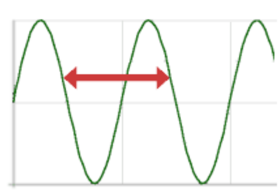
MODEL ANSWER

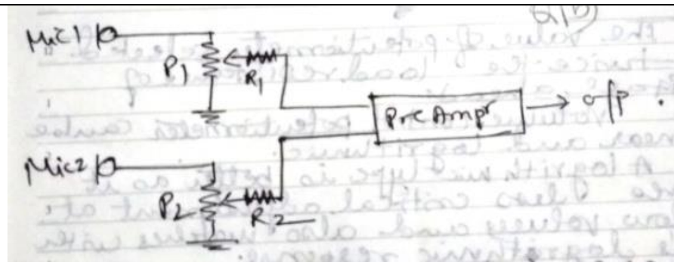
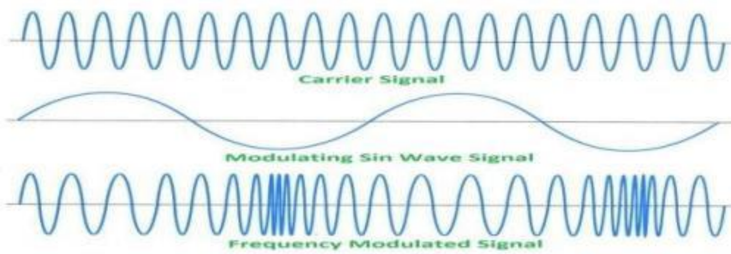
WINTER– 18 EXAMINATION

Subject Title: Fundamentals of Communication   Subject Code: 17316

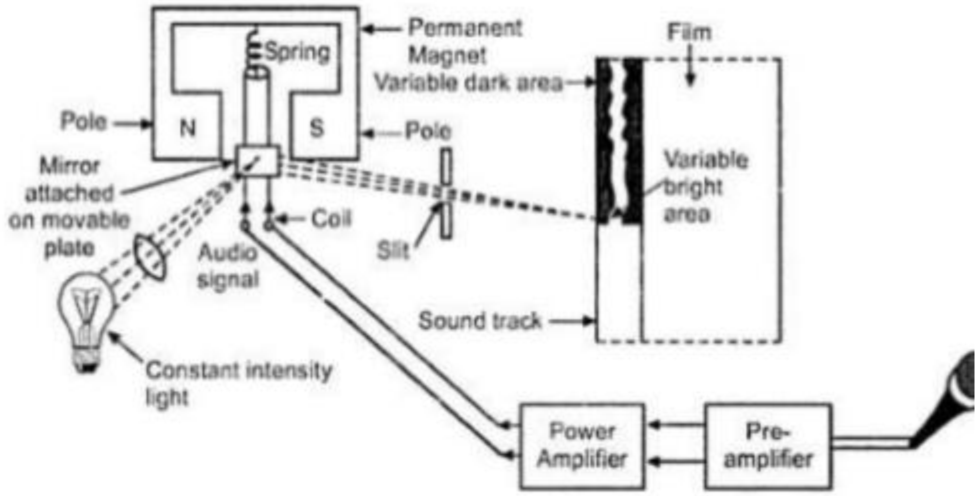
Important Instructions to examiners:

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 anyequivalent 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 judgement 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.

Q. No.	Sub Q.N.	Answer	Marking Scheme
Q.1		**Attempt any six :**	**12-Total Marks**
	a)	**Define amplitude and wavelength with respect to sound signal.**	**2M**
	Ans:	The maximum extent of a vibration as displacement of a sinusoidal oscillation measured from the portion of equilibrium is called amplitude. **Wavelength:** The length of space travelled by one cycle of variation is called wavelength and is represented in meters. Equation 1.3 gives the relationship between frequency (f in Hz), wavelength (λ in meters) and velocity (in meters per second). **V= fλ** 	**1M Each for Definition**
	b)	**Draw neat labeled circuit diagram of gain control in audio signal**	**2M**
	Ans:	It consist of a potentiometer in the output of microphone. The function of this control is to adjust the output of microphone depending upon the speaker's style of speaking.	**2M Diagram**

		 This control is present in pre-amplifier stage or mixer circuit.	
	c)	**Define frequency modulation and draw neat wave form of FM signal**	**2M**
	Ans:	This is the modulation technique in which frequency of carrier is changed according to amplitude variation in modulating signal. 	**1M Definition** **1M Waveform**
	d)	**List the different optical recording methods for sound recording**	**2M**
	Ans:	**1. Variable density optical recording** **2. Variable Area optical recording**	
	e)	**Define pre-emphasis and de-emphasis techniques**	**2M**
	Ans:	**Pre emphasis – this official boosting of higher audio modulating frequencies with preaarenged response curve is called pre-emphasis.** **De-emphasis – it is process which is used in receiver side to reduced the signal and get thje original signal.**	**1M Each**
	f)	**List the application of tie clip microphone.**	**2M**
	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.	**[Application of Tie Clip microphone-1M each, (any two)]**
	g)	**State the function of tone control circuit in audio amplifier.**	**2M**
	Ans:	**Some people like depth in sound which is given by bass and some of the people like sharp sound and want treble more than bass.** **To cater (provide) to the individual taste to the human ears and also to provide**	**2M Explanation**

		offset adjustment to reduce noise in the signal provision off bass and treble control is made. **The cambered control is known as "Tone Control".**		
	h)	**List any four characteristics of Hi – Fi amplifier.**	**2M**	
	Ans:	**Note: Any other relevant Characteristics can be considered.** **i) Gain** **ii) Voltage gain** **iii) Bandwidth pause gain** **iv) Frequency distortion**	**1/2 M Each.**	
	B)	**Attempt any TWO :**	**8 M**	8
	a)	**Define modulation. Explain need for modulation.**	**4M**	
	Ans:	**Amplitude modulation:** It is the technique of modulation in which the any one characteristics of carrier signal varies in accordance with instantenious amplitude of modulating signal by keeping other two characteristics of carrier signal constant. **Need for modulation:** 1. Reduction in height of antenna. 2. Avoids mixing of signals 3. Increase the range of communication 4. Multiplexing is possible 5. Improves quality of reception	**2M** **2M (any 4 Points)**	
	b)	**Calculate the band width requirement for an FM signal having a modulation freq. of 3.1 kHz and maximum daviation of 21.7kHz.**	**4M**	
	Ans:	**BW = ?** **FM = 3.1Khz** **Smax = 21.7 KHz.** **By using Carlons rull** **BW = 2 (Smax + FM)** **= 2 (21.7 KHz + 3.1 K)** **= 2 (24.8) k** **BW = 49.9 k**	**2 M formula** **2M Answer**	
	c)	**Explain variable area sound recording method with diagram**	**4 M**	
			Diagram-2M, Explanation: 2M	

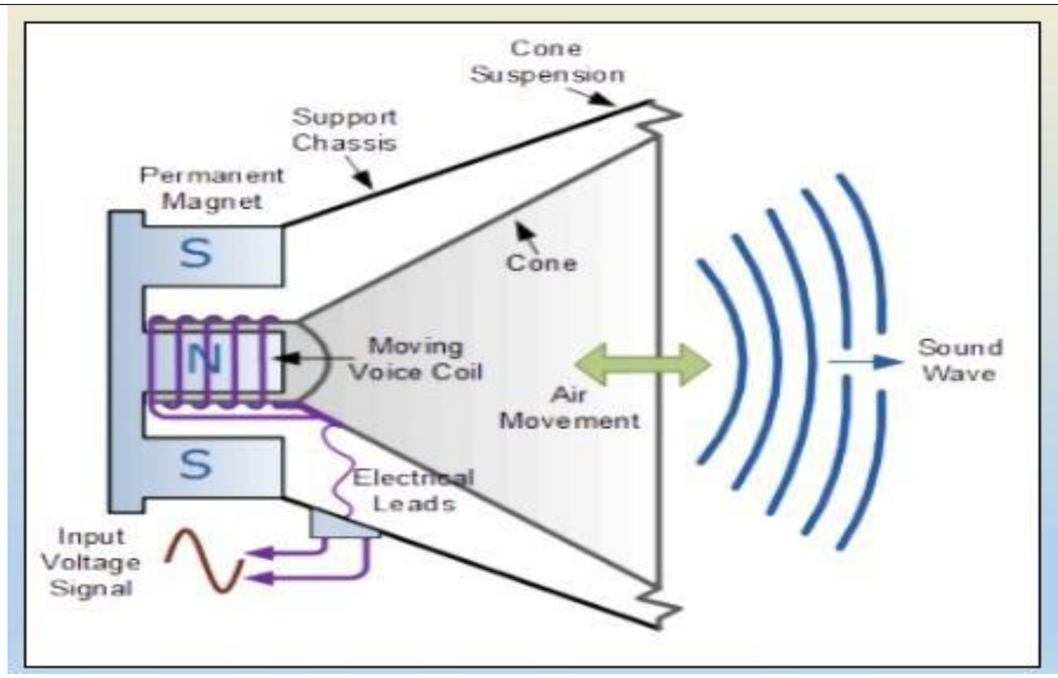


Explanation:

- **Variable Area Method:**

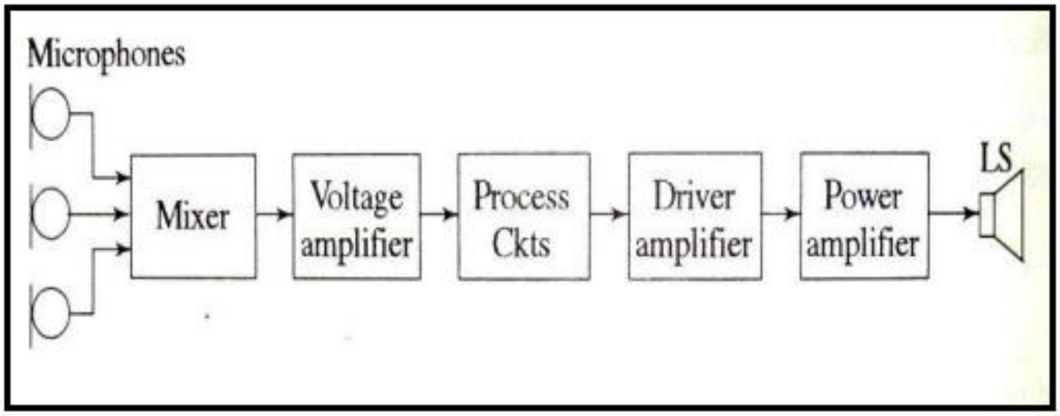
In this method,light of constant intensity falls on a slit.The area of slit opened for this light varies in accordance with the variation of sound pressur.hence,the light falls on the variable area on the soundtrack edge of the film.Thus,the area which is bright to light varies.The area of the slit is made variable with the help of a mirror or galvanometer.

Q 2		**Attempt any FOUR :**	**16-Total Marks**
	a)	**Explain working principle of moving coil type loud speaker.**	**4M**
	Ans:	**Construction of moving coil cone type loud speaker:-**	**2M diagram** **2M Explanation**



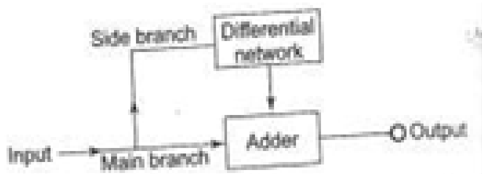
Working principle:
It works on the principle of interaction of magnetic field and current in the same way as an AC motor works. A coil called voice coil is placed in a uniform magnetic field, when the audio signal current passes through the voice coil there is an interaction between magnetic field and current resulting in a force working on the movable coil. This force is proportional to the audio current and hence vibratory motion in the coil.

b)	**Give any four advantages and four disadvantages of compact disk.**	**4M**	
Ans:	**Advantages:** CD makes use of digital storage technique & hence all the advantages of digital storage are applicable to CD. • When information is stored in the digital format, the problem of signal loss or disturbance in the signal is completely eliminated. • On CD the left & right channel information are stored separately one after another in fixed time interval. • Cross talk is eliminated between two channels & provides a real stereo output. • The capacity of storage on CD is high. • Available in small size. • Cost is less. • Makes use of interleaving process for error correction & detection. **Disadvantages:** • Not easy to change data • Require a burning software to record information to it and the shiny storage surface easily get damaged.	**2M Each (Any 2Advantages and disadvantages)**	

c)	**Explain working of P. A. system with neat block diagram.**	**4M**
Ans:	**Diagram:- Block diagram PA system**  **Block diagram PA system** **Working:-** **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 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.	**2M Diagram 2M Explnation**
d)	**Explain Dolby – A noise reduction technique with neat diagrams.**	**4M**
Ans:	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	**Diagram: 2M, Explanati**

main branch as shown in fig. the output of adder is the Dolby processed signal. **on: 2M**

 In playback, the differential network separates out the boosted signals in the side branch & subtracts from the input signal as shown in fig.



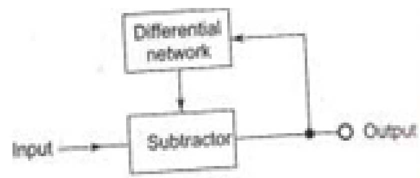


fig.(a) Coding of signal in Dolby method **fig. (b). Decoding of Dolby signal**

e)	**Explain generation of DSBSC AM signal using diode balance modulator.**	**4M**

Ans: A non-linear resistance or non-linear device may be used to produce Amplitude Modulation i.e. one carrier and two sidebands.

Diagram: 2M, Explanation: 2M

 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 .

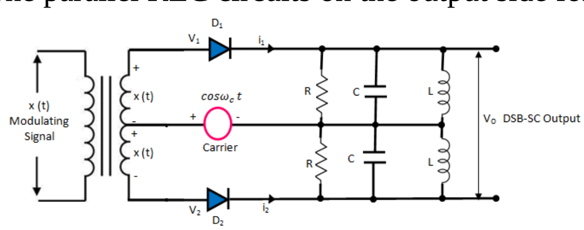
 Therefore, a balanced modulator may be defined as a circuit in which two non-linear devices are connected in a balanced mode to produce a DSB-SC signal.
Fig. shows the balanced modulator using diodes as non-linear device.
The modulating signal x(t) is applied equally with 180o phase reversal at the inputs 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 D1 is given by:

$$v_1 = cos\omega_c t + x(t)$$
…….. (1)

And the input voltage to D2 is given by:

$$v_2 = cos\omega_c t - x(t)$$
………. (2)

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



OR

The diode current i_1 and i_2 are given by:

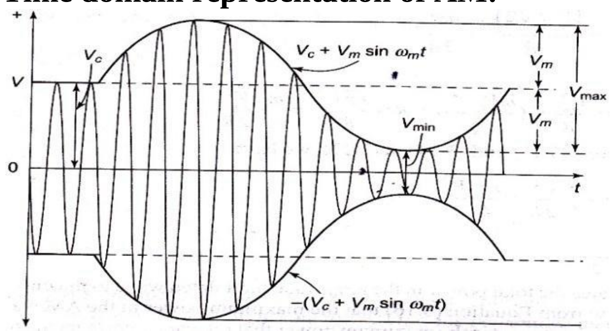
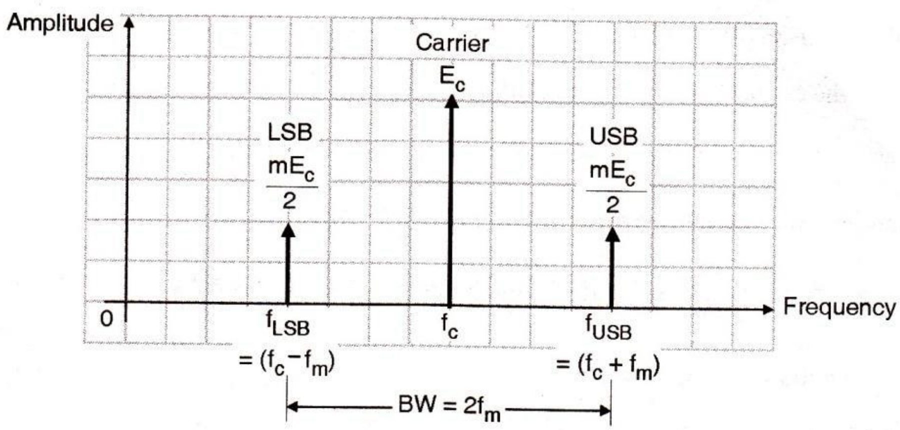
$$i_1 = av_1 + bv_1{}^2$$

The diode current i_1 and i_2 are given by:

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

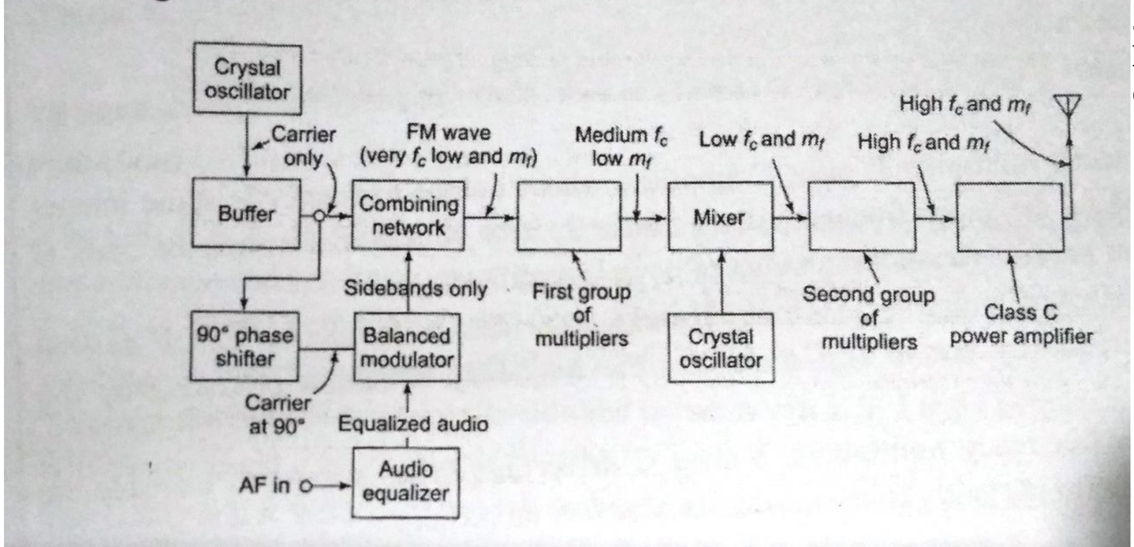
$$i_1 = ax(t) + acos\omega_c t + bx^2(t) + 2bx(t)cos\omega_c t + bcos^2\omega_c t$$

		$i_1 = a[x(t) + cos\omega_c t] + b[x(t) + cos\omega_c t]^2$ Similarly, $i_2 = av_2 + bv_2{}^2$ $i_2 = a[x(t) - cos\omega_c t] + b[x(t) - cos\omega_c t]^2$ $i_2 = av_2 + bv_2{}^2 = ax(t) - acos\omega_c t + bx^2(t) - 2bx(t)cos\omega_c t + bcos^2\omega_c t$ 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.	
	f)	**Define phase modulation and modulation index in phase modulation.**	**4M**
	Ans:	**Phase modulation:** 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:** The modulating index is defines as: Mp = δp is expressed in radiance where δp is maximum frequency deviation.	**(2M)** **(2M)**
Q. 3		**Attempt any four**	**16-Total Marks**
	a)	**Define modulation index of an AM wave and derive equation of modulation index for AM wave.**	**4M**
	Ans:	**Modulation Index**: It in AM is defined as the ratio of amplitude of modulating signal to the amplitude of carrier signal.	**2M Definition** **2M Equation**

		$$m = \frac{V_m}{V_c}$$ **Mathematical expression for amplitude modulated wave** $V_{AM} = V_c \sin \omega_c t + mV_c/2 \cos(\omega_c - \omega_m)t - mV_c/2 \cos(\omega_c + \omega_m)t$	
	b)	**Draw the time domain and frequency domain specturm of AM signal.**	**4M**
	Ans:	**Time domain representation of AM:**  **Frequency domain representation of FM:**  	**2 M Each Diagram**
	c)	**Draw block diagram of FM transmitter and explain its operation.**	**4M**

Ans:

Block diagram of FM transmitter:-



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.

Diagram: 2M, Explanation: 2M

d) **A 10 k W carrier wave is amplitude modulated at 60% depth of modulation by sinusoidal modulating signal. Calculate total power I the modulated wave.** **4M**

Ans: **Given :**

Pc = 10Kw

M = 60 %

P_T= ?

Formula :

Pt = Pc (1+ m^2a / 2)

= 10 Kw (1 + $(0.6)^2$/2 / 2)

= 10Kw (1 + 0.36/2)

= 10 Kw (2.36/2)

Pt = 11.8 Kw

2M Formula

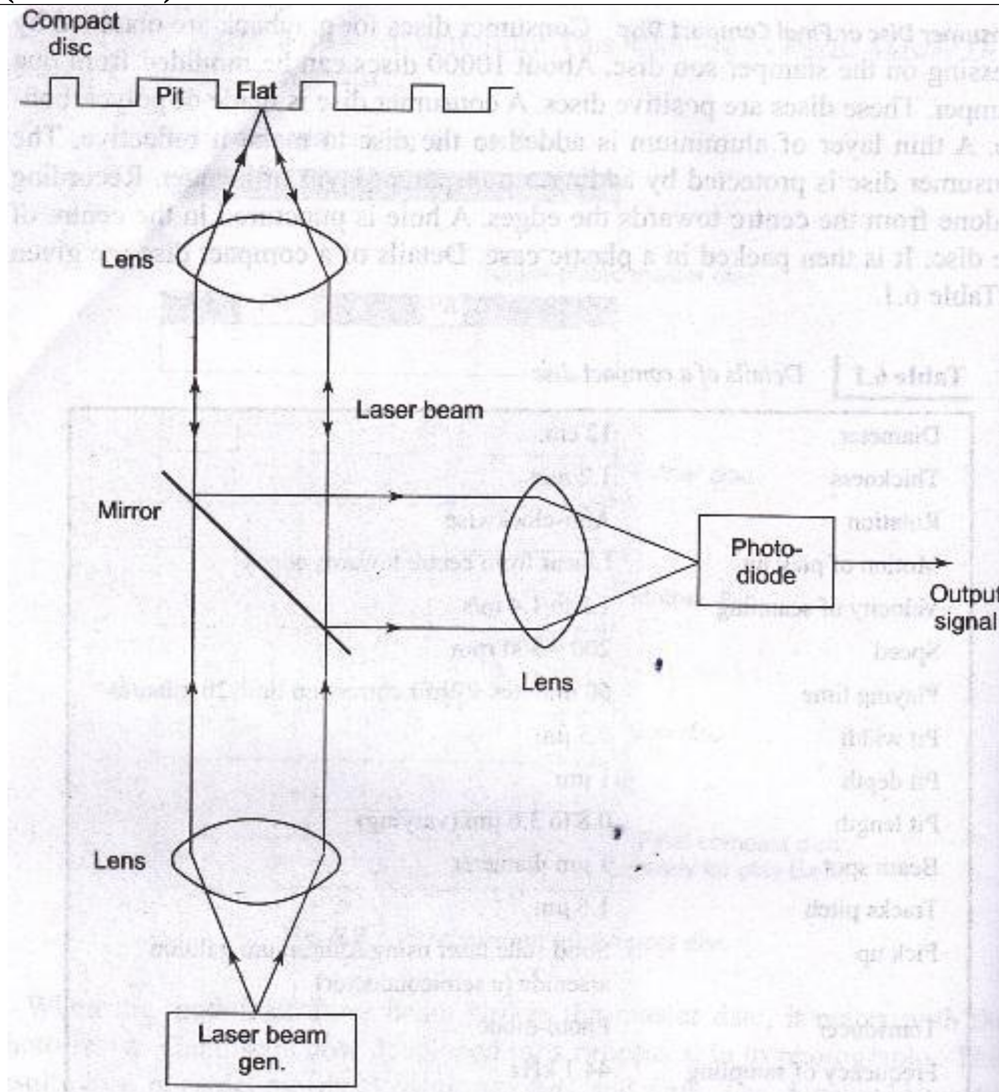
2M Answer

Pt = 11,800 wats.

e) **Explai preparation technique (sequense) of compact disk with diagram (waveforms).** **4M**

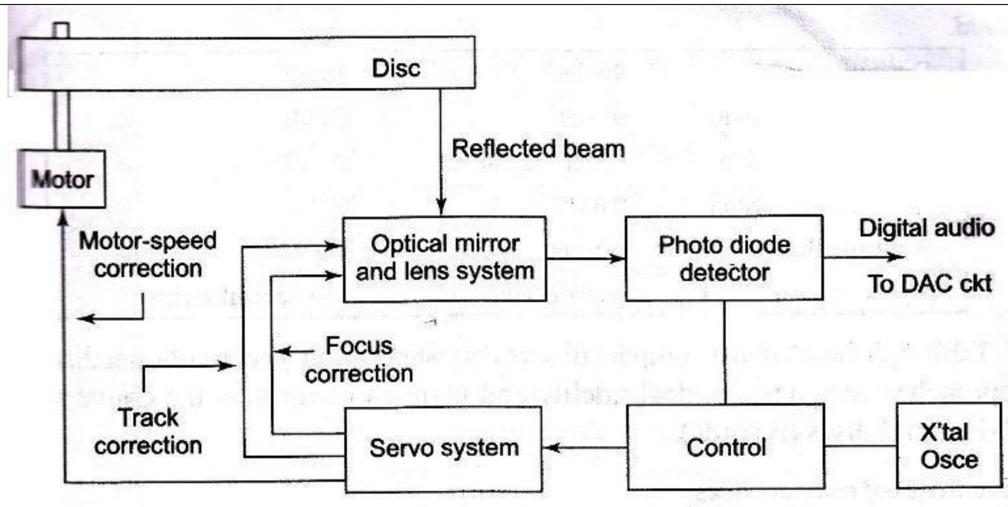
Ans:

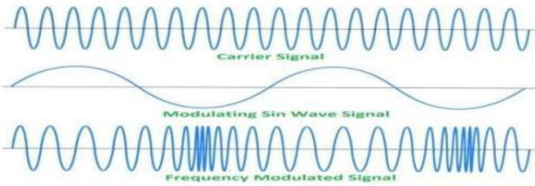
Diagram: 2M, Explanation: 2M



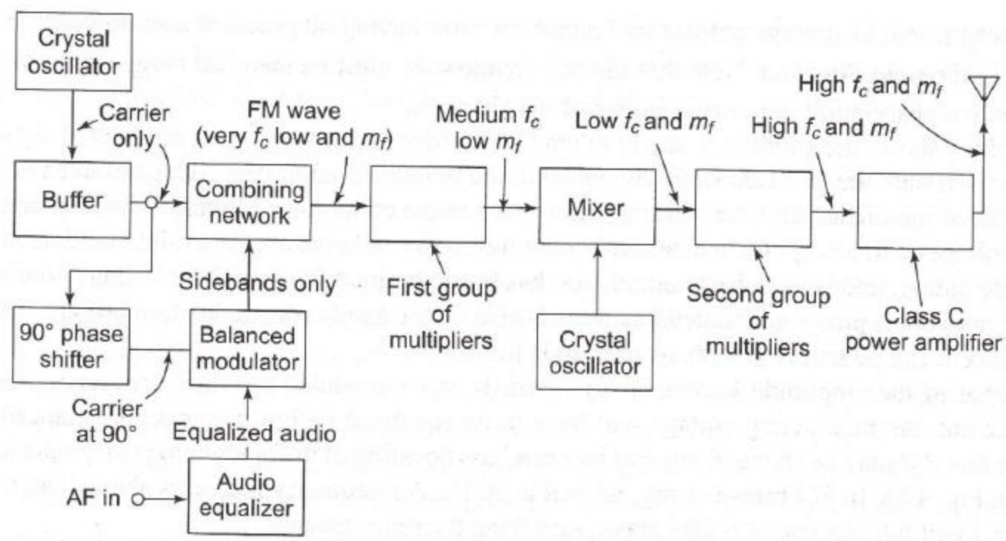
Explanation: A layer beam is incident on the compact disc through a half silver mirror. The returning beam is reflected from the aluminum flat surface and represents the logic 1. There is only a little reflection from a pit and it represents logic 0.
The binary digits are reproduces when this ON-OFF 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.	
	f)	**Give advangtes of FM over AM.**	**4M**
	Ans:	1. FM is higher noise immunity compared to AM. 2. FM modifies the frequency of the carrier 3. FM is much more complex compared to AM space wave is used for propagation do radius of transmission is limited to line of sight. 4. FM signal doesn't degrade as easily as AM 5. Application : Radio & TV broadcasting, police wireless, point to point communication Bandwidth is Twice the sum of the modulating frequency and the frequency deviation. **(20 times More as compared to AM)** All the transmitted power are useful.	**1M Advantage (any 4)**



Q. 4	**A)**	**Attempt any FOUR :**	**16-Total Marks**
	a)	**Explain generation of FM using Armstrong method.**	**4M**
	Ans:	**Diagram:-**	**2M**



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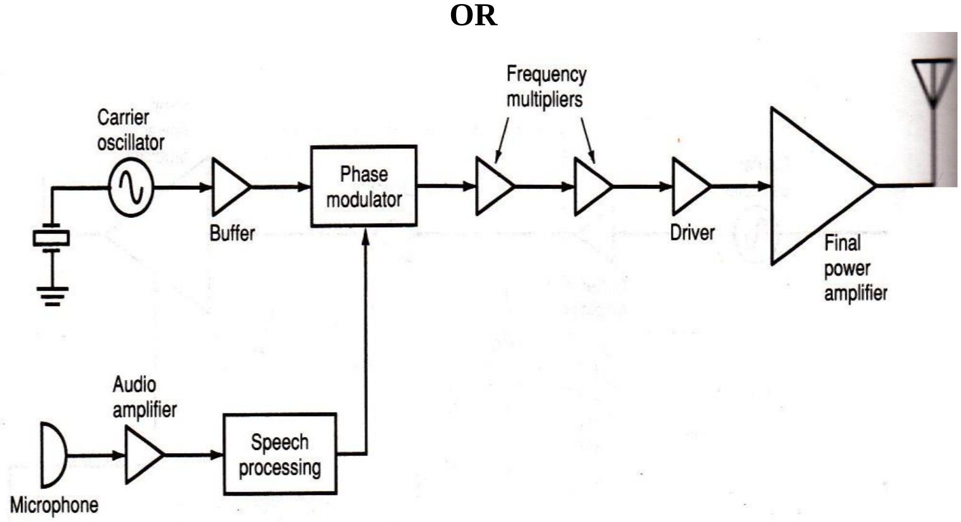

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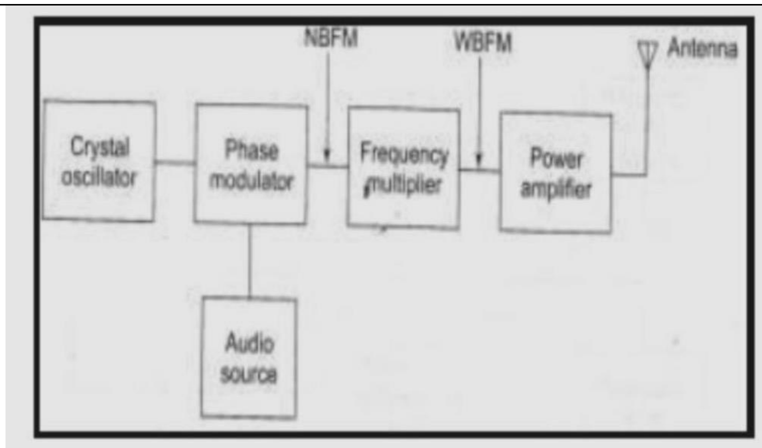
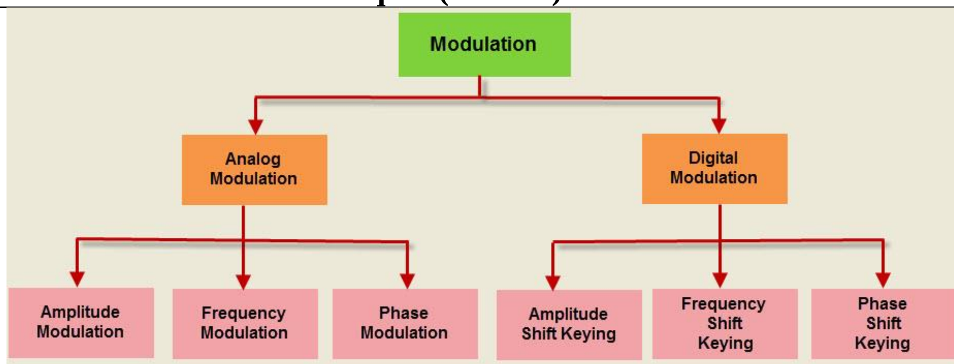
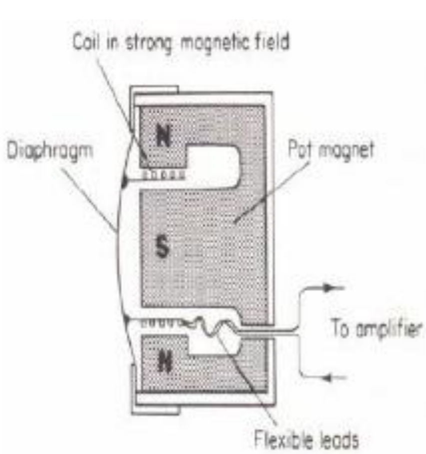
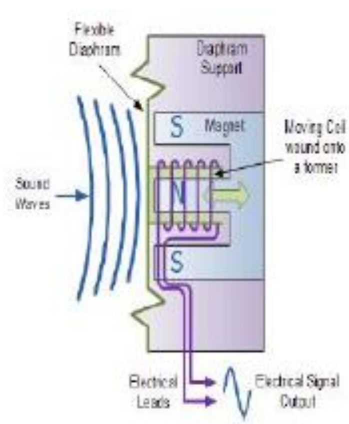


Explanation:

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.
- 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

2M

		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.	
	b)	**List different modulation techniques (method)**	**4M**
	Ans:		**4M**
	c)	**State need and application of PA system.**	**4M**
	Ans:	**Need of PA system:-** The intensity of sound decrease with distance. Hence when large gathering is to be addressed, sound needs to be amplified so that people at a distance from the stage may receive good intensity of sound for comfortable listening. **Application of PA system:-** 1) Sports meets 2) Public meetings 3)Auditoriums 4) Concerts & function. 5)To convey information to isolated locations as at railway station, airports, hospitals, factories etc	**2M** **2M**
	d)	**Explain construction and working principle of moving coil microphone.**	**4M**
	Ans:	**Construction :-** The dynamic microphone consists of a magnet, and a diaphragm to which a coil is attached. The assembly is held in place by an outer casing and the coil can move freely over the magnet.  OR 	**2M**

		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.	**2M**
	e)	**Explain working of complimentary symmetry push pull amplifier with neat circuit diagram.**	**4M**
	Ans:	**Circuit diagram:-**	**2M**

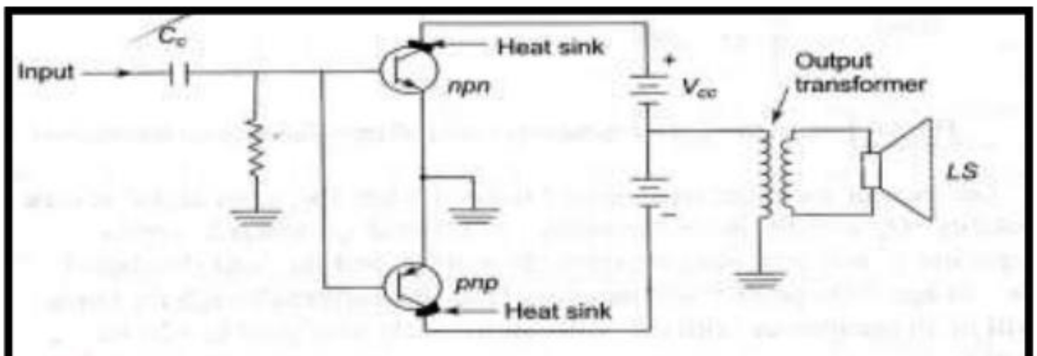


Fig. Complementary symmetry push-pull amplifier circuit with output transformer

Explanation: (**2M**)

• 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.

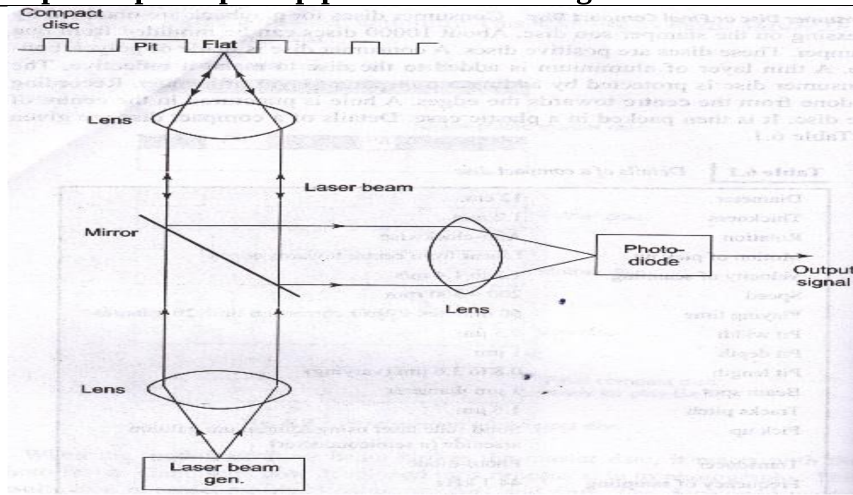
• Input signals to the two transistors are in the same phase. (Inter-Stage transformer for input is not required.)

• 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.

f) **Explain optical pickup process of sound signal with the block diagram.** **4M**

Ans: **2M**



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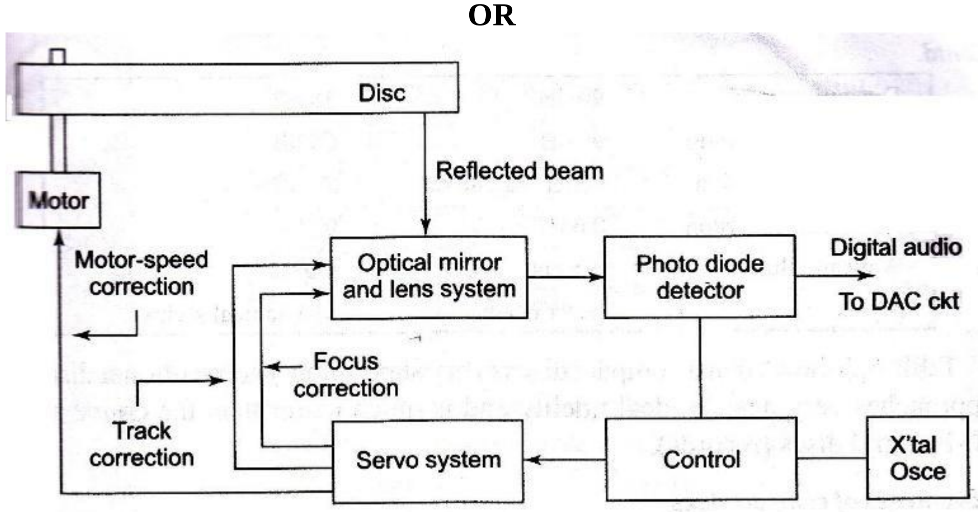
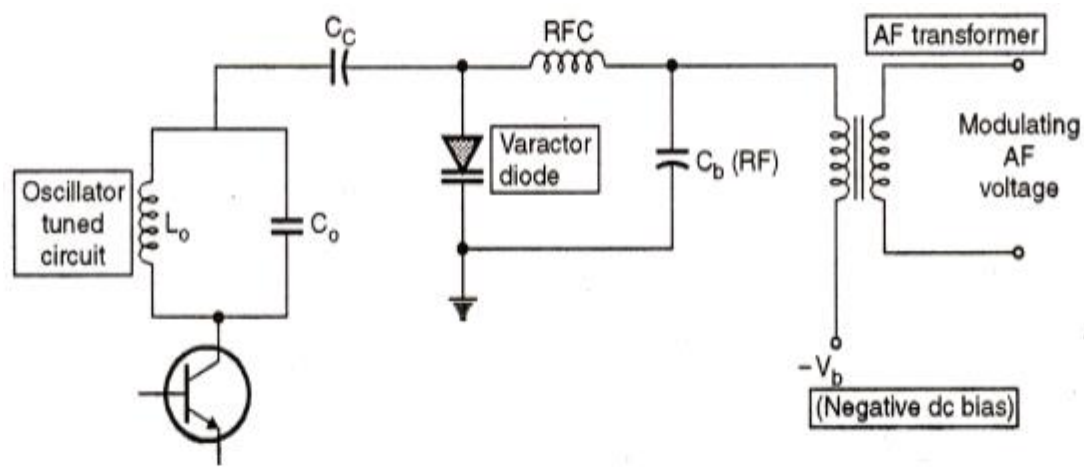


2M

Explanation:

- 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.
- 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.	
Q.5		**Attempt any FOUR :**	**16-Total Marks**
	a)	**Explain FM generation using varactor diode modulator.**	**4M**
	Ans:	**Diagram :-**  **Explanation:-** • Varactor diode modulator is the direct method of FM generation wherein the carrier frequency is directly varied by the modulating signal. • A varactor diode is a semiconductor diode whose junction capacitance varies linearly with applied voltage when the diode is reverse biased. • Varactor diodes are used along with reactance modulator to provide automatic frequency correction for an FM transmitter. The varactor diode modulator circuit is shown in Figure for generation of FM wave. • Varactor diode is arranged in reverse bias to offer junction capacitance effect. The modulating voltage which is in series with the varactor diode will vary the bias and hence the junction capacitance, resulting the oscillator frequency to change accordingly. • The external modulating AF voltage adds to and subtracts from the dc bias, which changes the capacitance of the diode and thus the frequency of oscillation. • Positive alternations of the modulating signal increase the reverse bias on the varactor diode, which decreases its capacitance and increases the frequency of oscillation. • Conversely, negative alternations of the modulating signal decrease the frequency	2M 2M

of oscillation.

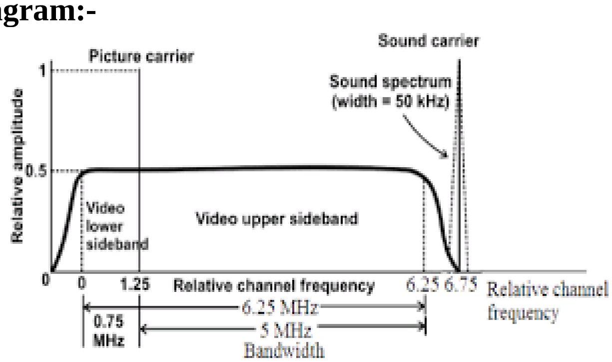
- The RFC and capacitor C b act as a filter which transmits only the AF variations to the varactor diode and blocks high frequency RF voltage from reaching the AF stage.
- The varactor diode FM modulators are widely accepted because they are simple to use, reliable and have the stability of a crystal oscillator.
- This method of FM generation is direct because the oscillator frequency is varied directly by the modulating signal, and the magnitude of frequency change is proportional to the amplitude of the modulating signal voltage.
- Varactor diode modulator isused for automatic frequency control and remote tuning.

b) **Define V.S.B. and draw V. S. B. spectrum.** **4M**

Ans: **2M**

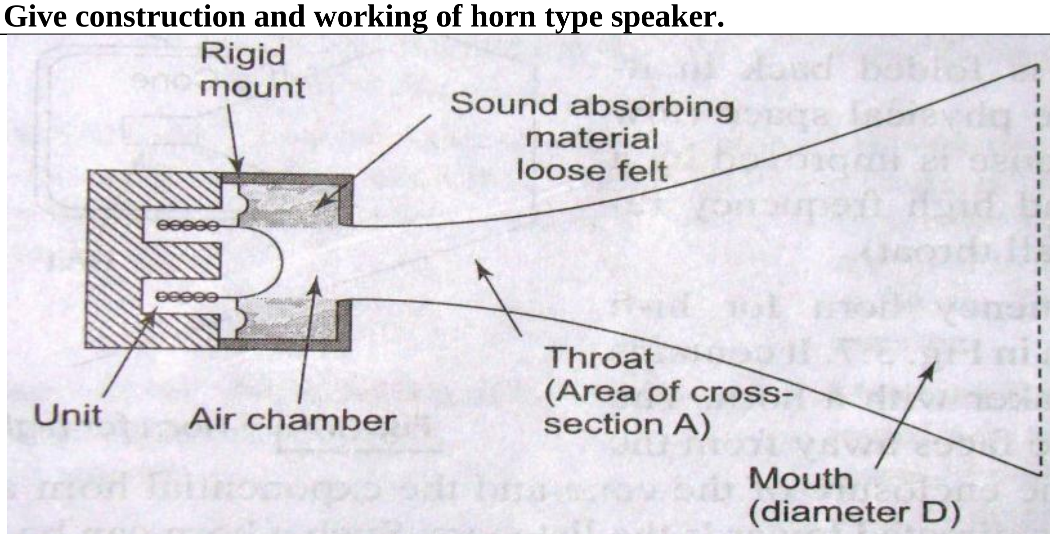
- Vestigial sideband is a type of Amplitude modulation in which one side band is Completely passed along with trace or tail or vestige of the other side band.
- VSB is a compromise between SSB and DSBSC modulation. In SSB, we send onlyone side band, the Bandwidth required to send SSB wave is loww.
- SSB is not appropriate way of modulation when the message signal containssignificant components at extremely low frequencies.
- To overcome this VSB is used.
- The main advantage of VSB modulation is the reduction in bandwidth.

Diagram:- **2M**



c) **Give construction and working of horn type speaker.** **4M**

Ans: **2M**

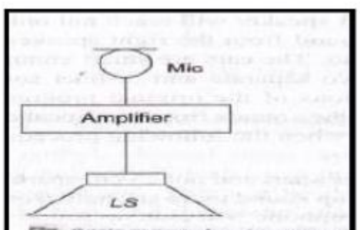
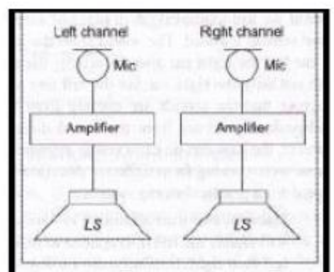


<u>Working Principle:-</u> **2M**

- 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.

d) **Compare monophony and stereophony systems.** **4M**

Ans: **1M each**

Monophony	Stereophony
1. Only one amplifier is used. Single amplifier stage is known as mono amplifier	1. At least two independent amplifiers are used. These part of amplifiers is called as stereo signal
2. No naturalness	2. Provides naturalness of sound Signal
3. Listener cannot judge the direction of sound	3. Listener can judge the direction of Sound
4. Low cost	4. Comparatively high cost.
Fig. Basic monophonic system	Fig. Basic stereophonic system

e) **Explain selection criterion of a good microphone as per application.** **4M**

Ans: **For Selection criteria of microphones, microphone should have**

Any 4 criteria 4M

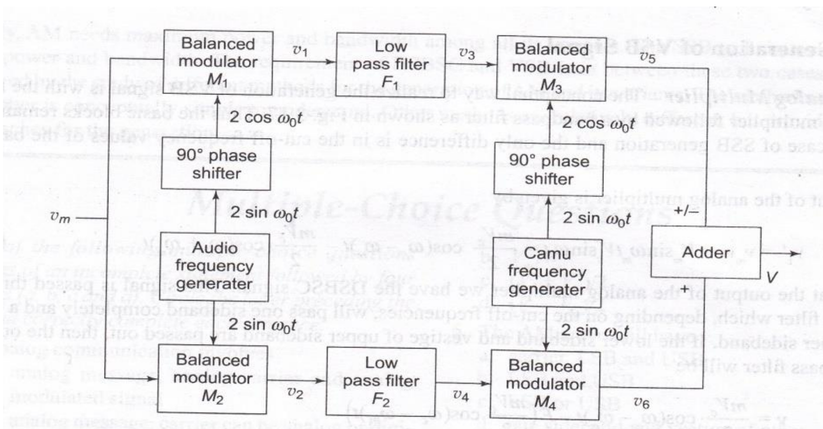
1. Sensitivity: It is an electrical output from microphone at certain sound pressure level. It is defined as output in milivolt for the sound pressure of 1 Pascal at 1 KHz.
2. S/N ratio: It is the ratio of level of the desired signal that a microphone records compared to the level of noise that it picks 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 patternsare:

   - Cardioids or uni-directional

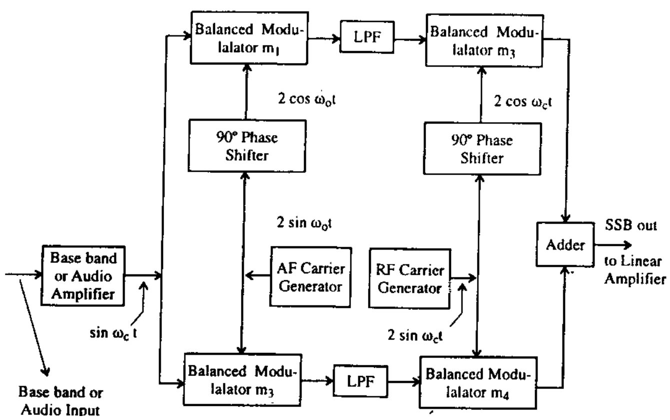
- Bi-directional or figure-of-eight
- Omni-directional

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).

f)	**Explain third method of generation of SSBAM with diagram**	**4M**
Ans:	**Diagram:-**	**2M**



OR



Explanation: - **2M**

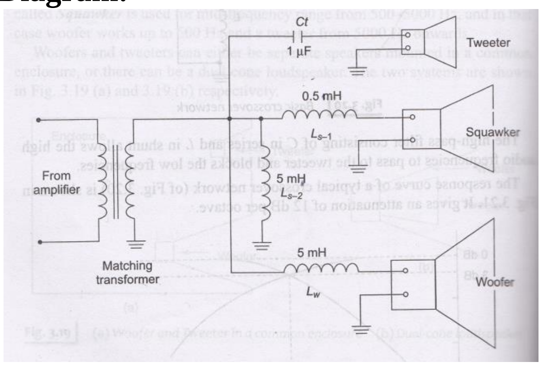
- It was developed by weaver to retain the advantage of the phase shift method, such as its ability to generate SSB at any frequency and use low audio frequencies.
- It is very complex and not often used method commercially.
- The later part of the block diagram is identical to phase shift method, but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

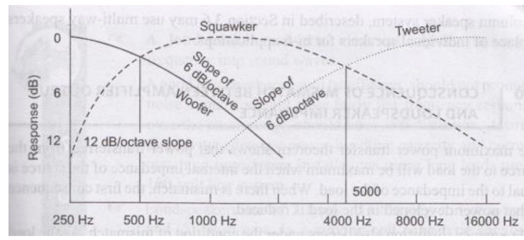
- If a lower sideband signal if required at the final output the phase of the carrier voltage being fed to M1 should be changed by 180^0

Q. 6 **Attempt any FOUR:** **16-Total Marks**

a) Explain working of 3-way cross over network with ckt diagram **4M**

Ans: Diagram:- **2M**





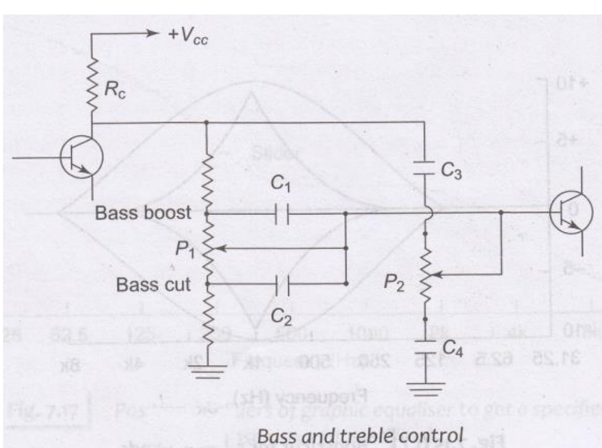
Three way cross over network **Response curve (*Optional to Draw*)**

Explanation: **2M**

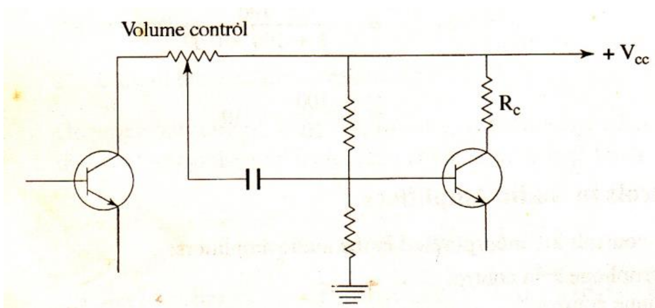
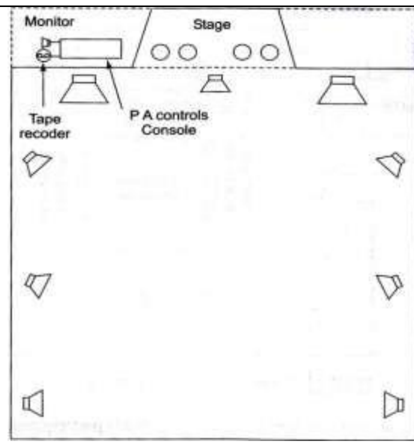
- When multi-way speaker system is used to get flat frequency response for the entire range of audio frequency it is essential to have a cross over network to divide the incoming signal into separate frequency ranges for each spectrum.
- In absence of cross over network, the speaker will suffer overheating 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. Lw of 5mH in series with woofer prevents high and mid frequencies reaching to woofer.
- Ls1 and Ls2 allows only mid frequency range to reach to squawker

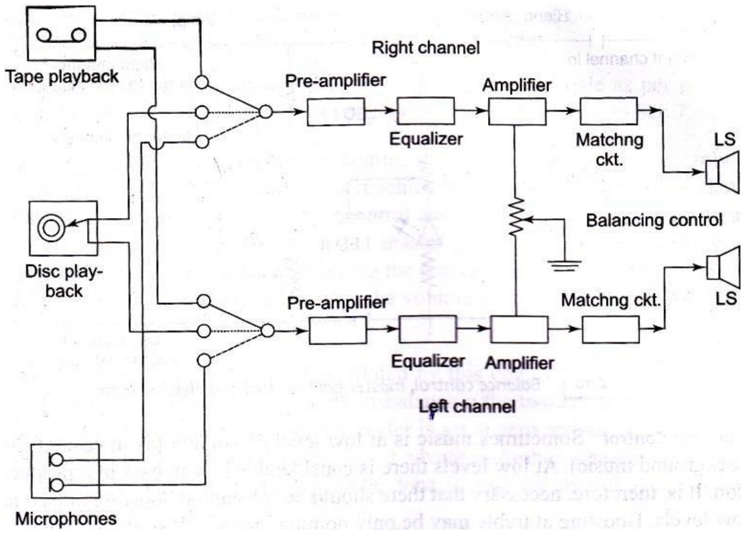
b) Draw the circuit of tone control and volume control for an audio amplifier. **4M**

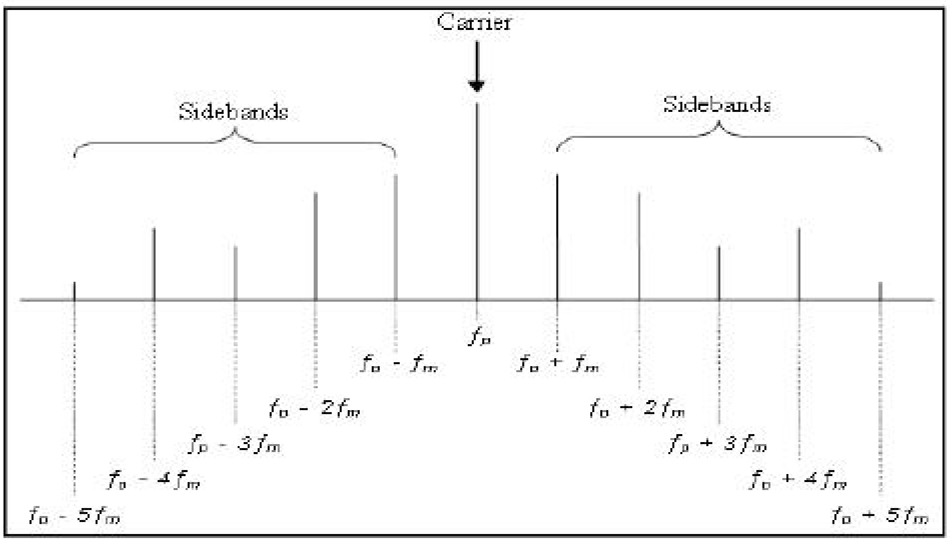
Ans: Tone control circuit:- **2M each**



Bass and treble control

		Volume control circuit:-  	
	c)	**Explain and planning and installation steps of typical public address system for an auditorium.**	**4M**
	Ans:	**Diagram :-**  	**2M**
		<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.	**2M**

	d)	**Draw and explain the block diagram of Hi-Fi amplifier.**	**4M**
	Ans:	**Diagram :-**  **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.	**2M** **2M**
	e)	**State characteristics of a good audio amplifier.**	**4M**
	Ans:	**Characteristics of audio amplifier:-** 1. Gain 2. Bandwidth 3. Distortion 4. Power output 5. Impedance	**Any 4 chara 1M each**
	f)	**Give mathematical equation of FM wave, FM modulation index and draw frequency spectrum of FM.**	**4M**
	Ans:	**Mathematical representation of FM** $$F= f_c + k_f V_m \sin\omega_m t$$ Where,	**1M**

		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 $m_f = \frac{(maximum)\ frequency\ deviation}{modulating\ frequency} = \frac{\delta_f}{f_m}$ **Frequency spectrum of FM:-**  FM signal's frequency spectrum	**1M** **2M**