
Important Instructions to examiners:

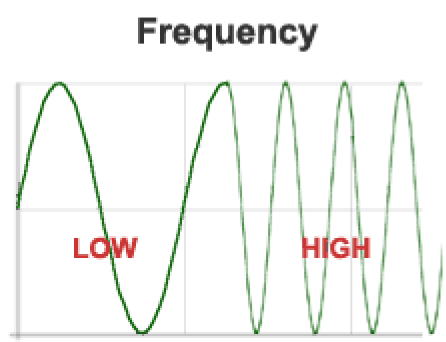
1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given more Importance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in the Figure. The figures drawn by candidate and model answer may vary. The examiner may give credit for any Equivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constant Values may vary and there may be some difference in the candidate's answers and model answer.
6) In case of some questions credit may be given by judgment on part of examiner of relevant answer based on candidate's understanding.
7) For programming language papers, credit may be given to any other program based on equivalent concept.

Q1. A) Attempt any six. (12M)

i) Define frequency and wavelength of a sound wave.

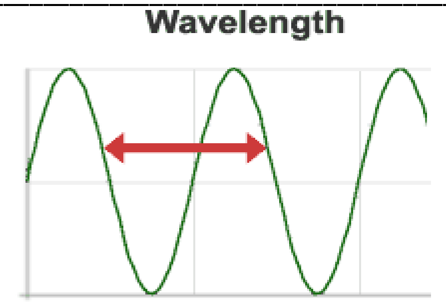
Ans: (Definition –1M Each, Diagrams are optional)

Frequency: It is defined as the number of successive compressions and rarefactions occurring in one second, and is expressed in hertz (or simply Hz). The frequency range of audible sound waves is 16 Hz to 20000 Hz.



Wavelength: The length of space travelled by one cycle of variation is called wavelength and is represented in metres. Equation 1.3 gives the relationship be-tween frequency (f in Hz), wavelength (λ in metres) and velocity (v in metres per second).

$$v = f\lambda$$



ii) State the need for graphic equalizer in an audio amplifier.

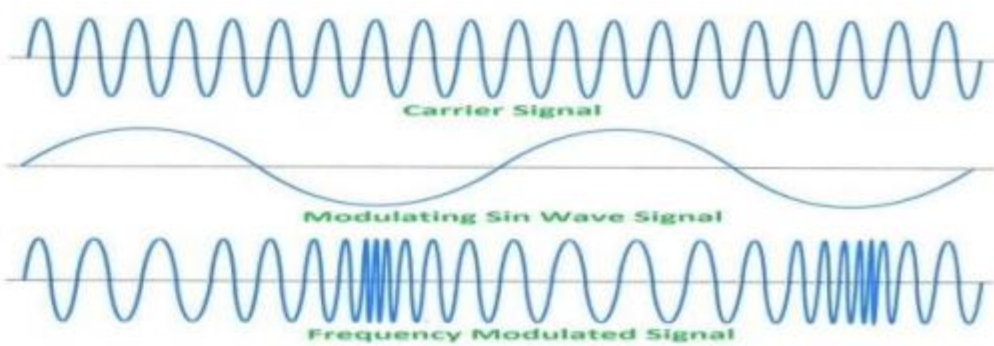
Ans: (Need of Graphic equalizer-2M)

- Graphic equalizers are special type of tone control system in which boosting is done up to +15dB and cut up to -15dB is done throughout the audio spectrum.
- It provides accurate selectivity.

iii) Define frequency in modulation.

Ans: (Definition- 2M)

FM: This is the modulation technique in which frequency of carrier is changed according to amplitude variation in modulating signal.



OR

Modulating Frequency: It is information signal low frequency. **(1M)**

Carrier Frequency: It is high frequency signal whose parameters are changing according to instantaneous value of information signal **(1M)**

iv) State the types of sound audio recording mechanisms.

Ans:(Types of sound recording mechanisms- 1M each, [Any Two])

1. Disc Recording (LP)
2. Magnetic Recording
3. Optical Recording
   a. Film Recording
   b. Compact Disc Recording

v) State the principle of Audio optical recording.

Ans: (2M)

<u>Principle of audio optical recording:</u>

In optical recording audio signals are converted to variable light intensity. This variable light intensity is recorded either on film or CD by different methods.

vi) State the application of Tie-Chip microphone (any two).

Ans: [Application of Tie Clip microphone-1M each, (any two)]

- Tie clip microphone is used for lecturers
- It is used as radio (wireless) microphones in sports meets.
- It is used in small P.A system for clubs and small halls.
- It is used in sound level meters.

vii) What is difference between parametric and graphic equalizer (any two)

Ans: (Any two points- 2M)

Sr. No.	Parametric equalizer	Graphic equalizer
1.	It provide variable boost or cut up to about 15dB.	Each band has individual slider control which can cut or boost the signal from +15 to -15dB.
2.	The parameters like frequency and bandwidth are varied throughout the audio spectrum (16Hz to 20KHz)	Here, complete audio spectrum is divided into narrow bands.
3.	The frequency response can be adjusted very precisely and selectively	The shape of the response curve is given by joining the slider positions

viii) State applications of Hi-Fi Audio amplifiers (any two).

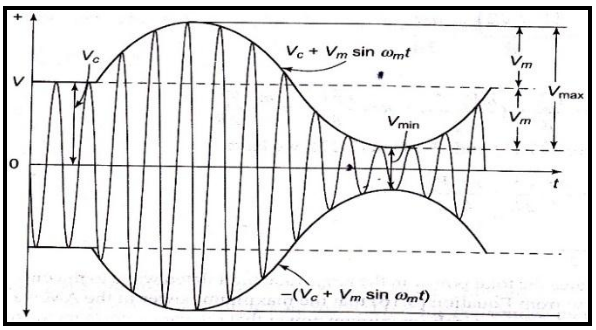
Ans: (Application of Hi-Fi Audio Amplifiers (any two)- 1M each)

1. Hi-fi audio system
2. PA system.
3. Sound recording Studios.
4. Theaters.

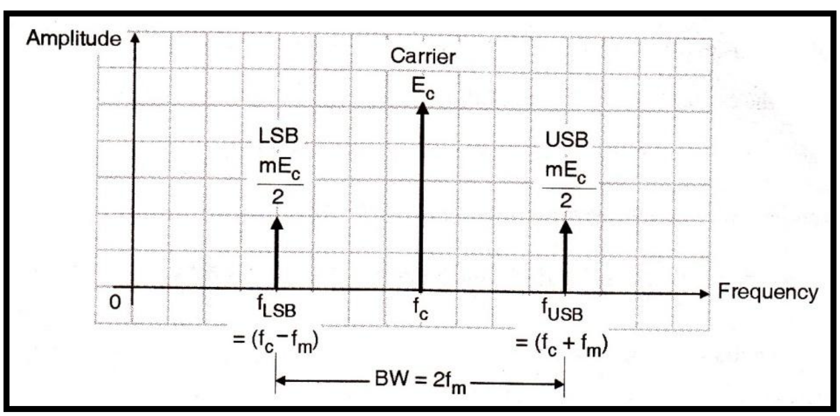
B) Attempt any two. (8M)

i) Draw a neat labeled time domain and frequency domain representation of AM wave.

Ans: <u>Time domain representation of AM:</u> (2M)



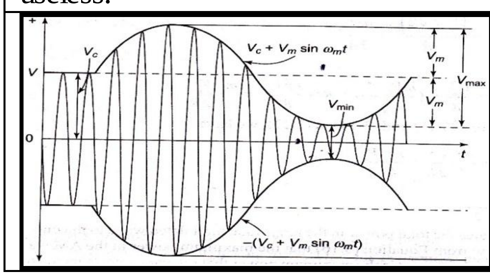
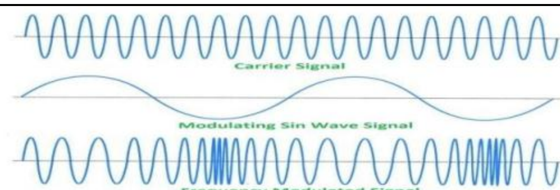
<u>Frequency domain representation of FM:</u> (2M)



ii) Compare FM and AM (Four points).

Ans: (Comparison - 1 M for each point, Any 4 points)

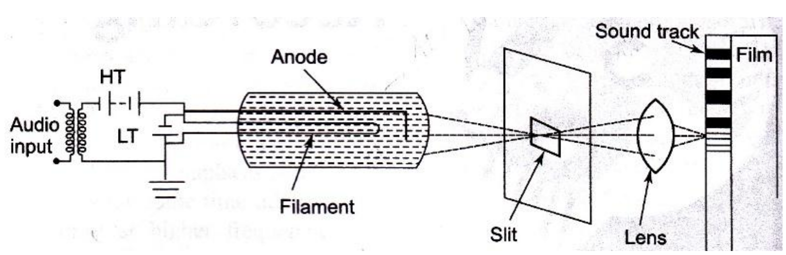
Comparison:

AM	FM
AM signal have low noise immunity	FM is higher noise immunity compared to AM.
AM modifies the amplitude of the carrier frequency	FM modifies the frequency of the carrier
AM is much more simpler compared to FM	FM is much more complex compared to AM
ground wave & sky wave propagation is used therefore large area is covered than FM	space wave is used for propagation do radius of transmission is limited to line of sight.
AM is more prone to signal distortion And degradation	FM signal doesn't degrade as easily as AM
Applications: Radio & TV broadcasting,	Application : Radio & TV broadcasting, police wireless, point to point communication
Bandwidth Required for AM is Twice the highest modulating frequency (less as compared to FM)	Bandwidth is Twice the sum of the modulating frequency and the frequency deviation. (20 times More as compared to AM)
Carrier power & one sideband power are useless.	All the transmitted power are useful.
	

iii) Describe the variable density optical recording of sound with diagram.

Ans: (Diagram-2M, Explanation: 2M)

Variable density method:



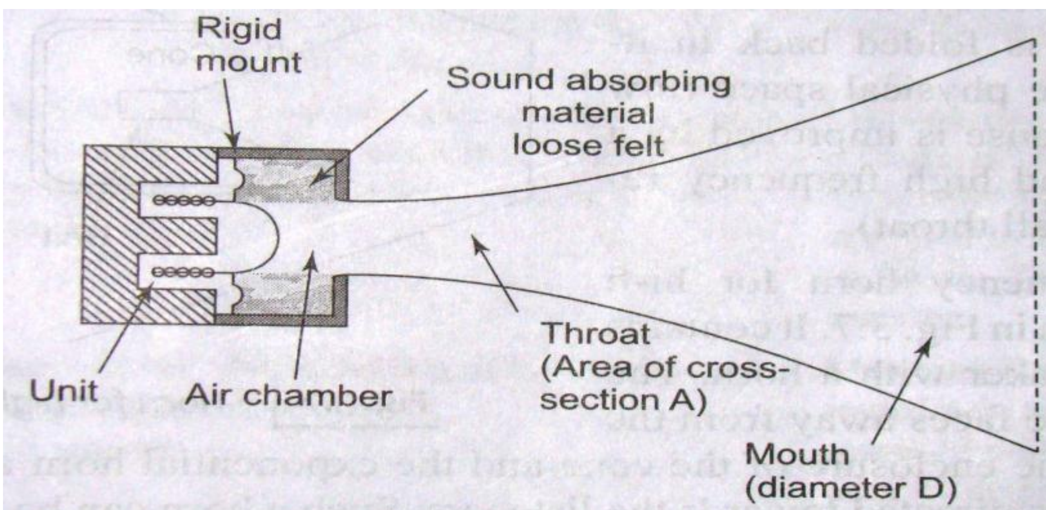
Explanation:

- In this method, sound is picked up by a microphone and converted into electrical signals which are amplified by audio amplifier & is fed to the anode of a special type of vacuum tube, called an AEO lamp. This lamp consists of a little quantity of helium gas.
- High DC voltage (HT) is applied to the anode in series with the audio voltage
- The filament of the lamp is connected to the low DC voltage (LT)
  The intensity of light coming from lamp varies in accordance with the audio signal. This varying light passes through a slit and a focusing lens. The focused light falls on a moving photographic film where the image is recorded in the form of bars of varying density and distance on the film.

Q2. Attempt any four. (16M)

i) Explain the working principle and construction of horn type Loudspeaker.

Ans: (Diagram- 2M, Working Principle -2M)



Working Principle:-

- A horn type loudspeaker uses a moving coil placed in a magnetic field similar to paper cone type, but instead of radiating acoustic power direct in open space of the listener's area, the power is first delivered to the air trapped in a fixed non vibrating tapered or flared horn and from there to the air in the listener's area.
- Thus it radiates sound power to the air in the space not direct from the diaphragm but indirectly through the horn.
- This is the reason why the horn type loudspeaker is called indirect radiating loudspeaker. The horn does acoustically what the cone does mechanically.

ii) Explain Dolby-A system for noise reduction with suitable diagram.

Ans: (Diagram- 2M, Explanation -2M)

- Dolby A was the company's first noise reduction system, presented in 1966.
- The output of four separate units is added. All this is done in side branch, and this branch is known as differential network. The output of differential network goes to the main branch as shown in fig. the output of adder is the Dolby processed signal.
- In playback, the differential network separates out the boosted signals in the side branch & subtracts from the input signal as shown in fig.

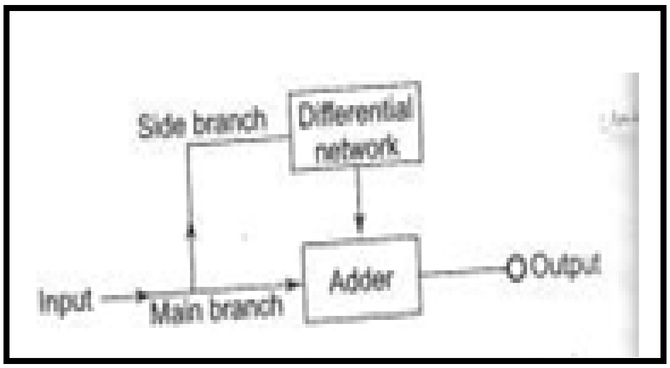


fig.(a) Coding of signal in Dolby method

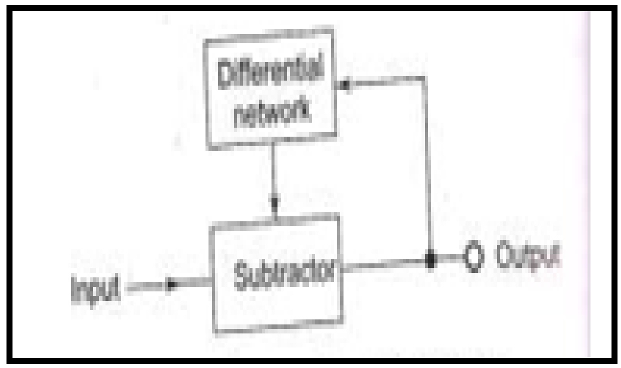


fig. (b). Decoding of Dolby signal

Explanation:

- Boosting is done in 4 bands:

1. Below 80Hz
2. 80Hz to 2999Hz
3. 3000 Hz and above
4. 9000 Hz and above

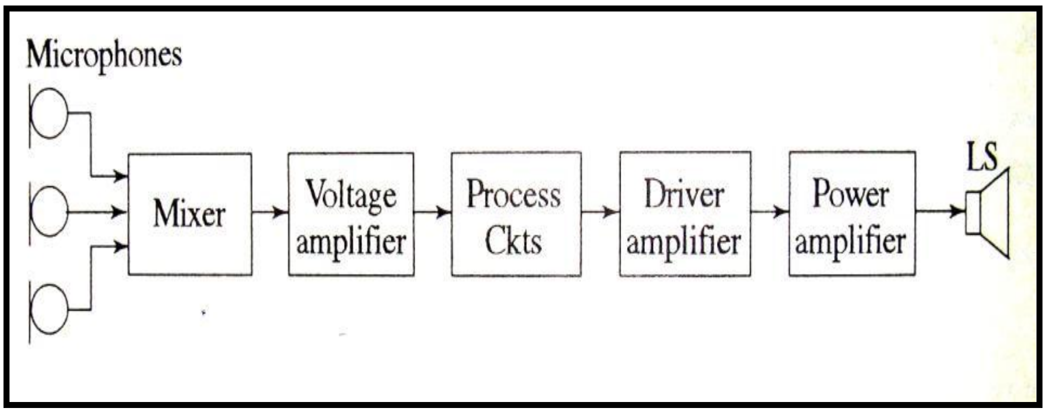
- Each band is processed separately by using low-pass, band-pass and high-pass filters and limiters.
- The 16 Hz to 80Hz signal goes to a low pass filter which causes improvement in signal to noise ratio with respect to hum and rumble.
- The 80Hz to 2999Hz signal goes to a band pass filter which deals with the mid band noise.
- Most of the sound energy for music is concentrated in this band. The 3000Hz and 9000Hz high pass filters improve signal to noise ratio with respect to hiss and modulation noise.
- The output of the four separate units is added. All this is done in a side branch, and this branch is known as the differential network.
- The output of the differential network goes to the adder of the main branch as shown in fig.(a)

- In playback, the differential network separates out the boosted signals in the side branch and subtracts them from the input signal as shown in fig. (b).

iii) Draw block diagram and explain the working of public address system.

Ans: (Diagram-2M, Explanation-2M)

Diagram:-



Block diagram PA system

Explanation:

1. Microphone - It picks-up sound wave and convert them to equivalent electrical signal called audio signals. Generally 2 or more microphones are used and in addition, an auxiliary input for tape/record player CD player.

2. Mixer- The output of microphones is fed to mixer stage. The function of the mixer stage is to effectively isolate different channels from each other before feeding to main amplifier. Function of preamplifier & amplifiers to amplify weak signals.

3. Voltage amplifiers- Amplifies the output of mixer stage.

4. Processing circuit- These circuits have master-gain control (volume control) and tone control Circuit.

5. Driver amplifier - It gives voltage amplification to the signal to such an extent that when feed to power amplifier (next stages) the into internal resistance of that stage is reduces. Thus drives the power amplifier to give more power.

6. Power amplifier - it gives desired power amplification to the signal generally push pull amplifier is used, so that harmonics are eliminated from the output and transformer core us bit saturated, The output of the power amplifier is connected to the loudspeaker through a matching transformer to match the low impedance of the L.S for max transfer of power

7. **Loudspeaker-** Converts electrical signal into pressure variation resulting in sound.

iv) Explain the principle of reproduction of sound from a recorded film.

Ans: (Diagram-2M, Explanation-2M)

Diagram:-

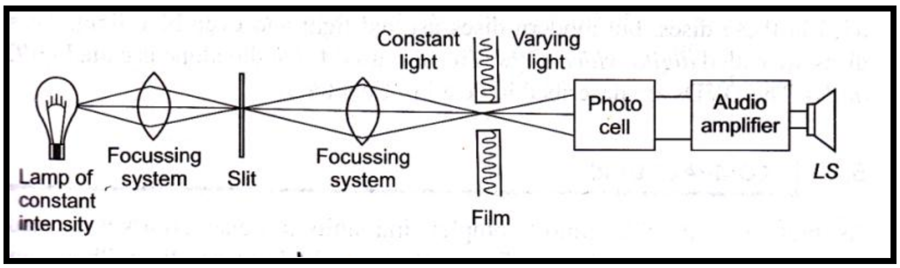


Fig. reproduction of sound

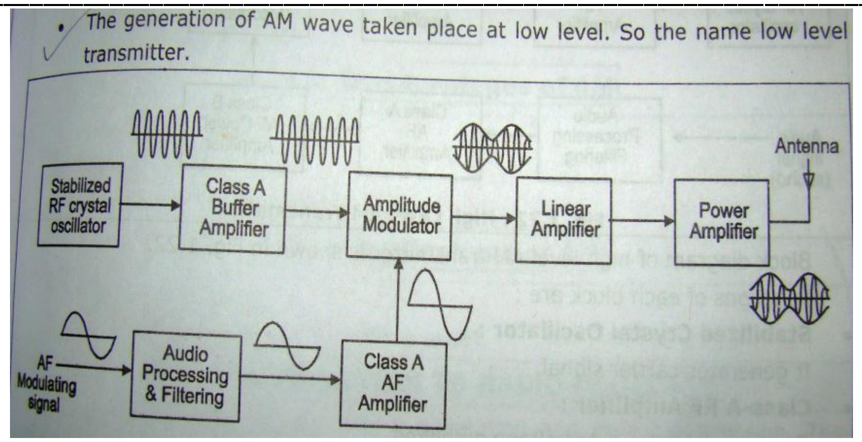
Explanation:

- A sharply focused narrow beam of light is made to fall on the soundtrack of film.
- As the film moves, light passing through bright and grey shaded portion in case of a variable-density record and through bright portions of variable area in case of a variable-area record, fall on a photocell which converts this light into electrical signals. In both types of recording (variable density as well as variable area), the quantity of light falling on the photocell will depend on the strength of the recorded audio signal.
- The output of the photocell will, therefore, be an audio voltage which can be amplified and fed to a loudspeaker which finally converts it into sound.
- The principle of reproduction is illustrated in above figure.

v) Draw the block diagram of a low level AM transmitter and explain its operation.

Ans: (Diagram-2M, Functions: 2M)

Block diagram of low level AM transmitter:-



Definition: In the modulation takes place prior to the output element of the final stage of transmitter.

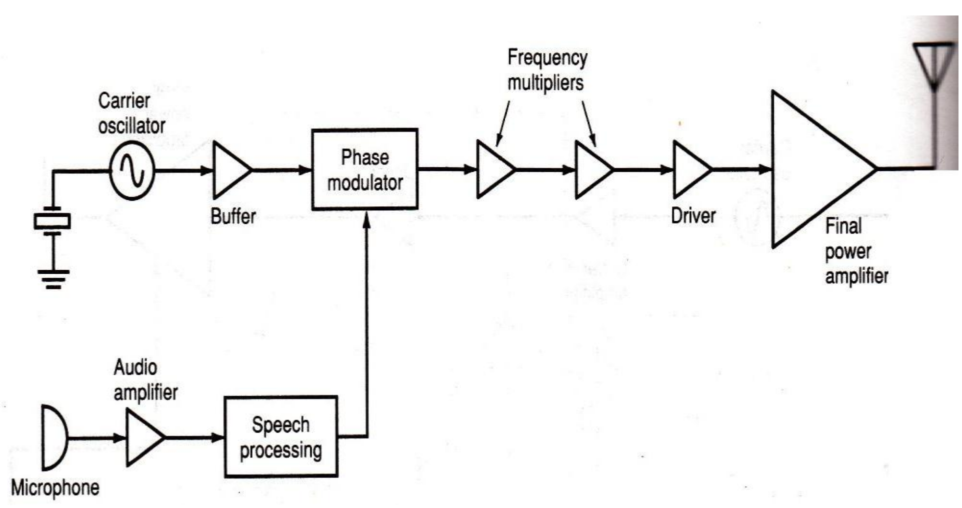
Functions of each block are :(Low level transmitter)

Stabilized RF crystal oscillator: RF oscillator generates the carrier signal. RF oscillator is stabilized to maintain carrier frequency deviation in limit

- Buffer amplifier : Carrier signal is amplified
- Audio processing and filtering: Sound information converted in to electrical signal. It is processed and filtered
- Class –A AF amplifier : It amplifies the modulating signal
- Amplitude modulator: Modulating and carrier signal applied to modulator to generate AM wave
- Linear amplifier: It avoids wave form distortion if any
- Power amplifier: Power of AM wave is amplified or increased
- Transmitting antenna: AM wave of high power is transmitted in free space

vi) Draw the block diagram of Armstrong frequency modulator and explain its operation.

Ans: (Block Diagram-2M, Explanation-2M)

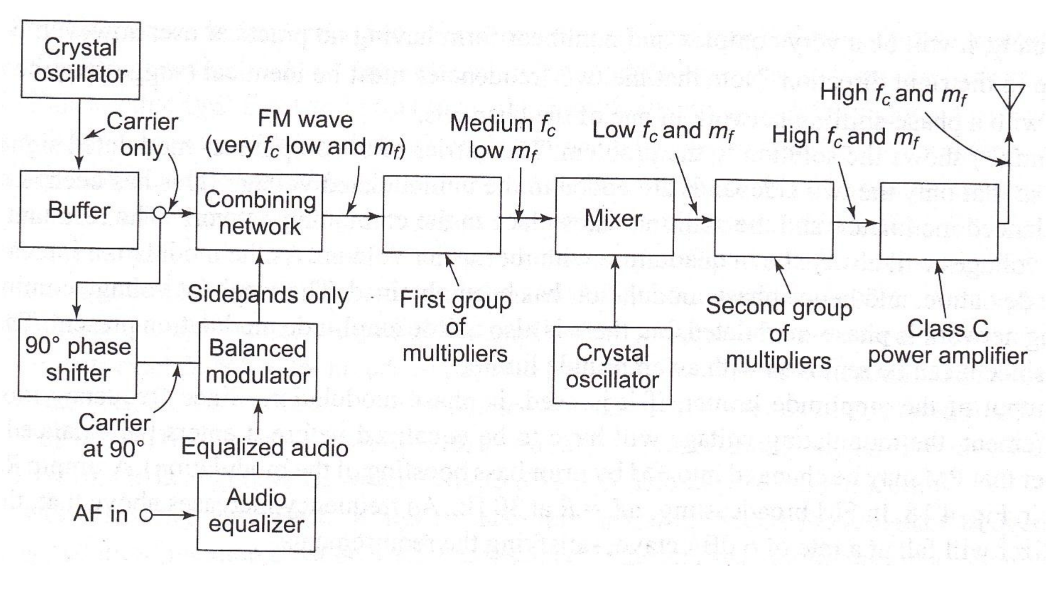


Explanation:

The indirect method of frequency modulation generation is used.

- A stable crystal oscillator is used to generate the carrier signal and a buffer amplifier is used to isolate it from the remainder of the circuitry
- The carrier signal is then applied to a phase modulator.
- The voice input is then amplified and processed to limit the frequency range & prevent over deviation. The modulator output is desired FM signal.
- Most FM transmitter are used in the VHF and UHF range and crystal are not available to generate those frequencies directly as result, the carrier is usually generated at frequency considerably lower than the final output frequency.
- To achieve the desired output frequency one or more frequency multipliers stage are used.
- A frequency multiplier is class C amplifier whose output frequency is some integer multiple of the input frequency by a factor 2, 3, 4 & so on. Because of class C amplifier provides a modest amount of power amplification.
- The frequency multiplier not only increases the carrier frequency to the desired output frequency but is also multiplies the frequency deviation produced by the modulator.
- After the frequency multipliers, a class C driver amplifier is used to increase the power level sufficiently to operate the final power amplifier.

OR



- The crystal oscillator generates the carrier at low frequency typically at 1 MHz this is applied to the combining network at 90 degrees phase shifter
- The modulating signal is passed through to an audio equalizer to boost the low modulating frequency. For the reason, high frequency modulating signals are attenuated but there is no change in the amplitudes of low frequencies modulating signals. Because in FM the frequency deviation is proportional to the modulating voltage regardless of its frequency.
- The balanced modulator produces two sidebands such that their resultant is 90 degrees phase shifted with respect to the unmodulated carrier.
- The unmodulated carrier and 90 degrees phase shifted side band are added in the combining network to generate FM wave. This FM wave has low carrier frequency F c and low value of modulation index mf.
- The carrier frequency & modulation index are raised by passing through FM to the first group of multipliers.

- The FM signal with high F_c and high m_f is then passed through class C power amplifier to raise the power level of FM signal.

Q3. Attempt any four: (16 M)

(i) Define modulation index of an AM wave and give the mathematical representation of AM wave.

Ans: (Definition –2M, Equation-2M)

Modulation Index: It in AM is defined as the ratio of amplitude of modulating signal to the amplitude of carrier signal.

$$m = \frac{V_m}{V_c}$$

Mathematical expression for amplitude modulated wave

$V_{AM} = V_c \sin \omega_c t + mV_c /2 \cos (\omega_c - \omega_m)t - mV_c /2 \cos (\omega_c + \omega_m)t$

(ii) A modulating signal 10 sin ($2\Pi x10^3 t$) is used to modulate a carrier signal 20 sin($2\Pi x10^4 t$). Find the modulation index, frequency of side band components and their amplitudes. What is the band width of the modulated signal?

Ans:

Given: **(1M)**

V_m= 10v

V_c= 20v

F_m= 10^3Hz.

F_c= 10^4 Hz.

Modulation Index $m_a = V_m / V_c$

$= 10/20 = 0.5$

$= 50\%$ **(1M)**

LSB frequency = $F_c - F_m = (10-1)\times 10^3$ = 9KHz. **(½M)**

USB frequency = $F_c + F_m = (10+1)\times 10^3$ = 11KHz. **(½M)**

Amplitude of side bands = $m_a V_c / 2 = 0.5\times 20/2 = 1$V **(1M)**

(iii) Explain the generation of FM wave using varactor diode.

Ans: (Diagram- 2M, Explanation- 2M)

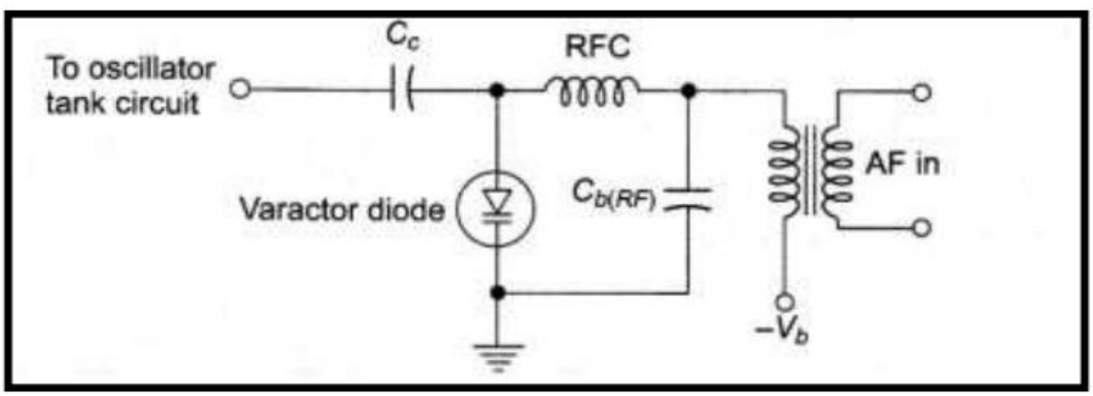


Fig. Generation of FM wave using varactor diode modulator

- 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 fig 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.

(iv) A 10 kW carrier wave is amplitude modulated at 80% depth of modulation by a sinusoidal modulating signal. Calculate sideband power and transmission efficiency of the AM wave.

Ans:

Given: **(1M)**

P_c =10 KW

m = 0.8

Side band power

$P_{LSB} = P_{USB} = P_c\, m^2/4$ **(1M)**

$= [10x10^3 x(0.8)^2]/4$

= 6.4KW **(1M)**

Efficiency $= [P_{LSB} + P_{USB}] / P_t$ **(1M)**

= [6.4+ 6.4]/ [6.4+6.4+ 10] = [12.8]/ [22.8]

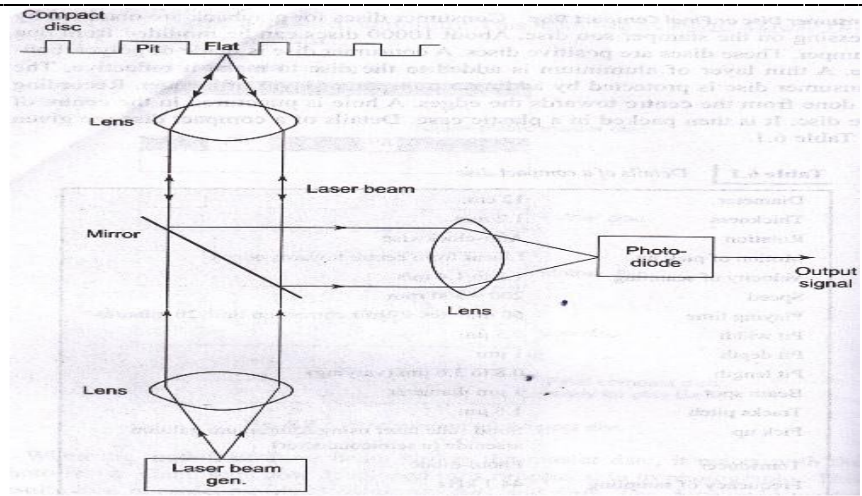
= 0.56 **OR**

Efficiency (η) = 56%

(v) Draw the block diagram of detection circuit in a compact disc player and explain its operation.

Ans: (Diagram- 2M, Explanation- 2M)

Diagram: (2M)

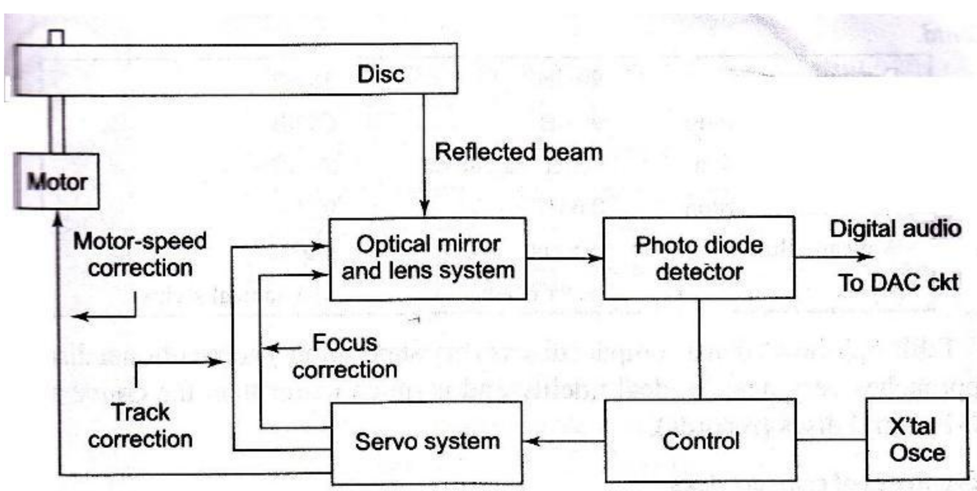


Explanation: **(2M)**

A layer beam is incident on the compact disc through a half silver mirror. The returning beam is reflected from the aluminum flat surface and represents the logic 1. There is only a little reflection from a pit and it represents logic 0.

The binary digits are reproduces when this ON-OFF reflected light falls on a photosensitive diode. The digital output of the diode is analog signal by using digital to analog converter.

OR



Explanation:

- Detection in optical recording is equivalent to playback process. In this a laser beam produced by a solid state laser of semiconductor aluminum gallium arsenide is made incident on the CD through half silver mirror the mirror allows the beam to pass through itself but does not allow the returning beam to pass.
- The returning beam is reflected from the aluminum flat surface & represents digit 1.there is only little reflection from a pit & it represents 0. Thus the laser beam is the replica of the original laser beam modulated by digits of audio signal.
- Light is not reflected from the pit fully reflected from flat surface. Thus binary digits are reproduces when this ON-OFF reflected light falls on a photodiode.

- The digital output of photodiode is processed & converted into the original signal by using DAC
- Control signals allow any combination of track to be played in any sequence with the help of keyboard.
- A clock signal is obtained from the disc itself. It is compared with a crystal oscillator signal. Any discrepancy result in generation of a correction signal which is applied to the servo system.

(vi) Define phase modulation. Write the expression of modulation index of a PM wave.

Ans:

Phase modulation: **(2M)**

The phase shift of the carrier signal is varied in proportional with the amplitude of the modulating signal. The amplitude of the carrier remains constant.

Modulated index: **(2M)**

The modulating index is defines as:

Mp = δp is expressed in radiance

where δp is maximum frequency deviation.

Q4. Attempt any FOUR: (16M)

(i) State the mathematical representation of a FM wave. Define modulation index and frequency deviation in FM wave.

Ans: (Expression-2M, Definition -1M each)

Mathematical representation of FM

$$F = f_c + k_f V_m \sin\omega_m t$$

Where,

f_c = unmodulated carrier frequency

k_f = proportionality constant

$V_m \sin\omega_m t$ = Modulating signal

Modulation index: The modulation index for FM, m_f is defined as

$$m_f = \frac{(maximum)\ frequency\ deviation}{modulating\ frequency} = \frac{\delta_f}{f_m}$$

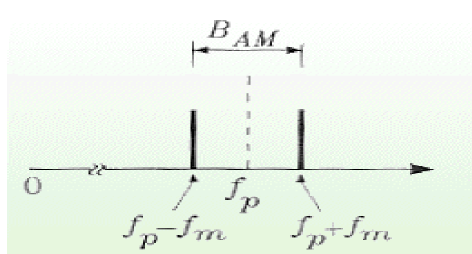
Deviation: The maximum instantaneous difference between an FM modulated frequency and the nominal carrier frequency.

(ii) What is DSBSC? Draw its time domain and frequency domain representation.

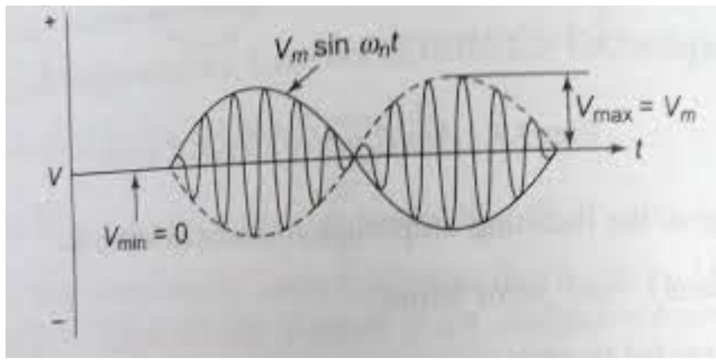
Ans: (Definition- 2M, Frequency Domain- 1M, time domain – 1M)

DSBSC- In AM when both the side bands are transmitted and carrier signal is suppressed before transmission in order to save power, then the signal is called as Double Side Band Suppress Carrier (DSBSC) Signal.

Frequency Domain Representation of DSBSC



Time Domain Representation of DSBSC:



(iii) State the need and application of Public Address system.

Ans: (Need- 2M, Application-2M)

Need:- **(2M)**

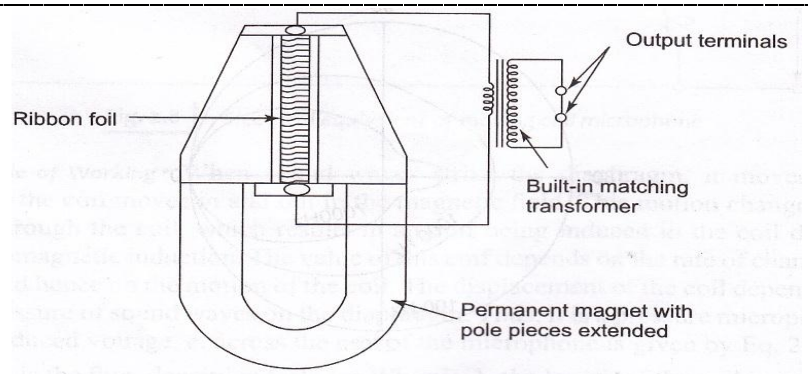
The intensity of sound decreases with distance. Hence when a large gathering is to be addressed sound needs to be amplified so that people at a distance from the rostrum or stage may receive good intensity of sound for comfortable listening. The system which fulfills this function is called public address system.

Applications: **(Any four- 2M)**

i. It is used in sports meet
ii. It is used in public meeting
iii. It is used in auditorium
iv. It is used in concerts & function.
v. It is also used to convey information to isolated locations as at railway station, airports, hospitals, factories etc.

(iv) Explain the construction and working principle of ribbon type microphone.

Ans: (Diagram-2M)



Explanation:

Construction:- **(1 M)**

- The main parts of ribbon microphone are permanent magnet, ribbon conductor
- The permanent magnet is specially designed horse shoe magnet with extended pole pieces. It provides a strong magnetic field.
- The ribbon is a light aluminum foil. it is corrugated at right angle to its length to provide greater surface area.
- The main feature is the lightness of the ribbon which is about 0.2 mg in weight less than 1 micron and about 20mm long and 3 mm wide.
- It is suspended in the magnetic field of the permanent magnet.
- The whole unit is enclosed in circular or rectangular baffle.

Principle of working:- **(1 M)**

- When the ribbon conductor placed in the magnetic field, is made to move at right angles to the magnetic field by the force of sound pressure, there is a change of magnetic flux through the ribbon conductor.
- Due to this change , an e. m. f is induced across the ribbon this e. m. f is proportional to the rate of change of magnetic flux which in turn proportional to the force of sound waves striking the ribbon.
- It is also called as pressure gradient or velocity microphone.

(v) Draw circuit diagram and explain the working of complementary symmetry push-pull amplifier.

Ans: (Diagram: 2M, Explanation: 2M)

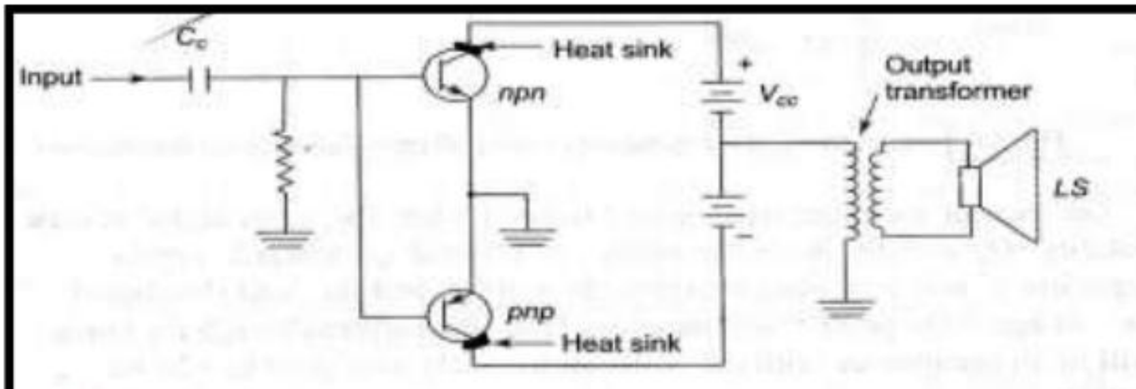


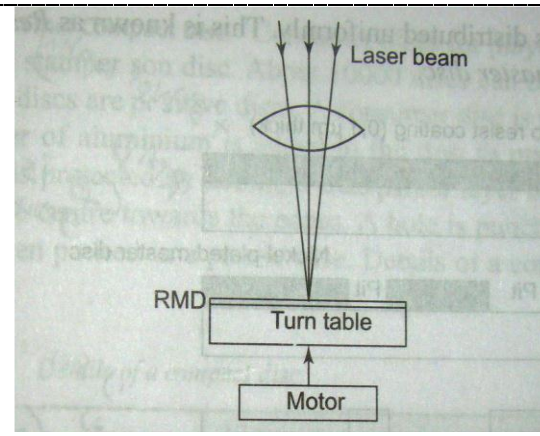
Fig. Complementary symmetry push-pull amplifier circuit with output transformer

Explanation:

- The circuit for a complementary symmetry push pull amplifier is shown in figure.
- It requires the same polarity at the input of two transistors.
- The circuit uses two transistors, one of NPN type and the other of PNP type.
- Input signals to the two transistors are in the same phase. (Inter-Stage transformer for input is not required.)
- The NPN collector gets positive dc voltage and the PNP collector, negative dc voltage.
- Direct current, through the primary of the transformer will be in the opposite directions. The audio currents from the two transistors will add in the primary and then will give all the advantages of push-pull configuration.

(vi) Explain the concept of optical recording on compact disc with block diagram.

Ans: (Diagram: 2 M, Explanation: 2 M)



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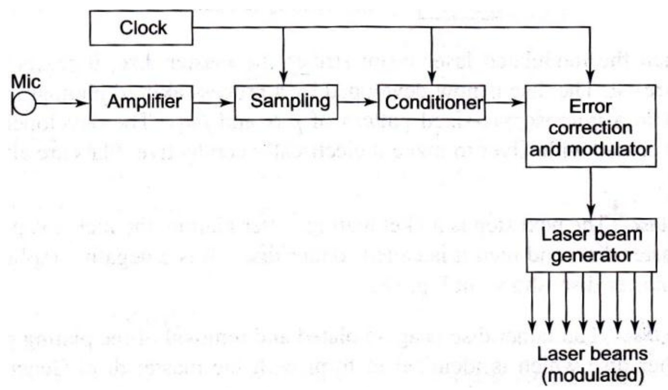


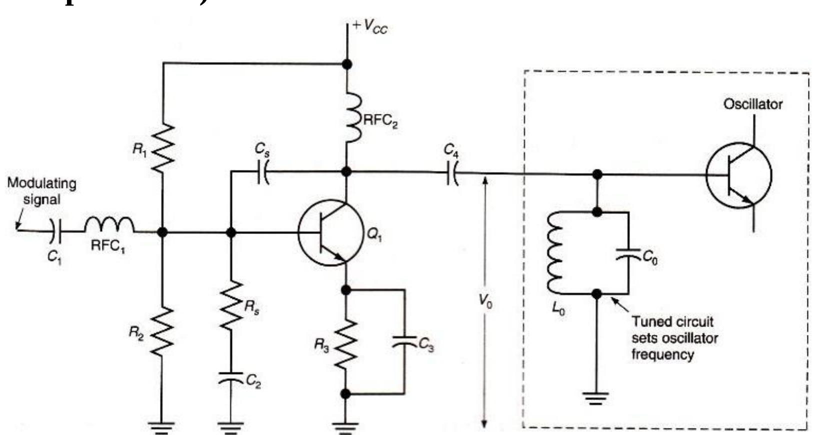
Recording on CD:

- This is done with the help of laser beams, made ON and OFF by digitized audio signals
- These beams fall on a photo resist material on a rotating disc and caused pits of varying width & fixed depth & thus records signals in binary form, flats & pits making 1s & 0s respectively.
- Recording is done on Resist Master Disc (RMD) with help of a powerful laser beam as shown as fig. The laser beam is modulated by digitized audio signals. The audio signal is sampled at rate of 44.1 KHz. the quantum level pertains to 16 bits.

Q5. Attempt any four:[16M]

i) Describe the generation of FM using reactance modulator.

Ans: (Diagram-2M, Principle: - 2M)



Principle:

In reactance modulator a transistor is operated as a variable reactance and it is connected across the tuned circuit of an oscillator. As the instantaneous value of modulating voltage changes, the reactance offered by the transistor will change proportionally. This will change the frequency of oscillator to produce FM wave.

Working:

The modulating signal is applied to the modulator circuit through C and RFC. The RFC helps keep the RF signal from the oscillator out of the audio circuit from which the modulating signal will vary the base voltage and current of Q will also vary in proportional.

As the collector current amplitude varies the phase-shift angle changes with respect to the oscillator voltage, which is interpreted by the oscillator as a change in the capacitance. So as the modulating signal changes the effective capacitance of the circuit varies and the oscillator frequency is varied accordingly the circuit produces direct frequency modulation.

ii) Explain the method for generation of DSBSC AM signal using diode balanced modulator.

Ans: (Diagram: 2M, Explanation: 2M)

- A non-linear resistance or non-linear device may be used to produce Amplitude Modulation i.e. one carrier and two sidebands.
- However, a DSB-SC signal contains only two sidebands. Thus, if two non-linear devices such as diodes, transistors etc. are connected in a balanced mode so that they suppress the carriers of each other, then only sidebands are left and a DSB-SC signal is generated .

- Therefore, a balanced modulator may be defined as a circuit in which two non-linear devices are connected in a balanced mode to produce a DSB-SC signal.

Fig. shows the balanced modulator using diodes as non-linear device.

The modulating signal x(t) is applied equally with 180^o phase reversal at the inputs of both the diodes through the input center tapped transformer .

The carrier is applied to the center tap of the secondary.

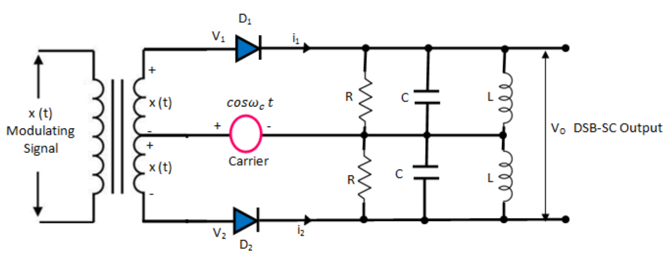
Hence, input voltage to D_1 is given by:

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

And the input voltage to D_2 is given by:

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

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



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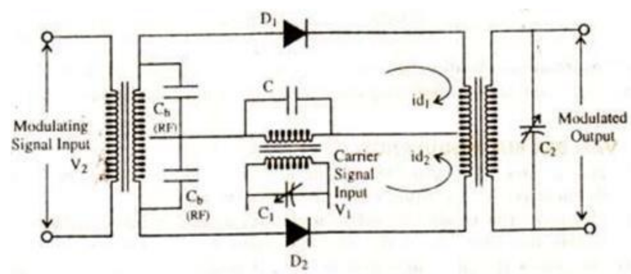


: Balanced Modulator circuit using diodes

The diode current i_1 and i_2 are given by:

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

The diode current i_1 and i_2 are given by:

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

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

$$i_1 = ax(t) + acos\omega_c t + bx^2(t) + 2bx(t)cos\omega_c t + bcos^2\omega_c t \qquad \text{\_\_\_\_\_\_\_\_(3)}$$

Similarly,

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

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

$$i_2 = av_2 + bv_2{}^2 = ax(t) - acos\omega_c t + bx^2(t) - 2bx(t)cos\omega_c t + bcos^2\omega_c t \qquad \text{\_\_\_\_\_\_\_\_(4)}$$

The output voltage is given by:

$$v_o = i_1 R - i_2 R$$

Substituting the expression for i1 and i2 from equations (3) and (4),we get

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

OR

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

Hence, the output voltage contains a modulating signal term and the DSB-SC signal .

The modulating signal term is eliminated and the second term is allowed to pass through to the output by the LC band pass filter section.

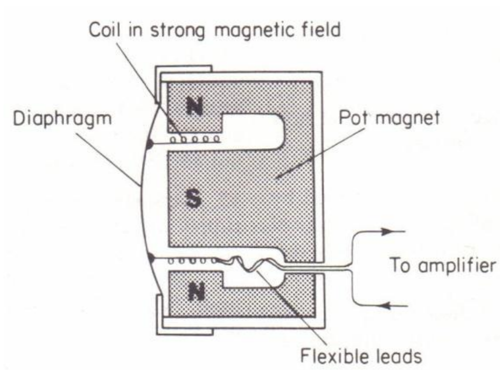
Therefore, final output = 4 b R x(t) cos ωct

= K x(t) cos ωct

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

iii) Explain the construction and working principle of moving coil microphone.

Ans: (Diagram: 2M, working: 2M)



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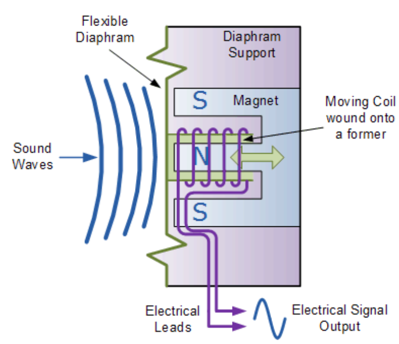


Working principle:

- Moving coil type microphone uses electromagnetic induction to convert the sound waves into an electrical signal. It has a very small coil of thin wire suspended within the magnetic field of a permanent magnet. As the sound wave hits the flexible diaphragm, the diaphragm moves back and forth in response to the sound pressure acting upon it causing the attached coil of wire to move within the magnetic field of the magnet.
- The movement of the coil within the magnetic field causes a voltage to be induced in the coil as defined by Faraday's law of Electromagnetic Induction. The resultant output voltage signal from the coil is proportional to the pressure of the sound wave acting upon the diaphragm so the louder or stronger the sound wave the larger the output signal will be, making this type of microphone design pressure sensitive.

iv) State the causes affecting fidelity and give the remedies for them.

Ans: (Causes-2M, Remedies- 2M)

Causes affecting fidelity: (Any four- 2M)

- High signal to noise ratio(S/N ratio)
- Flat frequency response
- Low nonlinear distortion
- Large dynamic range
- Creating sense of direction.

Remedies: (Any four -2M)

1. S/N ratio can be improved by using preamplifier of low noise figures proper shielding, grounding, Decoupling & filtering circuits, stabilized power supply, microphones
2. By using coupling capacitor and shunt capacitor in audio amplifier circuits
3. Nonlinear distortion can be reduced by using negative feedback in amplifier, designing bias circuit to keep Q point in the middle of linear portion of the characteristics curve.
4. Dynamic range can be increased by using solid-state amplifier; dynamic microphones & L.S. which are capable of withstanding the large change in loudness.
5. Creating sense of direction can be improved by using high fidelity system.

v) State and explain the selection criterion of microphones.

Ans: (Any four, 1 M each)

For Selection criteria of microphones, microphone should have

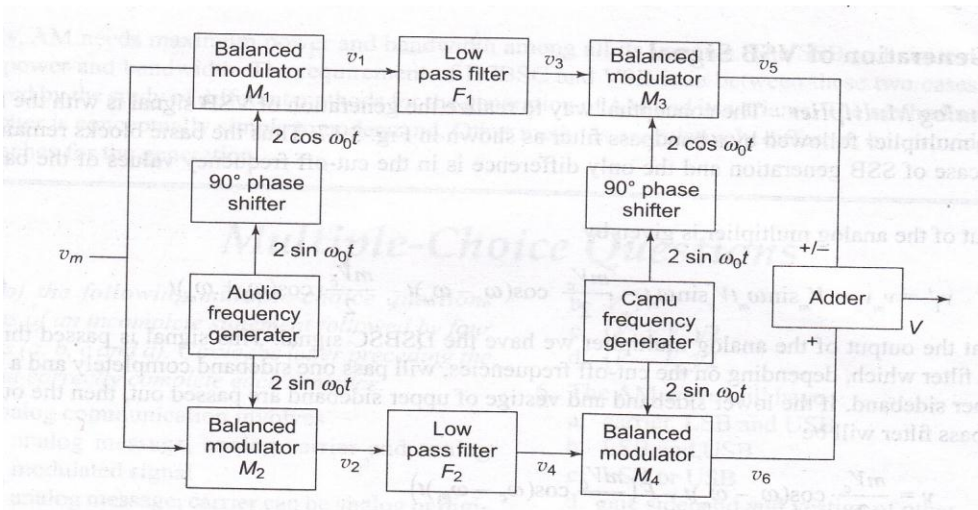
1. Sensitivity: It is an electrical output from microphone at certain sound pressure level. It is defined as output in milivolt for the sound pressure of 1 Pascal at 1 KHz.
2. S/N ratio: It is the ratio of level of the desired signal that a microphone records compared to the level of noise that it picks up from the background.
3. Frequency response: It is a plot of frequency vs. gain. which gives the flat frequency response for particular band of frequency for which it is designed
4. Distortion: It is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency.
5. Directivity: It is a response measured for various frequencies. Directivity is also called field pattem, polar pattern, or directional characteristic, and the major types of Directivity patterns are:

    - Cardioids or uni-directional
    - Bi-directional or figure-of-eight
    - Omni-directional

- **O/p impedance:** It specifies what load resistance is needed for the microphone to operate as designed. This value is generally at least 10 times greater than the internal source resistor of the microphone (open circuit).

vi) Explain third method for generation of SSB AM with suitable diagram.

Ans: (Diagram-2M, Explanation-2M)



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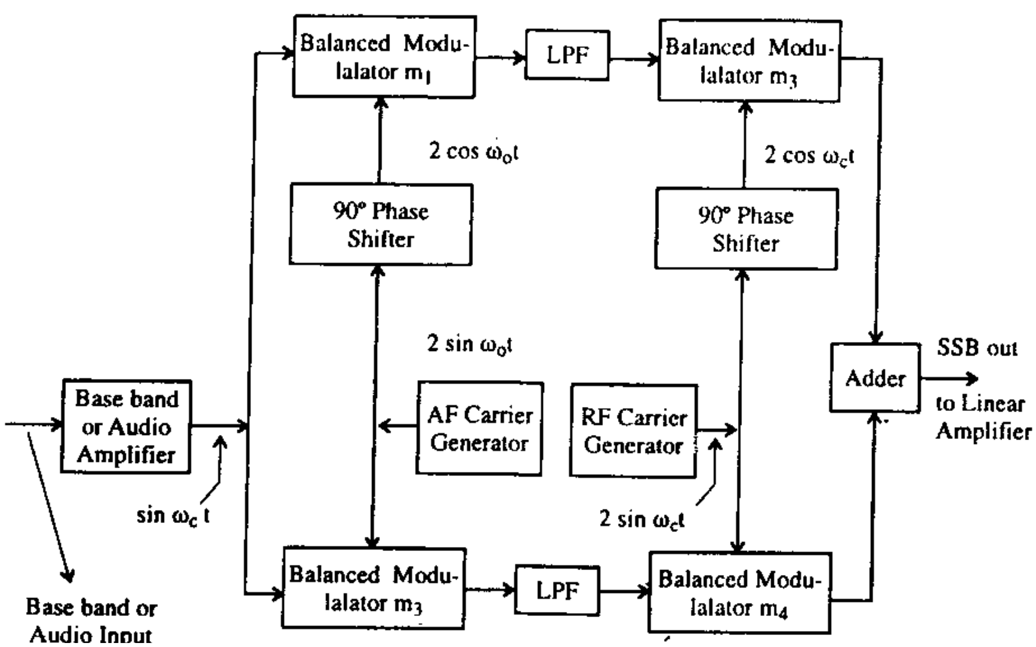


Explanation: -

- It was developed by weaver to retain the advantage of the phase shift method, such as its ability to generate SSB at any frequency and use low audio frequencies.
- It is very complex and not often used method commercially.

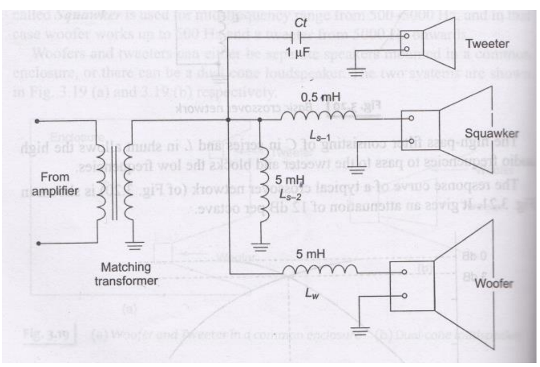
- The later part of the block diagram is identical to phase shift method, but the way in which the appropriate voltages are fed to the last two balanced modulators at points C & F has been changed that is instead of phase shifting the whole range of audio frequencies, this method combines them with an AF carrier Fo which is the fixed frequency in the middle of audio band.
- A phase shift is then applied to this fixed frequency only.
- The resulting voltage at the output of the balanced modulators M1 and M2 are applied to low pass filters whose cut off frequency is designed to be f0 to ensure that the input to the last stage of the balanced modulators i.e. M3 and M4 results in proper side band suppression.
- If a lower sideband signal if required at the final output the phase of the carrier voltage being fed to M1 should be changed by 180^0

Q6. Attempt any four: [16M]

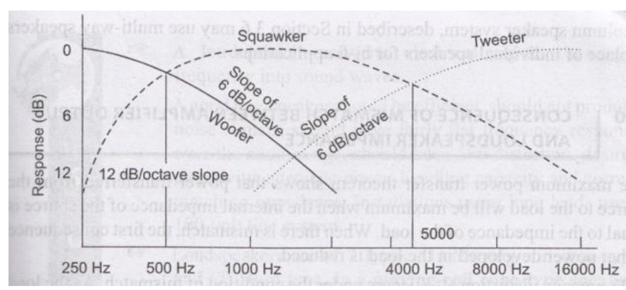
i) Explain the concept of multiway speaker with suitable diagram.

Ans: (Diagram-2M, Explanation-2M)

Diagram:-



Three way cross over network



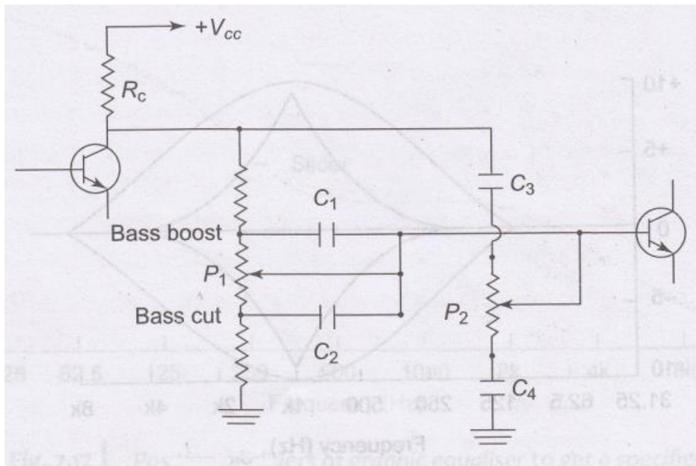
Response curve (*Optional to Draw*)

Explanation:

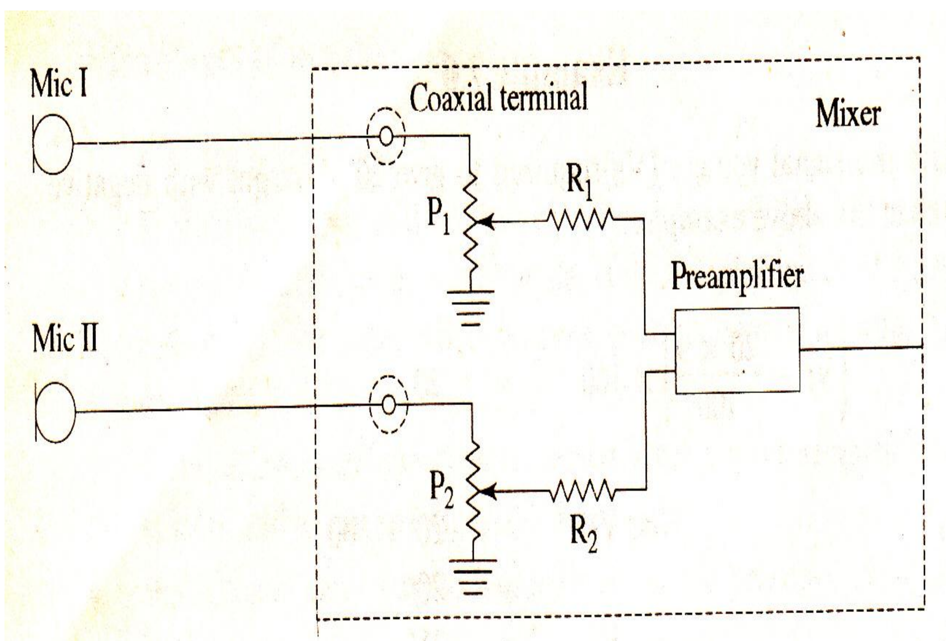
- When multi-way speaker system is used to get flat frequency response for the entire range of audio Frequency it is essential to have a cross over network to divide the incoming signal into separate frequency ranges for each spectrum.
- In absence of cross over network, the speaker will suffer overheating and output will be distorted when full power at frequencies outside the range in fed to them.
- As well as overall efficiency will be much reduced.
- Ct of 1µf in series with tweeter prevent 100 and mid frequencies reaching the tweeter. L_W of 5mH in series with woofer prevents high and mid frequencies reaching to woofer.
- L_{S1} and L_{S2} allows only mid frequency range to reach to squawker

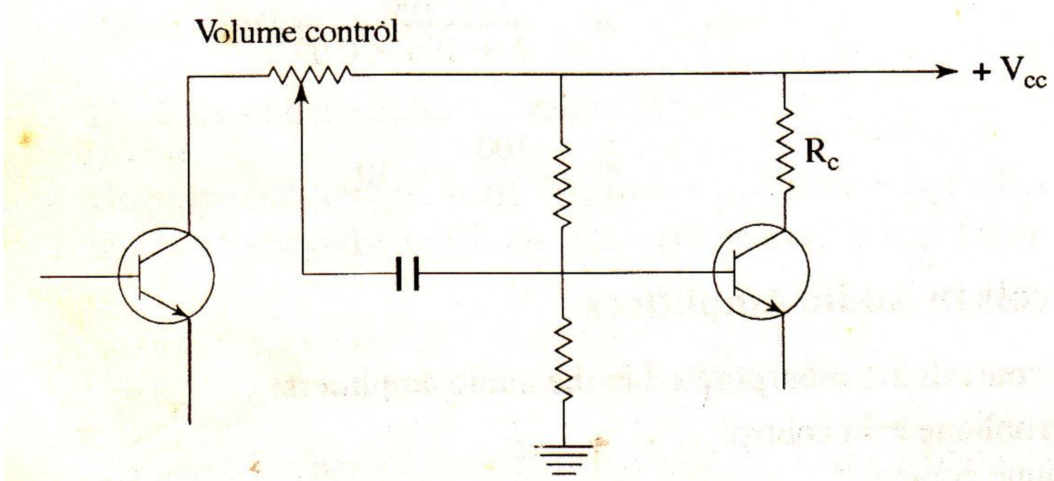
ii) Draw the circuit of microphone gain control, volume control and tone control (Bass and treble)

Ans: (Diagram- Microphone gain control: 1M, volume control: 1M, tone control (Bass and treble): 2M)



Bass and treble control



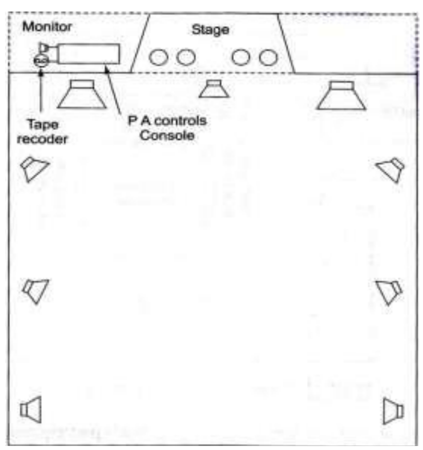


iii) Explain the planning and installation steps of a typical public address system.

Ans: (Diagram-2 M, Explanation-2 M)

(NOTE: Marks to be given if Students writes the answer considering any example of application of PA system like for auditorium, football stadium, public meeting, conferences, cultural Program etc)

Diagram:



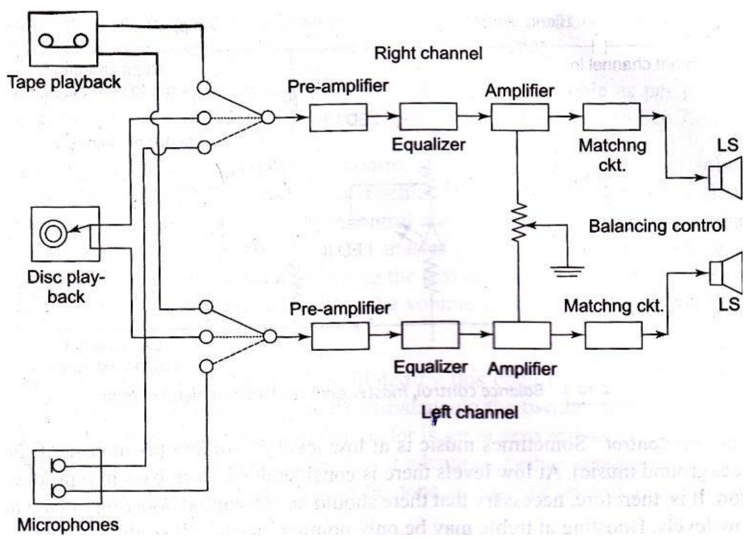
<u>Explanation:</u>

- An auditorium may be used for wide range of activities like public meeting, conferences, cultural
- Program etc. Hence the loudspeakers system should have a wide dynamic range and good frequency
- Coverage from 20 Hz to 16 KHz.
- Columns of loudspeakers having good bass and treble response should be mounted facing towards the front on the either side of the stage. If the hall is wide, a small column may also be mounted in the center of the front line. Another pair of small columns slightly inclined may be placed at about one third and two third ways down the hall from the front. The fourth pair placed last need not be inclined.

- A separate versatile mixer unit is desirable. It may have tape and disc input and several microphone inputs. The amplifiers should be 50 to 100 watt
- Microphone should be of moving coil type.
- If the hall has a line of pillars small loudspeakers may be mounted on the pillars facing the audience.
- The back wall should be furnished with good sound absorbing material.
- The height of the side speakers should be such that their axes point towards ear of the audience and not towards the flooring or ceiling.

iv) Draw and explain the block diagram of a Hi Fi system.

Ans: (Diagram- 2M, Explanation-2M)



Explanation:

- Fidelity means faithfulness. In audio system it is used to indicate faithful reproduction of sound. Figure shows block diagram of Hi- Fi system
- High fidelity sound can be obtained from the recorded stereo tape or in live system from the microphone or from record player.
- The stereo signal is fed to two independent amplification channels through a tape-mic switch . The amplifier consists of a low noise high gain preamplifier, equalizer, well designed amplifier giving flat frequency response & little distortion by using negative feedback circuit & then the matching transformer.
- A balancing circuit is incorporated to balance out any imbalance in the characteristics of identical circuits.
- The secondary of the matching transformer of each channel is connected to the respective loudspeaker column.
- For hi fi the L.S columns consisting of woofer, squawker & tweeter are used.

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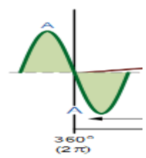
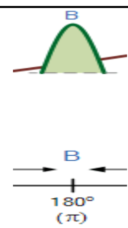
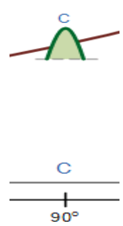
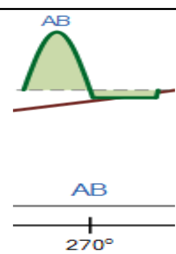
v) Explain classification of Audio amplifier on the basis of their efficiency and output waveforms.

Ans: (Classification of Audio amplifier: 1M each)

Note: If the student writes value of efficiency marks should be given.

Based on efficiency	**Output waveforms**
Class A power amplifier	
Class B power amplifier	
Class C power amplifier	
Class AB power amplifier	

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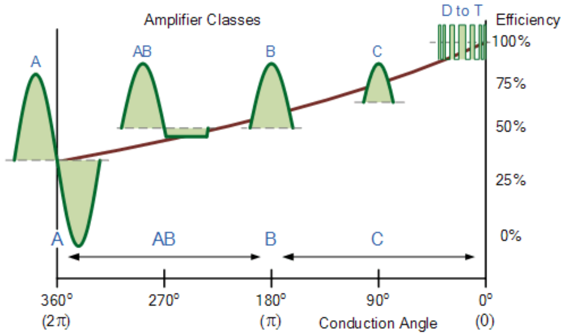


vi) Draw the block diagram of FM transmitter and explain its operation.

Ans: (Diagram-2 M, Explanation- 2 M)

Diagram:

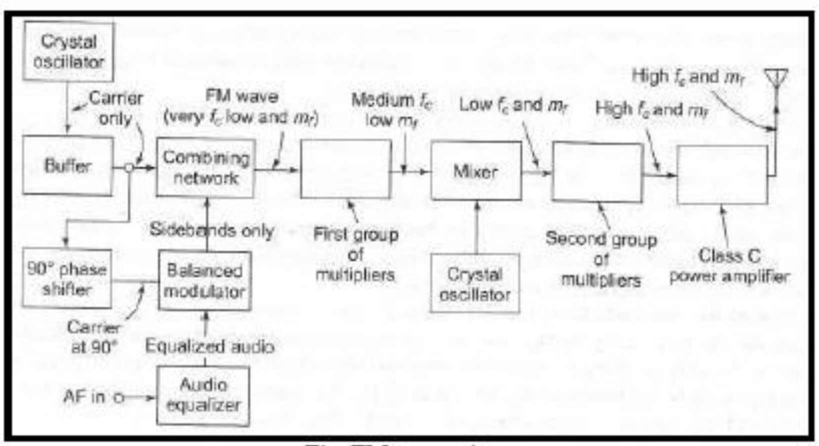
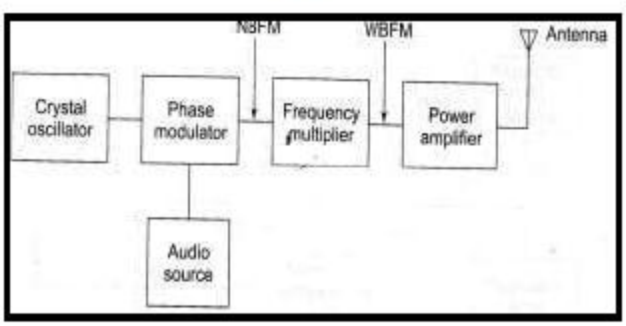


Fig. FM transmitter

OR



Explanation:

- The crystal oscillator generates the carrier at low frequency typically at 1 MHz This is applied to the combining network and a 90 degree phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies.
- The modulating signal is then applied to a balance modulator.
- The balance modulator produces two sidebands such that their resultant is 90 degree phase shifted with respect to the un-modulated carrier.
- The un-modulated carrier and 90 degree shifted sidebands are added in the combining network .The output of combining network is equivalent to FM wave. This FM wave has low carrier frequency Fc and low value of the modulating index mf.
- The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the Fc and mf both are raised to required high values using the second group of multipliers.
- The FM signal with high Fc and high mf is then passed through a class C power amplifier to raise the power level of the FM signal.