MODEL ANSWER

SUMMER– 17 EXAMINATION

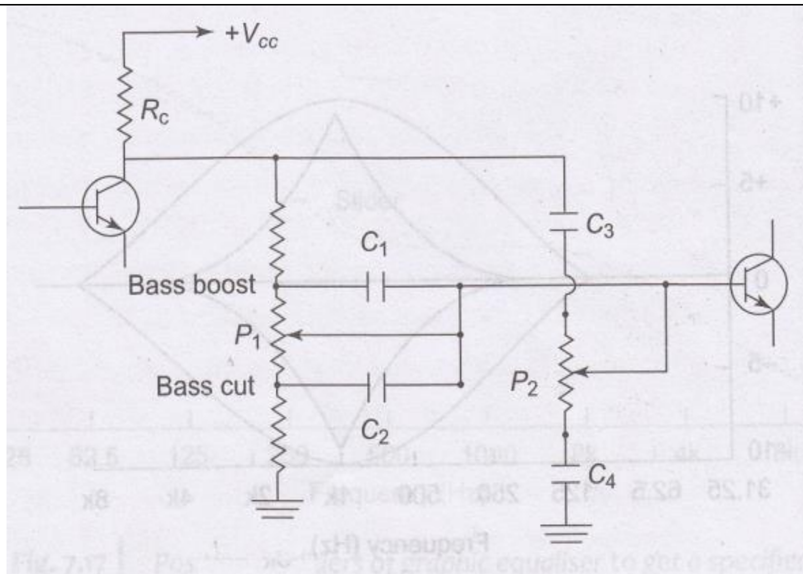
Subject Title: Fundamentals Of Communication Subject Code: 17316

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 Six of the following :**	**12-Total Marks**
	i)	**Write down formula relationship between frequency and wavelength. If the frequency of signal is 18 KHz, calculate the wavelength.**	2M
	Ans:	Velocity of Electromagnetic signal) C= Frequcy(n) * Wavelegth(λ) Given: C=3*10^8 m/s n= 18 KHz= 18*10^3 Hz Wavelegth(λ)= C/n = 3*10^8/18*10^3 =16.6 millimeter	2M
	ii)	**What is the function of Bass and Treble control in HIFI audio amplifier?**	2M

Ans:



Bass and treble control

Bass control: Controlling the low frequency of hi fi amplifier
Treble control: Controlling high frequency of hi fi amplifier

The potentiometer in series with capacitor forms the treble control. When slider P2 is at the lower end, maximum signal develops. As slider is moved upwards, less resistance of the potentiometer comes in series and there is more cut in the high frequency signals. Cut is maximum when slider is at the top end, short circuiting the potentiometer completely. This position is called treble cut. The other position where treble cut is minimum is called treble boost.
Bass would be cut if capacitive reactance in series of signal increases. Lower the capacitance greater will be reactance. When slider P1 of potentiometer R is at upper end, the capacitor C1 is shorted and signal directly goes to next stage, bypassing capacitor C1 and hence bass has minimum attenuation. It is called bass boosts when slider is at the lower end, capacitor C1 in parallel with whole resistance R of the potentiometer comes in series with the signal. In this position bass will have maximum attenuation. This position is called bass cut.

iii) **Define phase modulation and its modulation index.** **2M**

Ans:

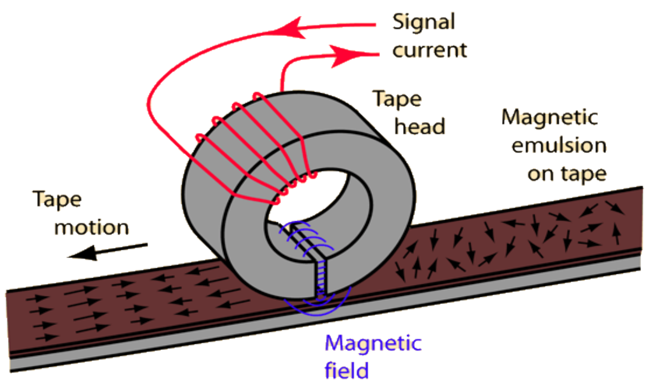
Phase modulation:
Phase modulation is the process of varying the instantaneous phase of Carrier signal accordingly with instantaneous amplitude of message signal.

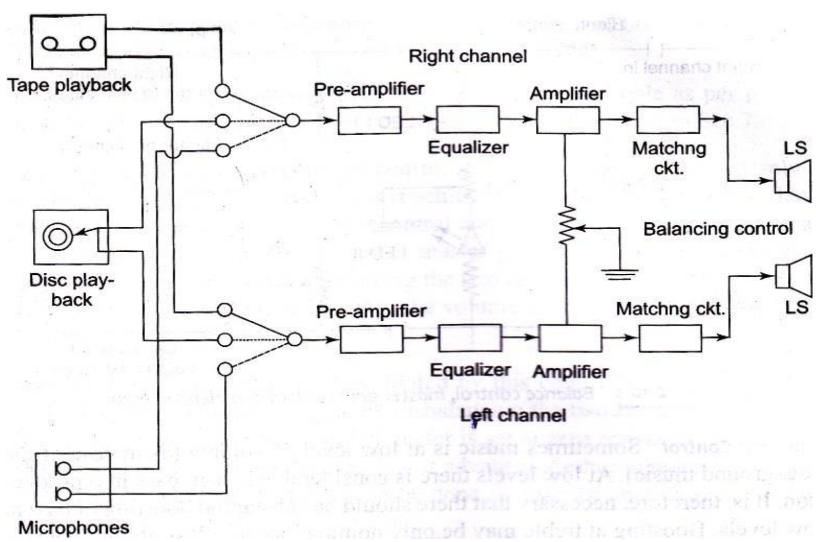
Modulation Index:
This quantity indicates by how much the modulated variable varies around its unmodulated level. It relates to the variations in the phase of the carrier signal.

The modulating index is defines as:

$Mp = \delta p$ is expressed in radiance

where δp is maximum frequency deviation.

	iv)	**Enlist specification of compact disc(4 points).**	**2M**
	Ans:	• Capacity: up to 700 MB • Diameter: 120 millimeters • Read mechanism: semiconductor laser(780 nm wavelength) • Makes use of interleaving process for error correction and detection	**2M**
	v)	**State principle of magnetic recording**	**2M**
	Ans:	 • When certain material like iron oxide comes in contact with magnetic field, get magnetized and retain that magnetism permanently until altered. • Sound pressure variations are converted into electrical variations by a microphone. The audio output of the microphone is amplified and fed to the coil of an electromagnet. • The electromagnet has a minute gap through which magnetic lines of force cannot pass easily due to high reluctance of air. • When a tape with a coating of a magnetic material is made to pass across the gap, the lines of force get an easy path through the iron oxide which is formed into elementary magnets. • The magnetic strength of electromagnet, and hence through the gap covered by the iron oxide of the tape depends on the audio current. • Thus the coating of iron oxide on the tape is magnetized in accordance with the audio current and hence, in accordance with the sound pressure variations. • The magnetism in the iron oxide can be retained for long time. This means that sound has been recorded in the form of varying magnetic field.	
	vi)	**Define timber, pitch.**	**2M**
	Ans:	**<u>PITCH :</u>** Pitch is a characteristic of sound mainly related to frequency. Pure tone pitch is determined by frequency alone. But in speech & music the pitch of sound depends on frequency as well as on intensity. **<u>TIMBRE :</u>** The proportion of tones & overtones in a sound form the special characteristics by which a particular sound can be recognized. When we hear the sound of a relative or a friend, even if the person is not visible. This quality of sound is called timbre & is related to the	

		proportion in which overtones are present in the sound.	
	vii)	**State the need of graphic equalizer.**	**2M**
	Ans:	A graphic equalizer is a high-fidelity audio control that allows the user to see graphically and control individually a number of different frequency bands in a stereophonic system. A typical graphic equalizer consists of several audio filter/amplifiers, each centered at a specific frequency in the audio range. Most graphic equalizers have two identical sets of filter/amplifiers, one for each channel in a stereophonic system.	
	viii)	**Draw the block diagram of HI-FI system.**	**2M**
	Ans:		**2M**
	B)	**Attempt any two of the following :**	**8M**
	a)	**A 500 watt carrier is modulation to a depth of 80%. Calculate the total power in modulated wave. What will be change in total power of modulated wave if we reduce modulation percentage from 80% to 70% by keeping same power of carrier?**	**4M**
	Ans:	Pc=500 W, ma=0.8, ma=0.7 Formula: Pt= Pc(1+ma^2/2) Total power for(ma=0.8) Pt= Pc(1+ma^2/2) =500(1+0.8^2/2) =660 watt Total power For(ma=0.7) Pt= Pc(1+ma^2/2) =500(1+0.7^2/2) = 622.5 watt	**1M** **1M** **1M**

		Total power change: 660-622.5=37.5 watt	1M
	b)	**Find the carrier frequency , modulating frequency, modulation index and maximum deviation of the FM wave represented by the voltage equation $\upsilon = 12 \sin(6 \times 10^8 t + 5\sin 1250t)$.**	**4M**
	Ans:	Comparing the above equation with V= Vc(fc t + mf sin ωm t) Carrier voltage Vc= 12 V Carrier frequecny(fc)= 6*10^8 Hz= 600 MHz Modulation index(mf)=5 ωm=2 π fm Modulating frequency(fm)= ωm/2 π = (1250/2 π)=198.9 Hz Maximum deviation= mf*fm= 5*198.5=992.5Hz	**1M** **1M** **1M** **1M**
	c)	**What is significant of low pass filter, band pass filter and high pass filter in Dolby-A system(with frequency range)?**	**4M**
	Ans:	• 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.	**1M each pt.**
Q 2		**Attempt any four in the following :**	**16M**
	i)	**Compare Woofer and Tweeter on the basis of'(a) definition (b) size (c) weight (d)frequency range.**	**4M**
	Ans:	(see table below)	**1M for each pt.**

Parameter	Woofer	Tweeter
Defination	A **woofer** is a speaker designed for low-frequency sounds	**A tweeter** is a speaker designed for high-frequency sounds.
Size	Big	small
Weight	More	less
Frequency range	16 Hz to 500 Hz	5 KHz to 20 KHz

	ii)	**ii) Enlist advantages of Compact Disc (4 points).**	**4M**
	Ans:	1.When information is stored in the digital format,the problem of signal loss or disturbance in the signal is completely eliminated	**1M for each pt.**

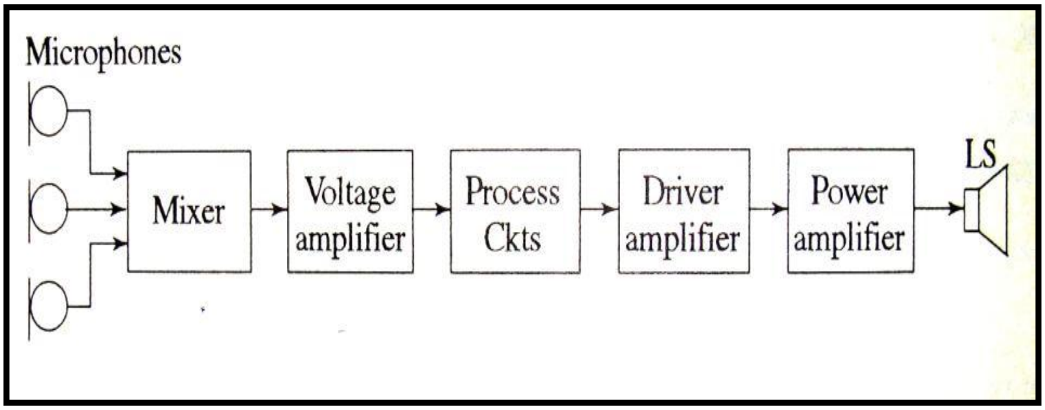
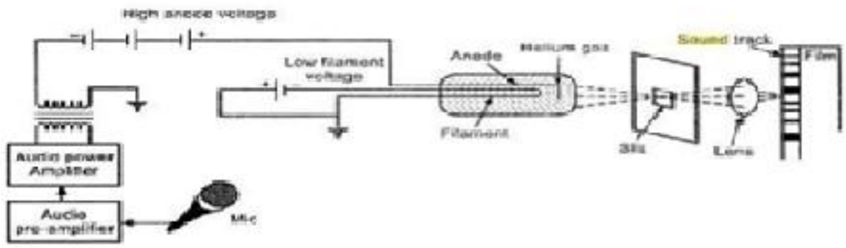
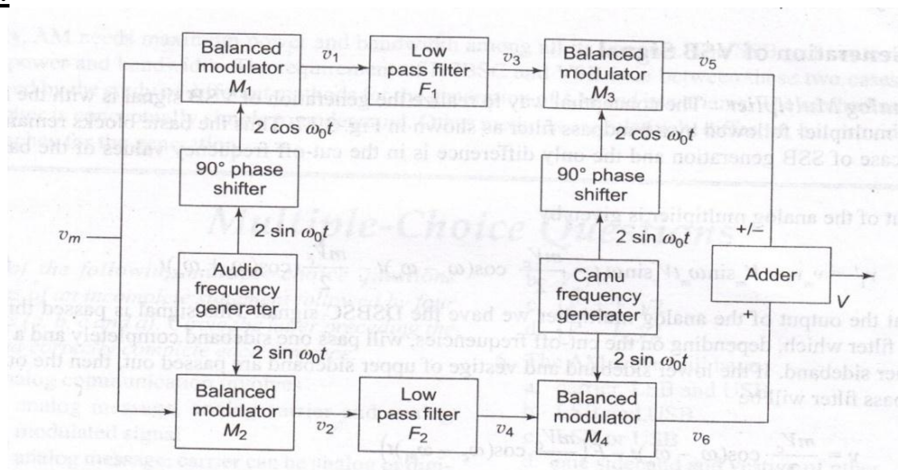
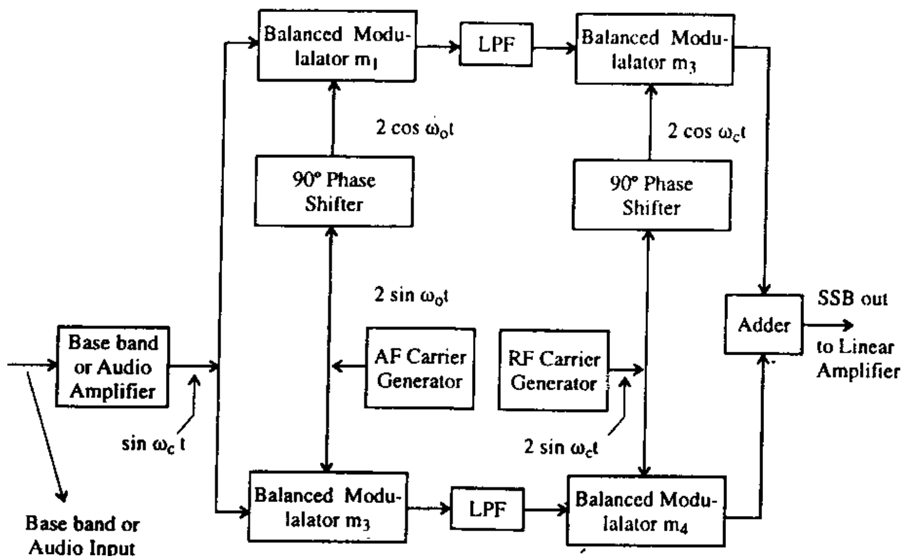
		2.The capacity of storage on CD is high 3.Small size 4.Cost is less	
	iii)	**Draw the block diagram of PA system. Explain function of each block**	**4M**
	Ans:	**<u>Block Diagram-</u>** 	**2M**
		<u>Explanation:</u> **1. Microphone -** It picks-up sound wave and convert them to equivalent electrical 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 sound.	**2M**
	iv)	**Explain the variable density method of optical recording of sound on film.**	**4M**
	Ans:	**<u>Diagram:</u>**	**2M**

		Diagram Of Variable-Density method  **Explanation:** • Variable-Density Method: In this method, sound is picked up by microphone, and converted into electrical signals which are amplified. Audio o/p of the amplifier is fed to the anode of special type of vaccumtube, called an AEO lamp. • The lamp contains a little quantity of helium gas. The anode gets high dc voltage in series with the audio voltage. • The filament of the lamp is connected to a low dc voltage The intensity of light coming out from the lamp varies in acordance with the audio signal. • This varying light passes through a slit and a focussinglens. The focussed light falls on moving photographic film where the image is recorded.	**2M**
	v)	**With neat block diagram explain the generation of SSB by "Third Method".**	**4M**

Ans: **Diagram:** **2M**

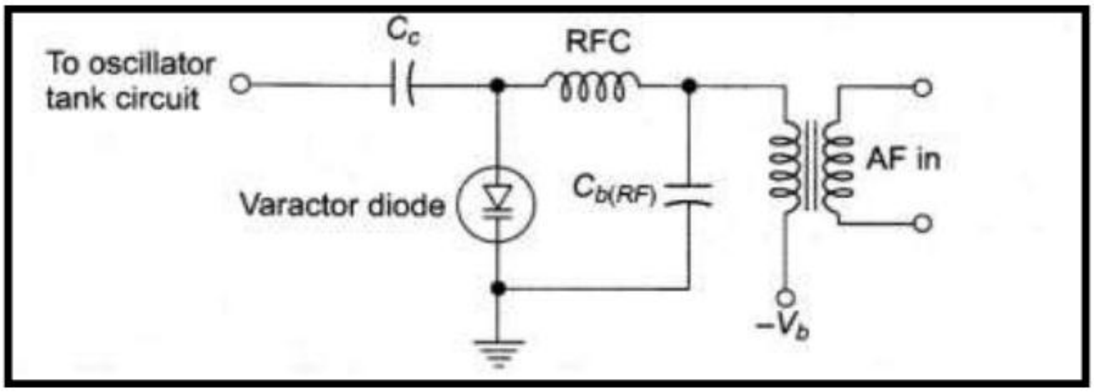
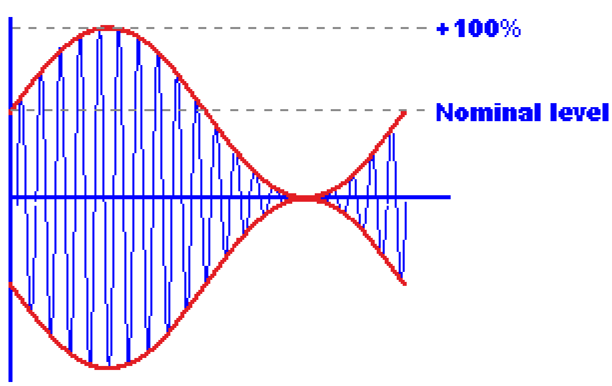


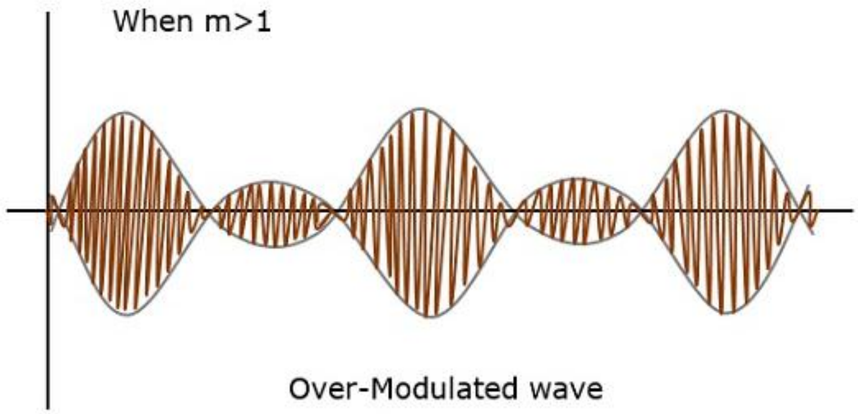
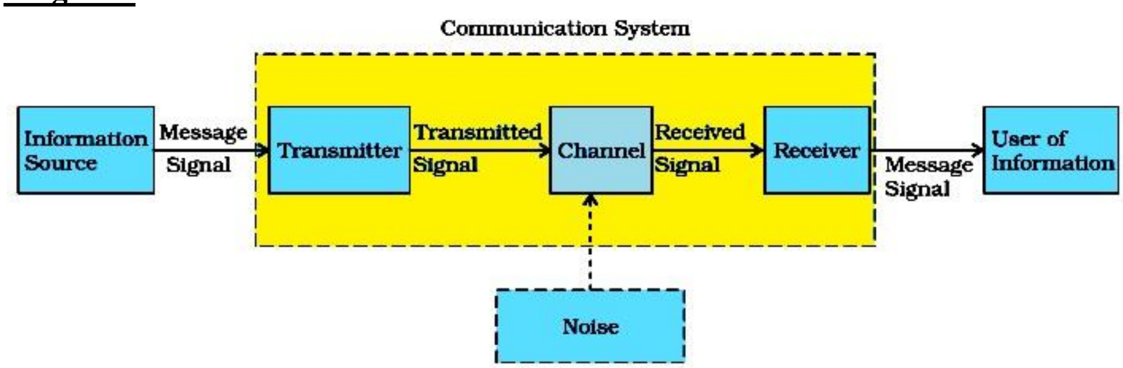
OR

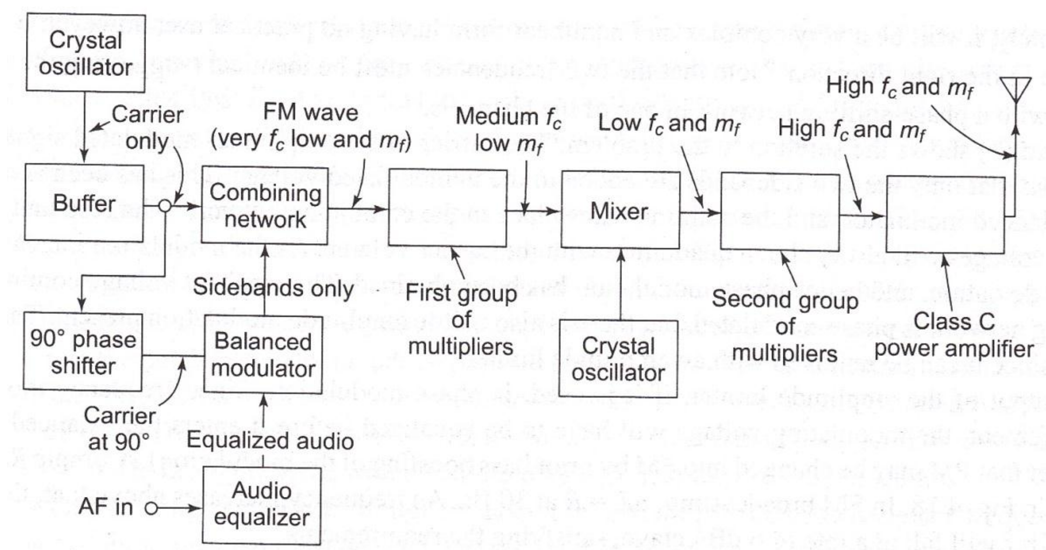


Explanation: - **2M**

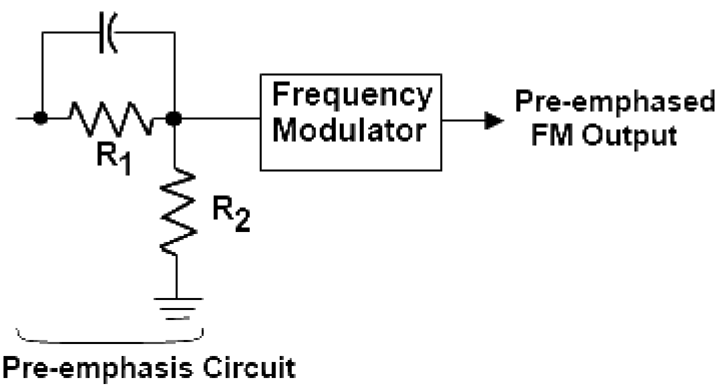
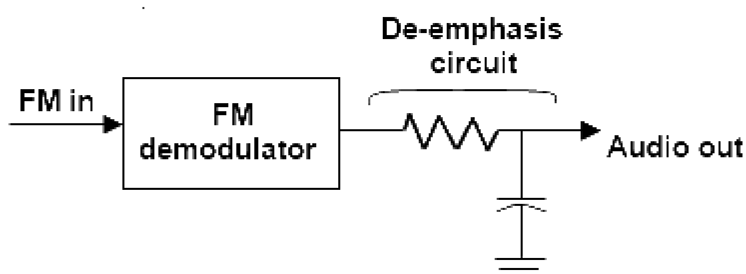
- It was developed by weaver to retain the advantage of the phase shift method, such as its ability to generate SSB at any frequency and use low audio frequencies.
- It is very complex and not often used method commercially
- The later part of the block diagram is identical to phase shift method, but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

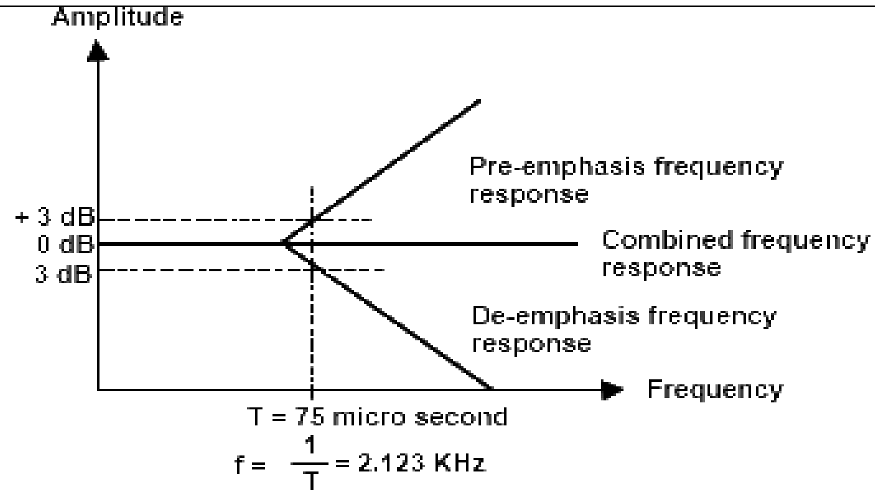
		suppression. • If a lower sideband signal if required at the final output the phase of the carrier voltage being fed to M1 should be changed by 180^0	
	vi)	**Draw the varacter diode modulator used for generation of F.M. Explain its working.**	**4M**
	Ans:	**Diagram:**  **Fig. Generation of FM wave using varactor diode modulator**	**2M**
		Working: • 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.	**2M**
Q. 3		**Attempt any four of the following:**	**16M**
	i)	**Draw the waveform of amplitude modulated envelope for-modulation index m = 1 What change will take place in amplitude modulated waveform if increase modulation index from 1 to 1.5? Explain with neat sketch.**	**4M**
	Ans:	**AM wave for m=1**  **If modulation Index change from 1 to 1.5**	**1M**

		Waveform:  If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an **over-modulated wave**. As the value of modulation index increases, the carrier experiences a 180° phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such over modulated wave causes interference, which cannot be eliminated.	**3M (Waveform : 1.5M, Explanation:1.5M)**
	ii)	**Draw the block diagram of communication system. Explain operation of each block.**	**4M**
	Ans:	**Diagram:**  **Explanation:** • **Information Source** As we know, a communication system serves to communicate a message or information. This information originates in the information source. • **Transmitter** The function of the transmitter is to process the electrical signal from different aspects. Modulation is the main function of the transmitter. In modulation, the message signal is superimposed upon the high-frequency carrier signal. All these processings of the message signal are done just to ease the transmission of the signal through the channel. • **The Channel and The Noise**	**2M** **2M**

		The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver. Noise is an unwanted signal which tends to interfere with the required signal. Noise signal is always random in character. Noise may interfere with signal at any point in a communication system. However, the noise has its greatest effect on the signal in the channel. • **Receiver** The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal. This reproduction of the original signal is accomplished by a process known as the demodulation or detection. Demodulation is the reverse process of modulation carried out in transmitter. User of information/ Destination Destination is the final stage which is used to convert an electrical message signal into its original form. For example in radio broadcasting, the destination is a loudspeaker which works as a transducer i.e. converts the electrical signal in the form of original sound signal.	
	iii)	**Draw the block diagram of Armstrong frequency modulator system. What is the function-of balanced modulator in Armstrong frequency modulator system?**	**4M**
	Ans:	**Block diagram:**  **Balance modulator:** **Function of Balance modulator:** A balanced modulator mixes the audio signal and the radio frequency carrier, but suppresses the carrier, leaving only the sidebands. The output from the balanced modulator is a double sideband suppressed carrier signal and it contains all the information that the AM signal has, but without the carrier. It is possible to generate an AM signal by taking the output from the balanced modulator and reinserting the carrier.	**2.5M** **1.5M**
	iv)	**Define modulation. Enlist different types of modulation. Explain need of modulation.**	**4M**
	Ans:	**Modulation:** Modulation is define as the process by which some charateristic (amplitude, frequency	**Definition: 1 M**

		and phase) of carrier signal is varied in acccordance with instnataneous value of modulating voltage. **Types of Modulation:** • Amplitude modulation • Frequency modulation • Phase modulation **Need of Modulation:** **1. Reduction in the height of antenna** For the transmission of radio signals, the antenna height must be multiple of λ/4 ,where λ is the wavelength . $\lambda = c / f$ where c : is the velocity of light f: is the frequency of the signal to be transmitted **2. Avoids mixing of signals** If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver cannot separate them from each other. Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals. **3. Increase the Range of Communication** The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted. They get heavily attenuated. The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance. The modulation process increases the frequency of the signal to be transmitted. Therefore, it increases the range of communication. **4. Multiplexing is possible-** Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously. This is possible only with modulation. The multiplexing allows the same channel to be used by many signals. Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time. **5. Improves Quality of Reception-** With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent. This improves quality of reception.	**Types: 1 M** **Need: 2 M(any four)**

	v)	**Explain the terms: preemphasis and deemphasis.**	**4M**
	Ans:	**<u>Pre-emphasis:</u>** In an FM system the higher frequencies contribute more tothe noise than the lower frequencies. Because of this all FMsystems adopt a system of preemphasis where the higher frequencies are increased in amplitude before being used tomodulate the carrier. pre emphasis is a process which is used in transmitter side to boosting the amplitude of higher modulating signal before modulator. IF we used it after the modulator than carrier &modulator signal will mixed and it is difficult to verify which one is the modulating signal.  (a) Pre-emphasis Circuit **<u>De-emphasis:</u>** De emphasis is process which is used in receiver side to reduced the signal & get their original signal. But it is used after demodulator ckt.  (c) De-emphasis circuit **<u>Pre-emphasis/ De-emphasis graph:</u>**	**Pre-emphasis-2M,** **De-emphasis: 2 M**



Combined frequency response

vi) **Define frequency modulation. Draw the waveform of frequency modulation. How many number of sidebands present in frequency modulated wave?** **4M**

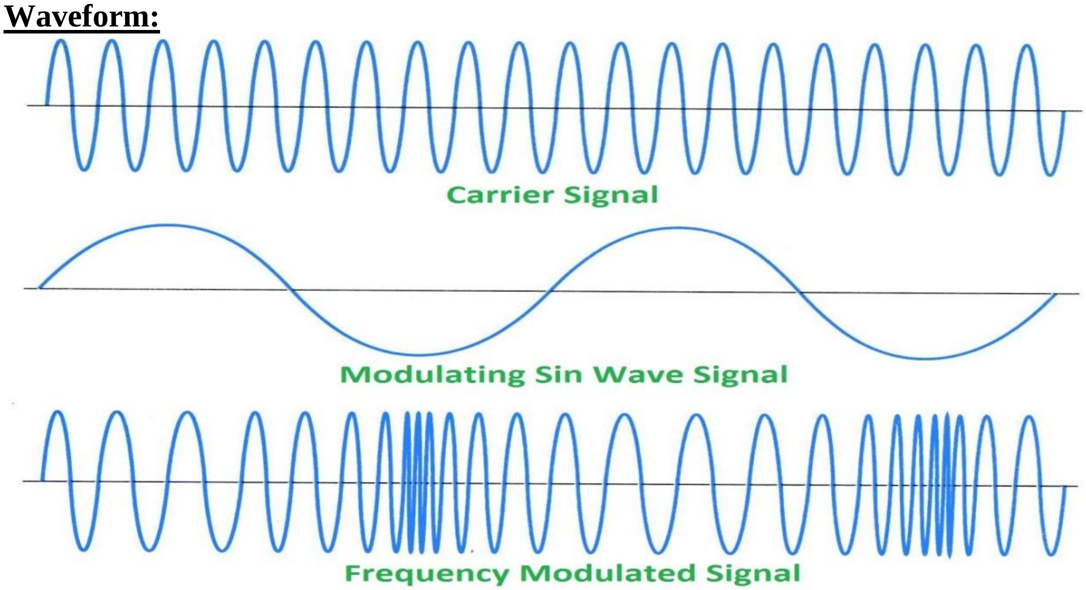
Ans:

<u>Frequency Modulation:</u>

The frequency of the carrier signal is vary with the instantanious value of amplitude of mudulating signal and amplitude and phase of modulating signal remains constant is called frequency modulation.

Definition:1 M, waveform:2 M, Number of sideband:1 M

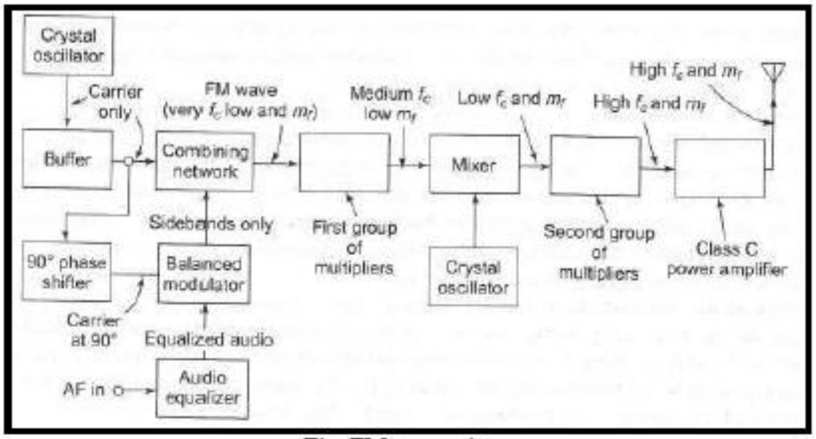
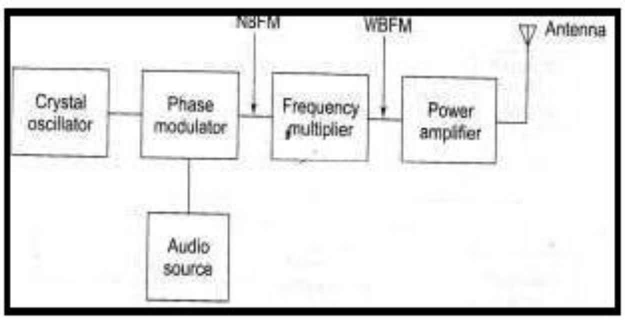
<u>Waveform:</u>

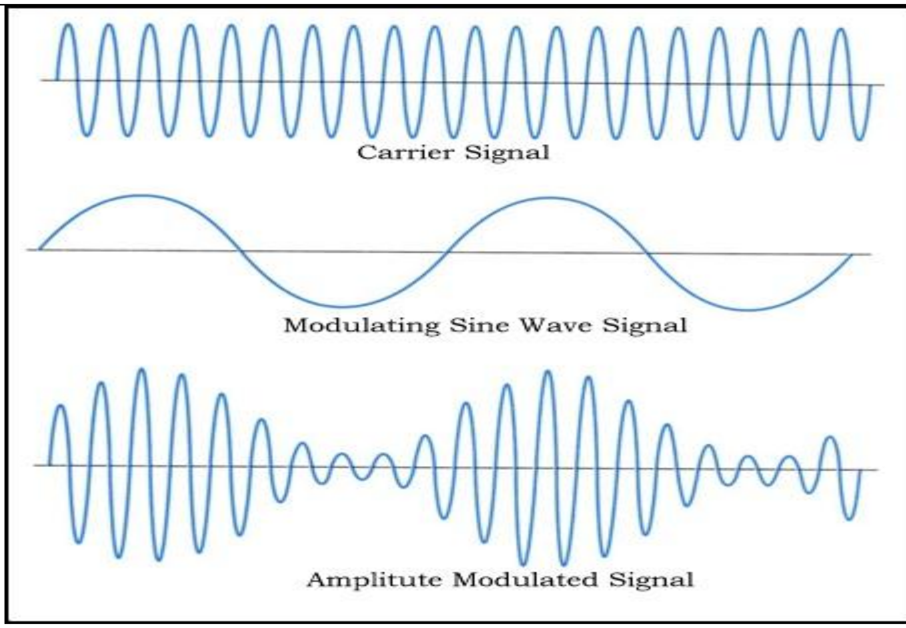


<u>Number of side band :</u>

The FM sidebands are dependent on both the level of deviation and the frequency of the modulation. In fact the total frequency modulation spectrum consists of the carrier plus an infinite number of sidebands spreading out on either side of the carrier at integral multiples of the modulating frequency.

The values for the levels of the sidebands can be seen to rise and fall with varying values of deviation and modulating frequency as seen in the diagram below. The parameters for the FM sidebands are determined by a formula using Bessel functions of the first kind.

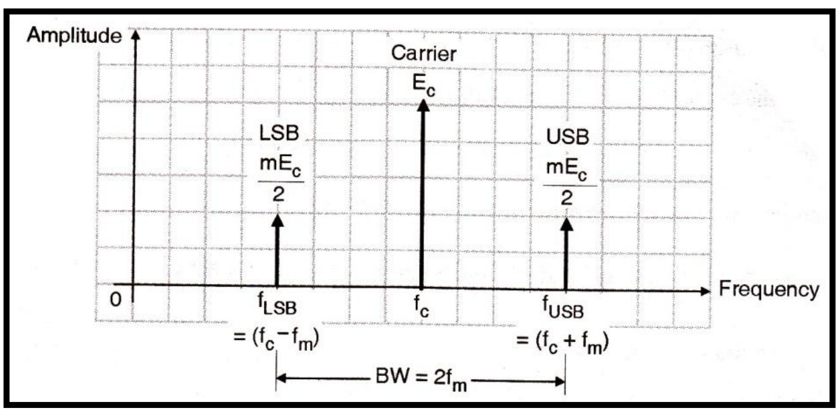
Q. 4	A)	**Attempt any four of the following:**	**16M**
	i)	**Draw the block diagram of FM transmitter. Enlist advantages of FM over AM.**	**4M**
	Ans:	**Diagram:**  **Fig. FM transmitter** **OR**  **Advantages FM over AM-** 1.FM is high noise immunity than AM 2.FM signal does not degrade easily than AM 3.Optimum power is utilized in FM than AM 4.Adjustant channel interference is less in FM than AM	**Diagram: 2M** **Advantages :2M(0.5 M for each pt.)**
	ii)	**Define amplitude modulation. Explain the term bandwidth of AM wave.**	**4M**
	Ans:	**Amplitude modulation:** The amplitude of the carrier signal is vary with the instantanious value of amplitude of mudulating signal and frequency and phase of modulating signal remains constant is called frequency modulation.	**Define AM: 2M, Bandwidth : 2M**



Bandwidth of AM wave:

The amplitude modulated signal consists of a carrier with two sidebands that extend out from the main carrier.

The B.W. of AM wave is twice the maximum modulating signal.

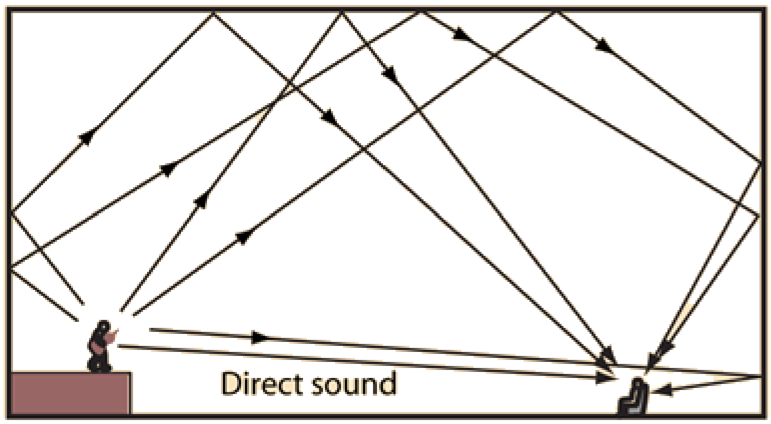


	iii)	**Enlist four specifications of PA system.**	**4M**
	Ans:	1.Acoustic feedback 2.Distribution of sound intensity 3.Reverberation 4.orientation of loudspeaker 5.Ambient noise 5.Dynamic range limitation 6.Selction of microphone 7.Sense of direction of source sound 8.Phase delay 9.Matching 10.Grounding	**1 M for each pt.**
	iv)	**Explain concept and necessity of reverberation.**	**4M**
	Ans:		

Reverberation:

Due to multiple reflection from walls, ceiling, floor etc. the sound in an enclosure fades away only gradually after the source of sound stops. This continuing eco is called reverberation**.**

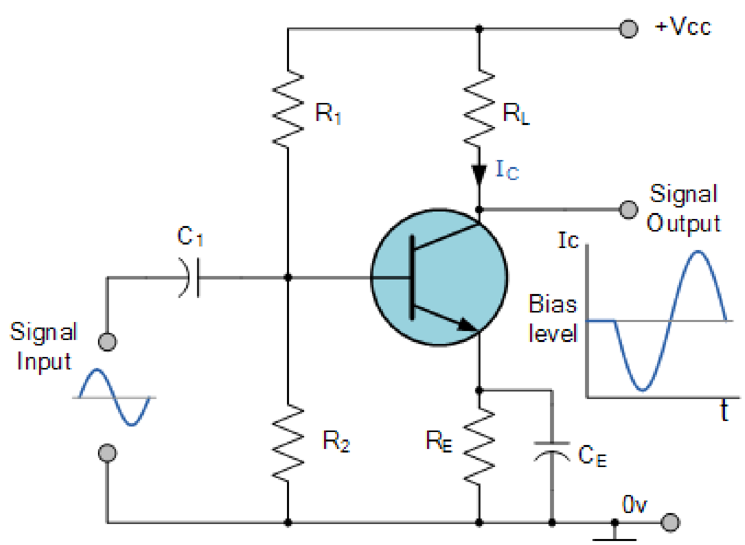
Concept: 2M
Necessity: 2M



Necessity of reverberation:
If all the echoes are eliminated by fixing sound absorbent material on the walls, ceiling, floor, the result would be a sound could be described as lifeless and unnatural.

Factor on reverberation time depends:
1.Volume of room
2.Surface area
3.Absorbtion coefficient of the surface area
4.Velocity of sound hence wavelength

v)	**Explain class-A voltage pre-amplifier.**	**4M**
Ans:	**Diagram:**	**2M**

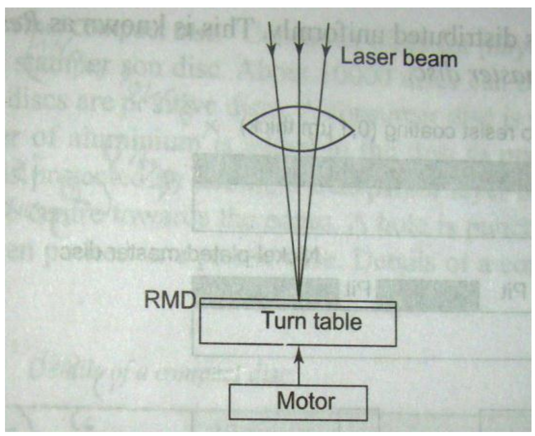


Explanation: **2M**

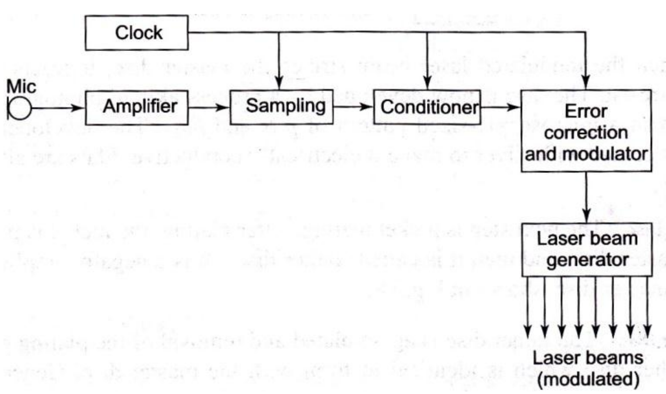
- Common Emitter (CE) amplifiers are designed to produce a large output voltage swing from a relatively small input signal voltage of only a few millivolt's and are used mainly as "small signal amplifiers"

- However, sometimes an amplifier is required to drive large resistive loads such as a loudspeaker or to drive a motor in a robot and for these types of applications where high switching currents are needed **Power Amplifiers** are required.
- The main function of the power amplifier, which are also known as a "large signal amplifier" is to deliver power, which is the product of voltage and current to the load.
- one of the main disadvantage of power amplifiers and especially the Class A amplifier is that their overall conversion efficiency is very low as large currents mean that a considerable amount of power is lost in the form of heat.

vi)	**Draw and explain optical recording on compact disc.**	**4M**
Ans:	**Diagram:**	**2M**



OR



Explanation: **2M**

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.

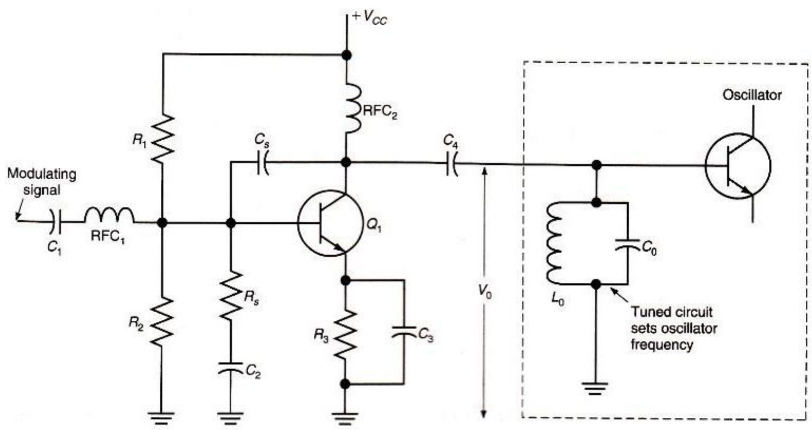
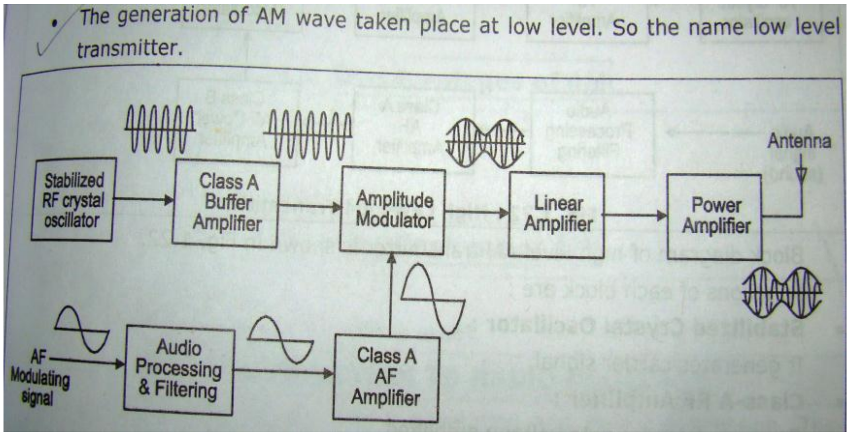
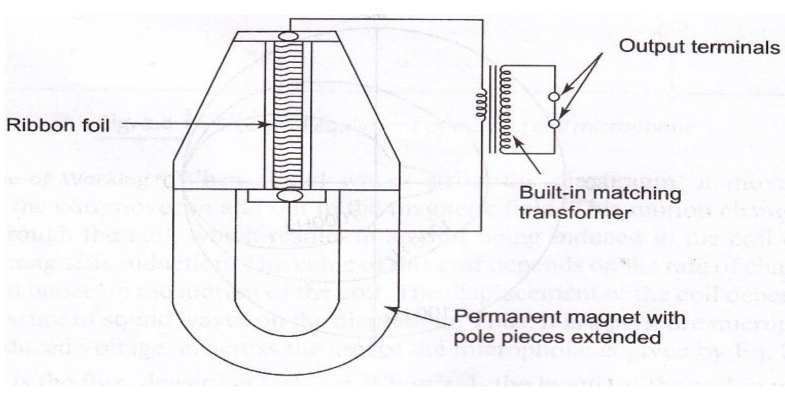
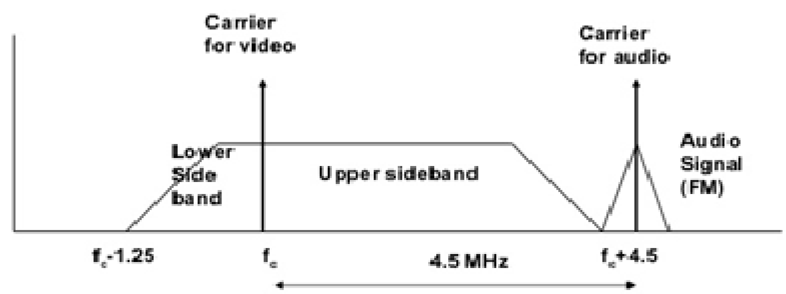
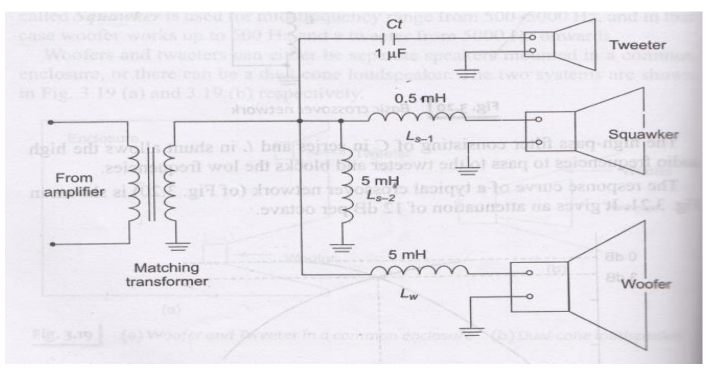
		• Recording is done on Resist Master Disc (RMD) with help of a powerful laser beam as shown as fig. The laser beam is modulated by digitized audio signals. The audio signal is sampled at rate of 44.1 KHz. the quantum level pertains to 16 bits.	
Q.5		**Attempt any**	**16M**
	i)	**Draw and explain reactance modulator for generation of FM.**	**4M**
	Ans :	**Diagram:**  **Explanation:** **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.	**2M** **2M**
	ii)	**With neat block diagram explain AM transmitter.**	**4M**
	Ans :		

		Diagram: **Low level AM transmitter OR High level AM transmitter.**  **Explaination:** **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.	**2M**
	iii)	**Explain ribbon microphone with construction and working principle.**	**4M**
	Ans:	**Diagram:**  	**2M**

		Explanation: **Construction:-** • 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:-** • 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.	**1M** **1M**
	iv)	**Compare monophony and stereophony (any 4 points).**	**4M**
	Ans:	(see table below)	**1 M for each pt.**

Monophony	Stereophony
1. Only one amplifier is used	1.At least two independent amplifier are used.
2.No naturalness	2.Provides naturalness of sound signal.
3.Listener cannot judge the direction of sound	3. Listener cann judge the direction of sound
4.Low cost	4.High cost

	v)	**Explain, how noise cancelling is done in radio noise cancelling microphones?**	**4M**
	Ans:	Noise cancellation is an electronic process. It was originally developed to improve radio communications in noisy environments like aircraft cockpits. There's a microphone built into each cup of the headphones that samples the ambient noise near the ear. That noise signal is fed into an electronic circuit that analyzes it and creates a mirror image of the noise, then adds the noise back into the music signal. Some of the real noise is cancelled out by the mirror image inverse noise. All such microphones have at least two ports through which sound enters; a front port normally oriented toward the desired sound and another port that's more distant. The microphone's diaphragm is placed between the two ports; sound arriving from an ambient sound field reaches both ports more or less equally. Sound that's much closer to the front port than to the rear will make more of a pressure gradient between the front and back of the diaphragm, causing it to move more. The microphone's proximity effect is adjusted so that flat frequency response is achieved for sound sources very close to the front of the mic – typically 1 to 3 cm. Sounds arriving from other angles are subject to steep midrange and bass roll off.	
	vi)	**Explain the concept of vestigial side band transmission.**	**4M**

	Ans :	**<u>Diagram:</u>**  Frequency spectrum of a Vestigial Sideband	**2M**
		<u>Explanation: -</u> • In the TV broadcasting the audio carrier in frequency modulated and video information in amplitude modulated. • The picture carrier is transmitted, but one sideband is partially A portion of the lower side band of the TV signal suppressed leaving only a small vestige of lower sideband. Such an arrangement is known as vestigial sideband signal. • Video signal above 0.75 MHz (750 KHz) are suppressed in the lower sideband while all video frequency are transmitted in the upper vestigial sideband suppressed.	**2M**
Q.6		**Attempt any four of the following:**	**16M**
	i)	**Draw the circuit diagram of 3 way speaker system. Explain its operation.**	**4M**
	Ans:	**<u>Diagram:</u>**  	**2M**
		<u>Explanation:</u> • 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.	**2M**

- L_{s1} and L_{s2} allows only mid frequency range to reach to squawker

ii) Draw the circuit diagram of different stereo controls. Explain balance control and blend control. — **4M**

Ans: **<u>Circuit diagram of Balance control and blend control-</u>**

Diagram: 2M; Explanation: 2M

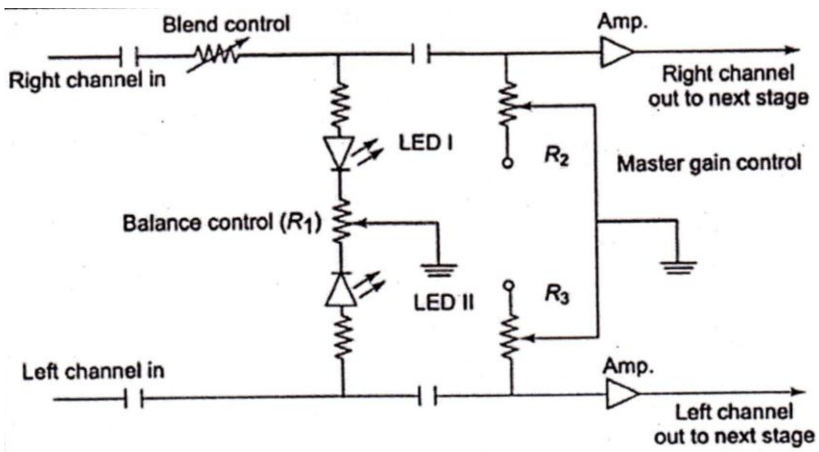


Fig. *Balance control, master gain control and blend control*

Balance control:

- Two amplifier of a stereo system, all through independent of each other, are built as matched pair to give equal output for the same input. In spite of the two amplifier being identical, there may be variation in the output of each channel due to variation in the characteristic of transistor and ICs.
- When the POT is set in center, the current through LED1 and LED2 should be identical, if the signals in the left and right channels are equal. In that case both LED will be equally bright.
- In case of any inequality, the two brightness level will also become unequal. When balance control is moved down, the output of the left channel will increase while that of right one will decrease, and vice-versa when moved up.

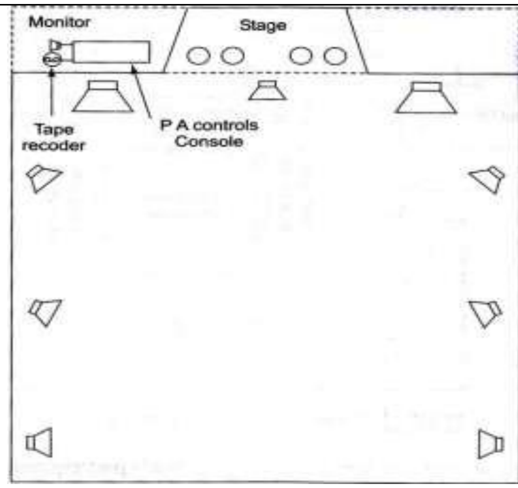
Blend Control:

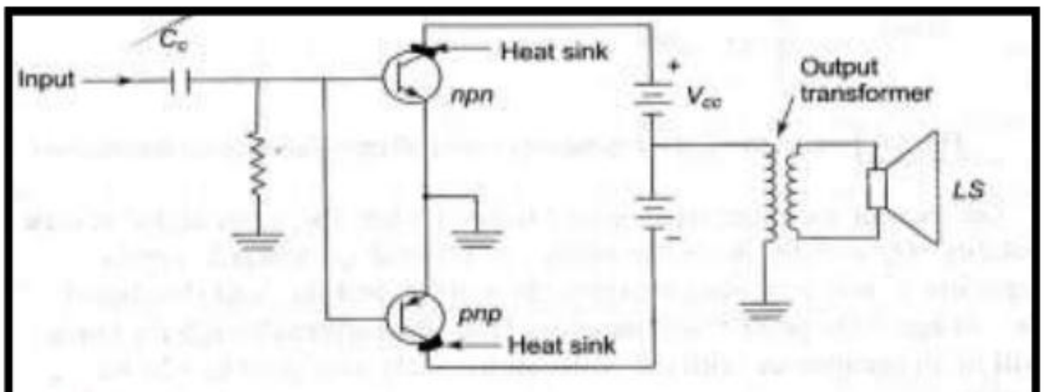
- The stereo effect is diluted by this control when there is too much left-right effect. Diluting is done by misbalancing the two channels.
- Blend control POT is set at zero resistance for balanced output. For disturbing the balance, this is advanced further to reduce gain of the left channel.
- Although blending can be done by balance control also, but once set, the balance control is not disturbed.

iii) Explain typical PA installation plan for public meeting. — **4M**

Ans: **<u>Diagram:</u>**

1.5M Explanation:

		 **<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.	**2.5M**
	iv)	**What is fidelity? Explain causes affecting fidelity.**	**4M**
	Ans:	**<u>Fidelity:</u>** High fidelity sound can be obtained from the recorded stereo sound. The stereo signal is fed to two independent amplification chennels through tape mic switch **<u>Causes affecting fidelity:</u>** • High signal to noise ratio(S/N ratio) • Flat frequency response • Low nonlinear distortion • Large dynamic range • Creating sense of direction. **<u>Remedies:</u>** • S/N ratio can be improved by using preamplifier of low noise figures proper shielding,	**1M** **(Any four-3M)**

		grounding, • Decoupling & filtering circuits, stabilized power supply, microphones • By using coupling capacitor and shunt capacitor in audio amplifier circuits • Nonlinear distortion can be reduced by using negative feedback in amplifier, designing bias circuitto keep Q point in the middle of linear portion of the characteristics curve. • Dynamic range can be increased by using solid-state amplifier; dynamic microphones & L.S.which are capable of withstanding the large change in loudness. • Creating sense of direction can be improved by using high fidelity system.	
	v)	**Draw and explain complementary symmetry push pull amplifier.**	**4M**
	Ans:	**<u>Diagram:</u>**  Fig. Complementary symmetry push-pull amplifier circuit with output transformer **<u>Explanation:</u>** • 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.	**2M** **2M**
	vi)	**Convert the carrier frequency 25MH$_Z$ and modulating frequency 400 H$_Z$ in rad/s. What is modulation index of FM if δ = 10 KH$_Z$ and fm = 400 H$_Z$?**	**4M**
	Ans:	**For fc= 25 MHz** $\omega_{(rad/s)} = 2\pi\times f_{(Hz)}$ $= 2\pi\times 25000000$ $=157079632.5$ **For fc= 400 Hz** $\omega_{(rad/s)} = 2\pi\times f_{(Hz)}$	**Convert the carrier frequency in rad/s: 3M(1.5 M each)** **Modulation index: 1M**

		$= 2\pi\times400$ $= 2513.27412$ Modulation Index = δ/ fm = 10 Khz/400 Hz = 25	