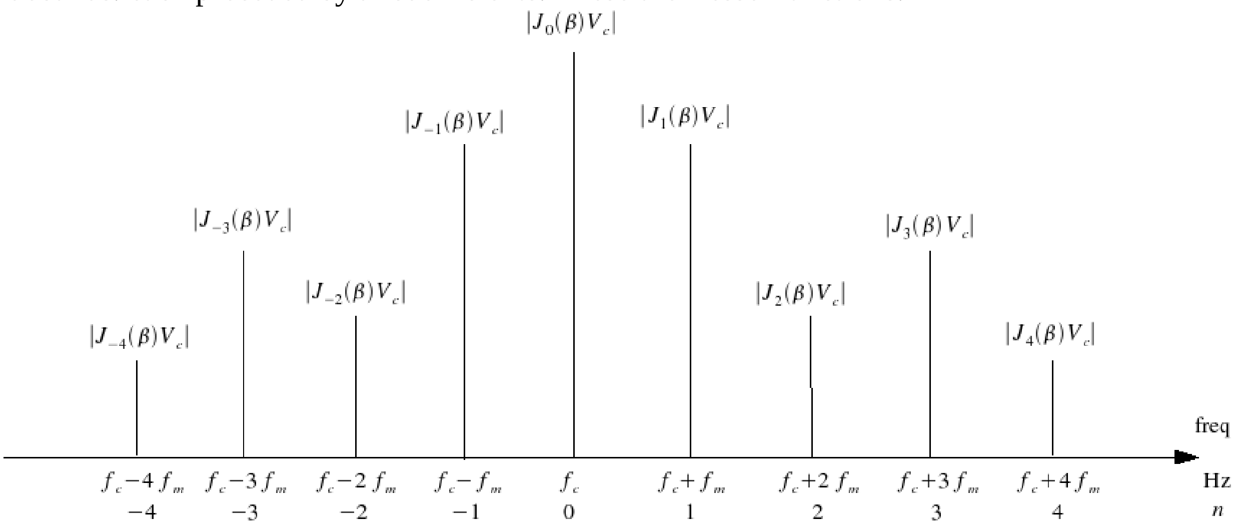


Figure 2:-Frequency Spectrum of FM

The amplitudes drawn are completely arbitrary, since we have not found any value for *Jn*(*b*) – this sketch is only to illustrate the spectrum.

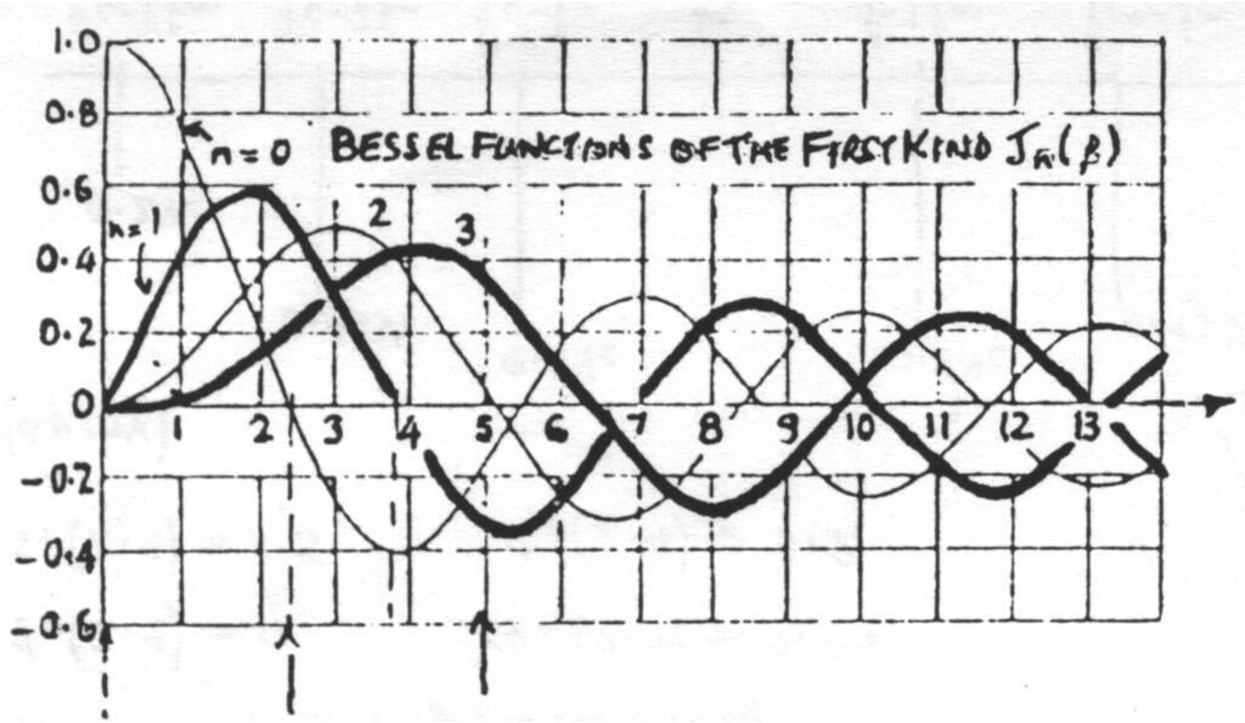


Figure 3:-Bessels Functions

Unlike AM, where there are only three frequencies (the carrier and the first two sidebands), FM has an infinite number of sidebands, as well as the carrier. They are separated from the carrier by fm, 2fm, 3fm, .., and thus have a recurrence frequency of fm. The J coefficients eventually decrease in value as n increases, but not in any simple manner. As seen in Figure 3, the value fluctuates on either side of zero, gradually diminishing. Since each J coefficient represents the amplitude of a particular pair of sidebands, these also eventually decrease, but only past a certain value of n. The modulation index determines how many sideband components have significant amplitudes.

Q.3 Differentiate between narrow band and wide band FM.

Answer:

Wideband FM: Wideband FM is typical used for signals where the FM modulation index is above about 0.5. For these signals the sidebands beyond the first two terms are not insignificant. Broadcast FM stations use wide-band FM which enables them to transmit high quality audio, as well as other facilities like stereo, and other facilities like RDS, etc. This frequency modulation has infinite bandwidth. The modulation index β is large, i.e., higher than 1. Its spectrum consists of a carrier and infinite number of sidebands, which are located around it. This is used in entertainment, broadcasting applications such as FM radio, TV, etc.

The wide bandwidth of wide band FM is enables high quality broadcast transmissions to be made, combining a wide frequency response with low noise levels. Once the signal is sufficiently strong, the audio signal to noise ratio is very good. Sometimes high fidelity FM tuners may use a wide-band filter for strong signals to ensure the optimum fidelity and performance. Here the quieting effect of the strong signal will allow for wide-band reception and the full audio bandwidth. For lower strength signals they may switch to a narrower filter to reduce the noise level, although this will result in the audio bandwidth being reduced. However on balance the narrower bandwidth will give a more pleasing sound when the received signal is low.

Narrowband FM: Narrow band FM, NBFM, is used for signals where the deviation is small enough that the terms in the Bessel function are small and the main sidebands are those appearing at ± modulation frequency. The sidebands further out are negligible. For NBFM, the FM modulation index must be less than 0.5, although a figure of 0.2 is often used. For NBFM the audio or data bandwidth is small, but this is acceptable for this type of communication. This frequency modulation has a small bandwidth when compared to wideband FM. The modulation index β is small, i.e., less than 1.Its spectrum consists of the carrier, the upper sideband and the lower sideband. This is used in mobile communications such as police wireless, ambulances, taxicabs, etc. Narrowband FM is widely used for two way radio communications. Although digital technologies are taking over, NBFM is still widely used and very effective. Many two way radios or walkie talkies use NBFM, especially those which conform to the licence-free standards like PMR446 and FRS radio communications systems.

Q.4 Find the carrier and modulating frequencies, the modulation index, and the maximum deviation of the FM wave represented by the voltage equation V = 12 sin (6 x 10^8t + 5 sin 1250 t). What power will this FM wave dissipate in a 10Ωresistor?

Answer:

$$fc = \frac{6 \times 10^8}{2\pi} = 95.5MHz$$

$$fm = \frac{1250}{2\pi} = 199Hz$$

mf=5

δ = mf* fm=5×199=995 Hz

P=$\frac{Vrms^2}{R}$=$\frac{72}{10}$ =7.2W

Q.5 A 25 MHz carrier is modulated by a 400Hz audio sine wave. If the carrier voltage is 4V and the maximum deviation is 10 KHz, write the equation of this modulated wave for a) FM and b) PM. If the modulating frequency is now changed to 2KHz, all else remaining constant ,write a new equation for c)FM and d)PM

Answer:

$$\omega c = 2\pi \times 25MHz = 1.57 \times 10^8$$

$$\omega c = 2\pi \times 400 = 2513 \times 10^8$$

$$m=mf=mp=\frac{\delta}{fm}=\frac{10000}{400}=25$$

(a) $v=4sin(1.57\times10^8t+25sin2513t)$ (FM)
(b) $v=4sin(1.57\times10^8t+5sin\,12{,}565t)$ (PM)

$$m=mf=\frac{\delta}{fm}=\frac{10000}{2000}=5$$

(c) $v=4sin(1.57\times10^8t+25sin\,12{,}565t)$ (FM)

(d) $v=4sin(1.57\times10^8t+25sin2513t)$ (PM)

Q.6 In an FM system When the audio frequency is *500* Hz and the **AF** Voltage is 2.4 V, the deviation is 4.8 kHz. If the AF voltage is now increased to 7.2 V. What is the new deviation? If the AF voltage is further raised to 10 V while the AF is dropped to 200 Hz, What is deviation? Find the modulation index in each case.

Answer:

As $\delta \propto$Vm, we may write

$$\frac{\delta}{V}=\frac{4.8}{2.4}=2$$

When Vm=7.2V

δ=2* 7.2 = 14.4 kHz

When Vm=10v

δ=2* 10 = 20kHz

$$mf1 = \frac{\delta 1}{fm1} = \frac{4.8}{0.5} = 9.6$$
$$mf2 = \frac{\delta 2}{fm1} = \frac{14.4}{0.5} = 28.8$$

$$mf3 = \frac{\delta 3}{fm2} = \frac{20}{0.2} = 100$$

Q7. Explain FM signal generation using reactance modulator method.

Answer:

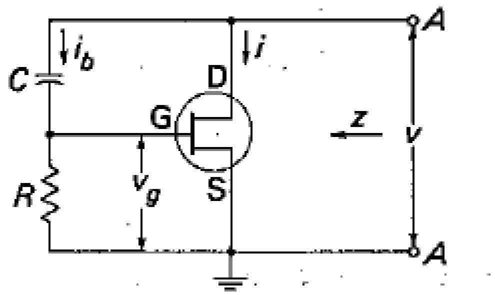


Figure 4:- Basic reactance Modulator

In order to determine z; a voltage vis applied to the terminals *A-A* between which the impedance is to be measured, and the resulting current i is calculated. The applied voltage is then divided by this current, giving the impedance seen when looking into the terminals. In order for this impedance to be a pure reactance (it is capacitive here), two requirements must be fulfilled. The first is that the bias network current *ib* musi be negligible compared to the drain current. The impedance of the bias network must be large enough to be ignored. The second requirement is that the drain-to-gate impedance (Xc here) mus(be greater than the gate to- source impedance *(R* in this case), preferably by more than 5: 1.

$$Vg=ib *R=\frac{Rv}{R-jXc} \qquad (1)$$

The FET drain current

$$i=gm*vg=\frac{gm*Rv}{R-jXc} \qquad (2)$$

Therefore, the impedance seen at the terminals A-A is

$$z=\frac{v}{i} = \frac{R-jXc}{gm*R} \qquad (3)$$

If Xc>>R

$$z=-j\frac{Xc}{gm*R}$$

$$Xeq=\frac{Xc}{gm*R} = \frac{1}{2*pi*f*Ceq} \qquad (4)$$

From above Equation it is seen that under such conditions the input impedance of the device at *A-A* is a pure reactance and is given by

$$Ceq= gm*R*C \quad (5)$$

The gate-to-drain impedance is, in practice, made five to ten times the gate-to source impedance. Let $Xc = nR$ (at the carrier frequency) in the capacitive *RC* reactance FET so far discussed. Then

$$Xc = \frac{1}{wc} = n*R$$

$$C = \frac{1}{\omega * n * R}$$

$$Ceq = gm * R * C = \frac{gm * R}{2 * \pi * f * n * R} \qquad (7)$$

Q.8 What is pre-emphasis? Why is it used? Sketch a typical pre-emphasis circuit and explain why de-emphasis must be used also.

Answer:

Pre-emphasis: The noise suppression ability of FM decreases with the increase in the frequencies. Thus increasing the relative strength or amplitude of the high frequency components of the message signal before modulation is termed as Pre-emphasis. The Fig3 below shows the circuit of pre-emphasis.

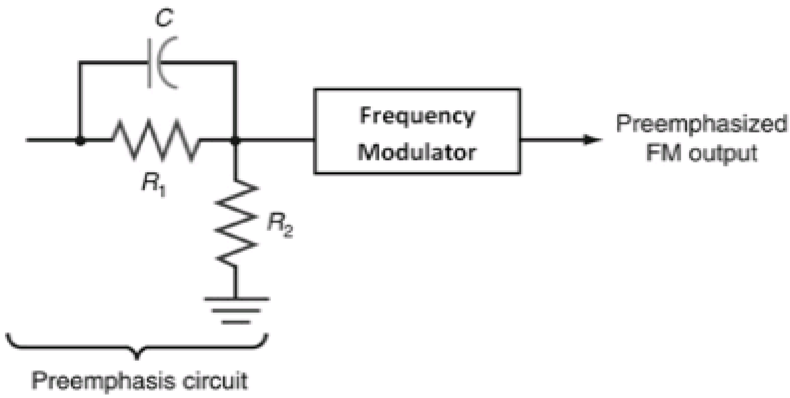


Figure 5:-Pre-emphasis circuit

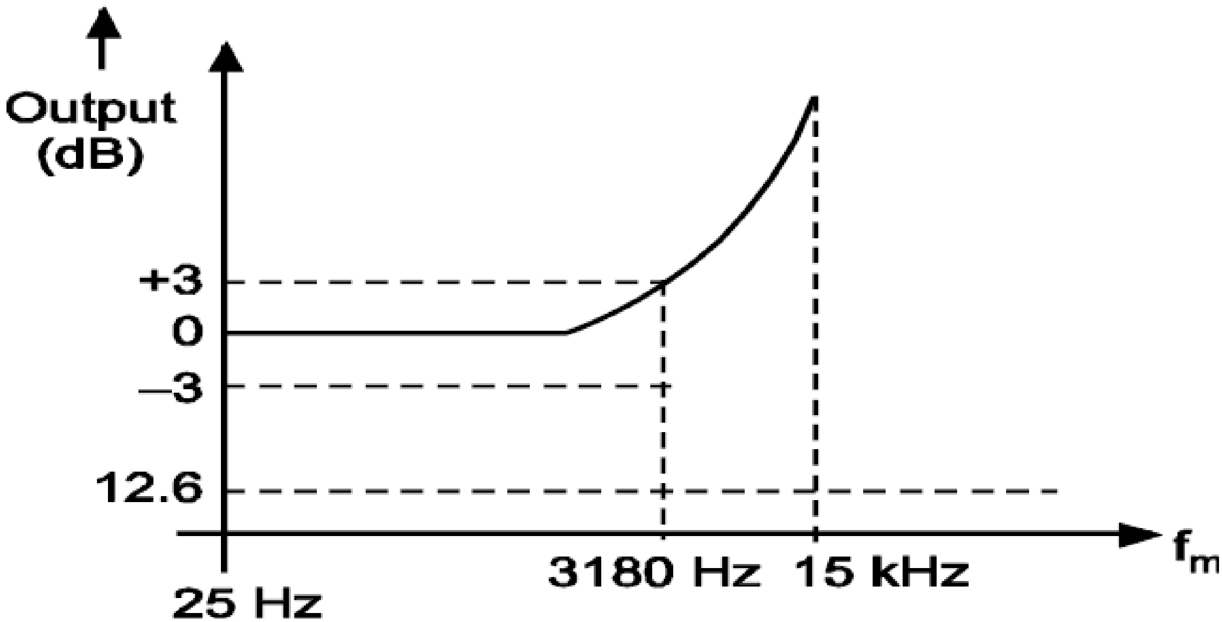


Figure 6:-Pre-emphasis Curve

De-emphasis: In the de-emphasis circuit, by reducing the amplitude level of the received high frequency signal by the same amount as the increase in pre-emphasis is termed as De-emphasis. The Figure below shows the circuit of de-emphasis.

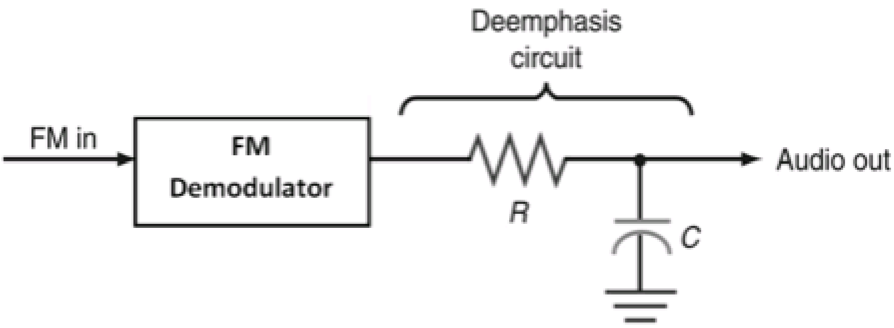


Figure 7:-De-emphasis Circuit

- The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side.
- Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation. To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed. Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant.
- The de-emphasis process ensures that the high frequencies are returned to their original relative level before amplification.
- Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator which allows only low frequencies to pass.

Q.9 Differentiate between AM and FM.

Answer:

Comparison Chart

Parameter	**Amplitude Modulation**	**Frequency Modulation**
Abbreviated as	AM	FM
Operation	Amplitude of carrier is changed according to the message signal.	Frequency of carrier is modified according to the message signal.
Constant factor	Frequency and phase.	Amplitude and phase.
Noise susceptibility	High	Low
Frequency range	Medium and high frequency range (nearly about 535 to 1705 KHz)	Very high and ultra high frequency range (nearly about 88 to 108 MHz)
Cost	Less	Comparatively more costly
Circuit complexity	The circuit of amplitude modulation is complex.	The complexity of a circuit used for frequency modulation is less as compared to amplitude modulation circuitry.
Bandwidth requirement	The bandwidth requirement is nearly about 30 KHz.	The bandwidth requirement in frequency modulation is 80 KHz.

Q.10 Draw the circuit & explain the working of frequency modulator using a varactor diode.

Answer:

A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied voltage when the diode is reverse biased. It may also be used to produce frequency modulation. Varactor-diodes are certainly employed frequently, together with a reactance modulator, to provide automatic frequency correction for an FM transmitter. The circuit of Figure shows such a modulator. It is seen that the diode has been back-biased to provide the junction capacitance effect, and since this bias is varied by the modulating voltage which is in series with it, the junction capacitance will also vary, causing the oscillator frequency to change accordingly. Although this is the simplest reactance modulator circuit, it does have the disadvantage of using a two-terminal device; its applications are somewhat limited. However, it is often used for automatic frequency control and remote tuning.

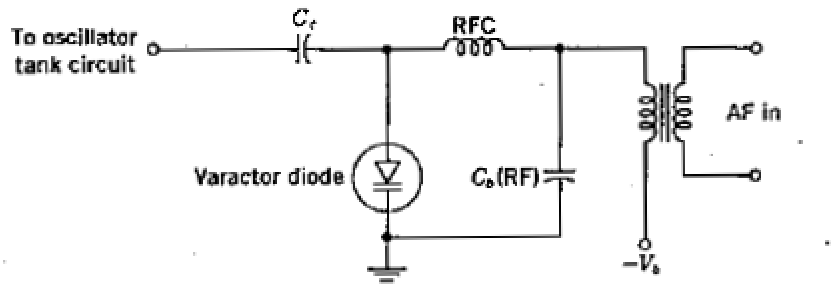


Figure 8:- Varacter diode Modulator

UNIT-3

TOPIC:-RADIO RECEIVERS

Q.1 Explain the working of TRF receiver. List out the advantages and disadvantages of TRF receiver

Answer:

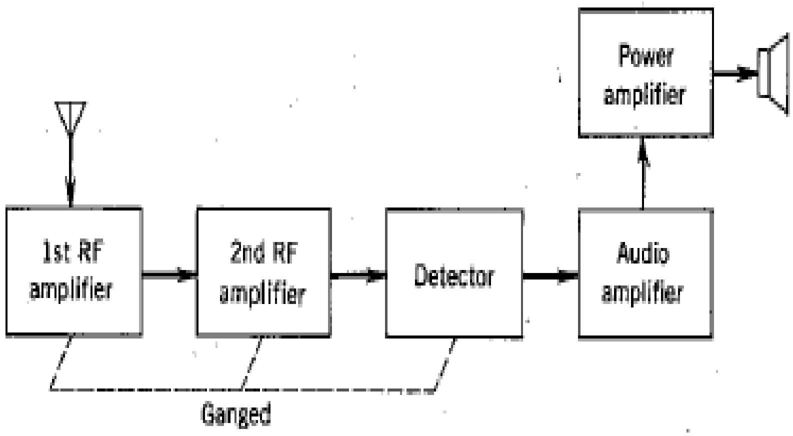


Figure 1:- Block diagram of a TRF Receiver

The TRF receiver is a simple "logical" receiver. A person with just a little knowledge of communications would probably expect .all radio receivers to have this form. The virtues of this type, which is now not used except as a fixed-frequency receiver in special applications, are its simplicity and high sensitivity. 'It must also be mentioned that when the TRF receiver was first introduced, it was a great improvement on the types used previously-mainly crystal, regenerative and super regenerative receivers.

Two or perhaps three RF amplifiers, all tuning together, were employed to select and amplify the incoming frequency and simultaneously to reject all others. After the signal was amplified to a suitable level, it was demodulated (detected) and fed to the loudspeaker after being passed through the appropriate audio amplifying stages. Such receivers were simple to design and align at broadcast frequencies (535 to 1640 kHz), but they presented difficulties at higher frequencies. This was mainly because of the instability associated with high gain being achieved at one frequency by a multistage amplifier. In addition the TRF receiver suffered from a variation in bandwidth over the tuning range. It was unable to achieve sufficient selectivity at high frequencies, partly as a result of the enforced use of single-tuned circuits. It was not possible to use double-tuned RF amplifiers in this receiver, although it was realized that they would naturally yield better selectivity.

Even though TRF receiver is simple in operation, but has some problems:

(i)Instability: Due to oscillatory nature of RF amplifiers.

(ii)Variation in bandwidth over tuning range: Variation due to variation in Quality factor ‘Q’. If BW of receiver increases, it will pick-up the adjacent channel along with the desired one.

(iii)Insufficient selectivity: Insufficient adjacent frequency rejection. Due to increased bandwidth at higher frequencies, the ability of the TRF receiver to select the desired signal and reject all other signals is affected.

Q.2 Draw a block diagram of a super heterodyne receiver and explain the function of each stage.

Answer: The problems in TRF radio receiver are solved in super heterodyne receiver.

Super heterodyne Principle:-In super heterodyne receivers the incoming signal is mixed with output of local oscillator and converted into a signal with lower fixed frequency called intermediate frequency (IF). This is known as super heterodyne principle**.** Super heterodyne means mixing of two frequencies.

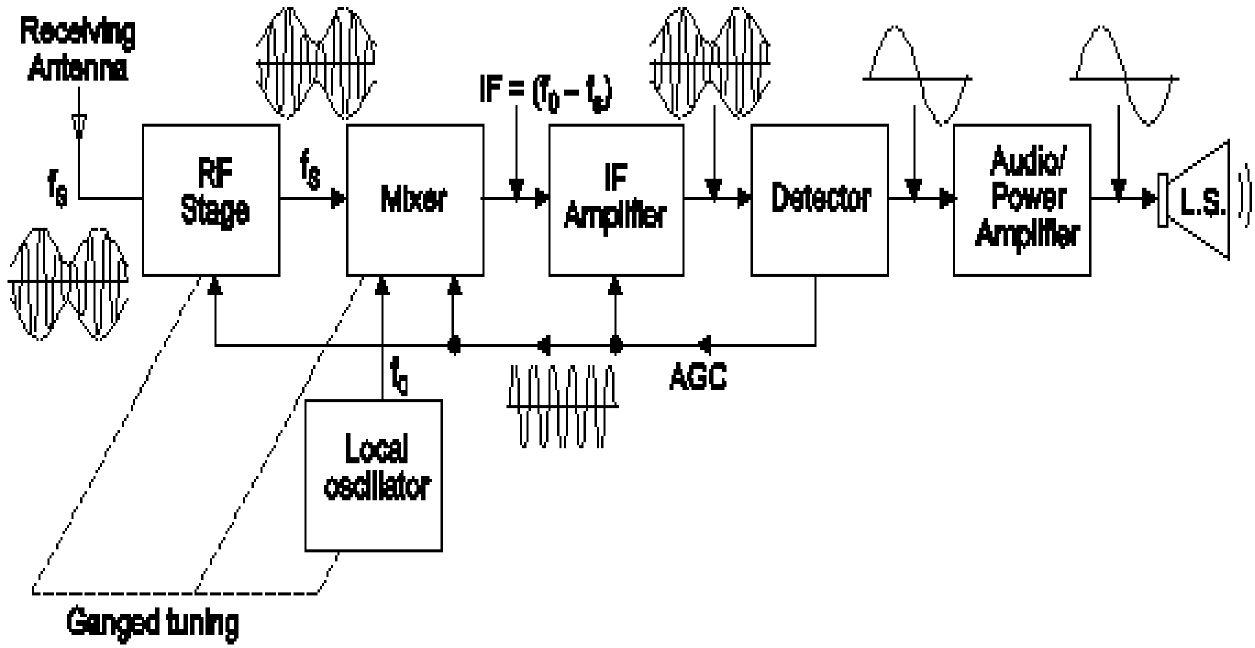


Figure 2:- Block diagram of a Super Heterodyne Receiver

RF Stage:-

- A radio receiver always has RF section, which is tunable circuit connected to the antenna terminals.
- It selects the wanted frequency and reject unwanted frequencies.
- Such a receiver need not have an RF amplifier.
- In the domestic radio receivers RF amplifier is not used for economic reason, however RF amplifier improves quality of receiver output.

Advantages of RF Amplifier (Characteristics)

- Greater gain i.e. better sensitivity.

- Improved image frequency rejection.
- Improved signal to noise ratio.
- Better selectivity
- Improves quality of receiver output.
- Better coupling of receiver to antenna.

Mixer:-

- The mixer or frequency changer is nothing but a non-linear resistance.
- It has two inputs at frequencies f_s and f_o.
- The mixing process is multiplication of incoming signal (f_s) and output of local oscillator signal (f_o). The product gives rise to two signals with f_s+f_o and $f_o - f_s$. The difference frequency produced ($f_o - f_s$) is taken as IF.

Local Oscillator:-

- In the receivers operating upto the limit of shortwave broadcasting, that is 36 MHz.
- Most commonly Colpits oscillators are used for higher operating frequencies.
- Where the frequency stability of the local oscillator must be high, AFC may be used.

IF Amplifier:-

- The IF amplifier is a fixed frequency amplifier, with very important function of rejecting adjacent unwanted frequencies.i.e. it decides sensitivity and selectivity of receiver.
- It provides maximum gain and selectivity in the receiver. AM receiver has IF value of 455 kHz.
- The two-stage IF amplifier as shown in Fig. is used to get a higher gain.
- All IF transformers (IFT) are single tuned.

- Note that neutralization may be used (capacitor C_n) in the transistor, IF amplifier depending on the frequency and type of transistor used.

Detector:-

- The process in which modulated signal is converted back into original modulating signal is called demodulation.
- Demodulation of AM signal is done by diode detector circuit.

Audio Amplifier:-

- It is low frequency amplifier which provides amplification at (20-20K)Hz.

Loud Speaker:-

- It converts electrical signal into sound or audio signal.

Q.3 Explain Receiver characteristics in details.

Answer:

Sensitivity, noise, selectivity, and fidelity are important receiver characteristics. These characteristics will be useful to you when performing receiver tests. They can help you to determine whether a receiver is working or not or in comparing one receiver to another.

Sensitivity:-

- The ability to amplify the weak signals is called sensitivity. It is the function of the overall receiver gain.
- Sensitivity of radio receiver is decided by the gain of the RF and IF amplifiers.

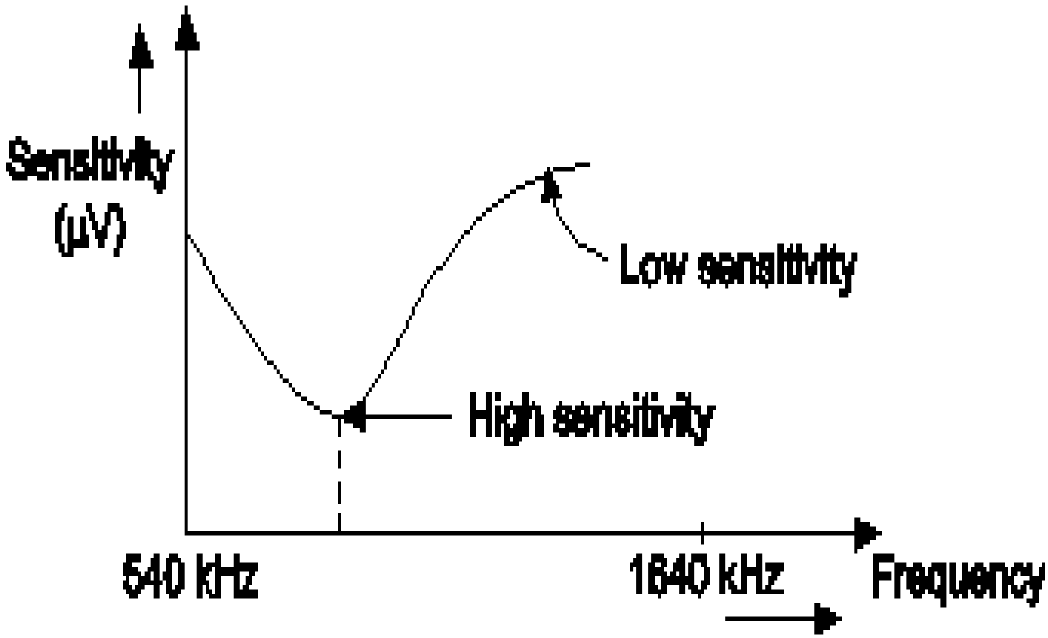


Figure 3:-Sensitivity Curve

- Practically, it is defined as the carrier voltage, which must be applied to the receiver input terminals to get standard output power at output terminals.
- The loudspeaker is replaced by load resistance of equal value of speaker.
- The sensitivity is expressed in m volt or millivolt.
- It may be measured at various frequencies in the radioband.
- Improvement in Sensitivity: The high gain IF amplifiers provides better sensitivity. Hence, smaller input signal is required to produce desired level of output.

Noise:-

All receivers generate a certain amount of noise, which you must take into account when measuring sensitivity. Receiver noise may originate from the atmosphere (lightning) or from internal components (transistors, tubes). Noise is the limiting factor of sensitivity. You will find sensitivity is the value of input carrier voltage (in microvolts) that must be applied from the signal generator to the receiver input to develop a specified output power.

Selectivity:-

- It is the ability of radio receiver to reject the unwanted signals.
- Selectivity depends on IF amplifier. Higher the 'Q' of the tuned circuit better is the selectivity.
- It is used to distinguish between two adjacent carrier frequencies.
- It shows how perfectly the receiver is able to select the desired carrier frequency and reject other frequencies

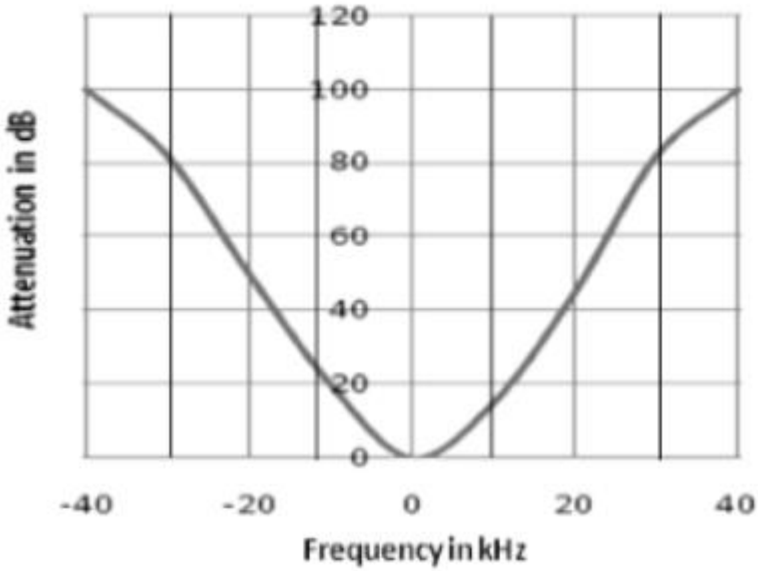


Figure 4:-Selectivity Curve

Fidelity:-

- Fidelity is the ability of the radio receiver to reproduce all the modulating frequencies equally.

- Fidelity depends on the frequency response of the audio frequency amplifier.
- The fidelity curve is shown in Fig.

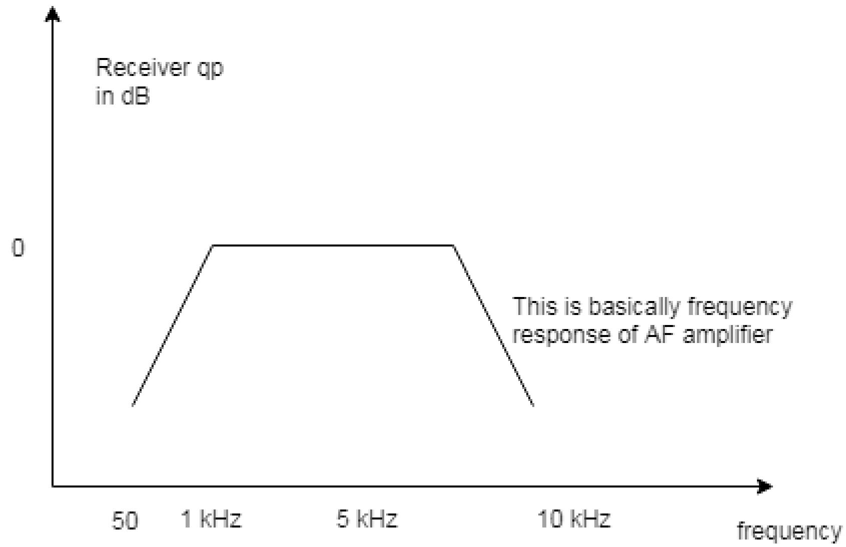


Figure 5:-Fidelity Curve

Q.4 Discuss the merits of delayed AGC as compared with simple AGC .Show AGC curves to illustrate the comparison and explain how delayed AGC may be obtained and applied. What does the delayed AGC control adjust?

Answer:

- The overall gain of receiver is decided by the weakest signal to be received. If a stronger signal is received it will result in higher output levels. Inability to handle these levels by receiver circuits can lead to distortions in the received signal.
- The solution to this problem is to provide gain control in the receiver. We can connect a potentiometer to control gains of RF and IF amplifier, so that larger signals can be taken care of it.

- A more logical and effective solution would be to adjust the gain as and when there is change in the received signal level. Large signal levels will cause gain of the receiver to be reduced whereas a weak signals will have higher gain.
- The dynamic range of receiver is the measure of receiver ability to receive both very strong and very weak signals, without having distortions. It is expressed in dB. The use of AGC increases dynamic range of receiver.

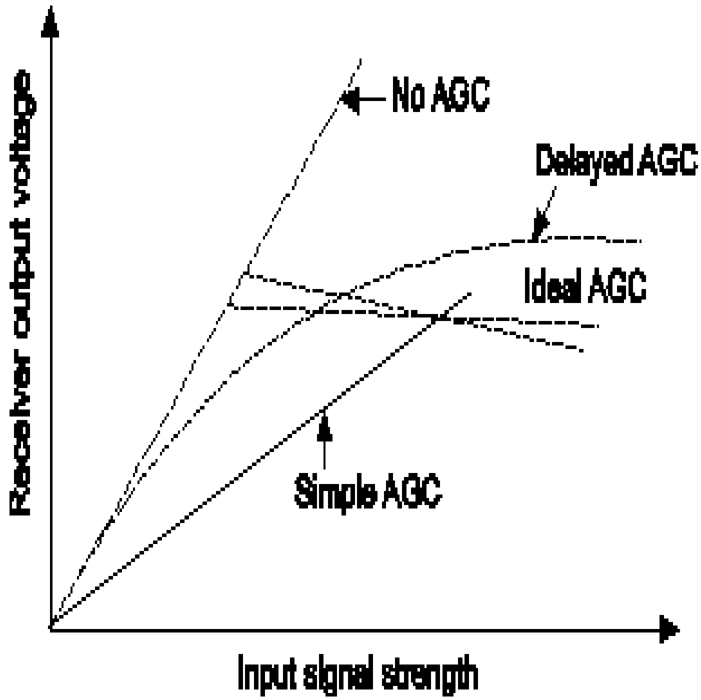


Figure 6:- AGC Curve

Advantages of Simple AGC:

1. Simple.
2. Low cost.
3. Improvement over No AGC.

Disadvantages:

Reduction in gain of the receiver will take place even for the weak signals.

Use:

It is Used in domestic radio receivers.

Q. 5 Draw and explain the block diagram of Ratio detector.

Answer:

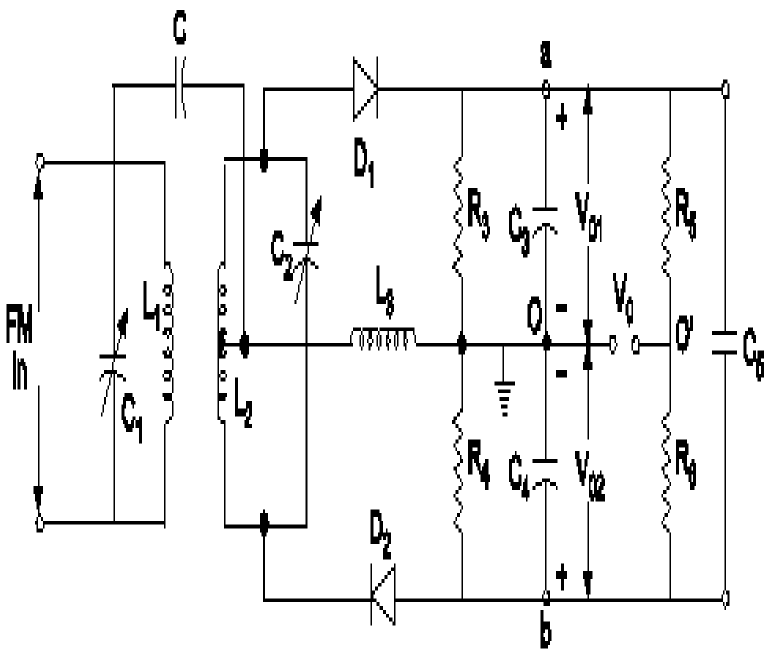


Figure 7:- Ratio detector

Circuit Operation:

With diode D_2 reversed, O (alphabet O) is now positive with respect to b, so that V_a is now sum voltage.

Large capacitor C_5 is connected to keep this sum voltage constant.

Output voltage V_0 is equal to half of the difference between the output voltages from the individual diodes.

$$\therefore \quad V_0 = (Vo_1 - Vo_2)/2$$

Thus, output voltage is proportional to the difference between the individual output voltages.

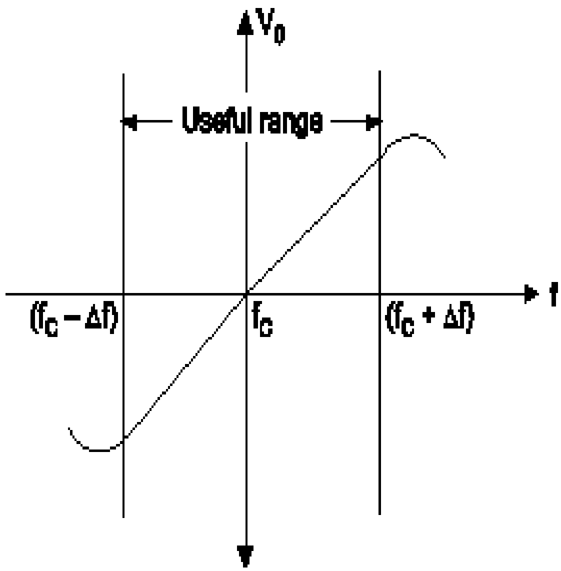


Figure 8:- Ratio Detector Response

Q. 6 Draw & Explain the circuit diagram of foster seeley discriminator circuit & state its advantages.

Answer:

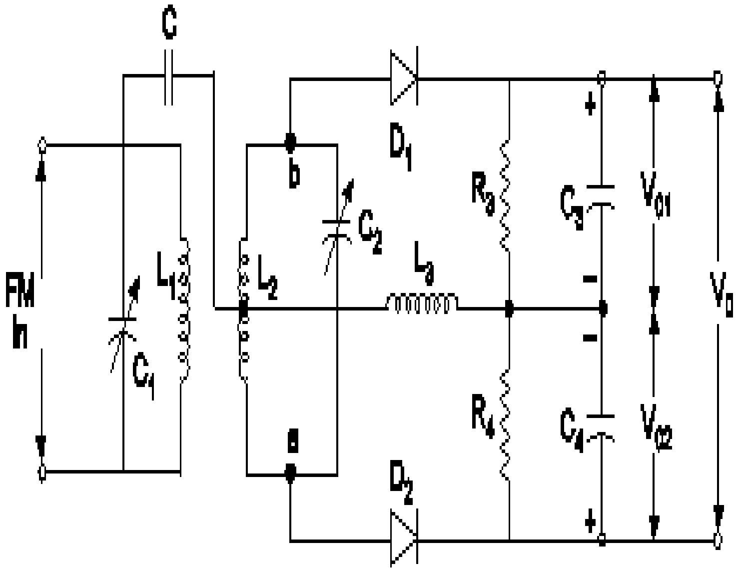


Figure 9:- Foster Seeley Discriminator

Primary and secondary windings both are tuned to the center frequency 'f_c' of the incoming signal.

Although the individual component voltages will be the same at diode inputs at all frequencies, but the vector sum will differ with the phase difference between primary and secondary windings.

Output voltage Vo= Vo1-Vo2

Circuit Operation:

(i) When $f_{in} = f_c$:

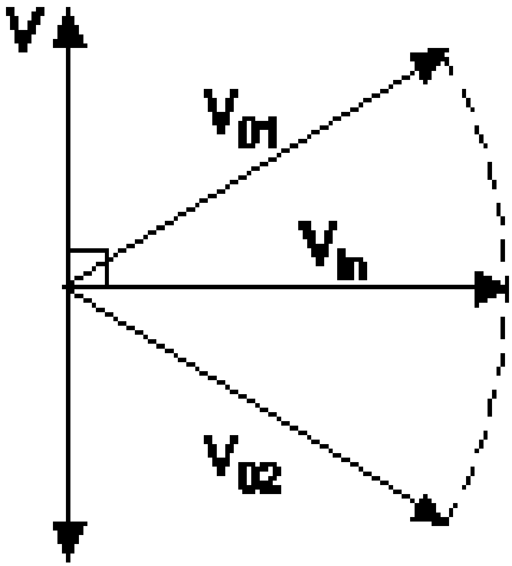
Primary and secondary voltages are exactly 90° out of phase.

As shown in vector diagram,

Input at D_1 = Input at D_2

V_{01} = V_{02}

Vo = 0



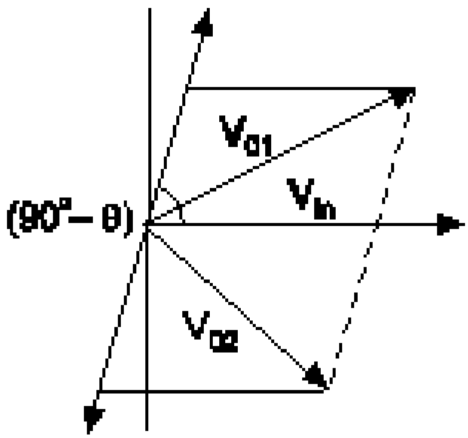
(ii) When $f_{in} > f_c$:

Primary and secondary voltages are less than 90° out of phase.

Input at D_1 >Input at D_2

$\therefore$ V_{01} >V_{02}

Vo1 is positive.



(iii) When $f_{in} < f_c$:

Primary and secondary voltages are more than 90° out of phase.

Input at D_2 > Input at D_1

$\therefore$ V_{02} > V_{01}

Vo2 is Positive.

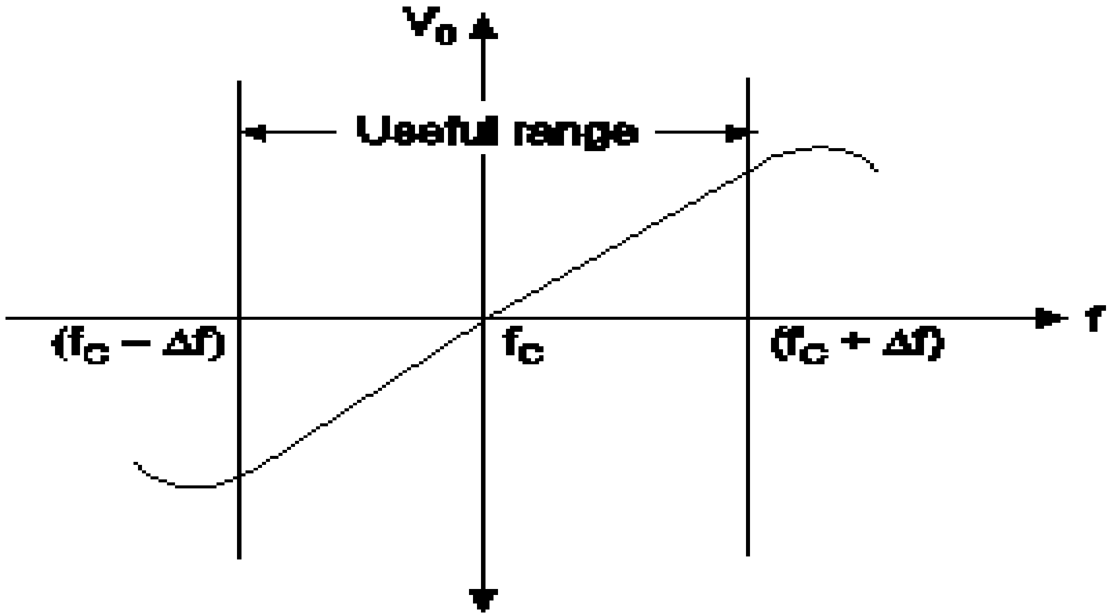


Figure 10:- Foster Seelay Detector Response

Advantages:

It simplifies the alignment (tuning) as both the tuned circuits are tuned to same frequency.

Better linearity.

Disadvantage:

It does not provide amplitude limiting. So that produces error at output.

Q.7 What do you mean by IMAGE frequency and image frequency rejection ratio? (IFRR). How it can be avoided.

Answer:

Image Frequency Rejection

- In the broadcast AM receives the local oscillator frequency is higher than the incoming by intermediate frequency i.e.

$$f_o = f_s + IF \qquad (1)$$

or

$$IF = (f_o - f_s) \qquad (2)$$

- Assume that the local oscillator frequency is set to 'f_o' and an unwanted signal at frequency $f_{si} = (f_o + IF)$ manages to reach at the input of the mixer. Then the mixer output consists of the four frequency components of

$$f_o,\ (f_o + IF),\ (2f_o + IF) \text{ and } IF$$

  Where the last component at IF is the difference between f_{si} and f_o

$$IF = f_{si} - f_o$$

  This component will also be amplified by the IF amplifier along with the desired signal at frequency f_s. This will create interference because both the stations corresponding to carrier frequencies f_s and f_{si} will be tuned at the same position.
  This unwanted signal at frequency f_{si} is known as Image frequency and it is said to be the image of the signal f_s. The relation between f_s and f_{si} is

$$\text{Image frequency} = f_{si} = fs + 2IF \qquad (3)$$

The rejection of an image frequency by a single-tuned circuit, i.e., the ratio of the gain at the signal frequency to the gain at the image frequency, is given by

$$\propto = \sqrt{1 + Q^2 \rho^2} \qquad (4)$$

Where
Q = loaded Q of tuned circuit

$$\rho = \frac{fsi}{fs} - \frac{fs}{fsi} \qquad (5)$$

If the receiver has an RF stage, then there are two tuned circuits, both tuned to fs. The rejection of each will be calculated by the same formula, and the total rejection will be the product of the two. Whatever applies to gain calculations applies also tothose involving rejection. Image rejection depends on the front-end selectivity of the receiver and must be achieved before the IF stage. Once the spurious frequency enters the first IF amplifier, it becomes impossible to remove it from the wanted signal.

Q.8 What factor governs the choice of Intermediate frequency?

Answer:

Choice of IF:

- The intermediate frequency (IF) of a receiving system is usually a compromise, since there are reasons why it should be neither low, nor high, nor in a certain range between these two.

The choice of IF depends on the factors:

1. If the intermediate frequency is too high, results in poor selectivity and poor adjacent channel rejection results.

2. High value of IF increases tracking difficulties.

3. As the IF is lowered, image-frequency rejection becomes poorer.

4. A very low IF can make the selectivity too sharp, cutting-off the sidebands.

5. If the IF is very low, the frequency stability of the local oscillator must be made corresponding higher.

6. The IF must not fall within the tuning range of the receiver, else instability will occur and heterodyne whistles will be heard, making it impossible to tune the frequency band immediately adjacent to the IF.

Q.9 In a broadcast super heterodyne receiver having an RF amplifier with Quality factor of the antenna coupling circuit (at the input to the mixer) is 100. If the intermediate frequency is 455 kHz, calculate

i) The image frequency and its rejection ratio at 1000 kHz.

ii) The image frequency and its rejection ratio at 25 MHz

Answer:

i) fsi=1000+2*455=1910kHz

$$\rho = \frac{1910}{1000} - \frac{1000}{1910} = 1.386$$

$$\propto = \sqrt{1 + 100^2 * 1.386^2} = 138.6$$

ii) fsi=25+2*0.455=25.91MHz

$$\rho = \frac{25.91}{25} - \frac{25}{25.91} = 0.0715$$

$$\propto = \sqrt{1 + 100^2 * 0.0715^2} \quad = 7.22$$

Q.10 Explain the types of AM detector and describe graphically the detection process performed by this circuit.

Answer:

The process in which modulated signal is converted back into original modulating signal is called demodulation.

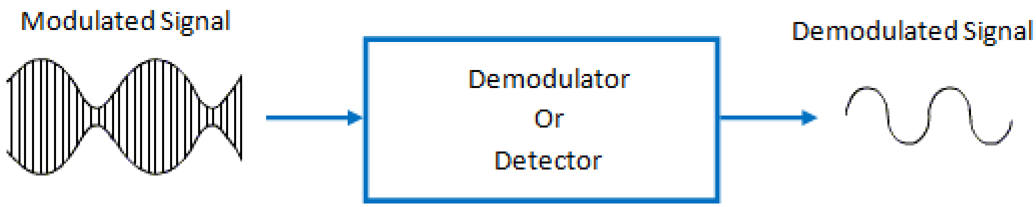


Figure 8:- Block diagram

1) Linear diode detector

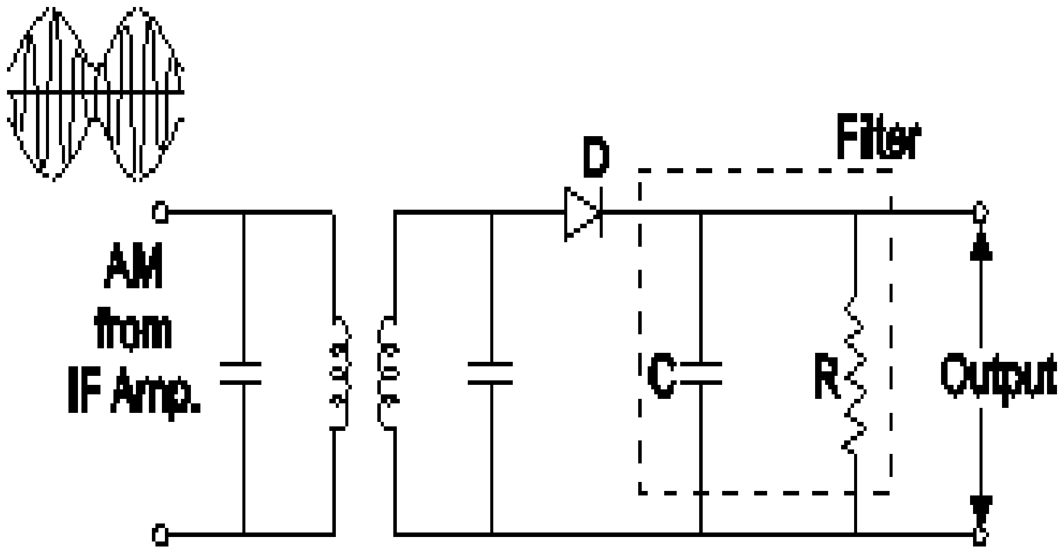


Figure 9:- Linear diode detector

Demodulation of AM signal is done by diode detector circuit.

- **Diode / rectifier:** The diode in the detector serves to that enhances one half of the received signal over the other. In many instances Schottky diodes are used for this form of detector, because signal levels may be low, and Schottky diodes have a much lower turn on voltage (typically around 0.2 V) than standard silicon diodes (typically around 0.7 or 0.7 V).
- **Low pass filter:** The low pass filter is required to remove the high frequency elements that remain within the signal after detection / demodulation. The filter usually consists of a very simple RC network but in some cases It can be provided simply by relying on the limited frequency response of the circuitry following the rectifier. As the capacitor in the circuit stores the voltage, the output voltage reflects the peak of the waveform. Sometimes these circuits are used as peak detectors.

2) Envelop Detector

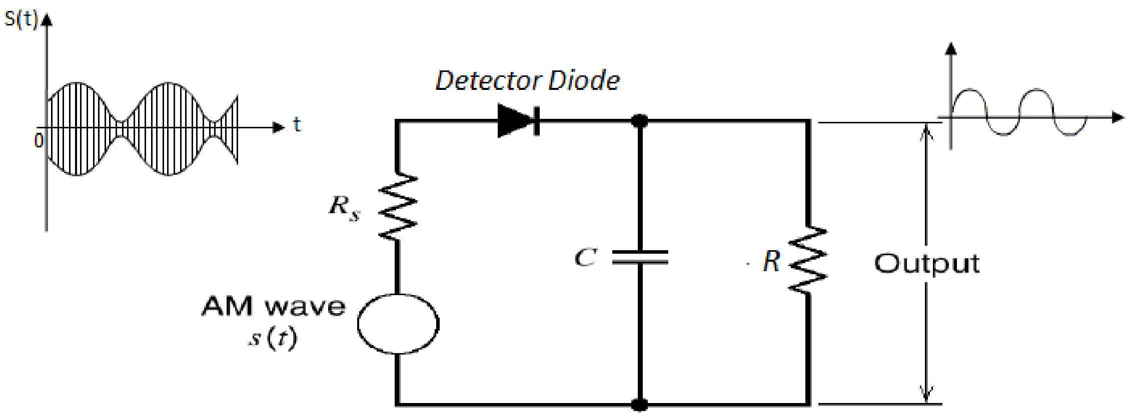


Figure 10 :-Envelope derector

- The standard AM wave is applied at the input of detector.
- In every positive half cycle of input diode is forward biased which charges capacitor ‘C’.
- When capacitor charges to peak value of input voltage,diode stops conducting.
- The capacitor discharges through ‘R’ between positive peaks.
- This process continuous and capacitor charges and discharges repeatedly.

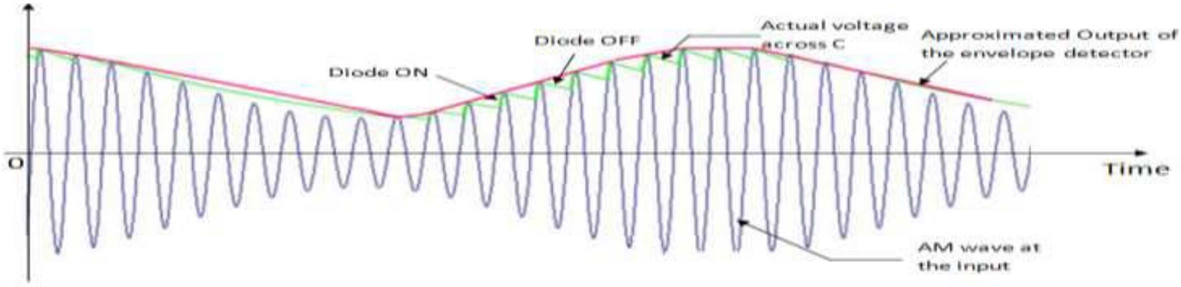


Figure 11 :-Input/output waveform

UNIT-4

Topic: Noise

Q.1 What are the different types of external noise? Explain.

Answer:

•Noise is unwanted signal that affects wanted signal. Noise is random signal that exists in communication systems

Noise level in system is proportional to

- Temperature and bandwidth
- Amount of current
- Gain of circuit
- The various forms of noise created outside the receiver come under the heading of external noise and include atmospheric extraterrestrial noise and industrial noise.

• Types of Noise
  Internal
  External

Internal:

- It is due to random movement of electrons in electronic circuits
- Major sources are resistors, diodes, transistors etc.
- Thermal noise or Johnson noise and shot noise are examples.

External:

- Man-made and natural resources
- Sources over which we have no control
- Examples are Motors, generators, atmospheric sources

.

1 Atmospheric Noise

- Perhaps the best way to become acquainted with atmospheric noise is to listen to show all waves on a receiver which is not well equipped to receive them. An astonishing variety of strange sounds will be heard, all tending to interfere with the program. Most of these sounds arc the result of spurious radio waves which induce voltages in the antenna. The majority of these radio waves come from natural sources of disturbance. They represent atmospheric noise, generally called static.
- Static is caused by lightning discharges in thunderstorms and other natural electric disturbances occurring in the atmosphere. It originates in the form of amplitude-modulated impulses, and because such processes are random in nature, it is spread over most of the RF spectrum normally used for broadcasting.

2 Extraterrestrial Noise

- Solar Noise:- The sun radiates so many things our way that we should no.t be too surprised to find that noise is noticeable among them, again there are two types. Under normal "quiet" conditions, tht::re is a constant noise radiation from the sun, simply because it is a large body at a very high temperature (over 6000°C on the surface). It therefore radiates over a very broad frequency spectrum whjch includes the frequencies we use for communication. However, the sun is a constantly changing star which undergoes cycles of peak activity from which electrical disturbances erupt, such as corona flares and sunspots. Even though the additional noise produced comes from A limited portion of the sun's surface, it may still be orders of magnitude greater than that received during periods of quiet sun.

- Cosmic Noise Since distant stars are also suns and have high temperatures, they radiate RF noise in the same manner as our sun. and what they lack in nearness they nearly make up in numbers which in combination

3. Industrial Noise

- Between the frequencies of 1 to 600 MHz (in urban, suburban and other industrial areas) the intensity of noise made by humans easily outstrips that created by any other source, internal or external to the receiver, Under this heading, sources such as automobile and aircraft ignition, electric motors and switching equipment, leakage from high-voltage lines and a multitude of other heavy electric machines are all included.

- Fluorescent lights are another powerful source of such noise and therefore should not be used where sensitive receiver reception or testing is being conducted. The noise is produced by the arc discharge present in all these operations, and under these circumstances it is not surprising that this noise should be most intense in industrial and densely populated areas.

Q.2 The noise output of the resistor is amplified by a noiseless amplifier having a gain of 40 and bandwidth of 40KHZ. A meter connected to the output of the amplifier reads 4mV rms (a) if the resistor is operated at 27°C. What is the Resistance? (b) If the bandwidth of the amplifier is reduced to 10KHz, its gain remaining constant what will the meter read now?

Answer:

Given Data: Gain (A)=40

Bandwidth (B): 40KHz

Vo=4mV

T=27°C=300K

We know that,

Vo=AV

V=Vo/A = 4mV/ 40

$V=1*10^{-4}$ V

Now ,

$R=Vn^2 / 4KTB$

= (1*10-4) / $(4*1.38*10^{-23}*300*40*10^{3})$

R= 15.09 MΩ

Now , Bandwidth is reduced to 10KHz

Vn=$\sqrt{4KTB}$TB

=$\sqrt{4 * 1.38 * 10^{-23} * 300 * 10^{4}}$

= $4.098*10^{-5}$ V

Now,

Vo=AVn

Meter reading(Vo) = $40*4.998*10^{-5}$ V

= 1.999Mv

Q.3 A parallel tuned circuit having Q=10 resonates at 10MHz with 10Pf capacito . If this circuit is maintained at 27°C , What noise voltage will a wide band voltmeter measure when placed across it.

Answer:

Given Data: Q=10

F=10MHz

C=10pF

T=27°C

We know that ,

$Q = \omega CR$

$R = Q / \omega C$

$= 10 / 2*3.14*10**10*10^{-12}$

$=15923.56\ \Omega$

Now,

$B = F / Q = 10MHz / 10 = 10^6\ Hz$

$Vn=\sqrt{4KTB}R$

$=\sqrt{4 * 1.38 * 10^{-23} * 300 * 10^6 * 15923.56}$

$Vn= 1.62*10^{-5}\ V$

$= 16.2\ \mu V$

Q.4 Compute the noise voltage at the input of a video amplifier using a device having 300Ω equivalent noise resistance and 400Ω input resistor. The bandwidth of the amplifier is 7MHz and the ambient temperature is 270C.

Answer:

Equivalent Noise resistance R = 300Ω

Input Resistor = 400Ω

B = 7Mhz

T= 27°C

R = 300+400= 700Ω

Vn=$\sqrt{4KTBR}$

=$\sqrt{4 * 1.38 * 10^{-23} * 300 * 10^{6} * 700 * 7}$

= 9 μV

Q.5 A parallel tuned circuit having Q of 20 is resonated to 200 MHz with a 10Pf capacitor. If this circuit is maintained at 17°C what is equivalent noise voltage?

Answer:

Given data-

Q = 20

F =200 MHz

C = 10PF

T = 17°C

We know that,

Q = ωCR

R = Q/ωC

= $20/2*3.14*200*10^{6}*10^{-12}$

= 1592.35

Now,

B = F/Q = $200*10^{6}/20$

B = 10^{7}

$$Vn=\sqrt{4KTBR}$$

$$=\sqrt{4 * 1.38 * 10^{-23} * 290 * 10^{7} * 1592.35} = 1.596*10^{-5}$$

$$Vn = 15.96\ \mu V$$

Q.6 A receiver is connected to an antenna whose resistance is 50 **Ω** has an equivalent noise resistance of 30 **Ω**. Calculate the receiver's noise figure in decibals and its equivalent noise temperature?

Answer:

Given data – Ra = 50 **Ω**

Req. noise = 30 **Ω**

F = 1+Req/Ra

F = 1+30/50

F = 80/50

F = 1.6

Therefore,

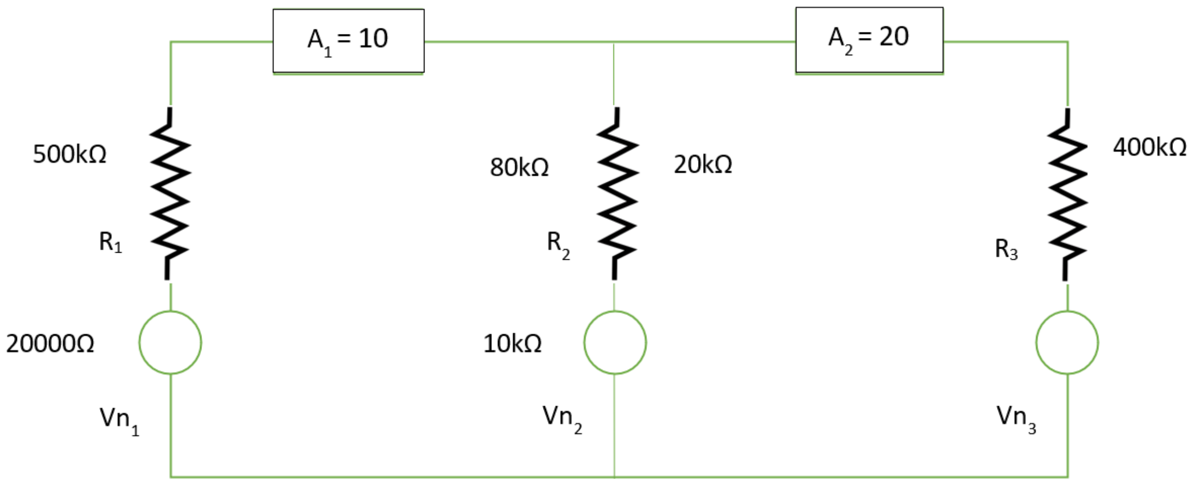
10log(1.6) = 10 * 0.204 = 2.04 dB

Teq = To(F - 1) = 290(1.6 - 1) = 290 * 0.6

= 174K

Q.7 The first stage of a two stage amplifier has output resistance of 20kΩ, voltage gain of 10, input resistance of 500Ωand equivalent noise resistance of 20000Ω . The second stage has voltage gain 20, input resistance of 80KΩ, equivalent noise resistance of 10KΩ and output resistance 400KΩ.Compute the equivalent input noise resistance of two stage amplifier. Also compute the voltage at the input of the first stage given that the bandwidth of amplifier is 10 kHz and the ambient temperature is 3000K.

Answer:



Calculate: - R_{eq} & V_n

1) For R_{eq}

$$R_3 = 400k\Omega$$

$$R_2 = (20k \parallel 80k) + 10k = 26k\Omega$$

$$R_1 = 500 + 20000 = 20.5k\Omega$$

$$R_{eq} = R_1 + \frac{R2}{A1^2} + \frac{R3}{A1^2 A2^2}$$

$$R_{eq} = 20.77k\Omega$$

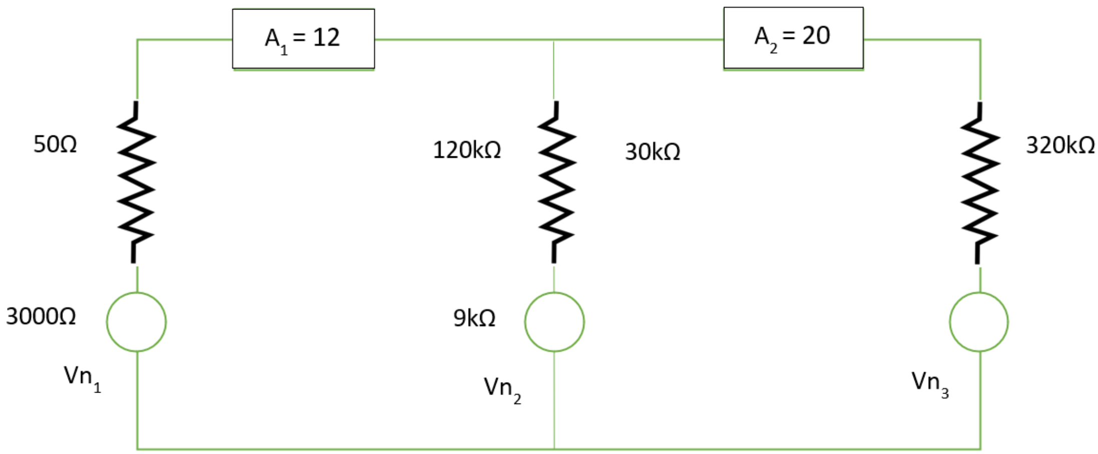
2) For V_n

$$K = 1.38 \times 10^{-23}\ ,\ B = 10kHz\ ,\ T = 3000K$$

$$Vn = \sqrt{4KTBR} = 5.866\mu V.$$

Q.8 The first stage of two stage amplifier has output resistor of 30KΩ, voltage gain of 12, input resistance of 500Ω and equivalent noise resistance of 3000Ω.The second stage has output resistor of 320KΩ, voltage gain of 20, input resistance of 120KΩ and equivalent noise resistance of 9 KΩ. The amplifier is driven by a generator of output impendence 40Ω.For this two stage amplifier compute i)equivalent input noise resistance ii)equivalent input noise voltage given that the bandwidth of the amplifier is 20KHz and ambient temperature is 3000K and iii) the noise figure.

Answer:



Calculate :- R_{eq}& V_n

1) For R_{eq}

$R_3 = 320k\Omega$

$R_2 = (120k \,||\, 30k) + 9k = 33k\Omega$

$R_1 = 500 + 3000 = 3.5k\Omega$

$$R_{eq} = R_1 + \frac{R2}{A1^2} + \frac{R3}{A1^2 A2^2}$$

$R_{eq} = 3.734k\Omega$

2) For V_n

$K=1.38 \times 10^{-23}$, $B = 10kHz$, $T = 3000$

$$V_n = \sqrt{4KTBR} = 3.517\mu V.$$

3)For Noise Figure F

$R_{eq} = 3.734k\Omega$, $R_t = 500\Omega$, $R_a = 40\Omega$

$R_{eq}' = R_{eq} - R_t = 3.234K\Omega$

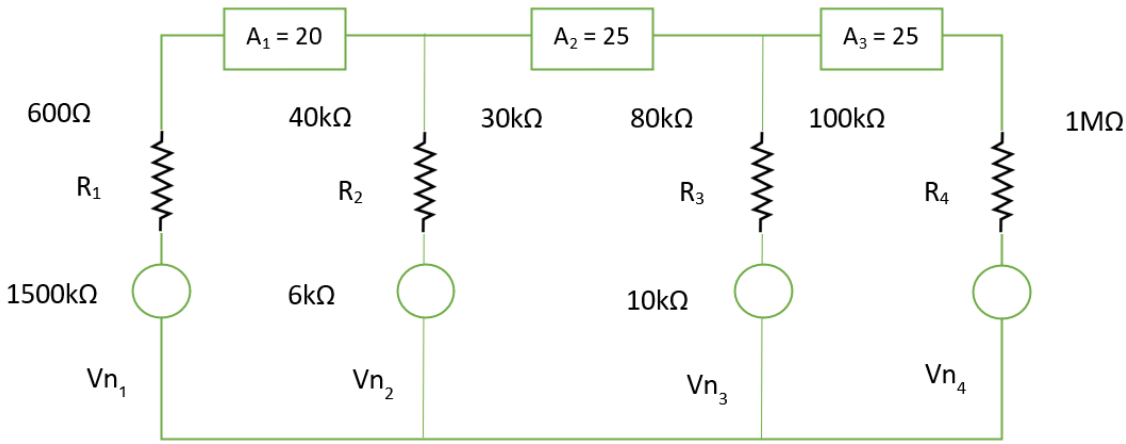
$$F = 1 + \frac{Req'(Ra+Rt)}{(Ra \times Rt)}$$

$$= 1 + \frac{3.234K(40+500)}{(40 \times 500)}$$

$F = 88.318\Omega.$

Q.9 The three stage amplifier operate over a bandwidth of 10 KHz at a temperature of 300 K .the three stages have respectively i) voltage gain of 20,25 & 25 ii) input resistance of 600Ω,40Ω,80KΩ iii) equivalent noise resistance of 1500Ω,6KΩ &10KΩ. iv) Output resistance 30KΩ, 100KΩ & 1MΩ.Compute equivalent i/p noise resistance of the overall three stage amplifier. Also compute the equivalent noise voltage of the first stage.

Answer:



ns :-

1) For R_{eq}

$R_4 = 1M\Omega$

$R_3 = (\ 80k \parallel 100k\ ) + \ 10k \ = \ 54.44k\Omega$

$R_2 = (\ 40k \parallel 30k\ ) + \ 6k \ = \ 23.142k\Omega$

$R_1 = 600 + 1500 = 2.1k\Omega$

$R_{eq} = R_1 + \frac{R2}{A1^2} + \frac{R3}{A1^2 A2^2} + \frac{R4}{A1^2 A2^2 A3^2}$

$R_{eq} = 2.158\text{k}\Omega.$

2) For V_{n1}

$V_{n1} = \sqrt{4KTBR}$

$V_{n1} = 0.5898\ \mu\text{V}.$

Q.10 The noise output of a resistor is amplified by a noiseless amplifier having gain of 40 and bandwidth of 40KHz.A meter connected to the output of the amplifier reads 4mV rms a) if the resistor is operated at, what is its resistance? b) If the bandwidth of the amplifier is reduced to 10KHz,its gain remaining constant, what will the meter read now?

Answer:

(a)

$Vn = \sqrt{4KTBR}$

Hence $R = \frac{Vn^2}{4KTB}$

The rms noise voltage generated in the resistor $= \frac{4mV}{40} =$ 100uV.

Hence $R = (100 \times 10^{-6})^2 / 4 \times 1.38 \times 10^{-23} \times (273+27) \times 40 \times 10^3$

$R = 15.1 \times 10^6\ \Omega$

(b) Initially

$B = 40\text{kHz}$

Then$Vn = \sqrt{4KTBR}$

$$Vn = A\sqrt{4KTBR}$$

Where A is the amplifier Gain, Next bandwidth is reduced to 10kHz, i.e. B' = B/4

Hence $$Vn = A\sqrt{4KTR(\frac{B}{4})} = (\frac{1}{2} \times 4)mV = 2mV.$$

UNIT No 5

TOPIC: PULSE MODULATION

Q.1 Explain the generation of PAM.

Answer:

Pulse Amplitude Modulation

Pulse Amplitude Modulation is the process in which the amplitude of the pulse is varied in accordance with the instantaneous amplitude of the message signal at sampling interval. The width and position of the pulse is kept constant. The PAM transmitter design is very simple in which we can get Natural Sampling output, Flat-top sampling output, and Sample and Hold output.

Working of PAM

A message signal of variable amplitude is applied at one end of the PAM system and at the other end fixed amplitude pulse train is applied. Before sampling the output these two signals are multiplied and we get Natural sampled output. Three test points are available on the kit with which we can easily observe that the amplitude of the fixed pulse train varies in accordance with the amplitude of the message signal. At the receiver section Sample and Hold output of PAM is passed through the 2^{nd} order Butterworth filter and we get back the Analog signal which is very close to the message input signal.

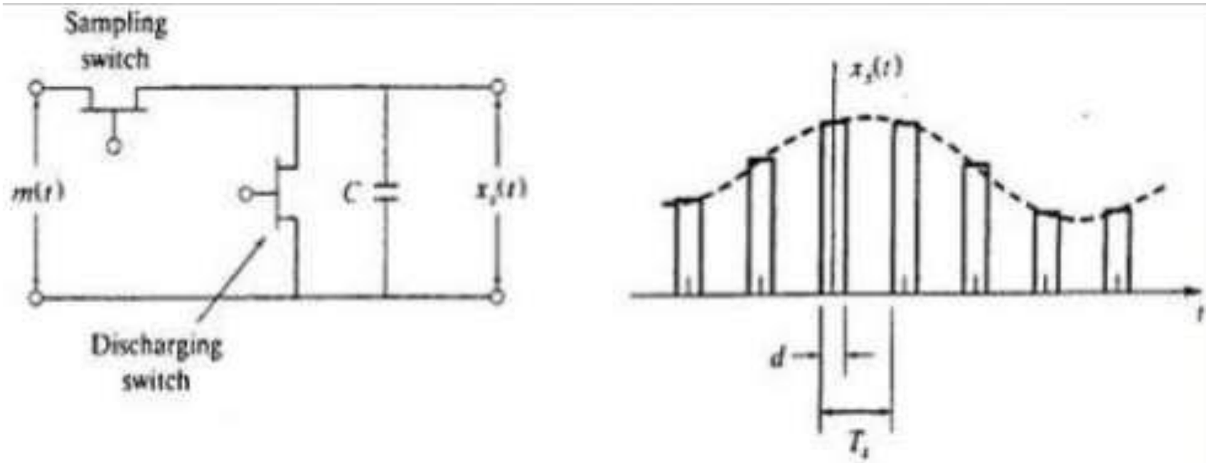


Figure 1:- PAM Generator

- The sample and hold circuit consists of two FETs and a capacitor.
- The sampling switch is closed for a short duration by a short pulse applied to the gate G1 of transistor.

- During this period,the capacitor is quickly charged to a voltage eequal to instantaneous sample value of incoming signalx(t)
- Now the sampling switch is opened and capacitor holds the charge.
- The discharge switch is then closed by a pulse applied to gate G2 of second transistor.
- Due to this the capacitor is discharged to zero volts. The discharge switch is then opened and the capacitor has no voltage.
- Hence the output of sample and hold circuit consists of sequence of flattop samples.

Q.2 What is pulse modulation? State advantages and disadvantage of PAM.

Answer:

Pulse Modulation:

It is the process of transmitting the signals in the form of pulses by using some special techniques. There are two types of pulse modulation systems,

1. Pulse Amplitude Modulation

2. Pulse Time Modulation

Pulse time modulation is further divided into,

•Pulse Width Modulation

•Pulse Position Modulation

1) Pulse Amplitude Modulation

Advantages:

•It is the simple process for modulation and demodulation

•Transmitter and receiver circuits are simple and easy to construct.

Disadvantages:

•Bandwidth requirement is high

• Noise Interference is maximum

•Power requirement is high

Applications:

•Used in microcontrollers for generating control signals

•Used as electronic driver for LED lighting

Q.3 Explain generation of PWM signal. How is it demodulated?

Answer:

PWM GENERATION

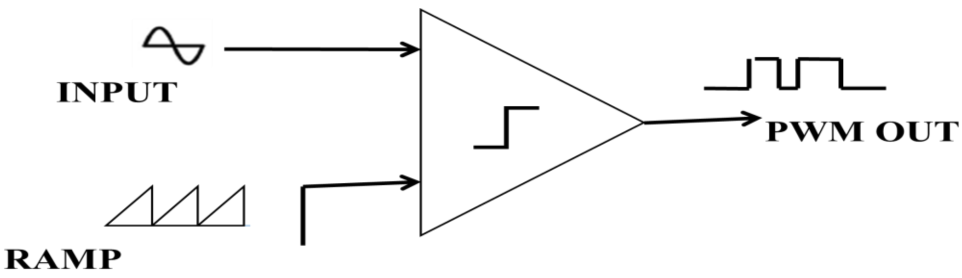


Figure 2: - Pulse width Modulator

•Modulating signal x(t) is applied to non-inverting terminal of comparator.

•Comparator output remains high as long as instantaneous sample amplitude of x(t) is higher than saw tooth signal.

•This gives the PWM output at the output of comparator.

•The leading edges of PWM waveform coincide with falling edges of ramp signal

•Therefore, leading edges of PWM signal are always generated at fixed time intervals

•Occurrence of falling edge of PWM signal is dependent on instantaneous amplitude of x(t)

Q.4 Explain detection of PWM signal.

Answer:

DETECTION OF PWM

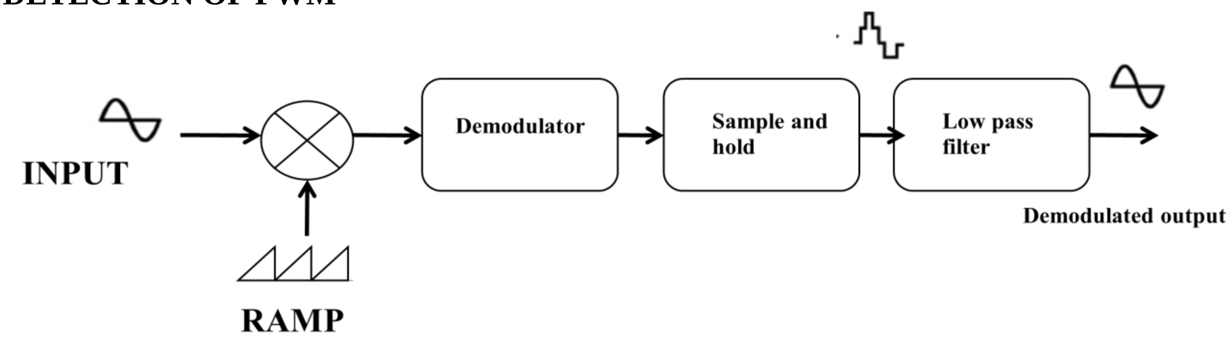


Figure 3:-Pulse width Demodulator

•The regenerated pulses are applied to a reference pulse generator.

•It produces train of constant amplitude and constant width pulses.

•These pulses are synchronized to the leading edges of regenerated PWM pulses but delayed by fixed intervals.

•There generated PWM pulses are also applied to a ramp generator whose o/p is a constant slope ramp for the duration of the pulse.

•At the end of the pulse a sample and hold circuit retains the final ramp voltage until it is reset at the end of the pulse.

•The constant amplitude pulses at the o/p of the reference generator are then added to ramp signal.

•O/P of the adder is then clipped off at a threshold level to generate a PAM signal.

•AlowpassfilterisusedtorecovertheoriginalmodulatingsignalbackfromPAMsignal.

Q.5 Explain generation of PPM with waveform. How is it derived from PWM?

Answer:

PULSE POSITION MODULATION (PPM)

It is the Modulation technique in which position of pulses of carrier pulse train is varied in accordance with amplitude of modulating signal.

Generation:

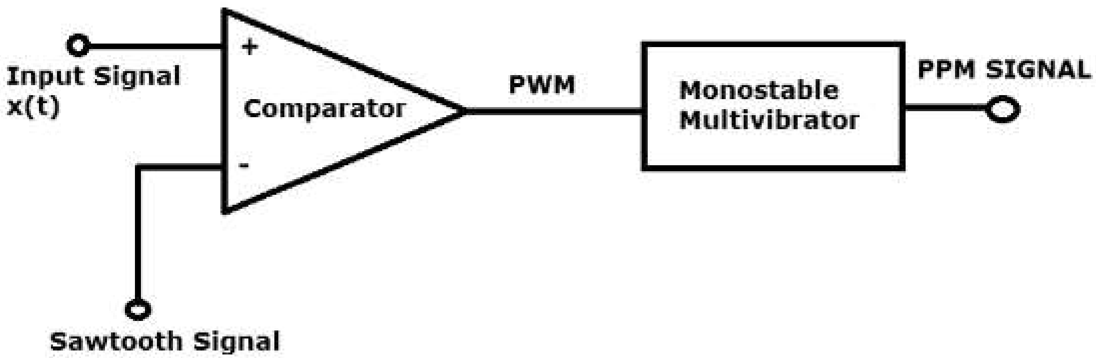


Figure 4:-Pulse Position Modulator

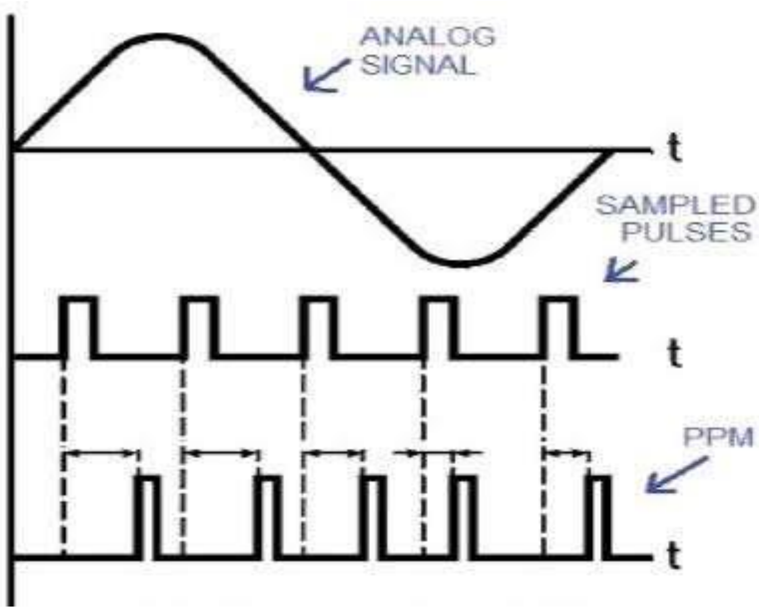


Figure 5:-Pulse position wavefoem

DETECTION OF PPM

•The circuit consists of S-R flip-flop which is set or gives high output when reference pulses arrive.

•Reference pulses are generated by a reference pulse generator.

•Flip-flop circuit is reset and gives low output at the leading edge of PPM signal.

•The process repeats and we get PWM pulses at the output of f lip-flop.

•PWM pulses are then demodulated in a PWM demodulator to get original modulating signal.

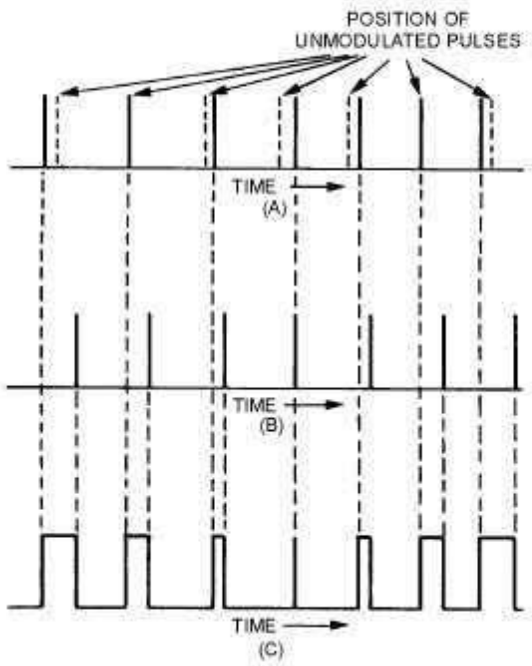


Figure 6:- DETECTION OF PPM

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

Q.6 Draw and Explain the block diagram of PCM transmitter and Receiver.

Answer:

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 PCMPCM.

A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., 1s and 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.

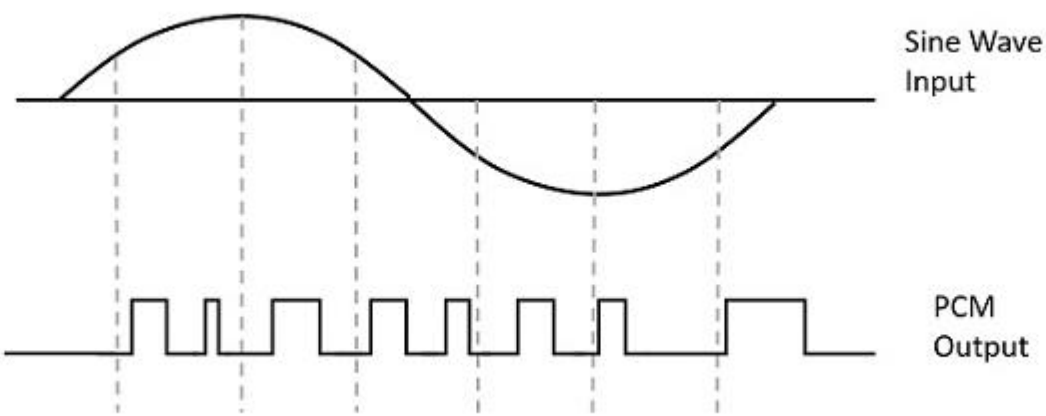


Figure 7:- Input/Output Waveform

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.

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.

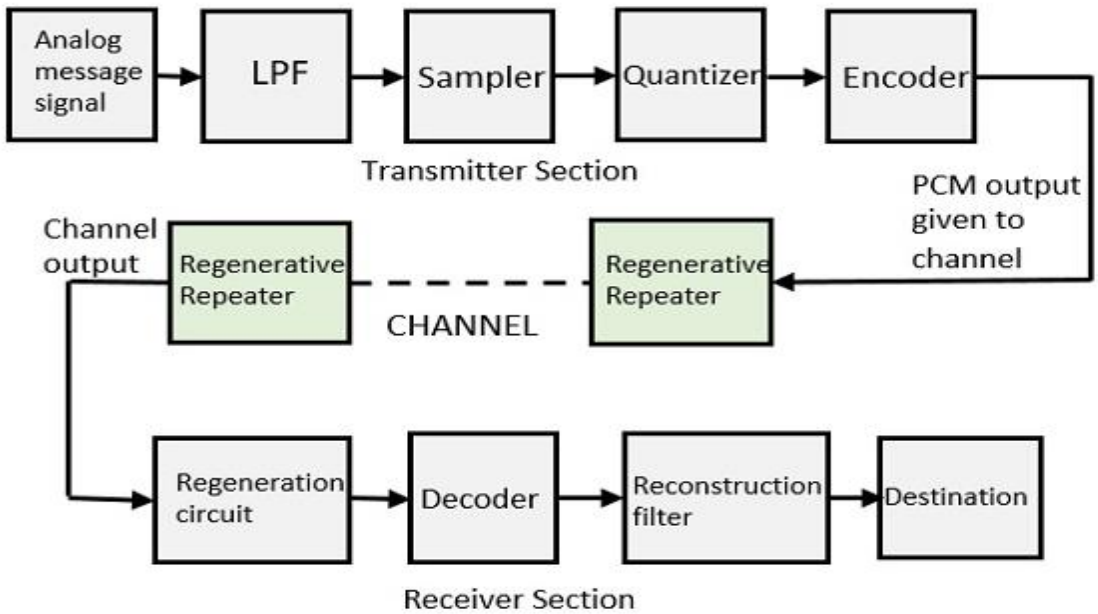


Figure 8:- Block Diagram of PCM System

1) **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.

2) **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.

3) **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.

4) **Encoder**

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

5) **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.

6) **Decoder**

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

7) **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.

Q.7 Explain Time Division multiplexing technique.

Answer:

- Time-division multiplexing (TDM) is a digital process that can be applied when the data rate capacity of the transmission medium is greater than the data rate required by the sending and receiving devices.
- TDM can be implemented in two ways: synchronous TDM and asynchronous TDM.
- In synchronous time-division multiplexing, the term synchronous means that the multiplexer allocates exactly the same time slot to each device at all times, whether or not a device has anything to transmit.
- Frames
- Time slots are grouped into frames. A frame consists of a one complete cycle of time slots, including one or more slots dedicated to each sending device.

Types of TDM

- Synchronous TDM does not guarantee that the full capacity of a link is used. Because the time slots are reassigned and fixed, whenever a connected device is not transmitting, the

  corresponding slot is empty.

- Asynchronous time-division multiplexing, or statistical time-division multiplexing, is designed to avoid this type of waste.
- Like synchronous TDM, asynchronous TDM allows a number of lower-speed input lines to be multiplexed to a single higher-speed line. However, in asynchronous TDM the total speed of the input lines can be greater than the capacity of the link.
- In an asynchronous system, if we have n input lines, the frame contains no more than m slots, with m less than n.
- The number of time slots in an asynchronous TDM frame (m) is based on statistical analysis of the number of input lines that are likely to be transmitting at any given time.
- In this case any slot is available to any of the attached input lines that have data to send.

Q.8 What is the concept of frequency division multiplexing?

Answer:

- In FDM signals generated by each device modulate different carrier frequencies. These modulated signals are combined into a single composite signal that can be transported by the link.
- Carrier frequencies are separated by enough bandwidth to accommodate the modulated signal.
- These bandwidth ranges are the channels through which various signals travel.
- Channels must separated by strips of unused bandwidth (guard bands) to prevent signal
- In this a number of signals are transmitted at the same time, and each source transfers its signals in the allotted frequency range. There is a suitable frequency gap between the 2 adjacent signals to avoid over-lapping. Since the signals are transmitted in allotted time so this decreases the probability of collision.
- The frequency spectrum is divided into several logical channels, in which every user feels that they posses a particular bandwidth.
- A number of signals are sent simultaneously on the same time allocating separate frequency band or channel to each signal. It is used in radio and TV transmission. Therefore to avoid interference between two successive channels Guard bands are used.

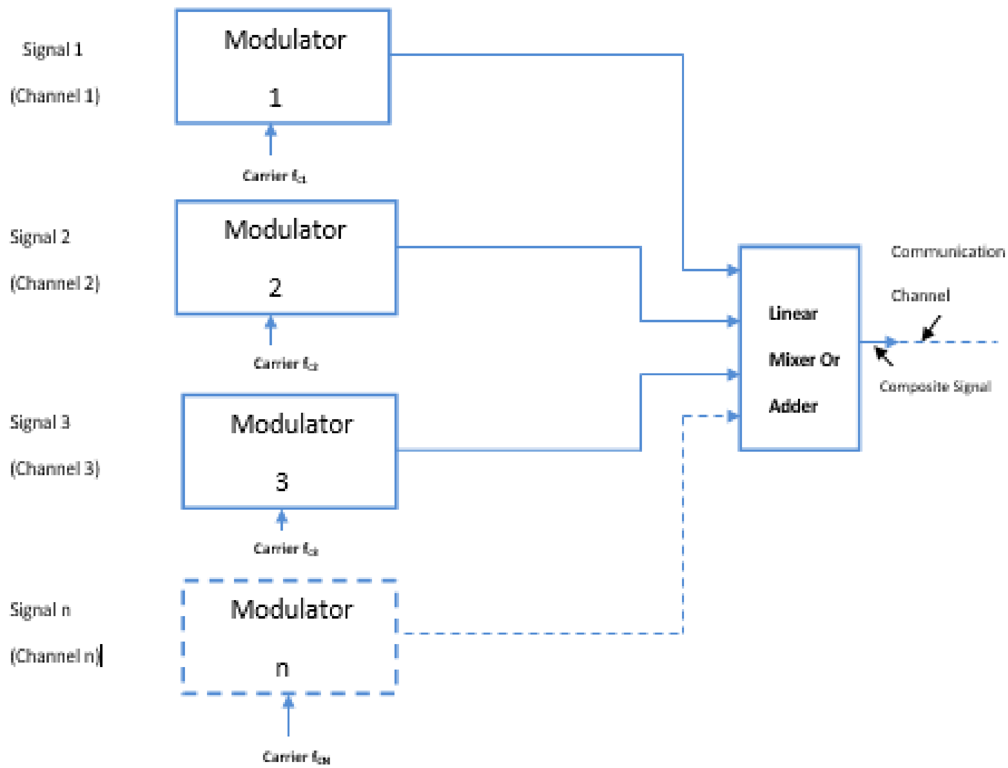


Figure 9:- FDM Transmitter for Modulation

Advantages:

- It does not need synchronization between its transmitter and receiver.
- Frequency division multiplexing (FDM) is simpler and easy demodulation.
- Due to slow narrow band fading only one channel gets affected.
- It is used for analog signals.
- A large number of signals (channels) can be transmitted simultaneously.

Disadvantages:

- It is suffers problem of cross-talk.
- It is used only when a few low speed channels are desired.
- Intermodulation distortion takes place.

Applications:

- It is used to public telephones and in cable TV systems.
- It is used in broad casting.
- It is used in AM and FM broadcasting.

Q. 9 Give the advantages of PCM. Also explain quantization in brief.

Answer:

Advantages of PCM

- Analog signals can be transmitted over a high- speed digital communication system.
- The probability of occurring error will reduce by the use of appropriate coding methods.
- PCM is used in Telkom system, digital audio recording, digitized video special effects, digital video and voice mail.
- PCM is also used in Radio control units as transmitters and also a receiver for remote-controlled cars, boats, planes.
- The PCM signal is more resistant to interference than normal signals.

Quantization.

- The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.
- The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. **Quantization** is representing the sampled values of the amplitude by a

finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

- Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation levels is called a quantum or step-size.overlapping.
- In FDM, signals are modulated onto separate carrier frequencies using either AM or FM modulation.

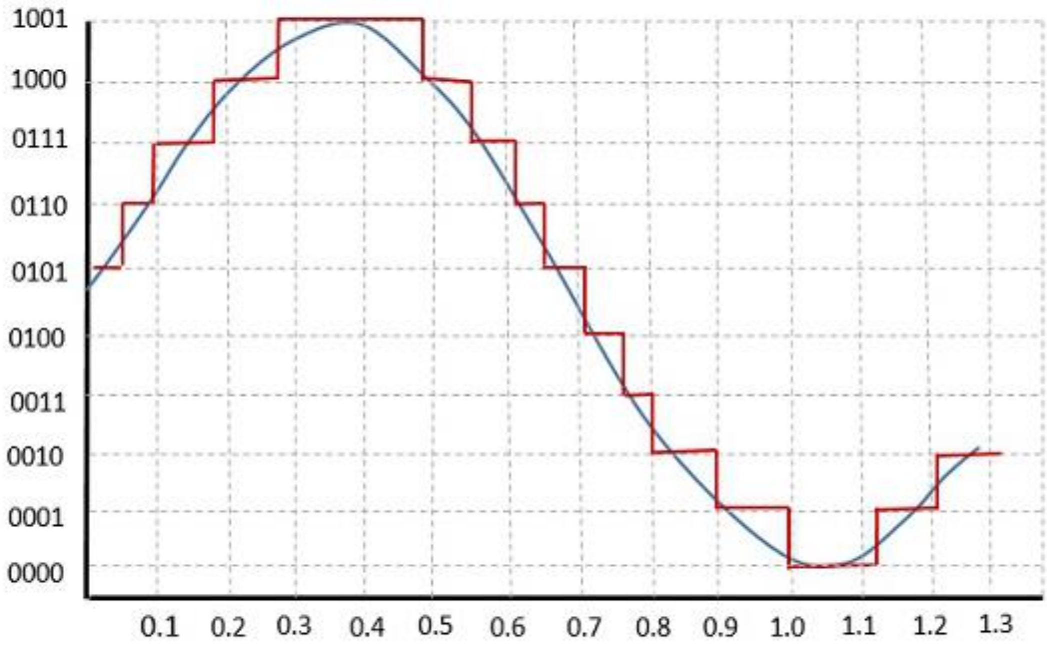


Figure 10:- Block Diagram of PCM System

Q. 10 Explain Companding process in PCM in brief.

Answer:

Companding in PCM:-

The word Companding is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques. They are −

A-law Companding Technique

- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and no compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.

μ-law Companding Technique

- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and no compression is done.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.
- μ-law is used in North America and Japan.

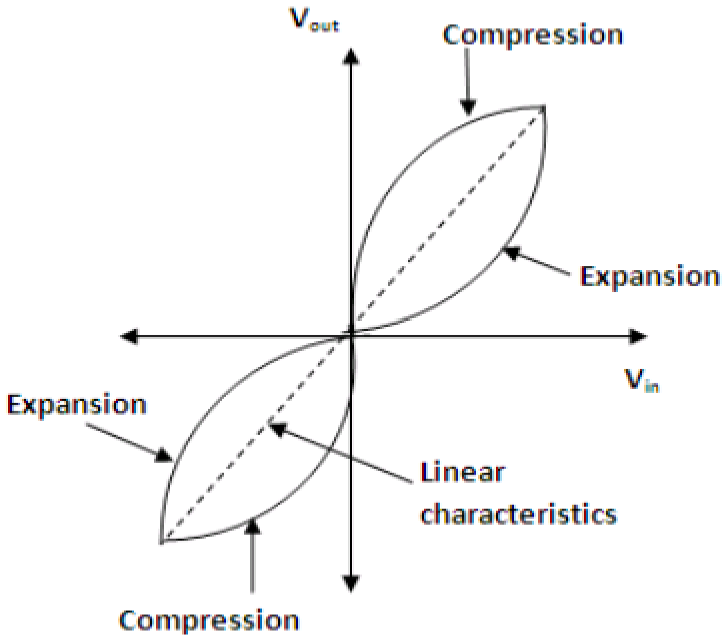


Figure 11:- Companding curve of PCM System

UNIT No 6

TOPIC:- RADIATION AND WAVE PROPAGATION

Q1. Describe ground wave propagation. How does it affect field strength at a distance from the transmitter?

Answer:
Ground wave Propagation:-

- The ground wave is a wave that is guided along the surface of the earth and is vertically polarized to prevent short circuiting the electric component.
- A wave induces currents in the ground over which it passes and thus loses some energy by absorption.
- This wave permits the propagation around the curvature of the earth.
- This mode of propagation exists when the transmitting and receiving antennas are close to the surface of the earth and is supported at its lower edge by the presence of the ground.
- It is called as medium wave propagation and
- Is used in local broadcasting.
- At high frequency, wave attenuation by ground is much more than at low frequency over the same ground.
- All the signals received during day time is due to ground wave propagation
- The earth attenuation increases as frequency increases. So this mode of propagation is suitable for low and medium frequency i.e. upto 2MHz only.

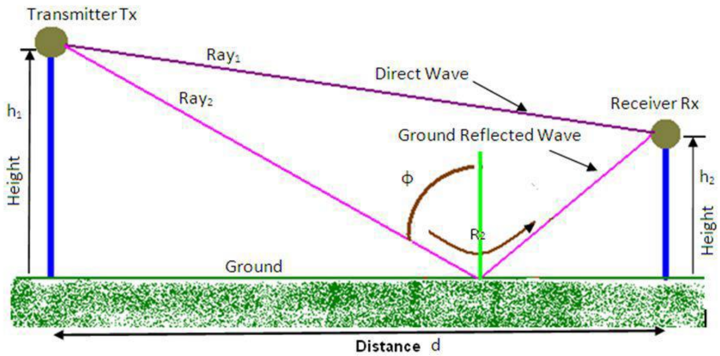


Figure 1:- Direct and ground reflected components of the space wave

Field Strength at a Distance
Electric field strength E at a distance from TX antenna due to ground wave,

E = 120 π ht /λ d (volt/meter) **in volts**
V = 120 π ht hr Is/λ d

where,
120 π –Intrinsic impedance of free space
ht, hr–Effective heights of transmitting and receiving antennas
Is –Antenna currents
d –Distance between TX and RX antennas
λ –Wavelength

Q.2 Prove that when electromagnetic waves are reflected from a perfectly conducting medium, the angle of refraction is equal to the angle of incidence.

Answer:

Reflection of waves

- Much similarity between the reflection of light by a mirror and the reflection of electromagnetic waves by a conducting medium.
- The angle of reflection is equal to the angle of incidence as shown in fig below
- As with the reflection of light ,the incident ray, the reflected ray and the normal at the point of incidence are in one plane.
- Reflection coefficient $\rho$is defined as the ratio of the electric intensity of the reflected wave to that of the incident wave.

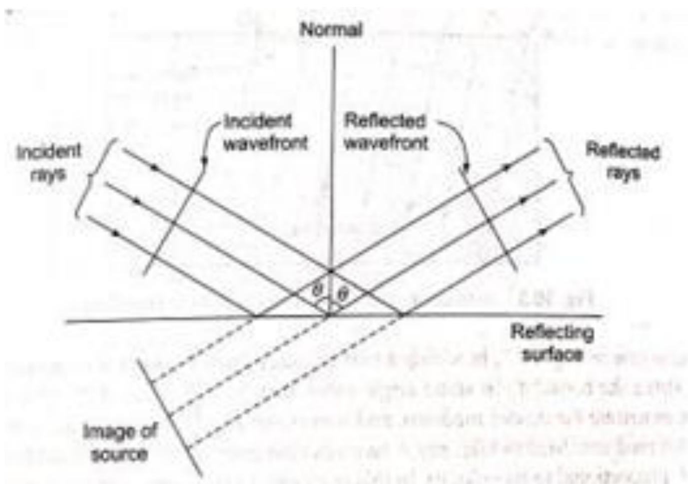


Figure 2:-Reflection of waves

There is much similarity between the reflection of light by a mirror and the reflection of electromagnetic waves by a conducting medium. In both instances the angle of reflection is equal to the angle of incidence, as illustrated in Figure 2. Again, as with the reflection of light, the incident ray, the reflected ray and the normal at the point of incidence are in the one plane. The concept of *images* is used to advantage in both situations. The proof of the equality of the angles of reflection and incidence follows the corresponding proof of what is known as *the second law of reflection* for light. Both proofs are based on the fact that the incident and reflected waves travel with the same velocity. There is yet another similarity here to the reflection of light by a mirror.

Q.3 Explain the concept of linear polarization with its types.

Answer:

POLARIZATION:

- The polarization of an antenna is the orientation of the electric field with respect to the Earth's surface and is determined by the physical structure of the antenna and by its orientation.
- Radio waves from a vertical antenna will usually be vertically polarized.
- Radio waves from a horizontal antenna are usually horizontally polarized.

Types of Polarization:-

Polarization refers to the physical orientation of the radiated waves in space. Waves. are said to be polarized (actually linearly polarized) if they all have the same alignment in space. It is a characteristic of most antennas that the radiatjon they emit is linearly polarized. A vertical antenna will radiate waves whose electric vectors will all be vertical and will remain so in free space. Light emitted by incoherent sources, such as the sun, has a haphazard arrangement of field vectors and is said to be randomly polarized.

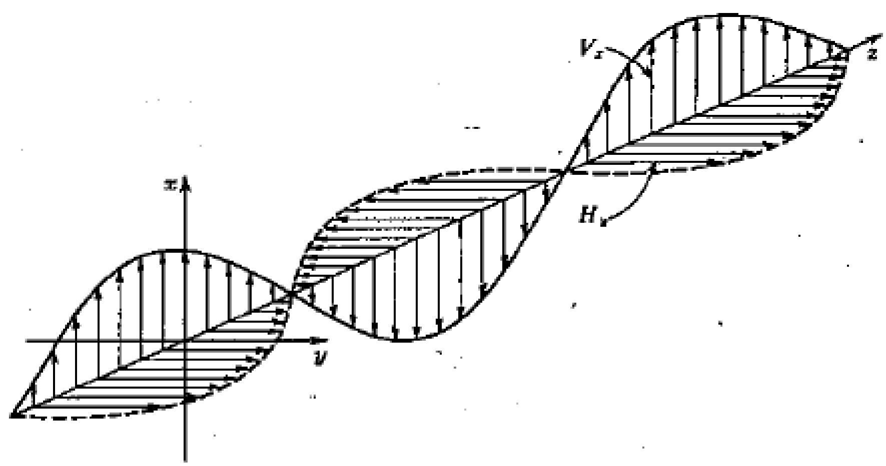


Figure 3:- Transverse electromagnetic wave in free space.

The wave of above Figure is, of course, linearly polarized and is also said to be vertically polarized, since all the electric intensity vectors are vertical. The decision to label polarization direction after the electric intensity is not as arbitrary as it seems; this makes the direction of polarization the same as the direction of the antenna. Thus, vertical antennas radiate vertically polarized waves, and similarly horizontal antennas produce waves whose polarization is horizontal. There has been a tendency, over the years, to transfer the label to the antenna itself. Thus people often refer to antennas as vertically or horizontally polarized, whereas it is only their radiations that are so polarized.

Q.4 What is refraction? Explain under what circumstances it occurs and what causes it.

Answer:

Refraction:-

Refraction takes place when EM wave pass from one medium to another medium with different density & different propogation velocity.

Index of refraction ɳ is used in refraction calculations

$$ɳ = \sqrt{€r}$$

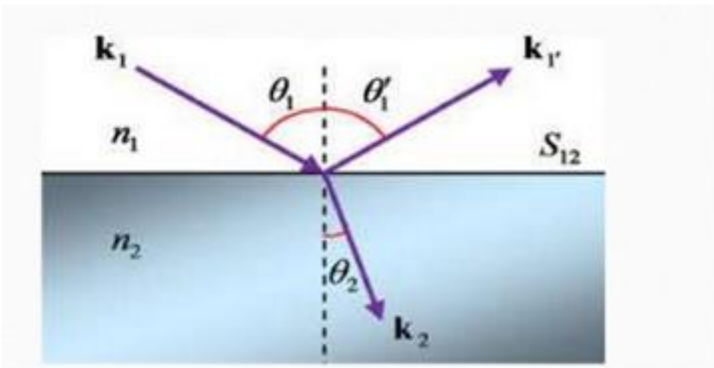


Figure 4:- Refraction of wave

As with light, refraction takes place when electromagnetic waves pass from one propagating medium to a medium having a different density. This situation causes the wavefront to acquire a new direction in the second medium and is brought about by a change in wave velocity. The simplest case of refraction, concerning two media with a plane, sharply defined boundary between them, is shown in Figure 4.

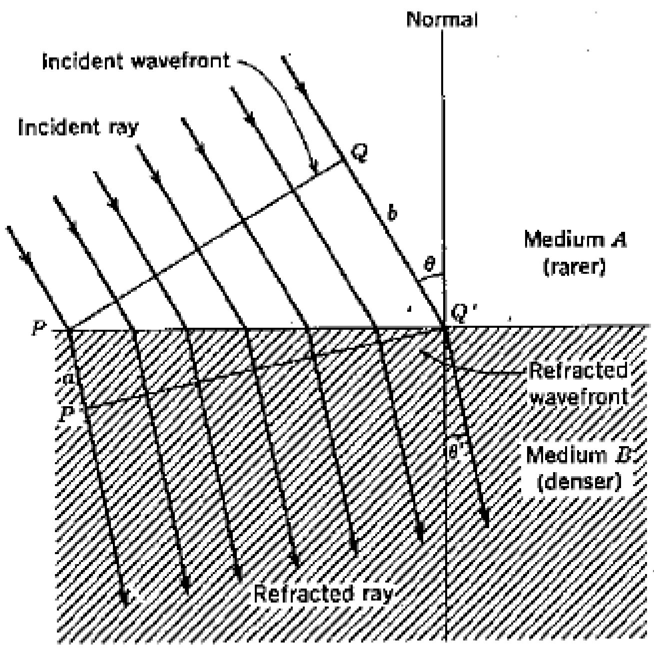


Figure 5:- Refraction at a plane, sharply defined boundary

Consider the situation in Figure 5, in which a wave passes from medium A to the denser medium B, and the incident rays strike the boundary at some angle other than 90°. Wave front P-Q is shown at the instant when it is about to penetrate the denser medium, and wave front P' -Q' is shown just as the wave has finished entering the second medium. Meanwhile, ray b has travelled entirely in the rarer medium, and has covered the distance Q-Q', proportional to its velocity in this medium. In the same time ray a, which travelled entirely in the denser medium, has covered the distance P-P'. This is shorter than Q-Q' because of the lower wave velocity in the denser medium. The in-between rays have traveled partly in each medium and covered total distances as shown; the wave front has been rotated.

Q.5 Explain in brief the following terms related to sky wave propagation :—

(i) Virtual height

(ii) Critical Frequency.

(iii) Maximum Usable Frequency (MUF).

(iv) Skip distance.

Answer:
(i) Virtual height

The virtual height of an ionospheric layer is best understood with the aid of Fig. This figure shows that as the wave is refracted, it is bent down gradually rather than sharply. However, below the ionized layer, the incident and refracted rays follow paths that are exactly the same as they would have been if reflection had taken place from .a surface located at a greater height, called the virtual height of this layer. If the virtual height of a layer is known, it is then quite simple to calculate the angle of incidence required for the wave to return to ground at a selected spot.

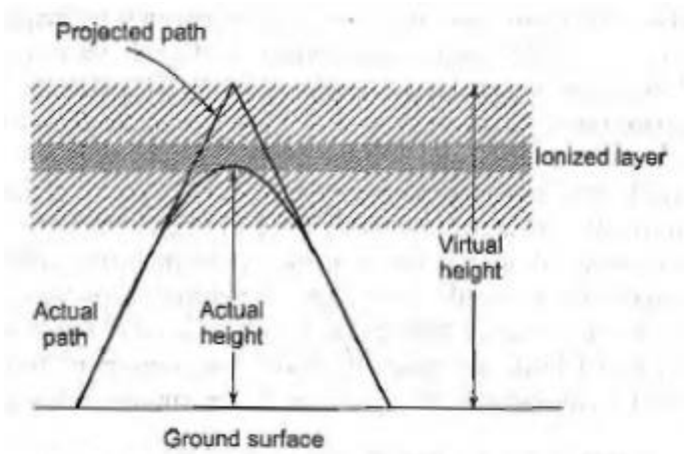


Figure 6 -Actual and virtual heights of ionized layer.

(ii) Critical Frequency.

The critical frequency for a given layer is the highest frequency that will be returned down to earth by that layer after having been beamed straight up at it. When the angle of incidence is nonnal the name given to this maximum frequency is critical frequency its value in practice ranges from 5 to 12 MHz for the F2 layer.

(iii) Maximum Usable Frequency (MUF).

The maximum usable frequency, or *MUF,* is also a limiting frequency, but this time for some specific angle of incidence other than the normal. In fact, if the angle of incidence (between the incident ray and the normal) is θ, it follows that

$$\text{MUF}=\frac{Critical\ frequency}{\cos\theta}$$

(iv) Skip Distance

It is defined as the minimum distance from the transmitter at which a sky wave of given frequency is returned to earth by the ionosphere or the minimum distance from the transmitter to a point where sky wave of a given frequency is first received. The minimum distance within a sky wave of given frequency fails to be reflected back distance.

Q.6 In connection with space-wave propagation, what is the radio horizon?

Answer:

Space waves generally behave with merciful simplicity. They travel in (more or less) straight lines! However, since they depend on line-of-sight conditions, space waves are limited in their propagation by the curvature of the earth, except in very unusual circumstances. Thus they propagate very much like electromagnetic waves in free space, as discussed in Section 8-1.1. Such a mode of behavior is forced on them because their wavelengths are too short for reflection from the ionosphere, and because the ground wave disappears very close to the transmitter, owing to tilt.

Radio horizon: -The radio horizon for space waves is about four-thirds as far as the optical horizon. This beneficial effect is caused by the varying density of the atmosphere, and because of diffraction around the curvature of the earth. The radio horizon of an antenna is given, with good approximation, by the empirical formula

$$dt = 4\sqrt{ht}$$

Where dt = distance from transmitting antenna, km

ht = height of transmitting antenna above ground, m

The same formula naturally applies to the receiving antenna. Thus the total distance will be given by addition, as shown in Figure 7, and by the empirical formula

$$d = dt + dr$$

$$= 4\sqrt{ht} + 4\sqrt{hr}$$

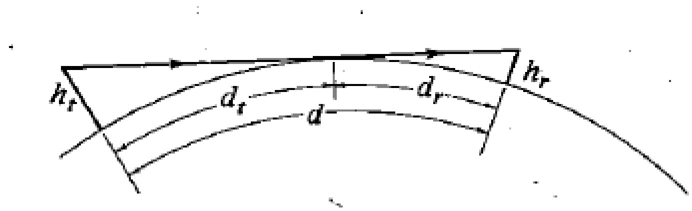


Figure 7 - Radio horizon for space waves.

Q.7 What do you mean by ionosphere? List its different layers along with their salient properties.

Answer:

The waves, which are transmitted from the transmitter antenna, are reflected from the ionosphere. It consists of several layers of charged particles ranging in altitude from 30-250 miles above the surface of the earth. Such travel of the wave from the transmitter to the ionosphere and from there to the receiver on Earth is known as Sky Wave Propagation. The ionosphere is the ionized layer around the Earth's atmosphere, which is suitable for sky wave propagation.

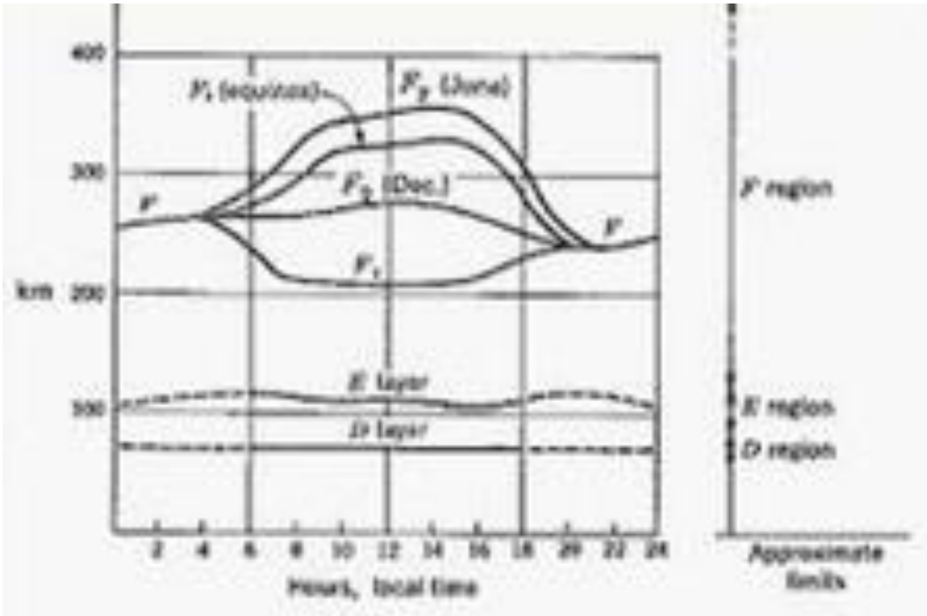


Figure 8:-Ionospheric layers and their regular variations

D Region:-

•Region below E layer

•Responsible for attenuation of high frequency in day time.
•Lower most region of Ionosphere.

•Height range of 50 km to 90 km.

•Present at day

•Disappears at night

•Critical frequency is 100 KHz.

•Electron density is ranging from 1014 to 1016 per cm3
•Reflecting –VLF Signals.

•Absorbing –HF signals.

•Ionization increases with solar activity.

Normal E-Region:-

•Layer occurs during day light hours.

•Electron density

•Day –105 to 4.5x105 per cm3

•Night –5x103 to 104 per cm3

•Max electron density at noon & summer (at 110 km)

•Useful in long distance communication in day hours

•Height range from 90 to 140 km.

•Critical frequency –3 MHz to 5 MHz.

•Also known as Kennelly Heaviside layer.

E Region:-

•Height-140 km to 400 km from earth surface.

•Top most layer & highly ionized layer.

•Remains ionized irrespective of hours.

•Critical frequency is 5MHz to 7MHz.

•During day, F layer is found to split up into two layers called,

•F1 Layer

•F2 Layer

Q.8 Explain what is meant by the terms isotropic source and isotropic medium.

Answer:

There are no interference or obstacles present in free space, electromagnetic waves will spread uniformly in all directions from a point source. The wave front is thus spherical, as shown in cross section in Figure . At the distance corresponding to the length of ray *P*, the wave has a certain phase. It may have left the source at an instant when its voltage and current were maximum in the circuit feeding the source, i.e., at an instant of maximum electric and magnetic field vectors. This is virtually the definition of a wave front; it is the plane joining all points of identical phase. Here, of course, it is spherical If the length of ray *Q* is exactly twice that of ray *P*, then the area of the new sphere will be exactly *four times* the area of the sphere with radius *P*. It is seen that the total power output of the source has spread itself over four times the area when its distance from the source has doubled. If *power density* is defined as radiated power per unit area, it follows that power density is reduced to one-quarter of its value when distance from the source has doubled.

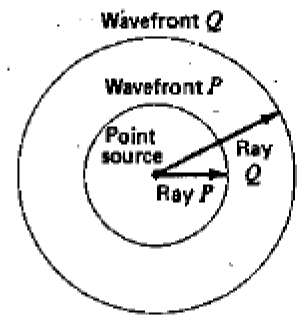


Figure 9:- Spherical wavefronts

It is seen that power density is inversely proportional to the square of the distance from the source. This is the *inverse-square law*, which applies universally to all forms of radiation in free space. Stating this mathematically, we have

$$Pd = \frac{Pt}{4\pi r^2} \quad (1)$$

Where *Pd* = power density at a distance *r* from an isotropic source
Pt, = transmitted power

An *isotropic* source is one that radiates uniformly in all directions in space. Although no practical source has this property, the concept of the isotropic radiator is very useful and frequently employed. As a matter of interest, it may be shown quite simply that .the inverse-square law applies also when the source is not isotropic, and students are invited to demonstrate this for themselves. However, for wave fronts to be spherical, the velocity of radiation must be constant at all points (as it is in free space). A propagation medium in which this is true is also called isotropic medium.

Q.9 At 20 km in free space from a point source, the power density is 200□W/m2.What is the power density 25 km away from this source?

Answer:

-We know that

$$P_{Di} = P_T / 4\pi d^2$$

P_{Di} – power density at a distance ‘d’ from an isotropic source

$P_T = P_{Di} \times 4\pi d^2$

$= 200\times10^{-6}\times4\times3.14\times(20000)^2$

$= 1004800\ \mu W/m^2$

$= 100.48\times10^4\ \mu W/m^2$

So power density 25km away from source is given by

$$P_{Di} = \frac{100.48\times10^4}{4\times3.14\times(⟦25000)⟧^2}$$

$= 128\ \times10^{-6}\ W/m^2$

P_{Di} at 25km is $= 128\ \mu W/m^2$

Q.10 A 150-m antenna, transmitting at 1.2 MHz (and therefore by ground wave) has an antenna current of 8A.What voltage is received by a receiving antenna 40 km away, with a height of 2 m? Note that this is typical MF broadcasting situation.

Answer:

Wavelength is given by

$$\lambda = \frac{c}{f}$$

where c= velocity of light through free space = 3×10^8 m/s

$$\lambda = \frac{c}{f} = \frac{3 \times 10^8}{1250 \times 10^3} = 240\text{m}$$

from equation

$$E_r = \frac{120\pi \text{ hT hR } I}{\lambda d}$$

$$= \frac{120 \times 3.14 \times 150 \times 5 \times 10}{240 \times 100 \times 10^3}$$

$$= 0.11775 \text{ V}$$

$$= 117.75 \text{ mV}$$

$$E_r = 117.75 \text{ mV}$$

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