Q.P CODE : 66003

MUMBAI UNIVERSITY

Principles Of Communication Engineering

SEMESTER 4 - DECEMBER 2019 – Choice Based

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

a) Compare AM and FM. **[5M]**

Ans :

Amplitude Modulation	Frequency Modulation
The amplitude of carrier wave is changed with respect to modulating wave is called amplitude modulation (AM).	Th frequency of carrier wave is changed with respect to modulating signal is called as frequency modulation (FM).
The radio wave is called a carrier wave and the frequency and phase remain the same	The radio wave is called a carrier wave, but the amplitude and phase remain the same
Has poor sound quality, but can transmit longer distance	Has higher bandwidth with better sound quality
The frequency range of AM radio varies from 535 to 1705 kHz	The frequency range of FM is 88 to 108 MHz in the higher spectrum
More susceptible to noise	Less susceptible to noise
It has simple circuit design.	It has complex circuit design.
It is a less costly method.	It is more costly than AM.
It requires low bandwidth in the range of 10 kHz.	It requires high bandwidth in the range of 200 kHz.
It operates in the medium frequency (MF) and high frequency (HF).	It operates in the upper VHF and UHF range where noise effects are less.
Wastage of power is more as a major part of the power carried by the carrier wave does not contain the information.	No wastage of power as all transmitted power is carried by the information signal.

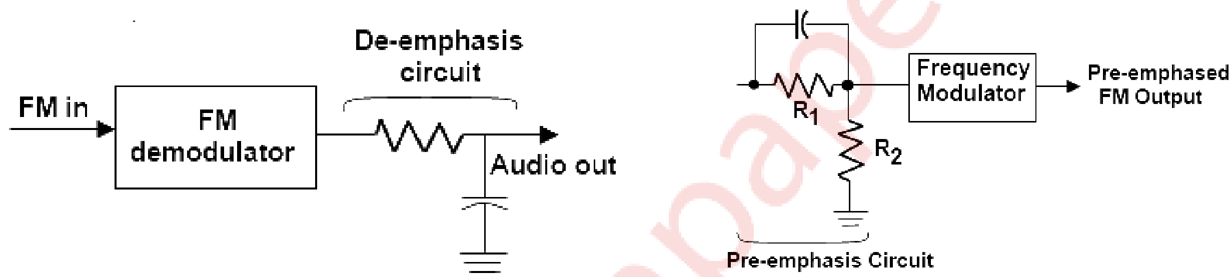
b) Explain the necessity of De-emphasis and pre-emphasis in frequency modulator.

[5M]

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. 2. 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 highfrequency noise component. This improves the signal-to-noise ratio. 3.

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 4.

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.

c) Define and explain selectivity and sensitivity for radio receiver. [5M]

Ans :i) Sensitivity is the ability to amplify weak signals.

ii) Minimum RF signal level that can be detected at the input to produce a usable demodulated information signal.

iii) Receivers should have high sensitivity so that it may have good response to the desired signal.

iv) But should not have excessively high sensitivity otherwise it will pick up all undesired noise signals. It is function of receiver gain and measures in decibels.

v) Sensitivity of a receiver is expressed in microvolt of the received signal.

vi) Typical sensitivity for commercial broadcast-band AM receiver is 50 µV. The best way to improve the sensitivity is to reduce the noise level.

vii) Selectivity of radio receiver is its ability to differentiate desired signal from unwanted signals.

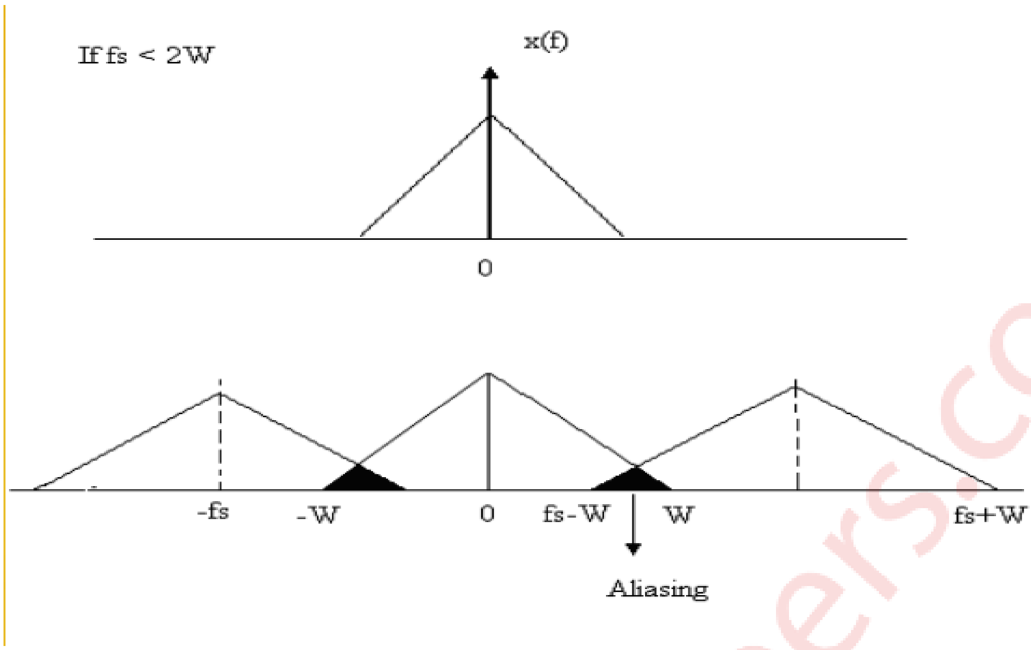
viii) Selectivity of super heterodyne receiver is determined by frequency response characteristics of the IF amplifier. Selectivity decides the adjacent channel rejection of a receiver.

d) What is aliasing ? How it can be prevented ? [5M]

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.

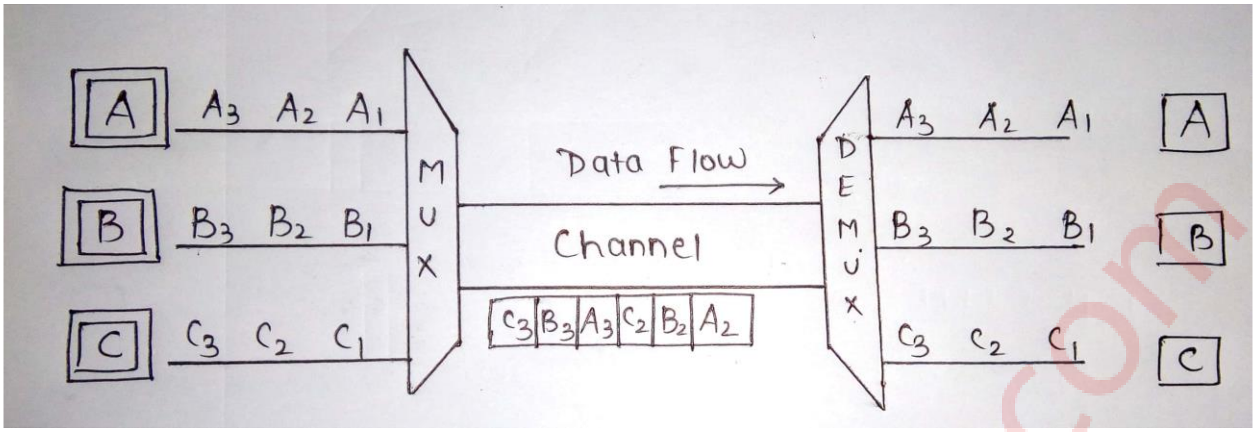
e) What is time division multiplexing ? Also give its application. [5M]

Ans : -Time division multiplexing (FDM) is a technique of multiplexing, where the users are allowed the total available bandwidth on time sharing basis. Here the time domain is divided into several recurrent slots of fixed length, and each signal is allotted a time slot.

- TDM is digital multiplexing technique. In TDM, the channel/link is not divided on the basis of frequency but on the basis of time.

-Total time available in the channel is divided between several users.

-Each user is allotted a particular a time interval called time slot or time slice during which the data is transmitted by that user.



-Thus each sending device takes control of entire bandwidth of the channel for fixed amount of time.

-In TDM the data rate capacity of the transmission medium should be greater than the data rate required by sending or receiving devices.

-Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.

-The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing. The TDM signal in the form of frames is transmitted on the common communication medium.

-Advantages of TDM :
1. Full available channel bandwidth can be utilized for each channel.
2. lntermodulation distortion is absent.

-Disadvantages of TDM :

1. Synchronization is essential for proper operation.
2. Due to slow narrowband fading, all the TDM channels may get wiped out.

- Applications : Digital telephony,Data communication , Satellite access , Cellular radio

Q.2 a) Explain balanced modulator using diode for the generation of DSBSC AM signal.

[10M]

Ans :i) Double Sideband Suppressed Carrier (DSBSC) is an amplitude modulation technique in which the modulated wave contains both the sidebands along with the suppressed carrier.

ii) Conventional AM consists of the two sidebands and a carrier where the major transmitted power is concentrated in the carrier which contains no information. Thus to increase the efficiency and to save power, the carrier is suppressed in DSBSC system.

iii)The DSBSC generation using balanced modulator based on nonlinear resistance characteristics of diode is given.

iv) The diode in the balanced modulator use the nonlinear resistance property for producing modulated signals.

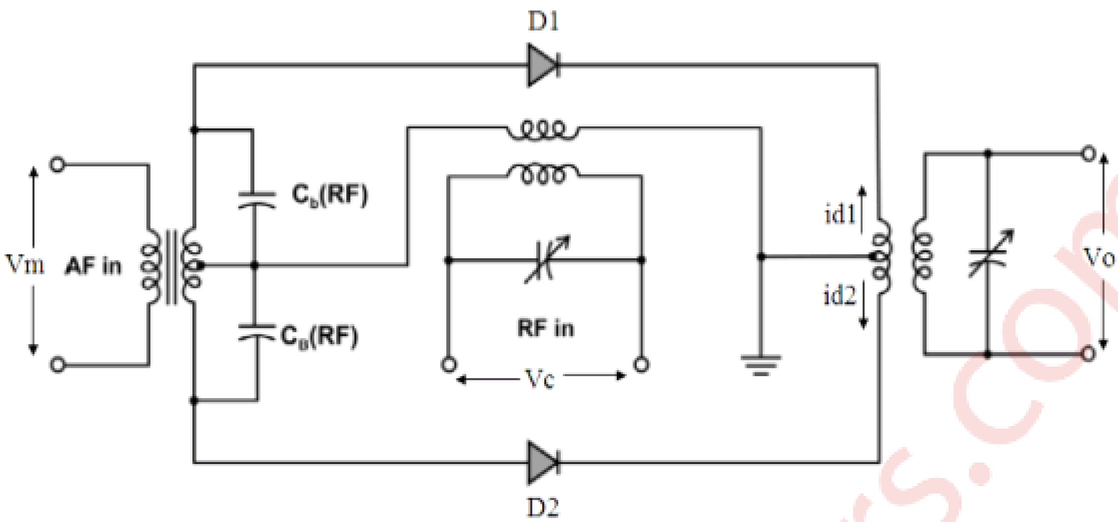
v) Carrier voltage is applied in phase at both the diodes, while modulating voltage appears 180° out of phase at the diode inputs as they are at opposite ends of a center tapped transformer.

vi) The modulated output currents of the two diodes are combined in the center tapped primary of the output transformer, which then gets subtracted.

vii) The output of the balanced modulator contains two sidebands and sum of the harmonic components.

viii) The input voltage at diode D1 is (vc+vm) and input voltage at diode D2 is (vc−vm).

ix) Balanced modulator circuit consists of two diodes with capacitors designed for RF and transformers.



x) The primary current of the output transformer is

$$i1 = av1 + b.v1^2 = a(x(t) + \cos(wc.t)) + b.(x(t) + \cos(wc.t))^2$$

Similarly ,

$$i2 = av2 + b.v2^2 = a(x(t) - \cos(wc.t)) + b.(x(t) - \cos(wc.t))^2$$

Final voltage is given by ,

$$vo = i1.R - i2.R = 2a.R.x(t) + 4.b.R.x(t).\cos(wct)$$

Which is comprises of two terms first term is modulating signal whereas second term is DSB SC signal.

b) How to generate SSB using filter method ? [10M]

Ans : The filter method can be used for generating the SSB modulated wave if the message signal satisfies the following conditions :

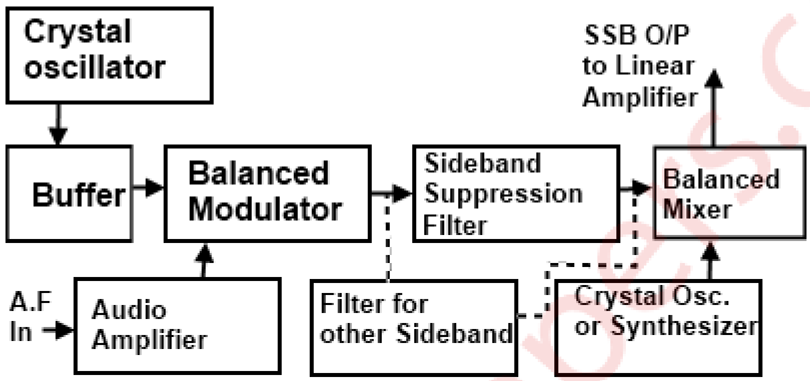
i)The message signal should not have any low frequency content . The audio signal posses this property, e.g. the telephone signal will have a frequency range extending from 300 Hz to 3.4 kHz . The frequencies in the range 0-300 Hz are absent .

ii)The highest frequency in the spectrum of the message signal i.e. W Hz should be much smaller than carrier frequency fc .

iii) A crystal controlled master oscillator produces a stable carrier frequency fc (say 100 KHz).

iv) This corner frequency is then fed to the balanced modulator through a buffer amplifier which isolates these two stages.

v) The audio signal from the modulating amplifier modulates the carrier in the balanced modulator Audio frequency range is 300 to 2800 Hz The carrier is also suppressed in this stage but allows only to pass the both sidebands (USB & LSB).



SSB Single Side Band Transmission Filter Method

vi) A band pass filter (BPF) allows only a single band either USB or LSB to pass through it. It depends on our requirements.

vii)Let we want to pass the USB then LAB will be suppressed. In This case fc = 100 KHz Audio range = 300 - 2800 Hz USB frequency range = fc + 300 to fc + 2800 = 100000 + 300 to 100000 + 2800 = 100300 to 102800 Hz. So this band of frequency will be passed on through the USB filter section.

viii) This side band is then heterodyned in the balanced mixer stage with 12 MHz frequency produced by crystal oscillator or synthesizer depends upon the requirements of our transmission.

ix)So in mixer stage, the frequency of the crystal oscillator or synthesizer is added to SSB signal. The output frequency thus being raised to the value desired for transmission

x) Then this band is amplified in driver and power amplifier stages and then fed to the aerial for the transmission.

xi) Advantages :

1. It allows better management of the frequency spectrum. More transmission can fit into a given frequency range than would be possible with double side band DSB signals

2. All of the transmitted power is message power none is dissipate as carrier power

3. The noise content when the bandwidth is reduced by half therefore single side band SSB signals have less noise contamination than DSB double side band. of a signal is an exponential function of the bandwidth the noise will decrease by 3dB.

xii) Disadvantages

1. The cost of a single side band SSB receiver is higher than the double side band DSB counterpart be a ratio of about 3:1

2.The average radio user wants only to flip a power switch and dial a station Single side band SSB receivers require several precise frequency control settings to minimize distortion and may require continual readjustment during the use of the system.

Q.3 a) List types of noise and explain any four types of internal noise. [5M]

Ans :) Noise : Noise is an unwanted signal which interferes with the original message signal and corrupts the parameters of the message signal.

ii) Noise limits the operating range of the systems.Noise affects the sensitivity of receivers.

iii)There are two main types of noise: a) External Source b) Internal noise

A) Atmospheric noise (due to irregularities in the atmosphere).
B) Extra-terrestrial noise, such as solar noise and cosmic noise.
C) Industrial noise

iv) Internal Source : This noise is produced by the receiver components while functioning. The components in the circuits, due to continuous functioning, may produce few types of noise. This noise is quantifiable.

v) **Shot Noise** : These Noise are generally arises in the active devices due to the random behaviour of Charge particles or carries. In case of electron tube, shot Noise is produces due to the random emission of electron form cathodes.

vi) **Partition Noise** : When a circuit is to divide in between two or more paths then the noise generated is known as Partition noise. The reason for the generation is random fluctuation in the division.

vii) **Low- Frequency Noise** : They are also known as FLICKER NOISE. These type of noise are generally observed at a frequency range below few kHz. Power spectral density of these noise increases with the decrease in frequency. That why the name is given Low-Frequency Noise.

viii) **High- Frequency Noise** : These noises are also known TRANSIT- TIME Noise. They are observed in the semi-conductor devices when the transit time of a charge carrier while crossing a junction is compared with the time period of that signal.

ix) **Thermal Noise** : Thermal Noise are random and often referred as White Noise or Johnson Noise. Thermal noise are generally observed in the resistor or the sensitive resistive components of a complex impedance due to the random and rapid movement of molecules or atoms or electrons.

b) What do you mean by noise factor and noise figure.How it can be improved ? [5M]

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 :

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

vi) Noise figure and noise factor can be improved by increasing signal to noise ratio of input side .

c) Draw the block diagram of super-heterodyne receiver and explain the operation. Write frequency componenets present at the output of each block if audio frequency is 1KHz and carrier frequency is 540KHz. [10M]

Ans : i)Heterodyne means to mix two frequencies together in a nonlinear device or to translate one frequency to another using nonlinear mixing. Essentially, there are five sections to a superheterodyne receiver: RF section, mixer/convertor section, IF section, detector section and the audio amplifier section.

ii)RF Section – The RF Section generally consists of a preselector and an amplifier stage. The preselector is a broad-tuned band pass filter with an adjustable center frequency used to reject unwanted radio frequency and to reduce the noise bandwidth. The RF amplifier amplifies the signal and also determines the sensitivity of the receiver.

iii)Mixer/ Converter Section – It consists of a radio-frequency oscillator and a mixer. The choice of oscillator depends on the stability and accuracy desired. Mixer/converter is a non-linear device that is used to convert radio frequency to intermediate frequencies. Although the frequencies are changed, the shape of the envelope, the bandwidth and the original information contained in the envelope remains unchanged.

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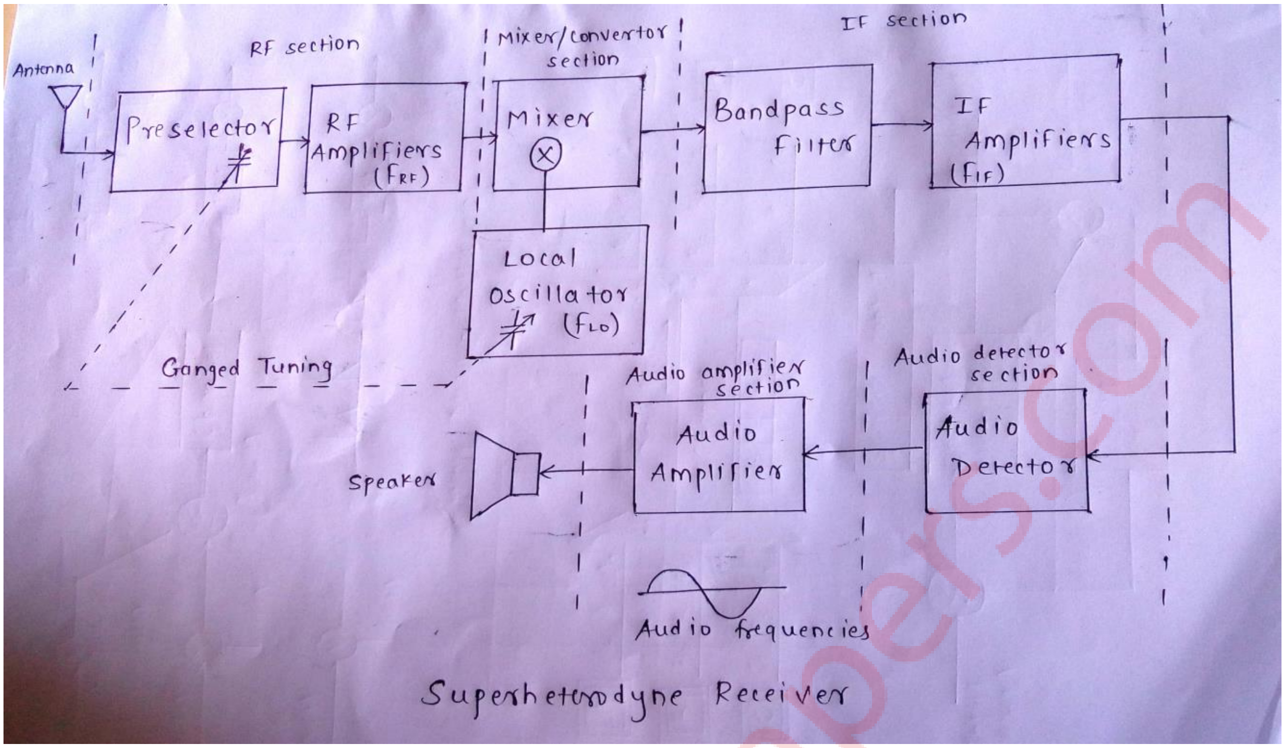


v)IF Section – It consists of a series of bandpass filters and IF amplifiers. Most of the receiver gain and selectivity is achieved in the IF section. The IF is always lower than the RF because it is easier and less expensive to construct high-gain, stable amplifiers for low frequency signals. IF amplifiers are also less likely to oscillate when compared to RF amplifiers.

vi)Detector Section – It converts the IF signals back to the original source information. The detector can be as simple as a single diode or as complex as a PLL or balanced demodulator.

vii)Audio amplifier section – It amplifies the information signal to a required level. It comprises of several cascaded audio amplifiers and one or more speakers. The number of amplifiers depends on the audio signal power desired.

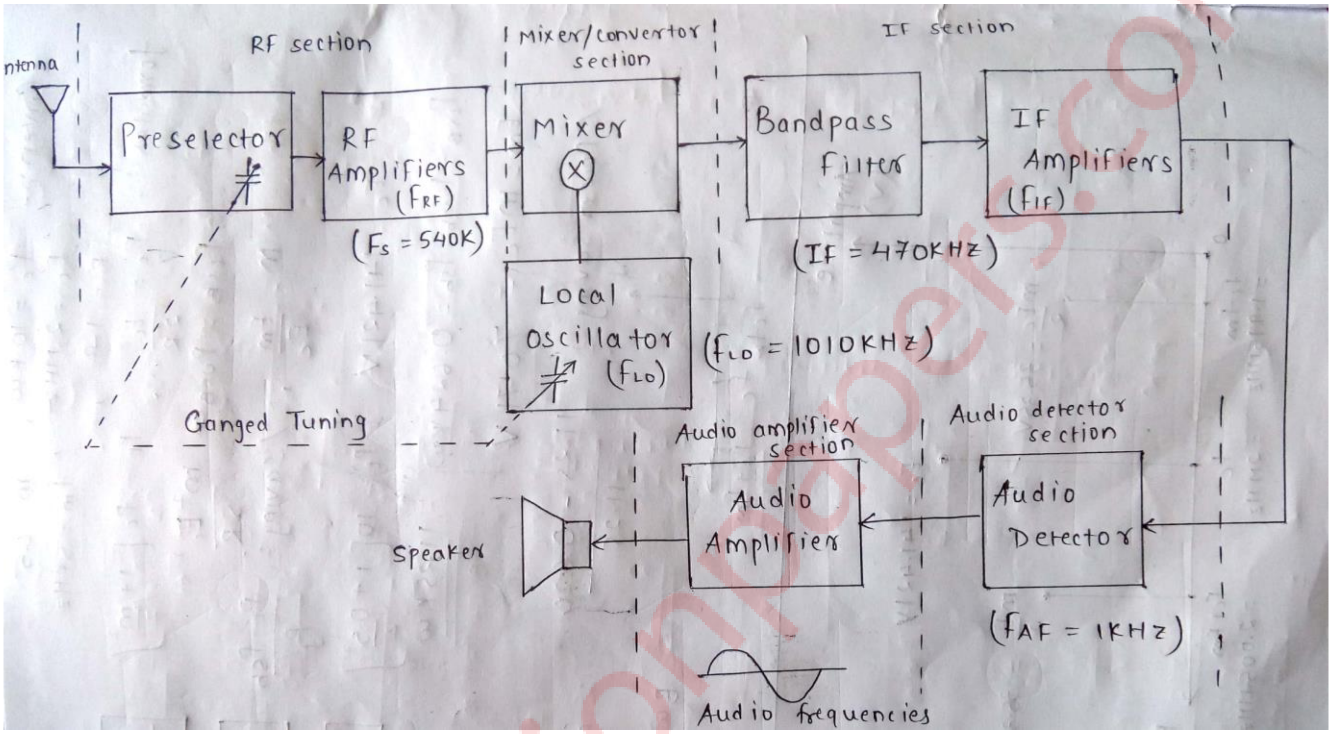
Advantages :

1. As high frequency is down converted to IF, there is no possibility of oscillations.

2. The bandwidth remains constant over the entire frequency range.

3. Better selectivity as no adjacent channels are picked due to variation in bandwidth.

vii) If audio frequency is 1KHz and carrier frequency is 540KHz , (local oscillator frequency 1010KHz) then frequency components involved in superheterodune receiver is ,



Q.4 a) With the help of neat digram and waveforms explain generation and demodulation of pulse position modulation. **[10M]**

Ans : i) Pulse position modulation : The amplitude and the width of the pulse remains constant. The time when the pulse occurs is varied in accordance with the modulating signal.

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

iii)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.

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

v)PWM & PPM generation PWM generation: – The output of the comparator is zero except when modulating signal waveform exceeds the sawtooth wave, when the output is high.

vi) PPM generation: – The output of the comparator (i.e. PWM) is given as input to negative edge triggered monostable pulse generator. – On the trailing edge (negative going edge) of the PWM signal produces a short pulse of fixed duration. This output is PPM signal.

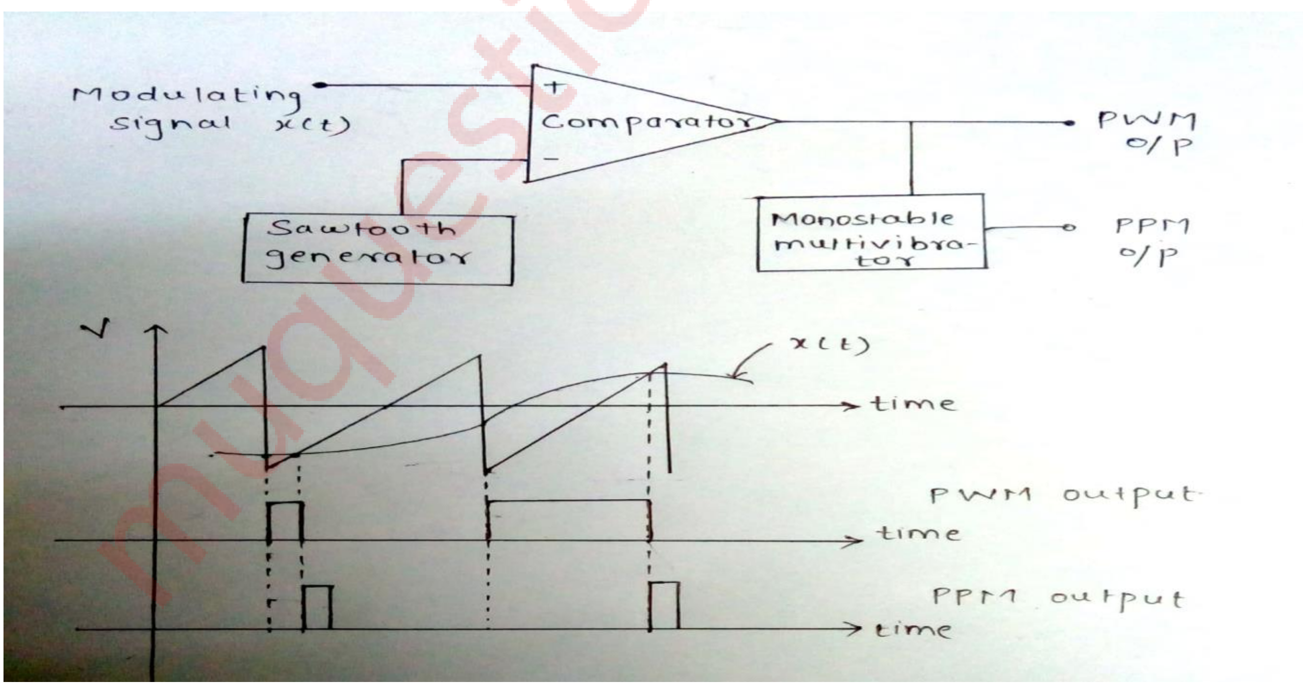
vii) Advantage: – Less noise interference due to constant amplitude. – Signal and noise separation is very easy. – Due to constant pulse width & amplitudes, transmission power for each pulse is same.

vii) Disadvantage – Synchronization between transmitter & receiver is required.

viii) Applications of PPM :

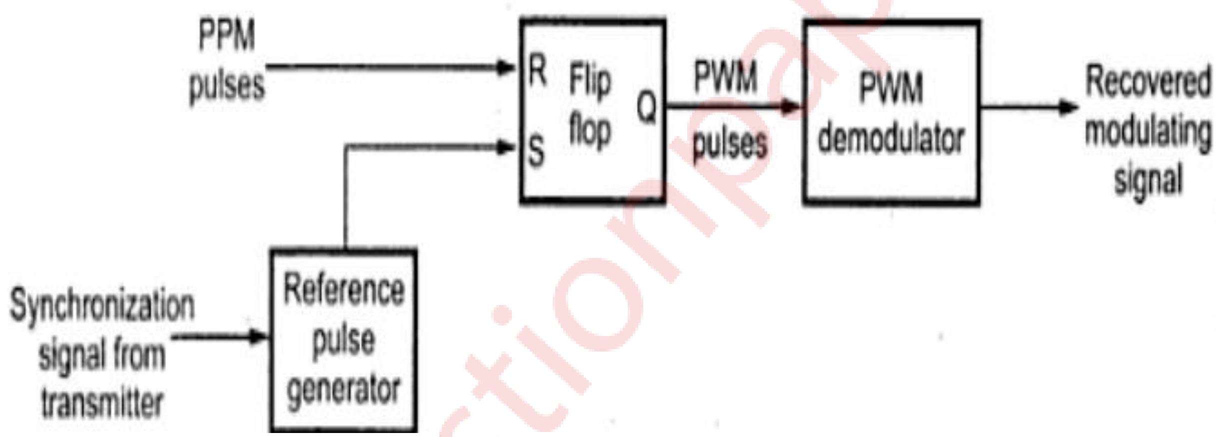
R/C transmitters,R/C receivers ,Autopilot/Stabilization system,PCTx

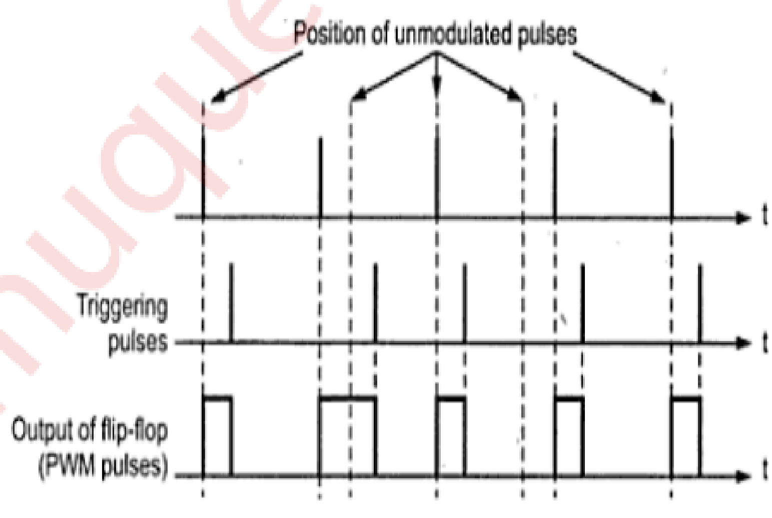
ix) PPM and PWM are used to send analog signals, not digital signals. They are analog protocols.



x) Role of monostable multivibrator : When triggered , a pulse of of predefined duration is produced.The circuit then returns to its stable state and produces no more output until triggered.In PPM generation the pulse is generated whenever the PWM pulse is high. And the duration of PPM pulse generated by monostable multivibrator is constant for each triggered pulse.

xi) Demodulation : It is necessary to convert the position varying pulses to length varying pulses. The flip flop is set when the reference pulse arrives. Reference pulse is generated by the generator with synchronization signal from the transmitter. The flip flop is reset at the leading edge of the position modulated pulse. This repeats and a PWM pulse is obtained at the output of the flip flop.The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.





b) A carrier wave of frequency 100MHz is frequency modulated by sine wave of amplitude 20 volts and frequency 100KHz.The frequency sensitivity of the modulation is 25KHz per volt.Determine the approximate bandwidth of FM wave using Carson's rule. [5M]

Ans : Given : Carrier wave frequency = fc = 100MHz

Amplitude (modulating wave) = Am = 20 V

Frequency (modulating wave) = fm = 100KHz

Frequency Sensitivity = kf = 25 KHz/volt.

To Find : Bandwidth of FM wave = B.W = ?

Formulas : Casron's Rule : B.W = 2(fd + fm)

fd = frequency deviation = kf.Am

Solution : We know that , frequency deviation is given by ,

fd = kf.Am = 25K (20) = 500 KHz

Now , using Carson's rule B.W = 2(fd + fm) = 2(500K + 100K) = 1200KHz.

c) A 360 W carrier is simultaneously amplitude modulated by two audio waves with modulation percentages of 55 and 65 respectively.What is total side band power ?

[5M]

Ans : Given : Carrier wave power = 360 W

Modulation prcentages = 55 and 65 => m1=0.55 & m2=0.65

To Find : Total side band power = ?

Formulas : mt = $\sqrt{m1^2 + m2^2}$, Pc=$\frac{Ec^2}{2R}$, Assume R=1

P(USB)=P(LSB)=$\frac{(m.Ec)^2}{8R}$

Solution : We know that , Pc=$\frac{Ec^2}{2R}$ => Ec = $\sqrt{360 \times 2} = 26.83\,V$

Now , total modulation index is = mt = $\sqrt{m1^2 + m2^2} = \sqrt{0.55^2 + 0.65^2} = 0.8514$

$$\therefore Total\ side\ band\ power = P(usb) + P(lsb)$$

$$= \frac{(m1.Ec)^2}{8} + \frac{(m2.Ec)^2}{8}$$

$$= \frac{(mt.Ec)^2}{4R}$$

$$= \frac{(0.8514\ \times 26.83)^2}{4}$$

$$\boxed{\text{Total side band power} = 130.45\ W}$$

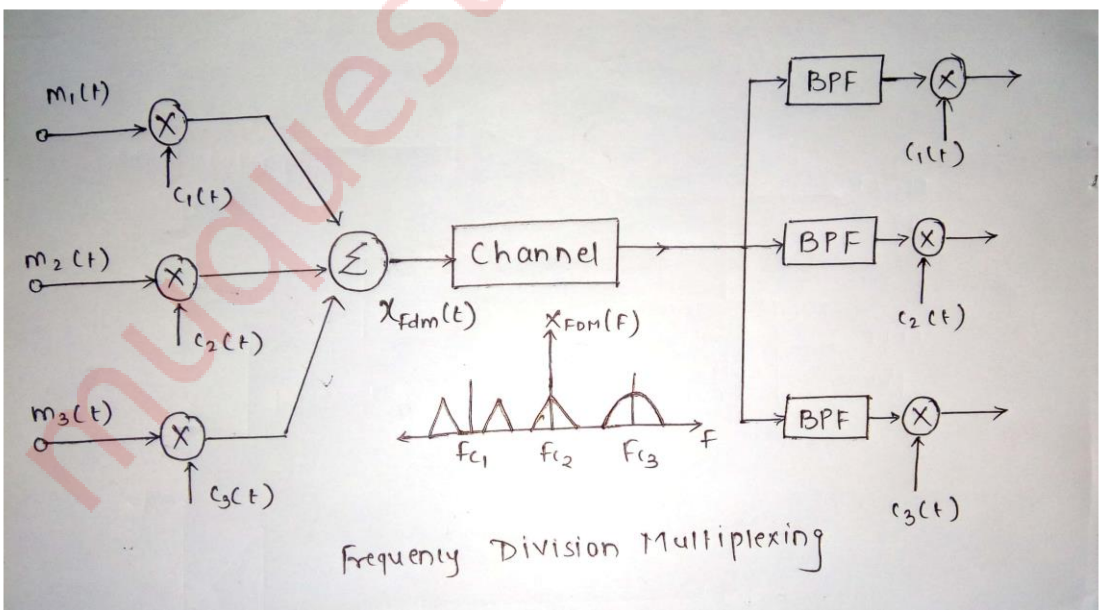
Q.5 Write short note on (Any Four)

a) Frequency division multiplexing **[5M]**

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.

b) Double spotting and fidelity of Radio Receiver **[5M]**

Ans : In double spotting same stations get picked up at two different nearby points, on the receiver dial.

ii) It occurs due to inadequate image frequency rejection.

iii) Double spotting is harmful, since a weak station can be masked by the reception of a strong station at the same point. Double spotting can be reduced by increasing front end selectivity of the receiver.

iv) RF amplifier stage helps in avoiding double spotting.

v) Fidelity is the ability of a communication system to produce an exact replica of the original source information at the output of the receiver.

vi) Radio receiver should have high fidelity or accuracy.

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

c) Wide band and narrow band FM **[5M]**

Ans :

Narrowband FM	Wideband FM
1.Modulation index is less than 1	1.Modulation is greater than 1.
2.Frequency Deviation = 5kHz	2. Frequency Deviation =75kHz
3.Modulating frequency = 3kHz	3. Modulating frequency = 30Hz to 15kHz
4.Supression of noise is very less.	4.Noise is more suppressed.
5.Applications : FM mobile communication,short range ship to shore communication,etc	5.Applications: Entertainment broadcasting,high quality music transmission,etc

d)Application of pulse communication **[5M]**

Ans : : i) Pulse amplitude modulation is used in Ethernet communication.

ii)Pulse amplitude modulation is used in many micro-controllers for generating the control signals.

iii)Pulse amplitude modulation is used in Photo-biology.

iv)PAM is used as an electronic driver for LED lighting.

v)The PCM is used in the satellite transmission system.

vi)It is used in space communication.

vii)It is used in telephony.

viii)The compact disc (CD) is a recent application of PCM.

ix)PWM is used in telecommunication systems.

x)PWM can be used to control the amount of power delivered to a load without incurring the losses. So, this can be used in power delivering systems.Audio effects and amplifications purposes also used.PWM signals are used to control the speed of the robot by controlling the motors.PWM is also used in robotics.

e) ISB Receiver **[5M]**

Ans : : i) Independent sideband is an AM single sideband mode which is used with some AM radio transmission.Normally each sideband carries identical information,but ISB modulates two different input signals , one on the upper side band , the other on the lower sideband.

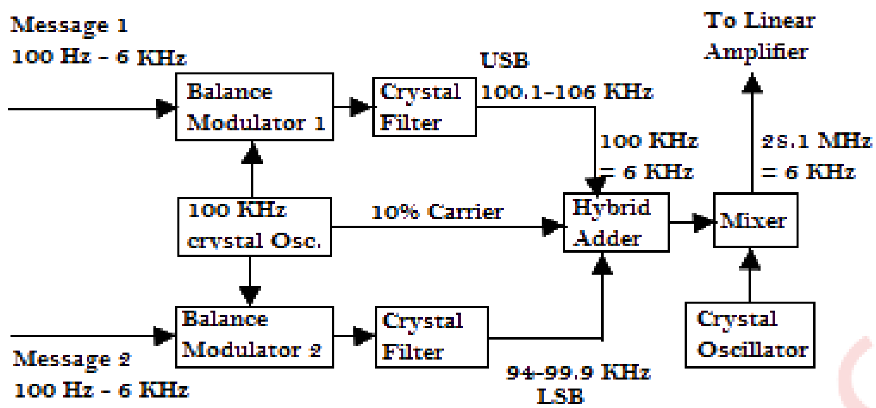
ii)Independent sideband is an amplitude modulated technique in whuch a single carrier frequency is independently modulated by two different modulating signals.

iii) The two sidebands are generated independently by the two balanced modulators. The carrier is suppressed by 45 dB or more in the balanced modulator and the following filter suppresses the undesired sideband. The output from both the filters are then combined in the adder along with 26dB carrier to form low frequency ISB signal which then leaves the drive unit and enters the main transmitter.

iv)Its frequency is raised through mixing to remove unwanted frequencies by the output filter and the resulting RF ISB signal is then amplified to the desired level.

v)ISB conserves both the transmit power and the bandwidth, since the two information signals are transmitted within the same frequency spectrum as against a single signal in traditional DSB transmission system.

vi) ISB Transmission : ISB essentially consists of two SSB channels added to form two side bands around the reduced carrier.Each sideband is quite independent of each other. It can simultaneously convey totally different transmission.



ISB Transmission Block Diagram

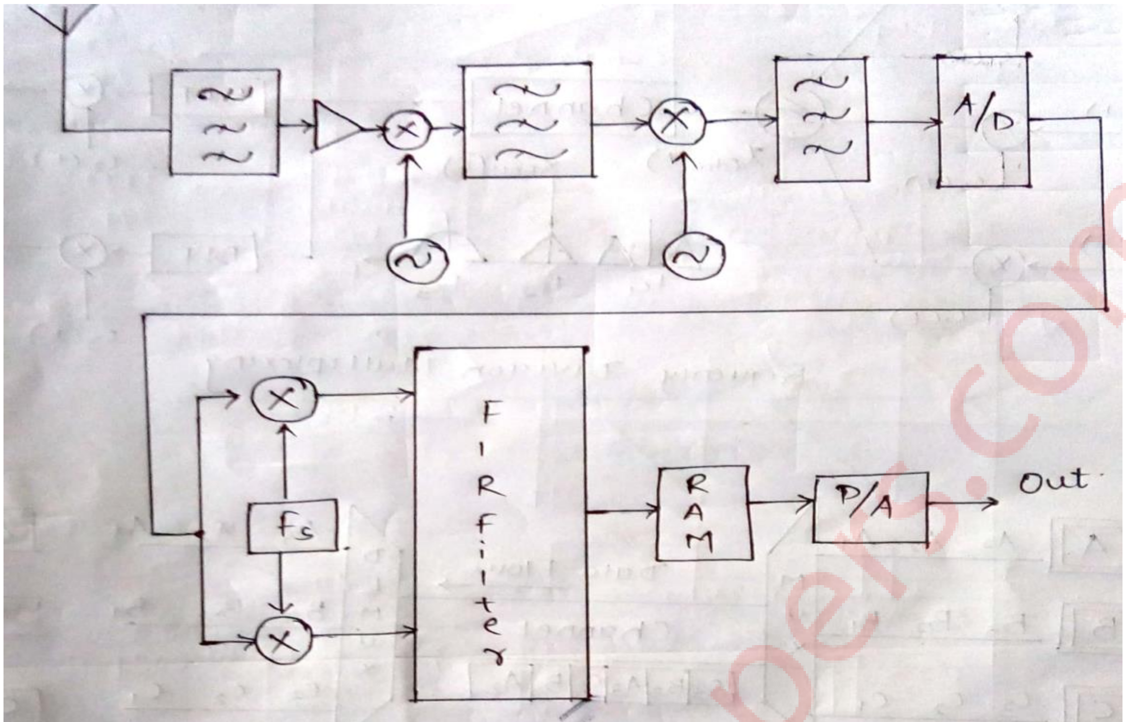
vii) Each 100 Hz - 6 KHz channel is fed to its own balanced modulator, each modulator also receiving the output of the 100 KHz crystal oscillator. Each modulator modulates each message (100Hz to 6KHz) on the frequency 100 KHz.

viii) The USB filter and LSB filter suppresses the unwanted side band in such a way that one filter suppresses the lower side band and the other filter suppresses the upper side band respectively.So USB = 100.1 - 106 KHZ & LSB = 94 - 99.9 KHz

ix)Both outputs are added at the hybrid adder with the 10% reduced carrier.The output is then fed to the balanced mier where it is mix with the output of the crystal oscillator, the frequency is then raised to 28.1 MHZ +- 6 KHz.The resulting RF ISB signal is then amplified by the linear amplifier, until it reaches the ultimate level then is fed to teh antenna for transmission.

x) ISB Receiver :

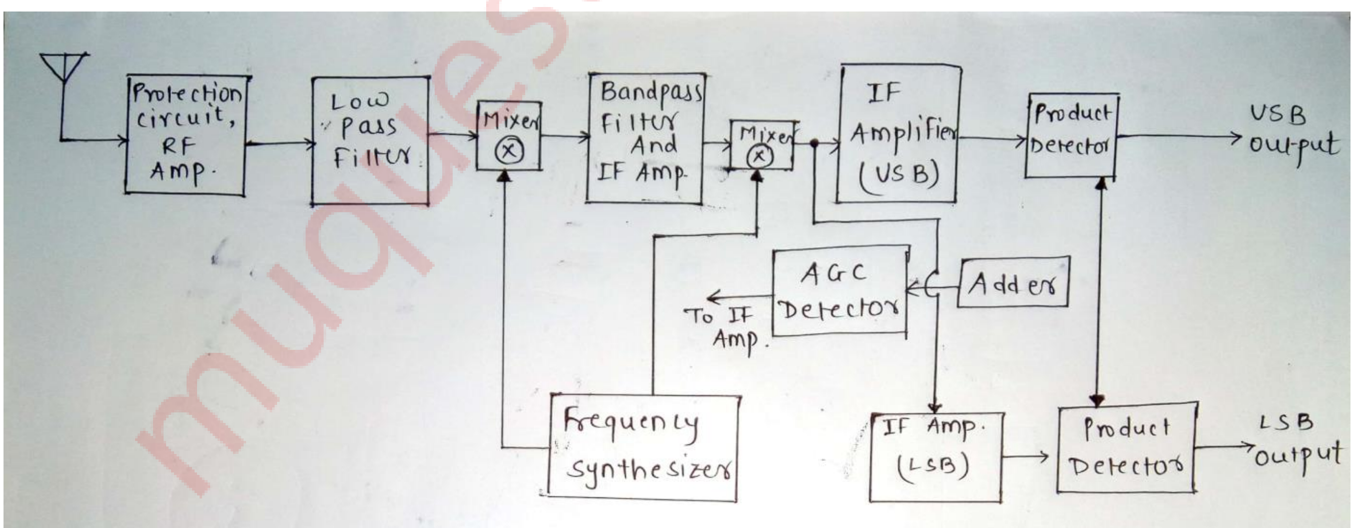
A receiver for independent sideband (ISB) signals comprises means for producing digitised quadrature related zero IF versions of the received signal which are applied to a complex FIR filter structure comprising respective real low pass filters in which alternate coefficients (C1 to CN-1, C0 to CN) are non-zero.



ISB Receiver

xi)The respective upper and lower sidebands (USB, LSB) are recovered by obtaining the sum and difference of the outputs of the respective filters.

xii)The respective sideband signals (USB, LSB) are stored in RAM and when it is desired to reproduce the stored signal it is expanded, equalised and converted to an analogue signal which is supplied to an audio transducer.



ISB Receiver with frequency synthesizer

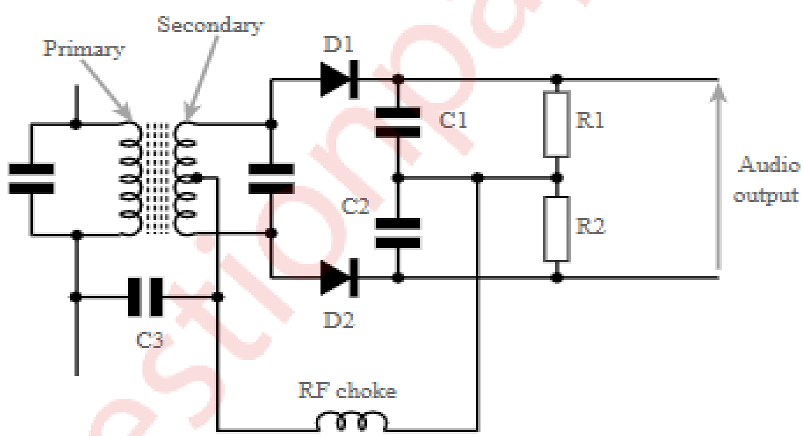
Q.6 a)Describe Foster-seeley Discriminator with neat circuit diagram and explain its principles with necessary equations.What are merits and demerits. [10M]

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.

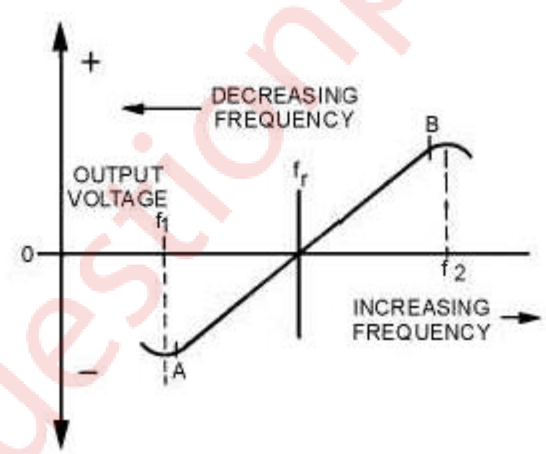
vii) If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.

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) Both the ratio detector and Foster-Seeley detectors are expensive to manufacture. Any wound components like the RF transformers are expensive to manufacture when compared with integrated circuits produced in vast numbers.

xiii) As a result the Foster Seeley discriminator as well as the ratio detector circuits are rarely used in modern radio receivers as FM demodulators.

We know that the equation of FM wave is

s(t)=Ac.cos(2πfct+2πkf∫m(t)dt)

Differentiate the above equation with respect to '**t**'.

$$\frac{ds(t)}{dt} = -Ac.\left(2\pi fc + 2\pi kfm(t)\right).sin(2\pi fct + 2\pi kf\int m(t).dt)$$

We can write sin in another terms as ,

$$\frac{ds(t)}{dt} = -Ac.\left(2\pi fc + 2\pi kfm(t)\right).sin\left(2\pi fct + 2\pi kf\int m(t).dt - 180\right)$$

$$\frac{ds(t)}{dt} = -Ac.2\pi fc(1 + \frac{kf.m(t)}{kc}).sin\left(2\pi fct + 2\pi kf\int m(t).dt - 180\right)$$

In the above equation, the amplitude term resembles the envelope of AM wave and the angle term resembles the angle of FM wave. Here, our requirement is the modulating signal m(t)m(t). Hence, we can recover it from the envelope of AM wave.

b) Explain generation of frequency modulated wave using Armstrong method. [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.

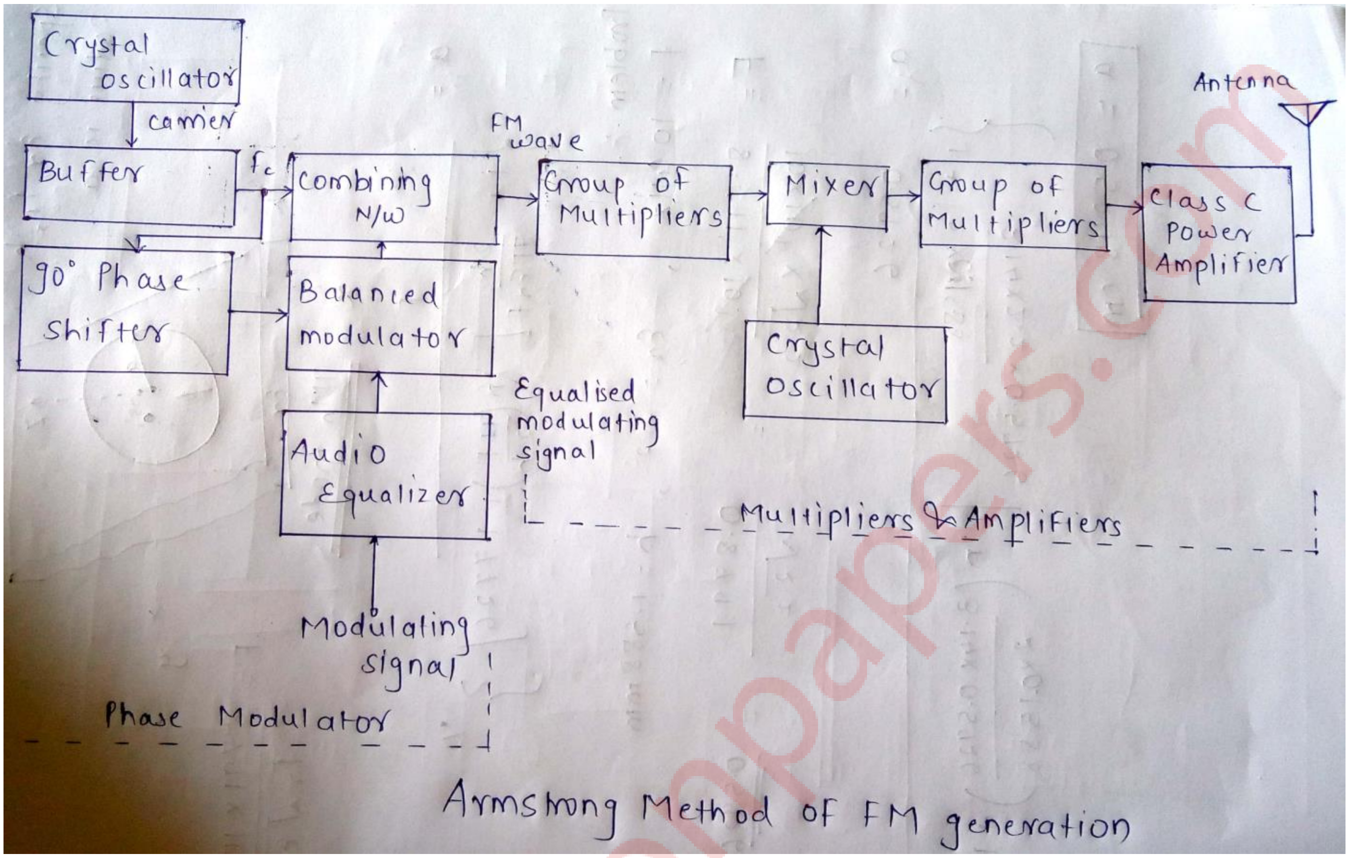
iii) Low frequency sub-carrier fc is phase shifted 90˚ and fed to a balanced modulator. It is mixed with the modulating signal fm .

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.

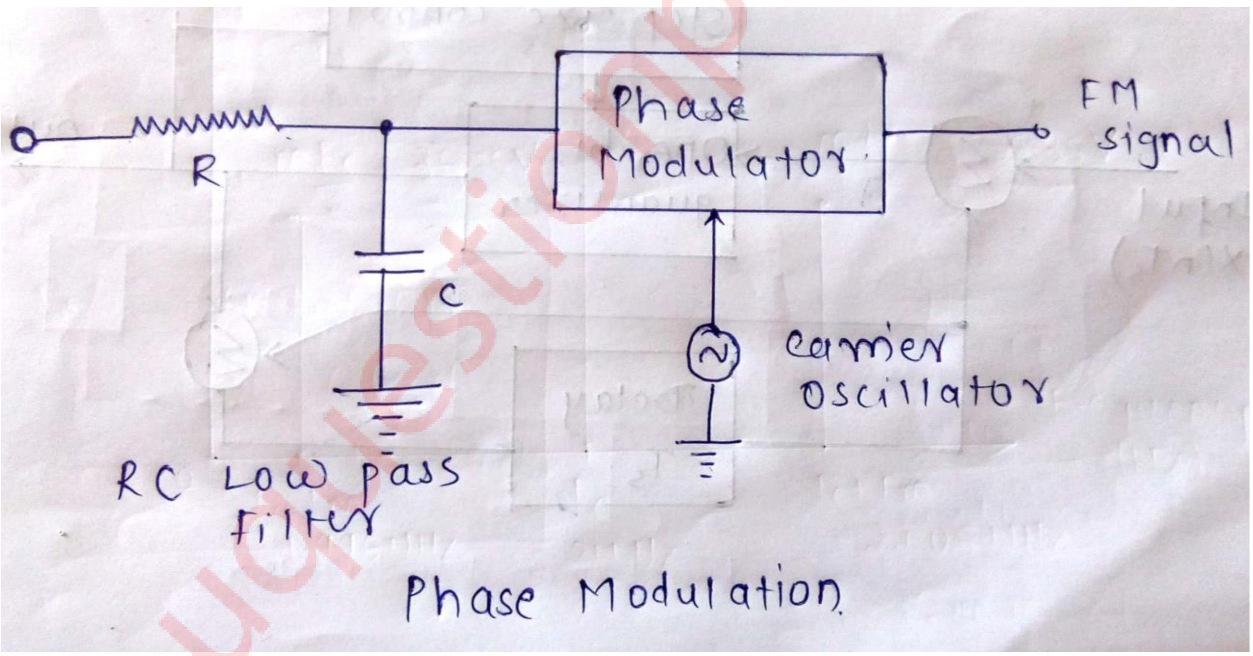
xiii)The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the fc and mf both are raised to required high values using the second group of multipliers.
xiv)The FM signal 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.

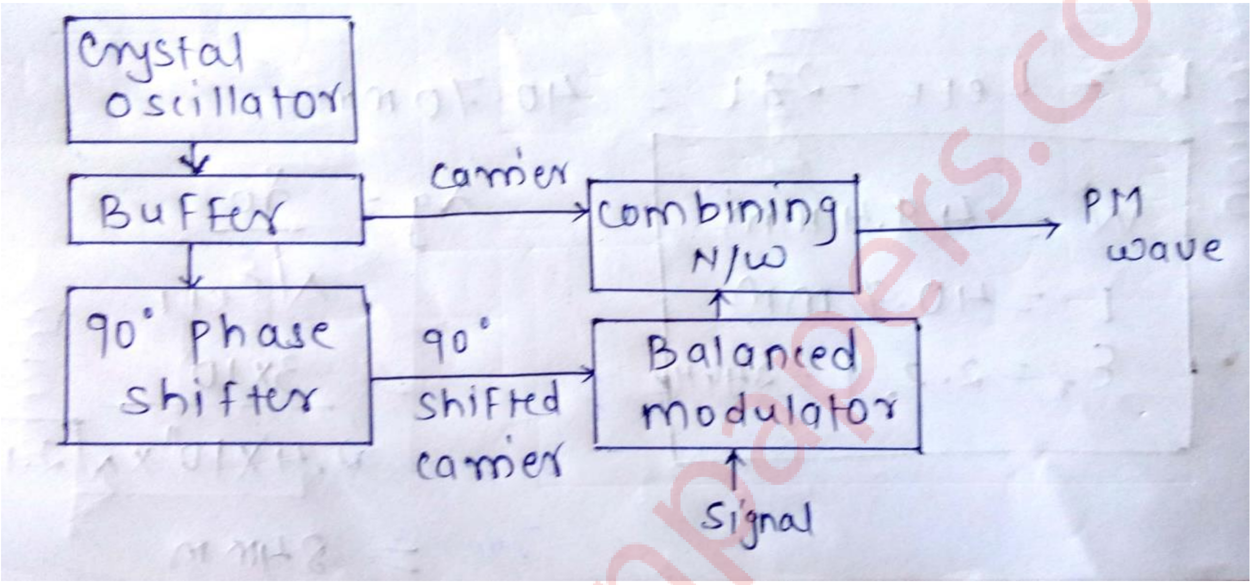


2.Implementation of phase modulator:

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