MUMBAI UNIVERSITY

Principles Of Communication Engineering

SEMESTER 4 - DECEMBER 2018 – Choice Based

Q.1 Solve any Four. **[20M]**

a) Modulation index for AM should be less than one. Justify/Contadict.

i) The ratio of maximum amplitude of modulating signal to the maximum amplitude of carrier signal is called as modulation index.

i) Amplitude modulation is the process of varying amplitude of carrier signal iin accordance with instantaneous value of modulating sihgnal.

ii) Modulation index for amplitude modulation, AM, indicates the amount by which the modulated carrier varies around its static unmodulated level.

iii) Modulation index (m) = Vm/Vc Where, Vc is the carrier peak amplitude , Vm is the modulating signal amplitude amplitude.Depending upon voltages the modulation index is divided into three parts i.e

a) m<1 Under modulation
b) m>1 Over modulation
c) m=1 Perfect modulation

iv) Its value is kept less than 1 to avoid overmodulation which leads to distortions in the modulated signal and makes it very hard to demodulate and extract the modulating signal.

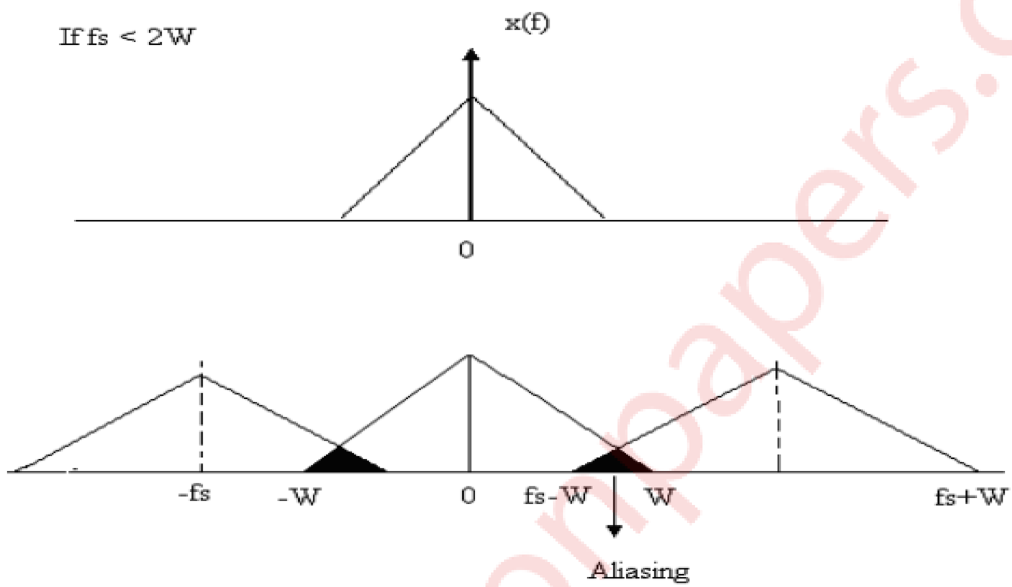
v) Over modulation also leads to phase reversal at zero crossings again leads to distortion in demodulation of signals. Hence modulation index for AM should be less than 1.

b) What is aliasing ? How it can be prevented ?

Ans : i) When the high frequency interferes with low frequency and appears as low frequency the effect is called as aliasing effect.

ii) Effects of aliasing –

1. Distortion is generated due to interfering frequencies.
2. Information in original signal is lost and cannot be recovered.



iii) Using "PRE–ALIAS" filter or "ANTIALIASING" filter . It is a band limiting LPF with cutoff frequency at fc = W hence it will limit the signal x(t) before sampling takes place.

iv) Let fs>2W. Even if x(t) is not strictly bandlimited the spectrum would not overlap. The gap between the neighboring x(f) spectrums is called as the "GUARD BAND" as it generates against error.

c) Why AGC is required in radio receivers ?

Ans : i) AGC is a departure from linearity in AM radio receivers.

ii) The AGC circuit keeps the receiver's output level from fluctuating too much by detecting the overall strength of the signal and automatically adjusting the gain of the receiver to maintain the output level within an acceptable range

iii) In simple AGC is a system which will change the overall gain of the receiver automatically, this is done in order to keep the receiver output constant even when the signal strength at the input of the receiver is changing.

iv) In AGC system a dc voltage (AGC bias) is derived from the detector. This AGC bias is thus proportional to the strength of received signal.

v) AGC bias is applied to a selected number of RF and IF amplifiers and mixer stage.

vi) The transconductance and hence the gain of the devices connected in these stages is dependent on the applied AGC bias.

d) Justify , why FM is more immune to noise ?

Ans : i) Variation of the frequency of the carrier signal with constant amplitude is called frequency modulation. Frequency variation is directly proportional to the amplitude of the modulating signal

ii)Noise effects are more on amplitude than on frequency.

iii)As the amplitude remains constant, total average power is equal to that of the unmodulated carrier power. So, the power = $Ac^2/2$. Although Am increases the bandwidth, it does not affect power.

iv)Therefore, the transmission power for FM is less compared to AM at the expense of higher bandwidth.

v)FM is more immune to noise than AM, since the power of transmission is independent of the modulation index

e) Define noise figure and noise factor.

Ans : i) Noise figure (NF) and noise factor (F) are measures of degradation of the signal-to-noise ratio (SNR), caused by components in a signal chain.

ii) It is a number by which the performance of an amplifier or a radio receiver can be specified, with lower values indicating better performance.

ii)iThe noise factor is defined as the ratio of the output noise power of a device to the portion thereof attributable to thermal noise in the input termination at standard noise temperature T0 (usually 290 K).

iv)The noise factor is thus the ratio of actual output noise to that which would remain if the device itself did not introduce noise, or the ratio of input SNR to output SNR.

v) Formulas :

$$\text{Noise Factor} = \frac{(S/N)in}{(S/N)out} \qquad \& \qquad \text{Noise Figure} = 10\log(\text{Noise Factor})$$

Q.2 a) State and prove sampling theorem for low pass bandlimited signals. [10M]

Ans : i) A continuous time signal can be represented in its samples and can be recovered back when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal. i. e.

$$Fs \geq 2.fm$$

ii) Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to fm Hz i.e. the spectrum of x(t) is zero for |ω|>ωm.

iii) Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period Ts. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) .

iv) The process of sampling can be explained by the following mathematical expression:

Sampled signal $\quad y(t)=x(t).\delta(t) \quad$ ......(1)

v) The trigonometric Fourier series representation of δδ(t) is given by

$$\delta(t)=ao + \sum_{n=1}^{\infty}(\text{ancos}n\omega st + \text{bnsin}n\omega st) \qquad ......(2)$$

Where $\quad ao=1/Ts \int_{-T/2}^{T/2} \delta(t)dt = 1Ts\delta(0) = 1/Ts$

$an=1/Ts \int_{-T/2}^{T/2} \delta(t) \cos(n\omega_s)\, dt = 2/T$

$bn=1/Ts \int_{-T/2}^{T/2} \delta(t)\sin(n\omega_s)dt = 0$

$$\delta(t)=1/T + \sum_{n=1}^{\infty} \left(\frac{2}{T}\right).\cos(n\omega_s t)$$

But we know that ,

$$y(t)=x(t).\delta(t)$$

$$\therefore y(t) = x(t).[\,1/T + \sum_{n=1}^{\infty} \left(\frac{2}{T}\right).\cos(n\omega_s t)]$$

$$\therefore y(t) = \frac{1}{T}[\,x(t) + \sum_{n=1}^{\infty}(2.x(t)).\cos(n\omega_s t)]$$

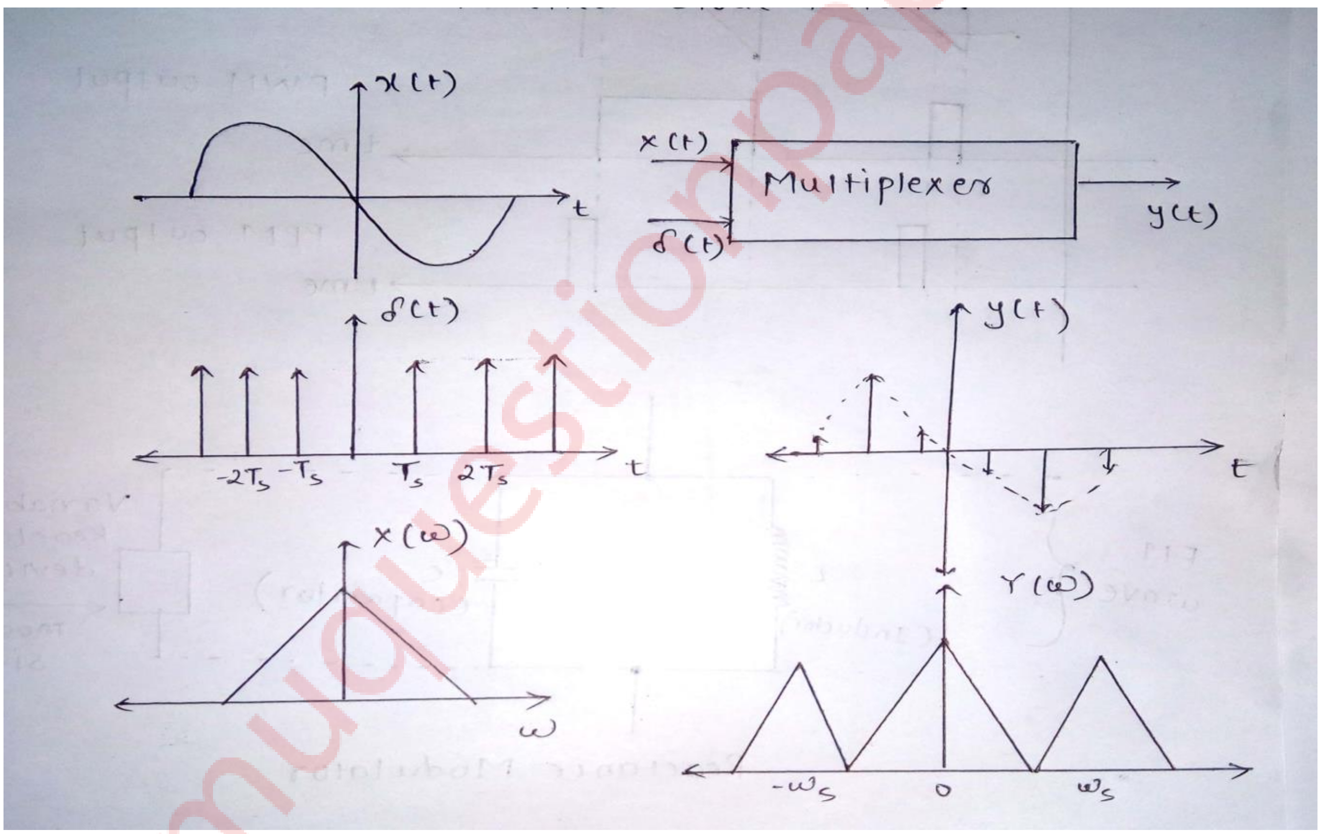
$$\therefore y(t) = \frac{1}{T}[\,x(t) + 2.\cos(\omega_s t).x(t) + 2.\cos(2\omega_s t).x(t) + 2.\cos(3\omega_s t).x(t) + \ldots]$$

Taking fourier transform ,

$$\therefore Y(w) = \frac{1}{T}[\,X(w) + X(w-ws) + X(w+ws) + \cdots\,]$$

$$\boxed{\therefore Y(w) = \frac{1}{T}[\,\sum_{-\infty}^{\infty} X(w - n.ws)]}$$

Where n=0,±1,±2,...



b) One input to AM modulation is 800 KHz carrier with an amplitude of 10 Vp.The second input is 10 KHz modulating signal that is of sufficient amplitude to cause a change in o/p wave of $\pm$ 5.5 Vp.Determine .

i) Upper and lower side frequency

ii) Modulation co-efficient and percent modulation

iii) Draw o/p frequency spectrum

iv) Draw modulated wave showing maxima and minima of waveforms [10M]

Ans : Given : Carrier frequency = fc = 800 KHz

Amplitude of carrier = Ac = 10 Vp

Modulating signal frequency = fm = 10 KHz

Amplitude of modulating signal = Am = $\pm$ 5.5 Vp

To Find : i) Upper and lower side frequency

ii) Modulation co-efficient and percent modulation

iii) Frequency spectrum

iv) modulated wave showing maxima and minima of waveforms

Solution :

i) Frequency of carrier = fc = 800 KHz and frequency of modulating wave is 10 KHz.

$\therefore$ Upper side band frequency = f(USB) = fc + fm = 810 KHz

$\therefore$ Lower side band frequency = f(LSB) = fc - fm = 790 KHz

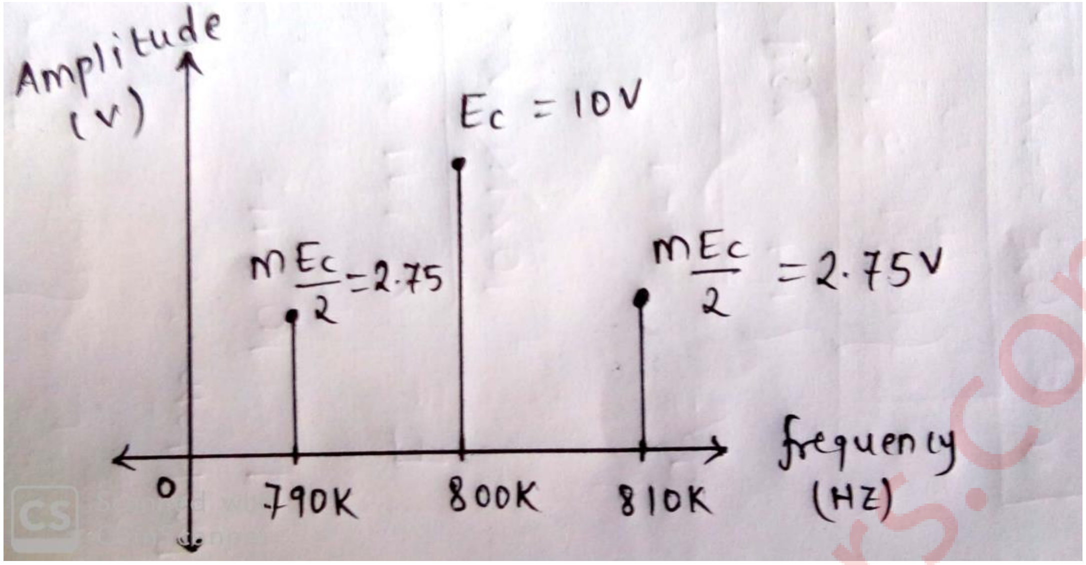
ii) Modulation co-efficient is given by , m= $\frac{Am}{Ac}$

$$\therefore m = \frac{5.5}{10} = 0.55$$

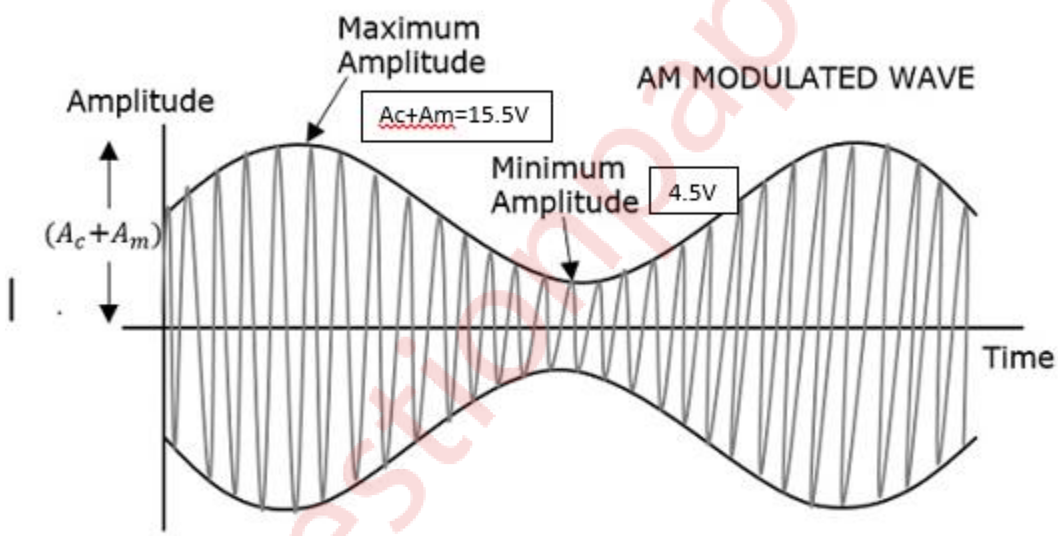
And modulation percentage = m*100 = 55 %

iii) Frequency spectrum :

Amplitude of sidebands = $\frac{m.Ec}{2} = \frac{0.55(10)}{2} = 2.75$



iv) Modulated wave showing maxima and minima :



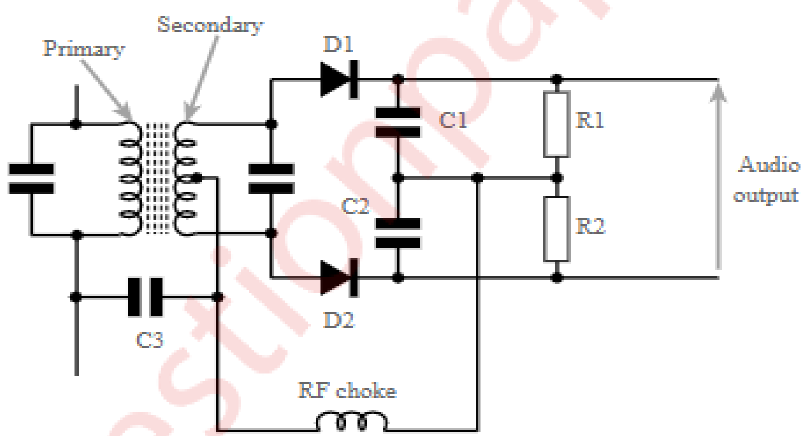
Q.3 a) Explain the operation of Foster-Seeley discriminator with the help of circuit diagram and phasor diagram. [10M]

Ans : i) The FOSTER-SEELEY DISCRIMINATOR is also known as the PHASE-SHIFT DISCRIMINATOR.

ii) 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.

iii) This voltage varies in both amplitude and polarity as the input signal varies in frequency. The output voltage is 0 when the input frequency is equal to the carrier frequency (fr).

iv) 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.



FM Foster Seeley discriminator / detector circuit

v) In many respects the Foster Seeley FM demodulator resembles the circuit of a full wave bridge rectifier - the format that uses a centre tapped transformer, but additional components are added to give it a frequency sensitive aspect.

vi) The basic operation of the circuit can be explained by looking at the instances when the instantaneous input equals the carrier frequency, the two halves of the tuned transformer circuit produce the same rectified voltage and the output is zero.

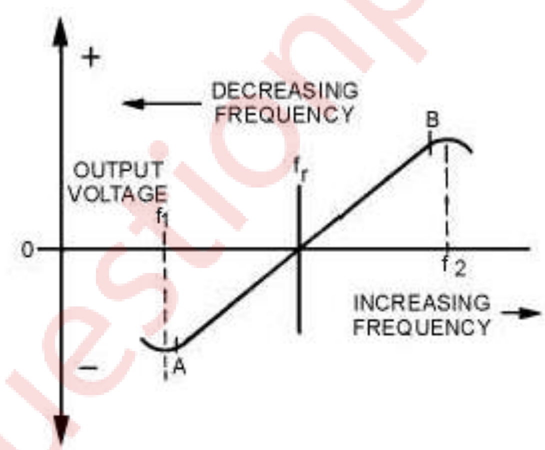
vii) If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.

viii) Looking in more detail at the circuit, the Foster-Seeley circuit operates using a phase difference between signals. To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90° out of phase.

ix) When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors.

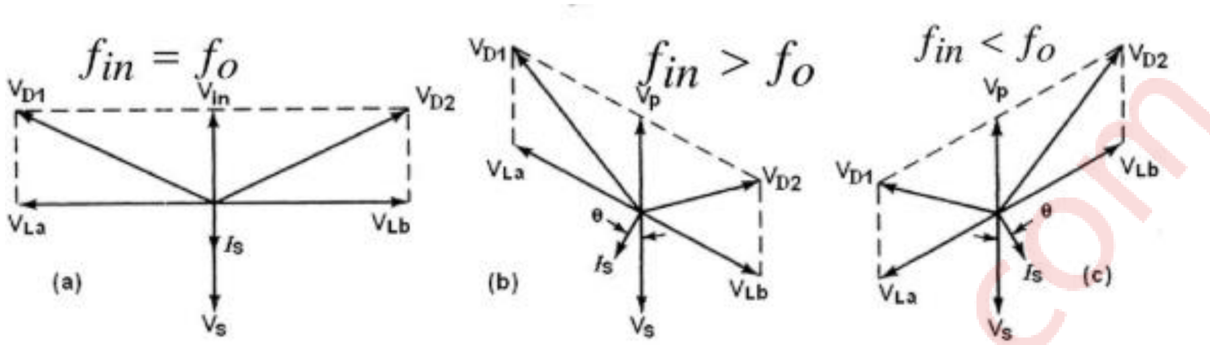
x)These voltages cancel each one another out at the output so that no voltage is present. As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.

xi) The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.



xii) Phasor diagrams :

---Circuit at resonance : The current flowing in the tank causes voltage drops across each half of the balanced secondary winding of transformer T1. These voltage drops are of equal amplitude and opposite 3-14 polarity with respect to the center tap of the winding. Because the winding is inductive, the voltage across it is 90 degrees out of phase with the current through it. Because of the center-tap arrangement, the voltages at each end of the secondary winding of T1 are 180 degrees out of phase and are shown as e1 and e2 on the vector diagram.



--- Circuit below resonance: When the input frequency is lower than the center frequency, the current and voltage phase relationships change. When the tuned circuit is operated at a frequency lower than resonance, the capacitive reactance increases and the inductive reactance decreases. Below resonance the tank acts like a capacitor and the secondary current leads primary tank voltage

--- Circuit above resonance : A phase shift occurs when an input frequency higher than the center frequency is applied to the discriminator circuit and the current and voltage phase relationships change. When a series-tuned circuit operates at a frequency above resonance, the inductive reactance of the coil increases and the capacitive reactance of the capacitor decreases. Above resonance the tank circuit acts like an inductor. Secondary current lags the primary tank voltage.

b) Explain the working of stabilized reactance modulator with suitable diagram. [10M]

Ans : Different methods of FM generation :

[A] Direct Method :

-- Reactance modulator

-- Varactor diode modulator

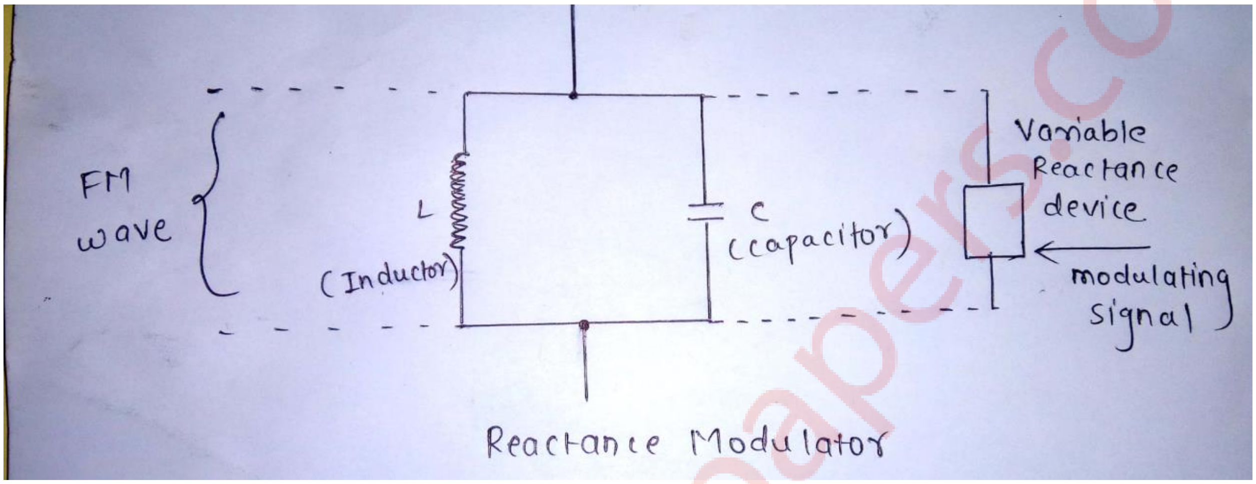
[B] Indirect Method

-- Armstrong method

Reactance Modulator : -

i)A reactance modulator changes the frequency of the tank circuit of the oscillator by changing its reactance.

ii) This is accomplished by a combination of a resistor, a condenser, and a vacuum tube (the modulator) connected across the tank circuit of the oscillator and so adjusted as to act as a variable inductance or capacitance.



iii)The net result is to change the resonant frequency of the LC circuit by amounts proportional to the instantaneous a.f. voltages applied to the grid of the modulator tube, without changing the resistance of the LC circuit or the amplitude of the oscillations.

iv)The voltages supplied to both the modulator and oscillator must be carefully stabilized to prevent undesired frequency changes.

v)The speech does not have to deliver any power and need supply only a small output voltage, say 10 or 15 volts.

vi)A diode, R-C coupled, will be sufficient even with a sensitive microphone and a high-powered oscillator. The frequency change of LC per volt change on the a.f. grid of the modulator tube will be greater when C1,is made smaller. The blocking condenser C2 has a comparatively high value, and hence offers but small reactance to r.f. currents.

vii)The radio-frequency voltages which are developed across the tank in the oscillator circuit also appear across the RC1 circuit and across the parallel 6L7 modulator tube.

viii) The resistance r has been replaced by the internal resistance of the modulator tube.

ix) The voltage drop across C1 is 90° out of phase with the tank voltage. It is applied to the control grid of the 6L7 whose r.f. plate current responds in the same phase. Thus this current is made to lag 90° behind the tank voltage.

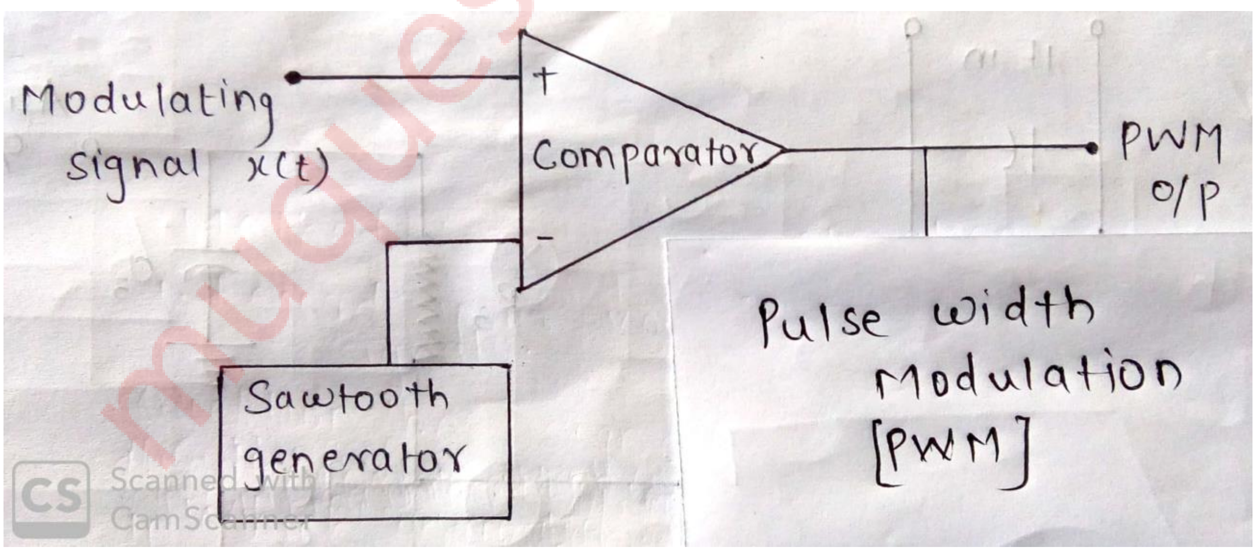
x) The r.f. plate current flows through the tank circuit and, combined with the current therein, is equivalent to a new current whose phase differs from the normal value just as though an additional reactance (not resistance) had been connected in with L and C. This, of course, changes the frequency of the LC circuit and hence of the transmitter. When a.f. is fed into the modulator tube, it causes proportionate changes in the r.f. plate current and hence in the equivalent reactance of the LC circuit.

Q.4 a) With the help of neat diagram and waveforms explain generation and demodulation of PWM . [10M]

Ans : i) In modulated signal (PWM), the amplitude and position of pulses are constant while the width (or duration) of pulses varies proportionally with the amplitude of analogical useful signal. Carrier signal is from a clock.

ii) Pulse Width Modulating signal can be generated using a Comparator as shown in the figure. Modulating signal forms one of the input to the Comparator and the other input is fed with a non-sinusoidal wave or sawtooth wave.

iii) It operates at carrier frequency. The Comparator compares the two signals and generates a PWM signal as its output waveform.

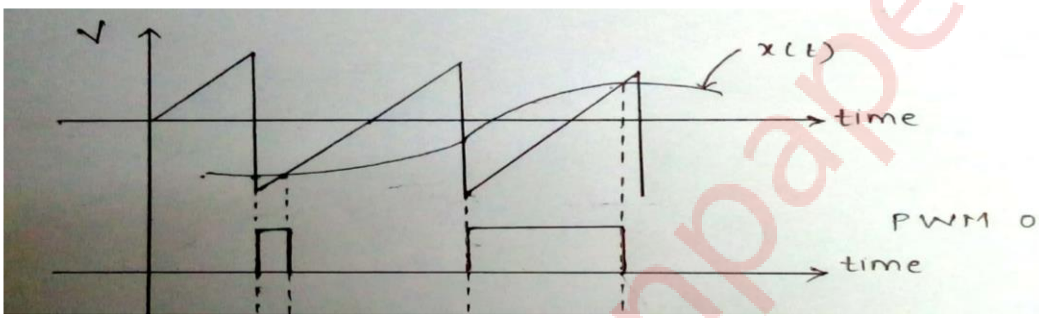


iv) If the value of the Sawtooth triangle signal is more than the modulation signal then the PWM output signal is at High else it's in Low state. Thus, the value of the input signal magnitude determines the comparator output which defines the width of the pulse generated at the output.

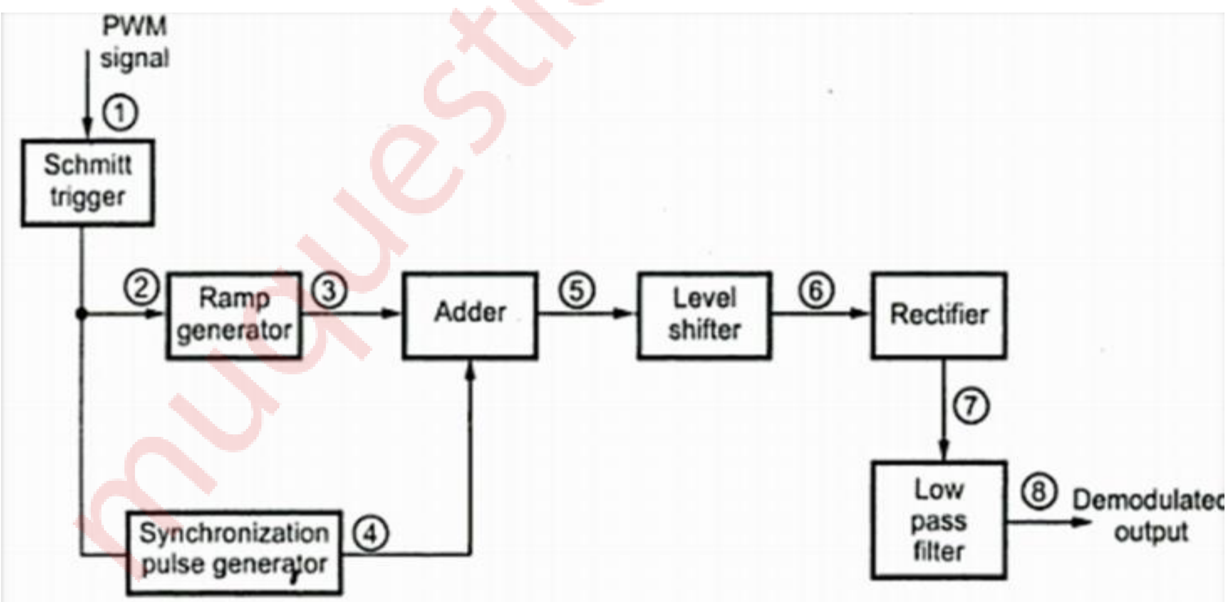
v)Pulse Width Modulation : The amplitude of the pulse is maintained constant but the width of each pulse is varied according to the modulating signal.

vi) Saw tooth generator: – The saw tooth generator is conneted to the inverting terminal of the operational amplifier (Opamp). The Op-amp is used in comparator mode.

vii) Modulating signal: – The modulating signal is given as input to the non inverting terminal of the Op-amp as comparator.



Demodulation process :

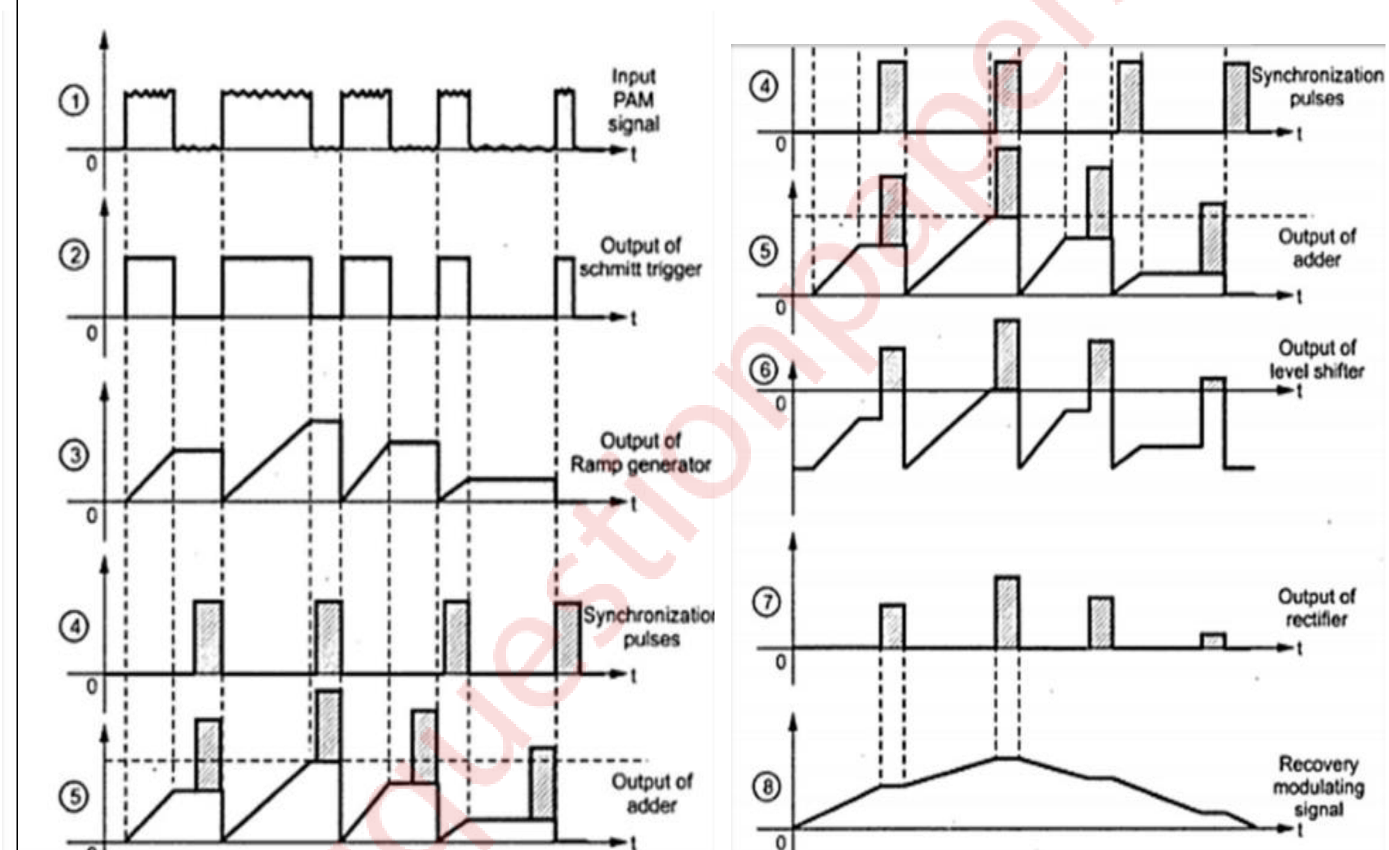


viii) Schmitt Trigger – Removes noise in the PWM waveform. Regenerated PWM is applied to ramp generator and synchronous pulse generator. Ramp Generator – Produces ramps such that height of ramps are proportional to the widths of PWM pulses. Maximum ramp voltage is retained till next pulse.

ix)Synchronous pulse generator – It Produces reference pulses with constant amplitude and pulse width which are delayed by a specific amount. Adder – Delayed reference pulses and output of ramp generator is added.

x) Level Shifter is used to shift the waveform to a specified level. Rectifier --Negative offset waveforms are clipped. Low Pass Filter recovers the modulating signal.

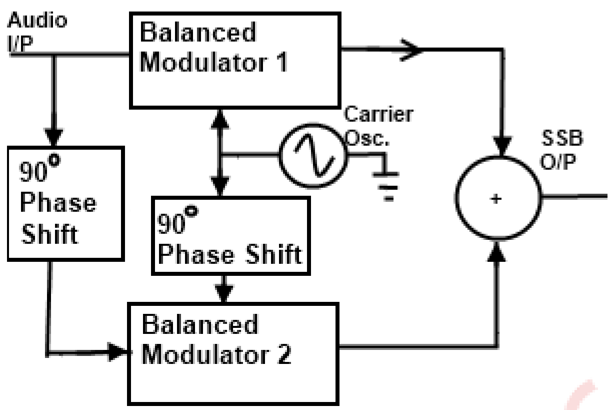
xi) Waveforms :



b) Explain phase shift method for suppression of unwanted carrier with neat block diagram. [10M]

Ans : i) Phase shift method of SSB generation in which carrier and one of the side bands is suppressed.

ii) Block diagram for the phase shift method of SSB generation is shown below :



SSB Single Sideband Transmission Phase Shift Method

iii) This system is used for the suppression of lower sideband .This system uses two balanced modulators M1 and M2 and two 90o phase shifting networks .

iv) The message signal x(t) is applied to the product modulator M1 and through a 90o phase shifter to the product modulator M2 .

v) Hence, we get the Hilbert transform $\hat{x}(t)$ at the output of the wideband 90 deg. phase shifter .The output of carrier oscillator is applied as it is to modulator M1 whereas it s passed through a 90o phase shifter and applied to the modulator M2 .

$$Output\ of\ M_1 = x(t) \times V_c cos(2\pi f_c t)$$

$$and\quad Output\ of\ M_2 = \hat{x}(t) \times V_c sin(2\pi f_c t)$$

vi) The outputs of M1 and M2 are applied to an adder .

$$Adder\ \ Output\ = x(t) \times V_c cos(2\pi f_c t) + \hat{x}(t) \times V_c sin(2\pi f_c t)$$

$$Or\quad Adder\ Output = V_c[x(t) \times cos(2\pi f_c t) + \hat{x}(t) \times sin(2\pi f_c t)]$$

vii) This expression represents the SSB signal with only LSB i.e. it rejects the USB .Adder polarities for the in-phase is positive and for the quadrature paths is negative .

Suppression of the upper sideband

viii)We can suppress the LSB and generate the SSB signal consisting of the USB by arranging the blocks.Here, the modulating and the carrier signals are applied to the upper balanced modulator directly (without any phase shift) .Whereas, both these signals are 90o phase shifted and then applied to the lower balanced modulator .

ix) Advantages of Phase Shift Method :

---The advantages of the phase shift method are as under :

---It can generate the SSB signal at any frequency, so the frequency up converter stage is not required .

---It can use the low audio frequencies as modulating signal .(In filter method, this is not possible) .

---It is easy to switch from one sideband to other .

x) Drawbacks of Phase Shift Method

--The main drawback is that the design of the 90o phase shfting network for the modulating signal is extremely critical .

---This network has to provide a correct phase shift of 90o at all the modulating frequencies which is practically difficult to achieve .

Q.5 a) Explain the following with reference to AM receiver [10M]

i) Double spotting

Ans : i)Double-spotting" is a term that means that the wanted station is tuned in at two spots on the dial. These spots would be just 60kHz apart if an IF of 30kHz is used.

ii) In a superheterodyne receiver, the local oscillator frequency is offset from the wanted station by the frequency of the IF amplifierFor example the wanted station is on 800kHz and the IF is 30kHz. This means that the local oscillator (which is usually higher in frequency than the tuned station) will be on 800 + 30 = 830kHz.

iii)However, if the selectivity of the RF stage is quite poor, a station on 860kHz will also give a 30kHz IF output when mixed with the local oscillator (on 830kHz).

iv) As a result, two stations - one on 800kHz and one on 860kHz - will be received at the same time. If the receiver is now tuned to 740kHz the oscillator will be on 770kHz. However, this will also give a 30kHz IF output from the 800kHz station. This means that the 800kHz station is heard at both the 800kHz and 740kHz positions on the dial.

ii) Three point tracking

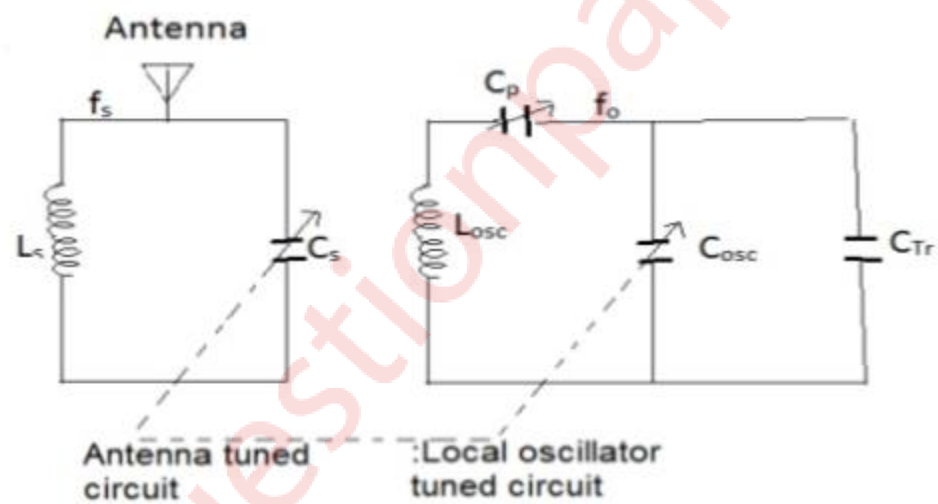
Ans :i) Three point tracking is a tracking error reduction technique in which the preselector and local oscillator have trimmed capacitors added in parallel to the primary tuning capacitor and the local oscillator has an additional padder capacitor in series with the tuning coil.

ii) Three-point tracking is achieved with the padder capacitor.

iii) The frequencies of correct tracking i.e at which zero tacking error exists at normally 600 KHz , 1500 KHz and geometric mean of both i.e 950 KHz.

iv) It is possible to keep tracking error below 3 KHz.

v) Three point tracking circuits :



iii) Image frequency rejection ratio

Ans : i) Image frequency rejection ratio is numerical measure of ability of pre selector to reject image frequency.

ii) Mathematically it is represented by ,

$$IFRR = \sqrt{1 + Q^2.\rho^2}$$

Where , $\rho = \left(\frac{f_{im}}{f_{RF}}\right) - (f_{RF}/f_{im})$

Q = quality factor of preselector

iii) Generally, local oscillator frequency is equal to the sum of signal frequency and intermediate frequency.

iv) When f s and f o are mixed, the difference frequency is equal to f i which is the only one passed and amplified by the IF stage.

v) This IF signal will also be amplified by the IF stage and provide interference. This has the effect of two stations being received simultaneously. The term fsi is called the image frequency and is defined as the signal frequency plus twice the intermediate frequency. fsi = fs + 2fi.

iv) Fidelity

Ans : i)Ability of a communication system to produce an exact replica of the original source information at the output of the receiver is called fidelity.

ii)Radio receiver should have high fidelity or accuracy.

iii)For high fidelity, it is essential to have a flat frequency response over a wide range of audio frequencies.

iv) Radio receivers are designed using preselector ,preselector gets signal from antenna .Preselector is used to select band of frequency which contains information.Here selectivity is major parameters and ability of system to produce exact replica of such signal is fidelity.

b) Explain Indirect FM transmitter with suitable diagram. [10M]

Ans :i) Indirect FM transmitters produce an output waveform in which the phase deviation is directly proportional to the modulating signal.

ii) Consequently, the carrier oscillator is not directly deviated. As a result, the stability of the oscillators can be achieved without using an AFC circuit. Armstrong transmitter is the most widely used indirect FM transmitter.

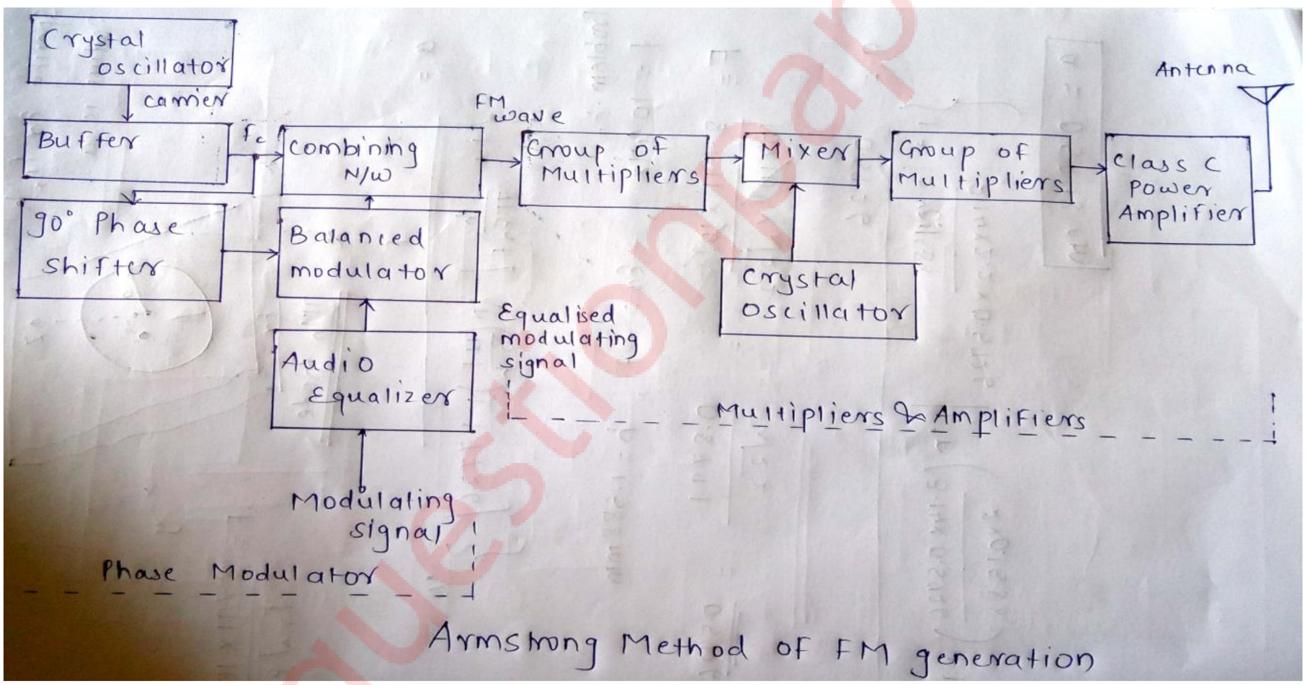
iii) Low frequency sub-carrier fc is phase shifted 90˚ and fed to a balanced modulator. It is mixed with the modulating signal fm .

iv)The output from the balanced modulator is DSBSC wave that is combined with the original carrier in a combining network to produce a low index, phase-modulated waveform

v)With Armstrong transmitter, the phase of the carrier is directly modulated in the combining network producing indirect frequency modulation.

vi) The magnitude of peak phase deviation (i.e. the modulation index) is directly proportional to the amplitude of the modulating signal but independent of its frequency (m = K.Vm).

vii) The modulation index remains constant for all modulating signal frequencies of given amplitude.



Operation:

viii)The crystal oscillator generates the carrier at low frequency typically at 1MHz. This is applied to the combining network and a 90° phase shifter.

ix)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.

x)The balanced modulator produced two side bands such that their resultant is 90° phase shifted with respect to the unmodulated carrier.

xi)The unmodulated carrier and 90° phase shifted sidebands are added in the combining network.

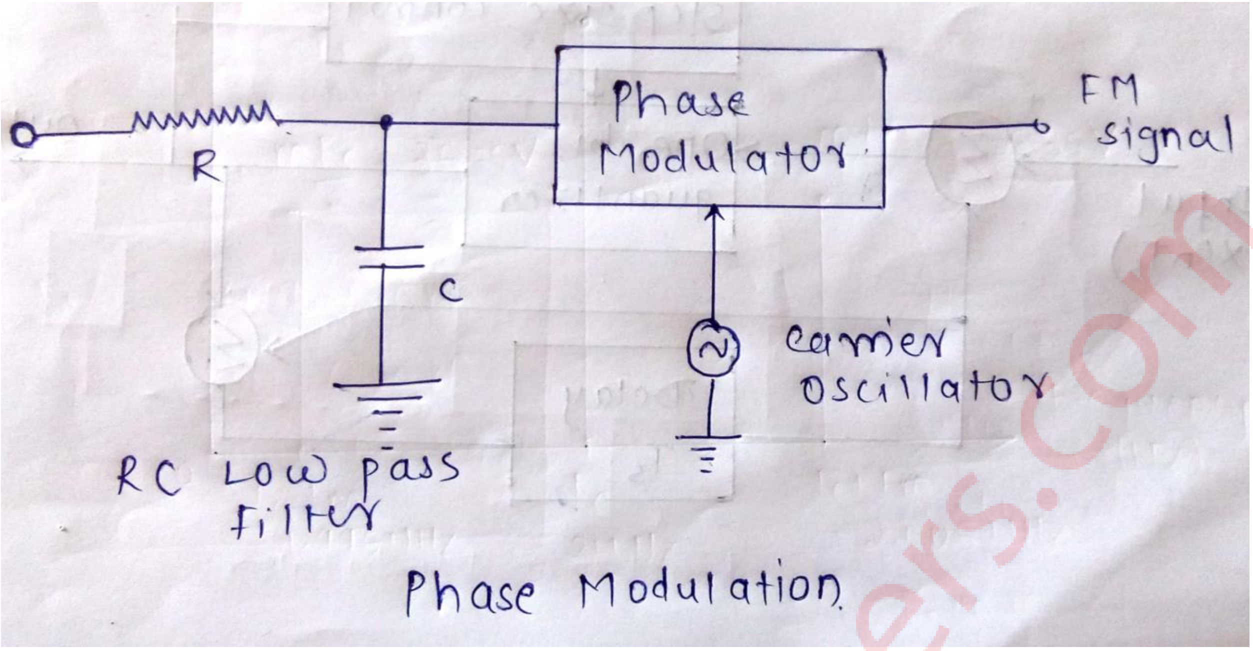
xii)At the output of the combining network we get Fm wave. This wave has a low carrier frequency fc and low value of the modulation index mf.
xiii)The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the fc and mf both are raised to required high values using the second group of multipliers.
xiv)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.
xv)The Armstrong method uses the phase modulation to generate frequency modulation. This method can be understood by dividing it into four parts as follows:

1.Generation of FM from phase modulator:

i)The modulating signal is passed through a low pass RC filter.

ii)The filter output is then applied to a phase modulator along with carrier.

iii)Hence the extra deviation in the carrier fc due to higher modulating frequency is compensated by reducing the amplitude of the high frequency modulating signals.
iv)Hence the frequency deviation at the output of the phase modulator will be effectively proportional only to the modulating voltage and we obtain an FM wave at the output of phase modulator.

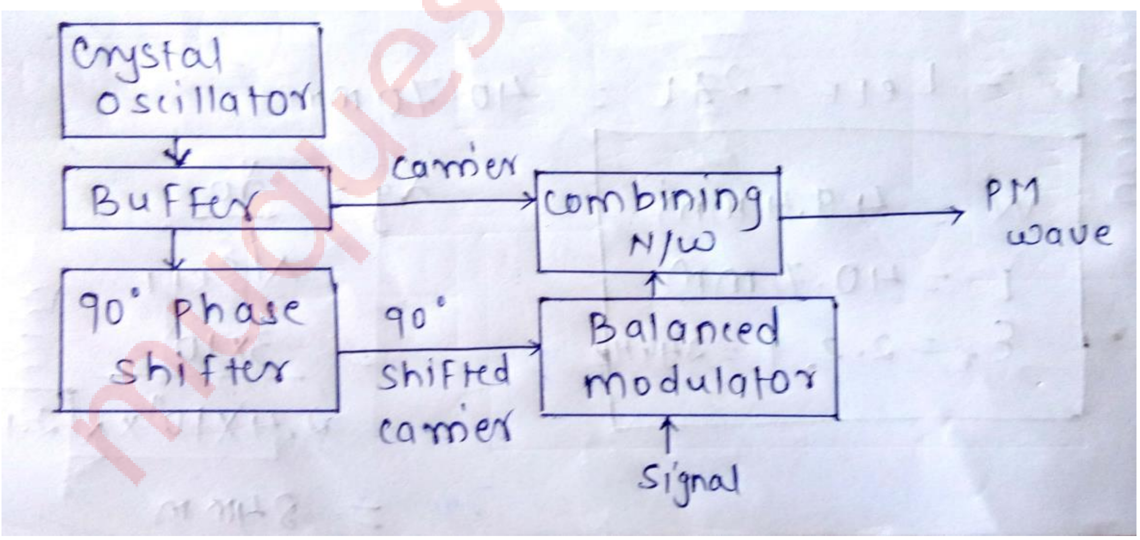


2.Implementation of phase modulator:

The crystal oscillator produces a stable unmodulated carrier which is applied to the “90° phase shifter” as well as the “combining network” through a buffer.

The 90° phase shifter produces a 90° phase shifted carrier. It is then applied to the balanced modulator along with the modulation signal.

At the output of the balanced modulator we get DSBSC signal i.e. AM signal without carrier. This signal consists of only two sidebands with their resultant in phase with their resultant in phase with the 90° phase shifted carrier.



3.Combining parts 1 and 2 to obtain The FM:

Combining the parts 1 and 2 we get the block diagram of the Armstrong method of FM generation

4.Use of frequency multipliers and amplifiers:

The FM signal produced at the output of phase modulator has a low carrier frequency and low modulation index. They are increased to an adequately high value with the help of frequency multipliers and mixer. The power level is raised to the desired level by the amplifier.

Q.6 Write short note on (Any Four) [20M]

a) Vestigial side band transmission (VSB) and its application.

Ans : i)Vestigial sideband (VSB) is a type of amplitude modulation (AM) technique (sometimes called VSB-AM) that encodes data by varying the amplitude of a single carrier frequency .

ii)Portions of one of the redundant sidebands are removed to form a vestigial sideband signal - so-called because a vestige of the sideband remains.

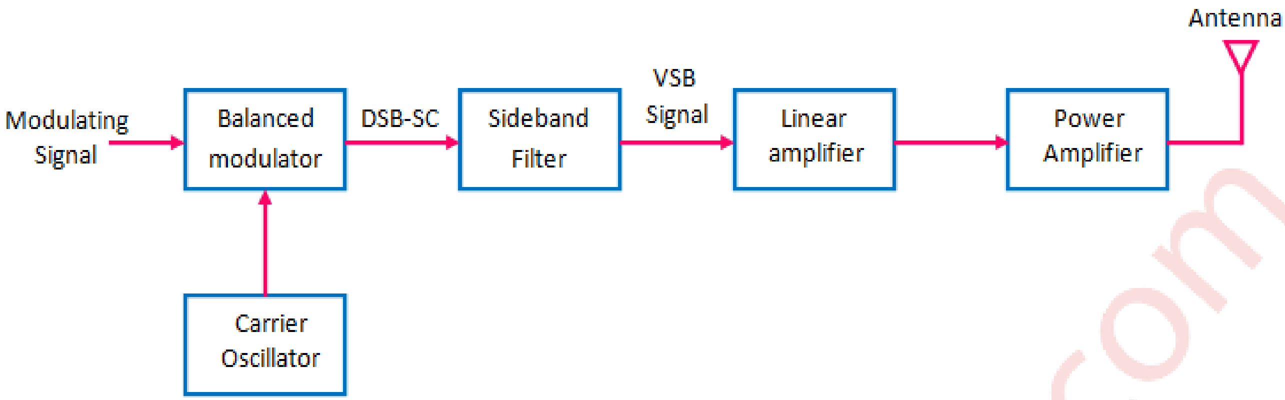
iii)In AM, the carrier itself does not fluctuate in amplitude. Instead, the modulating data appears in the form of signal components at frequencies slightly higher and lower than that of the carrier.

iv)These components are called sidebands . The lower sideband (LSB) appears at frequencies below the carrier frequency; the upper sideband (USB) appears at frequencies above the carrier frequency.

v)The actual information is transmitted in the sidebands, rather than the carrier; both sidebands carry the same information.

vi)Because LSB and USB are essentially mirror images of each other, one can be discarded or used for a second channel or for diagnostic purposes.

vii)VSB transmission is similar to single-sideband (SSB) transmission, in which one of the sidebands is completely removed. In VSB transmission, however, the second sideband is not completely removed, but is filtered to remove all but the desired range of frequencies.



viii) Advantages :-

--Highly efficient.

--Reduction in bandwidth.

--Filter design is easy as high accuracy is not needed.

--The transmission of low frequency components is possible, without difficulty.

--Possesses good phase characteristics.

ix)Disadvantages

--Bandwidth when compared to SSB is greater.

--Demodulation is complex.

b) μ-law and A-law companding

Ans : i) µ-law is used across US and Japan as companding standard. ii) By limiting linear sample value equivalent to 13 bits we can obtain µ-law equation as mentioned below.

iii) Here µ is known as compression parameter and its value is about 255 in US and Japan.

$$F(x) = \frac{sgn(x)*\ln(1+\mu|x|)}{\ln(1+\mu)} \qquad 0 \le x \le 1$$

iii) µ-law encoders operate on linear 13-bit magnitude data, whereas 12-bit magnitude data is used in A-law

iv) Before chord determination a bias value of 33 is added to the absolute value of the linear input data to simplify the chord and step calculations in µ-law.

v)In µ-law the definition of the sign bit is reversed

vi) The inversion pattern is applied to all bits in the 8- bit code in µ-law.

vii) A-law is used across Europe as companding standard recommended by CCITT.

viii) By limiting linear sample value equivalent to 12 bits we can obtain A-law equation as mentioned below.

ix) Here A is known as compression parameter and its value is about 87.7 in Europe while x is the normalized integer which need to be compressed.

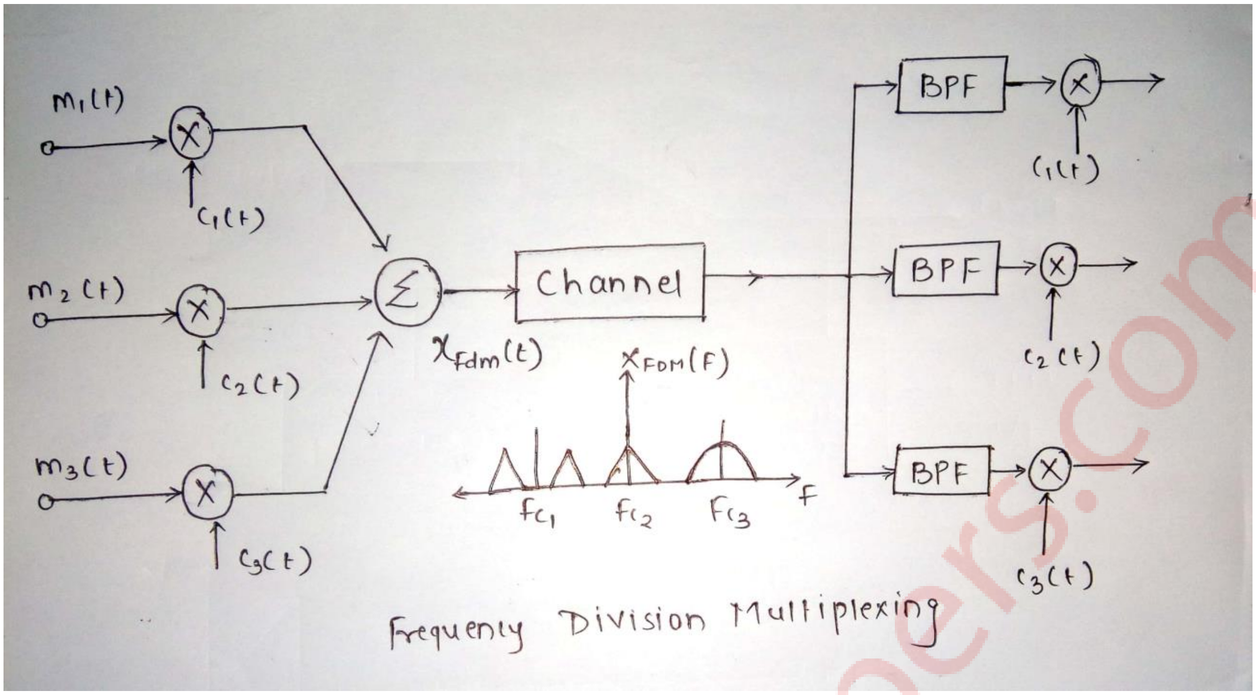
$$F(x) = \begin{cases} \frac{A*|x|}{1+\ln(A)} & 0 \le |x| \le \frac{1}{A} \\ \frac{sgn(x)*\ln(1+A|x|)}{\ln(1+A)} & \frac{1}{A} \le |x| \le 1 \end{cases}$$

c) Frequency division multiplexing (FDM)

Ans : -Frequency-Division Multiplexing (FDM) is a scheme in which numerous signals are combined for transmission on a single communications line or channel.

-It is analog multiplexing technique. Each signal is assigned a different frequency (sub channel) within the main channel. its requires channel synchronization. FDM multiplexing technique is based on orthogonality of sinusoids.

-FDM requires that the bandwidth of a link should be greater than the combined bandwidths of the various signals to be transmitted. Thus each signal having different frequency forms a particular logical channel on the link and follows this channel only.



-These channels are then separated by the strips of unused bandwidth called guard bands. These guard bands prevent the signals from overlapping

-A typical analog connection via a twisted pair telephone line requires approximately three kilohertz (3 kHz) of bandwidth for accurate and reliable data transfer.

-Twisted-pair lines are common in households and small businesses. But major telephone cables, operating between large businesses, government agencies, and municipalities, are capable of much larger bandwidths

Advantages of FDM:

1. A large number of signals (channels) can be transmitted simultaneously.

2. FDM does not need synchronization between its transmitter and receiver for proper operation.

3. Demodulation of FDM is easy.

4. Due to slow narrow band fading only a single channel gets affected.

Disadvantages of FDM:

1. The communication channel must have a very large bandwidth.

2. Intermodulation distortion takes place.

3. Large number of modulators and filters are required.

4. FDM suffers from the problem of crosstalk.

5. All the FDM channels get affected due to wideband fading.

d) Amplitude limiting and thresholding

Ans : i)Amplitude limiting is "a process in which the amplitude of output signal is limited to a desired level or margin irrespective of the variations in the input signal".

ii) Amplitude limiter is an electronic device which clips (removes) the amplitude of output signals to a desired margin irrespective of variations in the input signal

iii)The undesirable input amplitude is clipped by a limiter circuit and gives the desirable margin of output.

iv)Amplitude limiters are used in FM (Frequency modulation) receivers to eliminate the undesirable amplitude changes caused by noise.

v)Threshould effect is defined as the value of input signal to noise ratio below which the output signal to noise ratio decreases much rapidly than the input signal to noise ratio. It is the property of envelope detectors used for the demodulation of modulated signals.

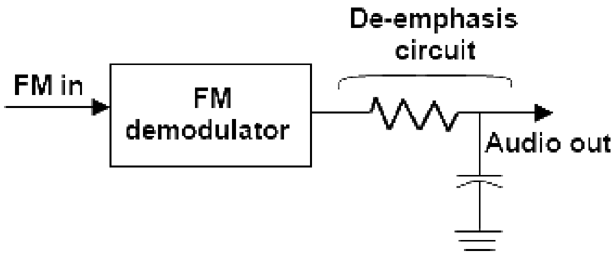
vi) It occurs due to presence of large noise and therefore causes loss in the message signal. When the noise is very large as compared to the input at envelope detector, the message signal at the output is mixed with noise.

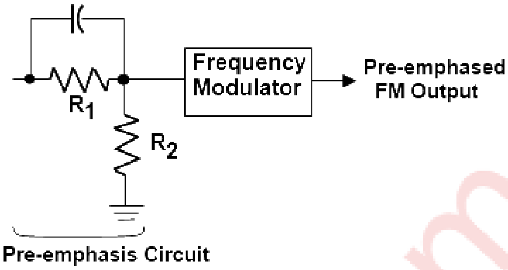
e) Pre-emphasis and de-emphasis circuits and its need

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

ii)De-emphasis means attenuating those frequencies by the amount by which they are boosted. However pre-emphasis is done at the transmitter and the de-emphasis is done in the receiver.

iii)The purpose is to improve the signal-to-noise ratio for FM reception. A time constant of 75µs is specified in the RC or L/Z network for pre-emphasis and de-emphasis.





iv)During the transmission over a channel, the received signal contains interference (high frequency noise). For demodulated FM signals, the interference power increases as the frequency ωi goes up. Thus, deemphasis is applied to the demodulated signal to decrease the power of the interference in high frequency.

v)However, in order to keep the high frequency component of the demodulated message, preemphasis must be applied to the message before going through the FM modulator.

vi) At the transmitter the modulating signal is passing through a simple network which amplifies the high frequency component more the low-frequency component.

vii) The simplest form of such circuit is a simple high pass filter. The pre-emphasis circuit increases the energy of the higher content of the higher-frequency signals so that will tend to become stronger than the high frequency noise component. This improves the signal-to-noise ratio.

viii) To return the frequency response to its normal level, a de-emphasis circuit is used at the receiver. This is a simple low-pass filter.

ix) The de-emphasis circuit provides a normal frequency response.

x) The combined effect of pre-emphasis and de-emphasis is to increase the high-frequency components during the transmission so that they will be stronger and not masked by noise.