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 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.1 Attempt any ten : **20 M**

a) What is timbre and state its unit?

Ans:-

Timbre :- **01 M**

Sound waves consists of fundamental frequencies (tones) their harmonies other frequencies (over tones). The proportion of tones, overtones in a sound is called Timbre. This is a special characteristics by which a particular sound can be recognized.

It is a unit less quantity **01 M**

b) State the limit of audibility dynamic range?

Ans:-

Audibility dynamic range :-

i) Threshold of hearing = 20 * 10^{-6} pa (Lower limit) **01M**

ii) Threshold of pain = 63 pa (upper limit) **01 M**

c) What is multiway speaker system?

Ans:- **Multiway Speaker system :-** **02 M**

A single loudspeaker cannot have flat frequency response from 16 Hz to 20 KHz. Therefore audio frequency spectrum is divided into two or three parts. Separate speakers are designed for each part. Thus each speaker covers a small frequency range. Such system is called multiway speaker system.

d) List different types of audio amplifier?

Ans:-

Different types of audio amplifiers:-

- Audio amplifiers are classified as:- **01 M**
  i) Voltage amplifiers
  ii) Power amplifiers
- Amplifiers are also classified as :- **01 M**
  i) Class A
  ii) Class B
  iii) Class C amplifiers

e) What is Surround sound?

Ans:-

Surround sound: - **02 M**

In digital system artificial delay is introduced in the signal produced by two channels of stereophonic system. The delay so introduced simulates natural delay in sound reflected from walls ceiling. Such signals are then fed to various audio channels that surround the listener.

The sound appears around the listener by 360^0.

This enables the listener to experience sound coming from all directions depending on the source material.

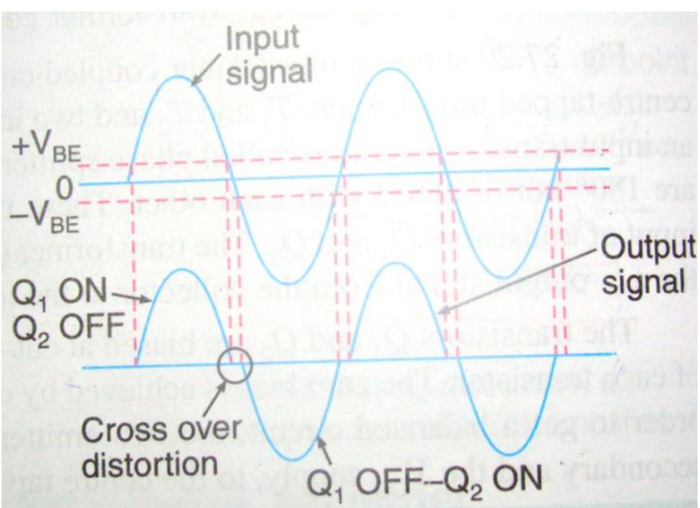
f) Define cross over distortion?

Ans:-

Cross over distortion:- **02 M**

In class B push pull amplifier two transistors are used. But only one conducts at a time. One amplifies positive half cycle & other negative half cycle.

But due to voltage drop across B –E junction of each transistor, they do not 'conduct' for complete half cycle, thus nature of o/p is shown below:



The signal is distorted when it crosses ‘ Zero’ level. Therefore distortion is called cross over distortion

g) What is PA system?

Ans:- **PA System:-** **02 M**

When a large gathering is to be addressed, sound needs to be amplifies so that people at a distance from stage may receive good intensity of sound for comfortable listening.

The system which fulfills this function is called PA system or public address.

h) State the need of biasing and list its types?

Ans:- **Need of Biasing:-** **01 M**

Biasing is needed to avoid the distortion while recording the sound signal.

Types of biasing :- **01 M**

1. AC biasing
2. DC biasing

i) State the principle of optical recording?

Ans:- Principle of optical recording :- **02 M**

The audio signal is sampled and coded in digital form. This signal is used to make the laser beam ‘ON’ & ‘OFF’.

This beam is used to make ‘pits’ flats on CD surface. Thus optical recording is carried out.

j) State the need of modulation ?

Ans:- Need of Modulation:- **02 M**

- Low frequency signals cannot be transmitted for long distance that's why there is a need of modulating the information signal in short, it improves the signal strength.
- To reduced antenna heights, noise & distortion
- To narrow banding the signal, to reduce the complexity of the equipment
- To increase the bandwidth of the signal & to multiplexed more number of the signal

k) Define AM and FM?

Ans:- **AM (Amplitude Modulation) :-** **01 M**

The modulation technique in which amplitude of carrier signal is changed according to amplitude variations in modulating signal is called amplitude modulation.

FM (Frequency Modulation) :- **01 M**

This is a modulation technique in which frequency at carrier signal is changed according to amplitude variations in modulation signal.

l) What are different controls of audio amplifiers?

Ans:- Different controls of audio amplifiers:- **02 M**

i) Microphone gain control

ii) Volume control

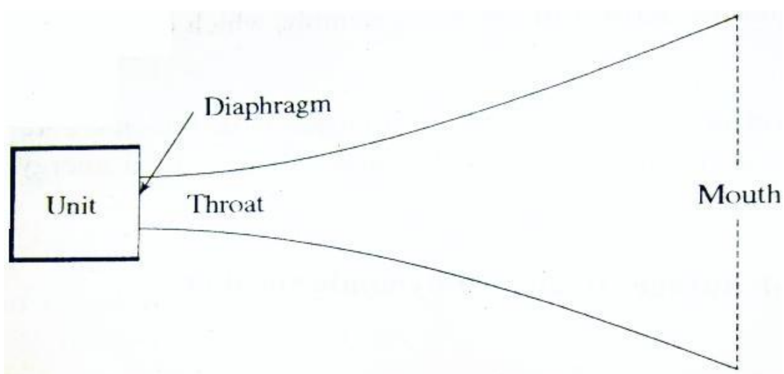
iii) Tone control

The tone control comprises of Bass control & treble control.

Q.2 Attempt any four : 16 M

a) Draw the construction of Horn type loudspeaker and explain its working?

Ans:-



The horn is tapered enclosure whose diameters increases from a small value at one end (called 'throat') to a large value at the other end (called 'mouth')

There is an air chamber trapped between throat and diaphragm. The chamber is lined with sound absorbing material like loose felt. The driver unit is similar to direct radiating type except that the paper cone is not present. The cross sectional area of the cone increases along its length from its throat to mouth.

Working:- Here, the unit consist of moving coil microphone. The acoustic power from mic is not radiated directly in listeners area. The power is first delivered to the air trapped in a horn and then from there it is delivered to the air in the listeners area.

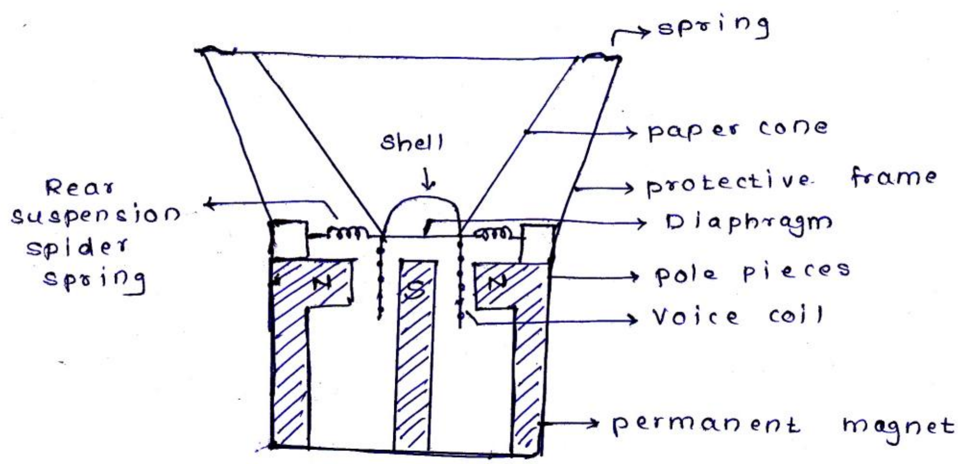
The horn acts as acoustic transformer. It allows better impedance match between low impedance of free air & high impedance of vibrating voice coil assembly.

(Note:- Even if students draw only diagram & do not write constructional details in words then also 02 marks should be given)

b) Draw the construction sketch of moving coil cone type of loudspeaker and explain its working?

Ans:-

Constructional sketch :- **02 M**



Working:- **02 M**

When an audio current flow through the voice coil placed in a magnetic field, a force equal to B1 i newton acts on the coil & moves it to and fr0. The paper cone attached to the coil also moves causes compression rare faction cycles in the air. Thus audio current is converted in to sound waves.

The amplitude & frequency of vibration of paper cone depends on amplitude & frequency of audio signal.

c) Explain the term reverberation and state the factors on which it depends?

Ans:- Reverberation :- **02 M**

A person receives sound directly from the source as well as sound reflected from the walls, ceiling, floor etc. The reflected sound is heard as a distinct echo if the time gap between the original (direct) wave and the reflected wave is more than 60ms.

Reflections over shorter distance shall simply prolong the sound due to multiple reflections in the hall. It is because the listener receivers direct sound as well as reflected waves.

The sound persists even after the source of sound has stopped sounding and fades away gradually. This gradual fading of continuing echo is called reverberation.

Factors on which reverberation depends:- **02 M**

i) Volume of the room V

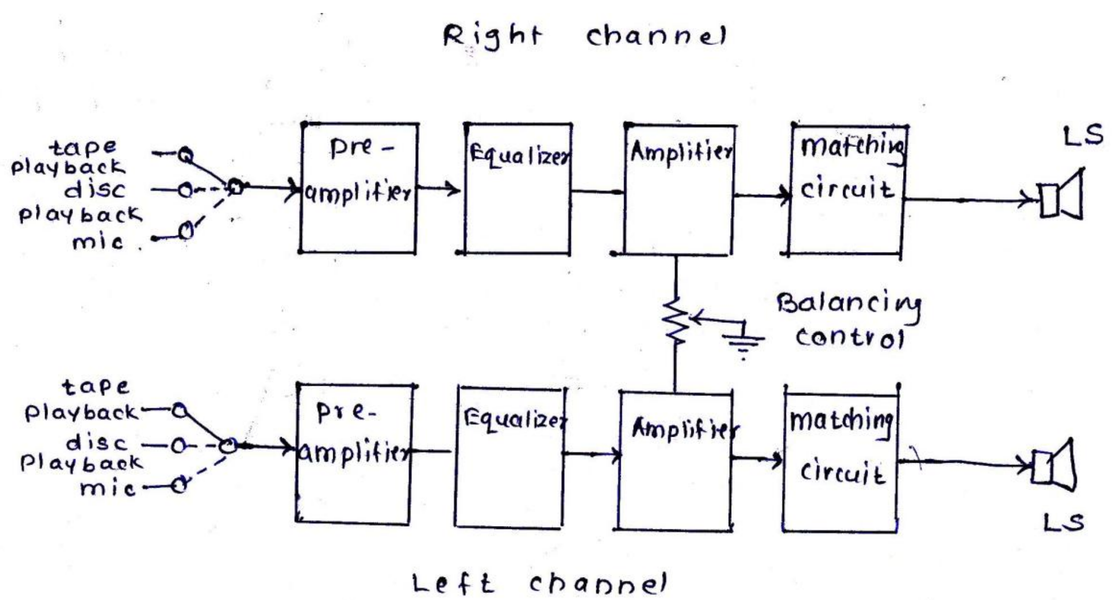
ii) Surface area

iii) Absorption coefficient of surface area

iv) Velocity (and hence wavelength) of sound.

d) Draw the block diagram of Hi – Fi reproducing system and explain its working?

Ans:- Block Diagram of Hi – Fi system:- **02 M**



Working:- **02 M**

In this system, the stereo signal is fed to two independent amplification channels through the tape – mic switch.

The amplifier system consists of low noise high gain preamplifier, equalizer, amplifier with flat frequency response, matching transformer.

A balancing control is added to balance out any imbalance in the characteristics of otherwise identical circuits.

The secondary of matching transformer is connected to respective loudspeaker column.

All the blocks are designed to get

i) Flat frequency response from 40Hz – 15000Kz

ii) Little distortion (< 1%)

iii) High signal to noise ratio (> 50dB)

iv) High dynamic range (100dB)

e) Explain the concept of Graphic equalizer?

Ans:- Concept of Graphic equalizer :- **04 M**

- Graphic equalizer is used to eliminate unwanted peaks in the frequency response of audio system
- The complete audio spectrum is segmented into narrow bands.
- Each band has an individual slider control which can boast or cut the signal from +15dB to –15dB.
- Simple audio equalizer can have 5 to 10 bands whereas professional equalizer can have 25 – 30 bands.

f) State the working of mono – stereo amplifier and state its characteristics?

Ans:- Working of mono – stereo amplifier:-

{Note :- The question is not clear Mono & stereo these are two different types of amplifiers. But there does not exist any mono – stereo amplifier. So even if student write working of mono amplifier or stereo amplifier full marks should be given.}

Mono amplifier:- **02 M**

- It consist of only one amplifier it combines all signals from all sources and produces resultant signal.
- The amplifier o/p is fed to a signal or several speakers.
- All loudspeakers shall give the same resultant sound. The ears will interpret the reproduced sound to be coming only from one source of sound.

Characteristics:- **02 M**

i) Uses only one amplifier.

ii) Less expensive

iii) Easy to record. Requires only basic equipment

iv) Do not create an illusion of directionality & audible perspective.

OR

Stereo amplifier:- **02 M**

- Such system uses independent amplifiers. These amplifiers have their own set of loudspeakers for o/p
- The two sets of amplifiers are called as left channel & right channel.
- In these amplifiers, actual sources of sound are virtually transferred to respective loud speakers.

Characteristics:- **02 M**

i) Uses two or more amplifiers

ii) More expensive for recording & reproduction .

iii) Requires technical knowledge & skill to record

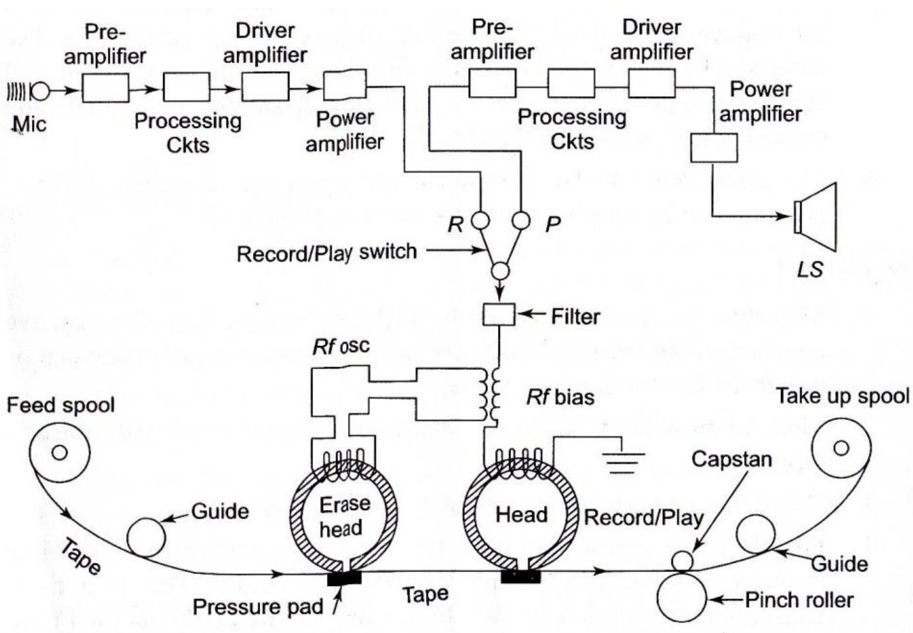
iv) Require special equipment's for recording

v) Create the impression of sound heard from various directions as in natural hearing.

Q.3 Answer any two **16 M**

a) Draw the block diagram of magnetic tape recording and reproducing system and explain its working?

Ans:- Block Diagram :- **04 M**



Block diagram of a tape recorder

Function of block :- **04 M**

Microphone :- Sound waves strikes the diaphragm of the microphone which converts the sound – pressure variations into electrical signals, called audio signals.

Pre - amplifier :- It amplifies the work output of the microphone. Its noise figure is low.

Processing circuits:- These circuits in the record section control the gain and level of recording and also provide de – emphasis and pre – emphasis for low – frequency and high frequency audio signals, respectively.

Driver - Amplifier:- It gives further voltage amplification to the signal so as to reduce the internal resistance of the power amplifier and hence, to drive it to give power amplification.

Power Amplifier:- It amplifies the power of the audio signal so as to drive the record head.

Filter:- It is a trap circuit which does not allow the bias oscillator's signal to go to the amplifier as it will unnecessarily get overloaded.

Heads:- There is one erase head before the record head, pressing on the tape. It erases all previous recordings.

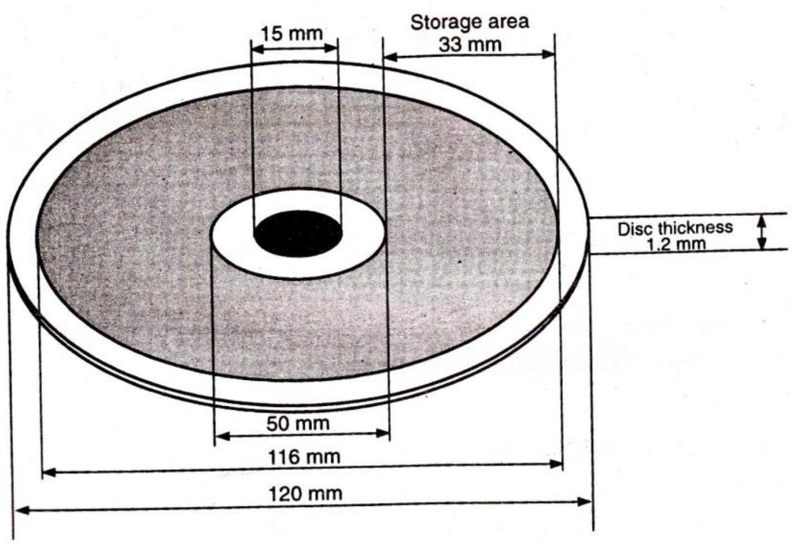
The next to erase head is the record head **(**which is also generally the play back head as shown in block diagram. For professional hi – fi recording, a separate play back head is used after the record head**).** The record head records audio current in the form of a varying magnetic field in the magnetic material of the tape.

Record/ Play switch:- It connects the recording power amplifier to the record/play head for recording through record (R) terminal of the record/play switch. When this switch is connected the terminal marked (P), the head is connected to the pre – amplifier of the playback section.

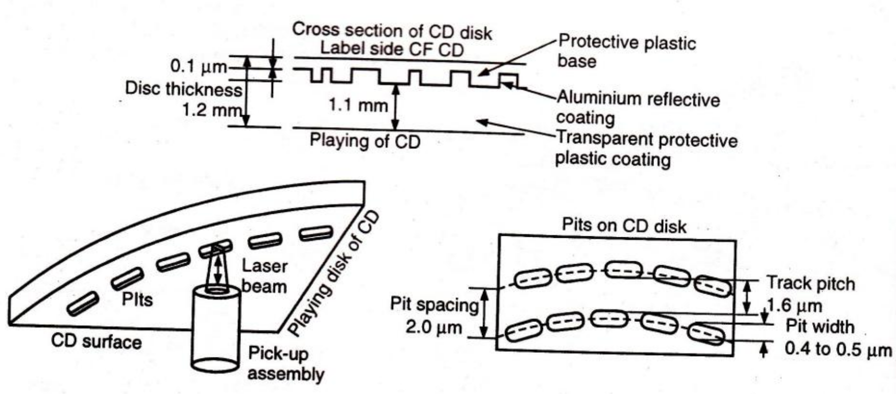
Pre – amplifier, Driver amplifier and power amplifier:- In the playback path have the same function as described above for the recording path. The processor does just the opposite of what has been done by the processor in the recording chain. It de – emphasises the high audio frequency signals and emphasizes the low – frequency signal to restore the original proportionalities of sound. It also contains volume control, and treble and bass controls.

b) Draw the structure of Compact Disc and explain the principal of recording.

Ans:- Diagram:-



Construction of a Compact Disc (CD) **01 M**



Compact Disc (CD) structure **03 M**

Explanation:- **0 4 M**

The track on the CD is read by the CD player from the innermost track to the outer most track. This is reverse of the process followed by normal LP records, where the grooves are read from the outermost groove to the innermost groove.

We know that the information is stored on the CD surface in the form of number of small pits in circular tracks. As shown in figure, these pits have a depth of around 0.1µm.

The distance between each spiral track is around 1.6µm. This distance is known s "track pitch". The distance between centerline of one track and the centerline of adjoining track is around 2.0µm and this distance is known as pit spacing.

A typical CD consists of reflective surface coated with aluminium layer.

This reflective surface is covered with transparent plastic or lacquer coating. This coating acts as a protective layer for the reflective aluminium surface.

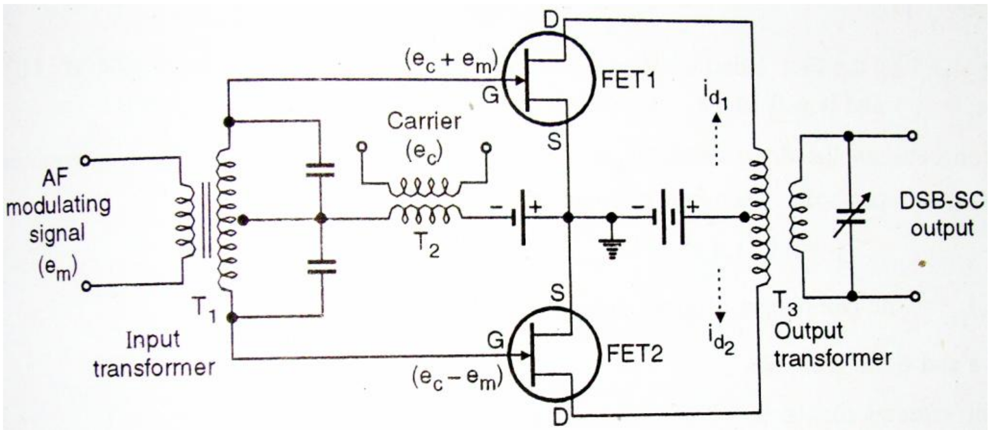
Because of this protective coating, even if the disc becomes dirty one can clean it with the soap water and use the disc without any problem.

If there is some permanent damage such as a deep scratch on the surface of the disc, which will block the laser beam from reaching the reflective surface of the disc, then only the CD player will not be able to read the information stored at that area of the disc.

c) Draw the diagram of SSB generation by balanced modulator and explain its working?

Ans:- **Balance modulators (using FETS):**

Circuit diagram **03 M**



To balanced modulators are used. To suppers the unwanted carrier in an AM wave. The carrier and modulation signals an applied to the input of the balanced modular. And we get the DSBSE O/P

Principle of operation - **05 M**

If two signal at different frequencies are parsed through a "nonlinear"-

Resistance then at the output waged an AM signal with supplement carrier.

The device can be diode or JFET or transistor.

The carrier voltage in applied "in phase "to the two gates via the transformers T2 & T1 however the modulating signal appears 180 out of phase at the fates, this is because the input transformer T1 is a center tapped transformer.

i) **Operation in the absence of modulating signal**-:

In the absence of modulating signal both the FETs conduct simultaneously due to the in-phase carrier voltage applied to their gates. The drain currents are equal in magnitude but opposite in direction through the primary of output transformer T3. Due to this their magnetic fields cancel each other inducing a zero secondary voltage. Thus the output of transformer T3 in zero the carrier is thus suppressed.

ii) Operation when carrier and modulating signal both are present -

When modulating signal in applied, the drain current of the two FETS flow due to the combined effect of carrier and the modulating signal. The FET currents due to carrier are equal and opposite and hence each cancels each other. But FET currents due to the modulating signal are equal but not opposite so they do not cancel out.

This is because the modulating signals in applied 180 out of phase to the two FETS.

At the output the current we get DSBSC signal.

For 100% suppression of carrier, both the FETS must have identical characteristics

I.e. it should be a matched pair and the transformer center taps must be exactly at the center of the winding .

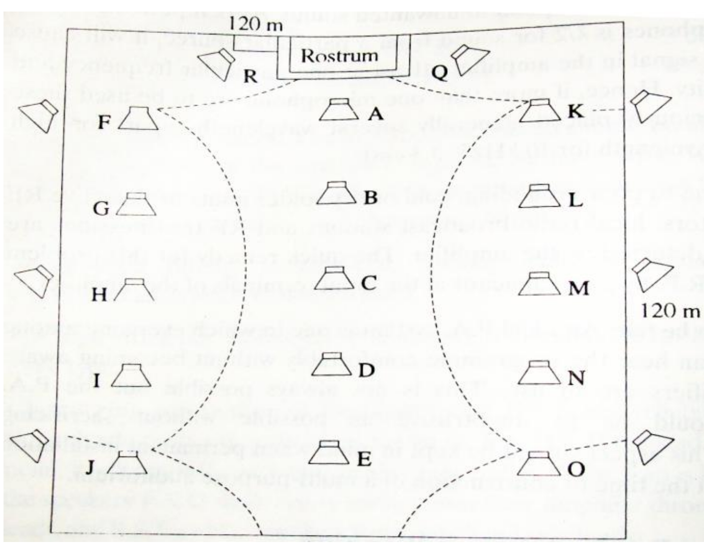
Practically this is not possible hence carrier will be heavily. Suppressed but not completely removed.

For SSB one side band can be suppers/ filleted by filler method.

Q.4. ATTEMPT ANY TWO **16 M**

a) Explain how will you install PA system for public meeting

Ans: PA system configuration Public meeting- **(01 M for each point)**



1) The loudspeakers A, B, C, D and E in the center line will give the sense of direction to most of the audience and can be mounted on poles.
2) Loud speakers F, G, H and I on one side and K, L, M and N on the other side will give full coverage to meeting ground on both sides of the central area.

3) To cover the remote semicircular side and corner areas, loud-speakers J. and O are used. These will throw sound power towards corners.
4) The loudspeakers Q and R will cover the left and right sides, respectively near the rostrum.
5) There may be some loudspeakers to give coverage to audience standing outside the meeting park. These may be slightly inclined.
6) Microphones should be of cardioids type and the loudspeakers may be of horn type.
7) The output audio power of the amplifier may be calculated by using the formula given
8) It is preferable to use hot standby with batteries.

b) Predict what will happen if

I. The pinch roller in the tape recorder is worn out.

II. The tension pulley in the tape recorder is broken

Ans:

I) **Pinch roller:-** **04 M**

It is a plastic cylinder coated with rubber. it has a rough surface. During play and recording the tape moves in a controlled manner between the pinch roller and the capstan. The pressure of the pinch roller on the capstan determines the speed of the tape. The capstan revolves at a very precise rate to pull the tape across the head at exactly the right speed. The roller simply applies the pressure so that the tape is tight against the capstan. If the pressure is less than the prescribed value then the tape may start slipping resulting in an increase in speed, whereas if it is more then it may crush the tape .hence if pinch roller in the tape recorder is worn out then the pressure is less than required value and tape may start slipping results in increase in speed the sound is like fluttered sound

II) Tension pulley:- **04 M**

It is used to move the tape with proper pressure towards the R/P head. if it is broken then the magnetic tape will not move and the sound will not be heard at the loudspeaker

c) Compare

i. FM and PM(any 4 points)

ii. SSB and VSB(any 2 points)

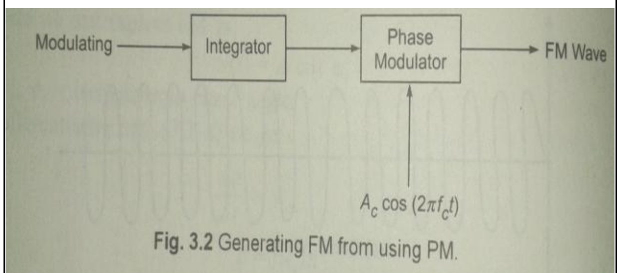
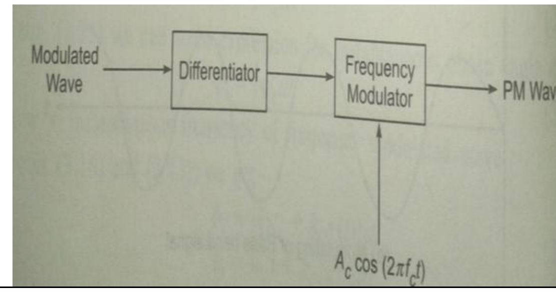
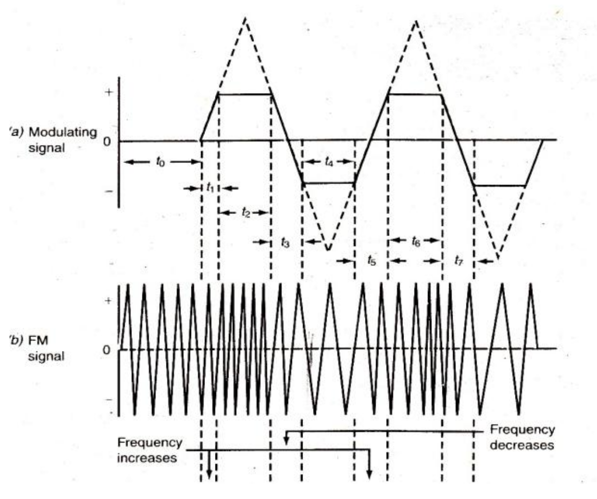
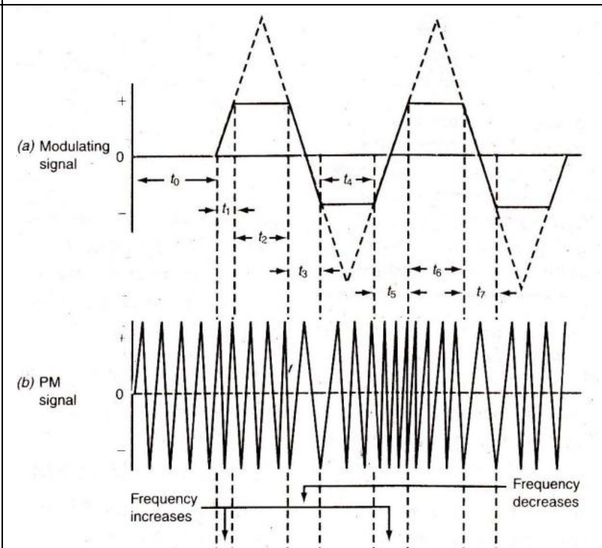
Ans:-

SSB and VSB:- (any 2 points) **02 M for each point**

SR NO	SSB	SR NO	VSB
1	Most difficult modulation and demodulation	1	less difficult modulation and demodulation as compared to SSB-AM
2	Less power and most efficient bandwidth	2	More bandwidth
3	Application: long distance transmission of voice signal	3	Application: to transmit video signal of TV ie. in TV broadcasting
4	SSB is too Expensive to implement	4	Easy to implement

i. **FM and PM :- (any 4 points) :-** **01 M for each point**

SR NO	**FM**	**SR NO**	**PM**
1	Frequency of the carrier is varied according to the modulating signal.	1	Phase angle of the carrier is varied according to the modulating signal.
2	FM can be generated by integrating PM and then using the resulting signal to phase modulate the carrier.	2	PM signal can be generated by first differentiating modulating signal and then using the resulting signal as the input to a frequency modulator to modulate the carrier.

3	 Fig. 3.2 Generating FM from using PM.	3	
4	In FM wave the maximum deviation occurs at the peak positive and negative amplitude of the modulating signal.	4	In PM the maximum frequency deviation modulator occurs during the ‘time’ that the modulating signal is changing at its most rapid rate.
5		5	
6.	The Fm Modulation Index Will Increase As Modulating Frequency Is Reduced And Vice Versa	6.	The Pm Modulating Index Will Remain Constant As Modulating Frequency Is Change.

Q.5 ATTEMPT ANY FOUR **16 M**

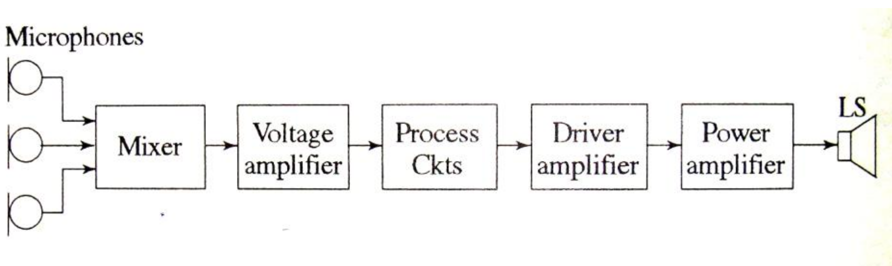
a) Explain the necessity of line transformer

Ans: Necessity of line transformer in PA system: **01 M for each point**

- Line transformer is a transformer connected between o/p of PA system & loud speaker.
- It is used for impedance matching to allow maximum power transfer.
- It is a step up transformer which also boosts signal voltage to some extent.
- It is placed very close to the loudspeaker usually in the loudspeaker cabinet only to avoid signal loss in the connecting wires between transformer & loudspeakers.

b) With neat block diagram, explain the function of PA system.

Ans: Public Address system- **01 M**



Explanation :- **03 M**

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. It may be built in unit or a separate plug-in unit.

   Three type of mixers

   1) Simplest – no amplifiers only gain controls (faders) and isolating services resistors.
   2) Little sophisticated- common amplifiers after isolating resistors.
   3) Most sophisticated – Has separate pre amplifier for separate channels then after gain control potentiometers and isolation resistor. There is a common amplifier followers
   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 drivers 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.

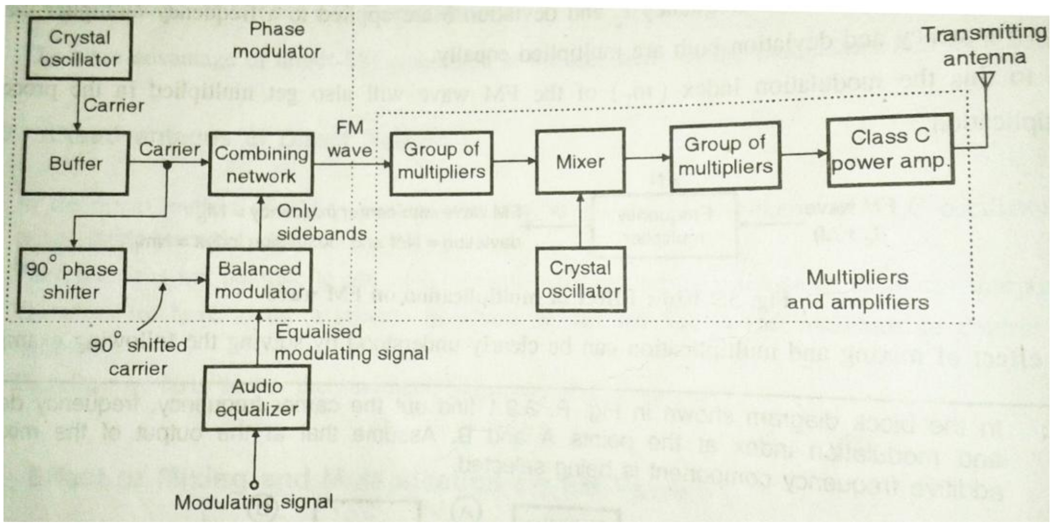
c) Write the uses of PA system.

Ans: Uses of PA system :- **04 M**

1. The intensity of sound decreases with distances. 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.
2. It is used in sports meeting, public meeting auditorium, concerts, functions etc.
3. It is also used to convey information to isolated location like, railway station airport, hospitals, Factories, schools etc.

d) With neat block diagram, explain the operation of FM transmitter using Armstrong method

Ans:- Block diagram:- **02 M**



Operation:- **02 M**

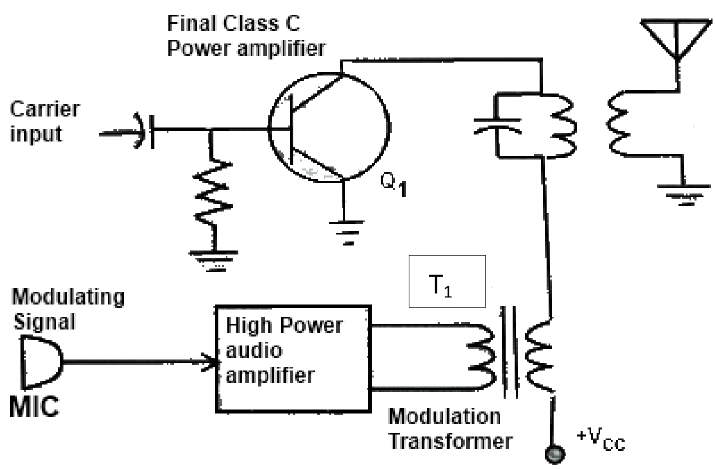
- The crystal oscillator generates the carrier at low frequency typically at 1 MHz. This is applied to the combining network and a 90 degree phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies. The modulating signal is then applied to a 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 F_c and low value of the modulating index m_f .
- 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 F_c and m_f both are raised to required high values using the second group of multipliers.
- The FM signal with high F_c and high m_f is then passed through a class C power amplifier to raise the power level of the FM signal

e) Explain the method of AM generation by using collector modulation method using neat circuit diagram

Ans: Collector Modulator:- **02 M**

1. This circuit shows the output stage of the transmitter. It is a high power radio frequency class C amplifier. The Class C amplifiers conduct for only a portion of the positive half cycle of carrier signal applied at the base of transistor Q_1.thus collector current of Q_1 is in the form of current pulses The collector current pulses cause the tuned circuit to oscillate or ring at the desired output frequency. The tuned circuit, therefore, reproduces the negative portion of the carrier signal

2. The modulator is a linear power amplifier that takes the low level modulating signal and amplifies it to a high power level. The modulating output signal is coupled through modulation transformer T_1 to the class C amplifier. The secondary winding of the modulation transformer is connected in series with the collector supply voltage Vcc of the class C amplifier.



02 M

3. With zero modulation input signal. There will be zero modulation voltage across the secondary of T_1. Therefore, the collector supply voltage will be applied directly to the class C amplifier, and the output carrier will be a steady sine wave.
   When the modulation signal occurs, the AC voltage across the secondary of the modulation transformer will be added to and subtracted from the collector supply voltage.
4. This varying supply voltage is applied to the class C amplifier. Naturally, the amplitude of the current pulses through transistor Q_1 will vary. As a result, the amplitude of the carrier sine wave varies in accordance with the modulated signal. For example, when the modulating signal goes positive, it adds to the collector supply voltage, thereby increasing its value and causing higher

current pulses and a higher amplitude carrier. When the modulating signal goes negative, it subtracts from the collector supply voltage making it less. For that reason, the class C amplifier current pulses are smaller, thereby causing a lower amplitude carrier output. Hence amplitude modulated wave is obtained which is then transmitted through antenna

f) Explain the pre-emphasis and de-emphasis

Ans: Concept of pre-emphasis & de-emphasis with respect to audio recording:-

Emphasising low intensity sound before recording is called pre-emphasis the process of de-emphasising the playback circuit to bring originality is called equalization.

Pre-emphasis :- **02 M**

Noise signal becomes more significant during quiet passage of music .therefore it is desirable to emphasize a low power notes before recording so that these are at much higher level than noise.

D**e-emphasis** :- **02 M**

At the receiver, it is essential that the reproduced sound possess the same proportions of intensities for low & high notes as were present in the original sound, De-emphasis will bring back the originality.

Application:

It is needed to improve signal to noise ratio to maintain high fidelity in the reproduced sound.

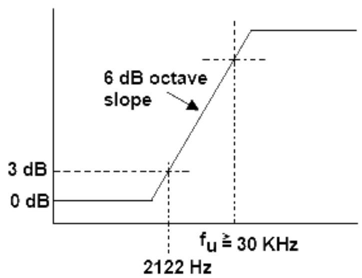
OR

Concept of pre-emphasis & de-emphasis with respect to modulation:

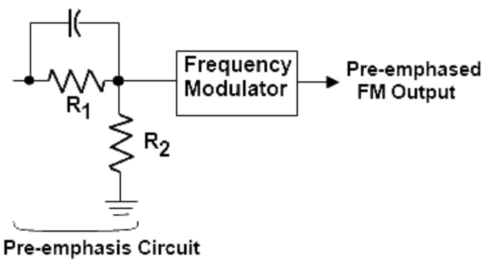
Pre-emphasis:- **02 M**

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

Pre-emphasis circuit:-



(b) Pre-emphasis Curve



(a) Pre-emphasis Circuit

At the transmitter, the modulating signal is passed through a simple network which amplifies the high frequency, components more than the low-frequency components. The simplest form of such a circuit is a simple high pass filter of the type shown in fig (a). Specification dictate a time constant of 75 microseconds (μs) where t = RC. Any combination of resistor and capacitor (or resistor and inductor) giving this time constant will be satisfactory. Such a circuit has a cutoff frequency fco of 2122 Hz. This means that frequencies higher than 2122 Hz will he linearly enhanced. The output amplitude increases with frequency at a rate of 6 dB per octave. The pre-emphasis curve is shown in Fig (b). This pre-emphasis circuit increases the energy content of the higher-frequency signals so that they will tend to become stronger than the high frequency noise components. This improves the signal to noise ratio and increases intelligibility and fidelity.

The pre-emphasis circuit also has an upper break frequency fu where the signal enhancement flattens out.

See Fig (b). This upper break frequency is computed with the expression.

$fu = R1 + (R2/2\pi R1R1C)$

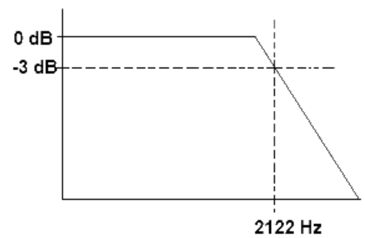
It is usually set at some very high value beyond the audio range. An fu of greater than 30KHz is typical.

De-emphasis:- **02 M**

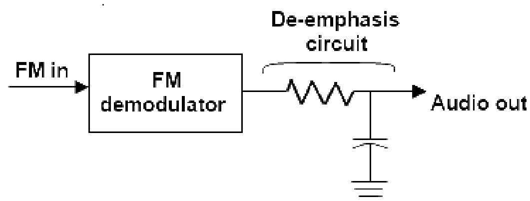
De-emphasis means attenuating those frequencies by the amount by which they are boosted.

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

De-emphasis Circuit



(d) De-emphasis Curve



(c) De-emphasis circuit

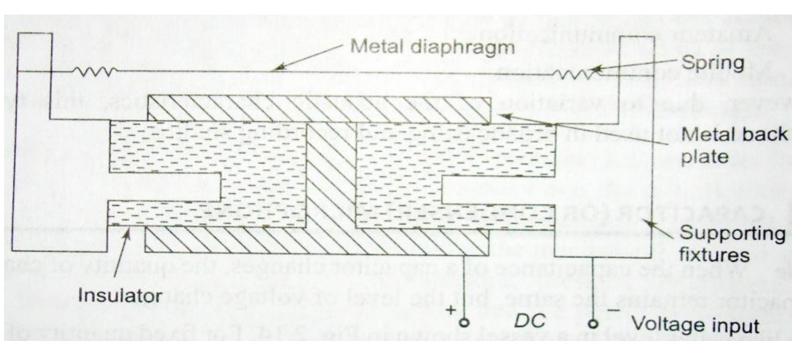
To return the frequency response to its normal level, a de-emphasis circuit is used at the receiver. This is a simple low-pass filter with a constant of 75 πs. See figure (c). It features a cutoff of 2122 Hz and causes signals above this frequency to be attenuated at the rate of 6bB per octave. The response curve is shown in Fig (d). As a result, the pre-emphasis at the transmitter is exactly offset by the de-emphasis circuit in the receiver, providing a normal frequency response. The combined effect of pre-emphasis and de-emphasis is to increase the high-frequency components during transmission so that they will be stronger and not masked by

Q.6 Answer any Four **16 M**

a) Draw the sketch of condenser microphone and explain its working?

Ans:-

Condenser microphone: **02M**



Working principle:- **02 M**

- When sound waves strikes the diaphragm, it moves. During compression, it moves towards the fixed back plate and increases capacitance.
- During rarefaction, it moves away from the back plate and therefore decreases the capacitance.
- The change in capacitance changes the DC voltage across the capacitor plates. as the distance between the plates changes, its capacitance changes as per the below equation

$$C = kA/d$$

Where,

“ k ”is a dielectric constant of the medium between the plates.

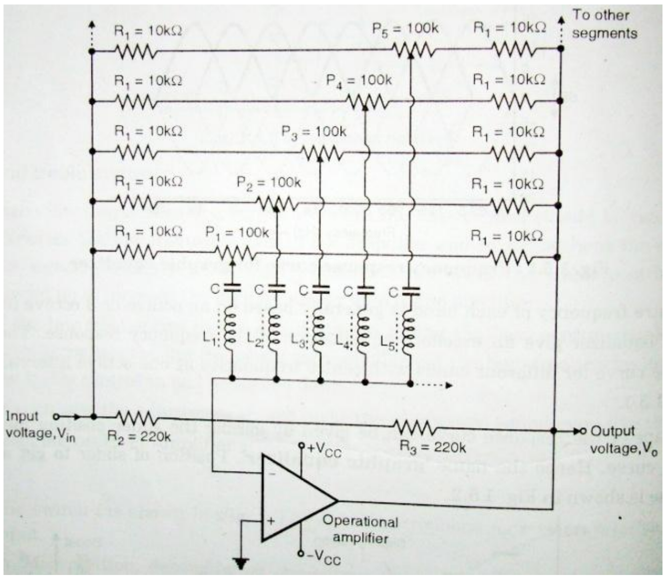
“A” is area of the cross section of the plates.

“d ”is the distance between the plates.

b) Elaborate the concept of graphic equalizer with neat circuit diagram?

Ans:-

Circuit diagram- **02M**



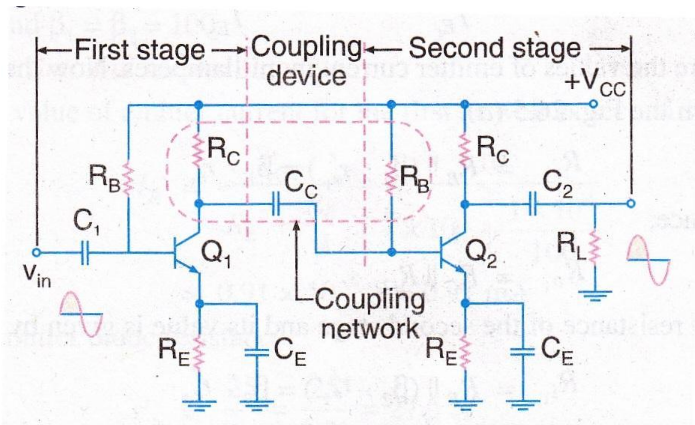
Concept of graphic Equalization:- **02M**

- Graphic equalizer is used to eliminate unwanted peaks in the frequency response of audio systems.
- Here, complete audio spectrum is segmented into narrow bands. Each band has an individual slider control which can boost or cut the signals from +15 dB to -15dB.
- Above circuit diagram of graphic amplifier consist of one amplifier with multiple feedback paths.
- Every feedback path consist of gain controls (P1,P2,P3,P4 & P5 here) and LC tuned circuit.
- Thus one can adjust gain of every octave band separately by adjusting corresponding feedback path gain.
- The centre frequency of the octave band is selected by inductors L1, L2, L3, L4, and L5.
- The combination of individual control setting will provide the required frequency response.
- The above circuit consist of only one amplifier for one octave band. Many such amplifiers are used for different octave bands. Such amplifiers are connected in parallel to cover complete frequency range.

c) Draw the circuit of RC coupled amplifier and explain its working with frequency response?

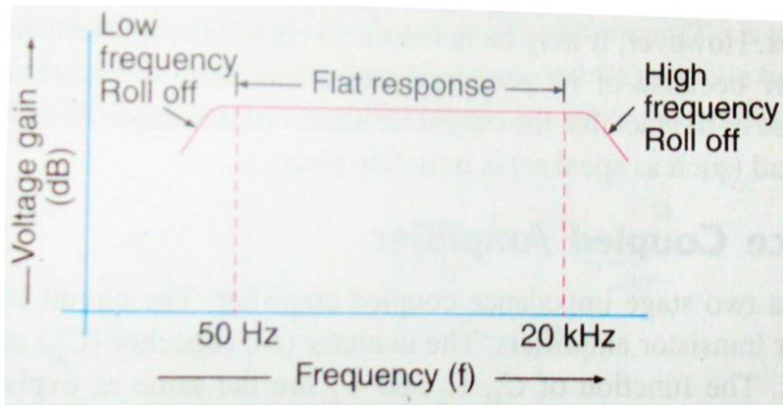
Ans:-

RC coupled Amplifier: **02 M**



RC coupled amplifier.

Frequency Response :- ½ M



Frequency response of RC coupled amplifier.

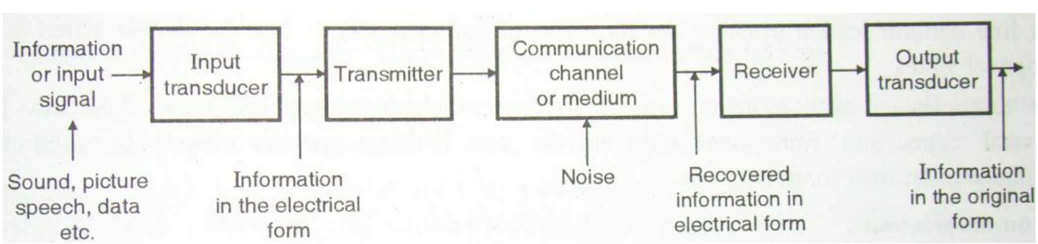
Working:- 1 ½ M

When the Ac signal is applied to the input of first stage , it is amplified by a transistor and appear across the collector resistor R_c. This signal is given to the input of second stage through a coupling capacitor c_c. The second stage further does the amplification of the signal. In this way, the cascaded stages amplify the signal and the overall gain is equal to the product of individual stage gains

d) Draw the block diagram of communication system and explain its working?

Ans:-

Communication system 02M



Explanation:- **02 M**

Information or input signal –

The communication system has been developed for communication useful information from one place to the other.

Input transducer-

The information in the form of sound, picture or data signals cannot be transmitted as it is. First it has to convert into a suitable electrical signal. E.g. Microphones, TV camera

Transmitter-

The function of the transmitter block is to convert the electrical equivalent of the information to a suitable form.

- In addition to that it increases the power level of the signal. The power level should be increased in order to corer a large range.
- The transmitter consists of the electronic circuits such as amplifier, mixer, oscillator and power amplifier.

Communication channel or medium

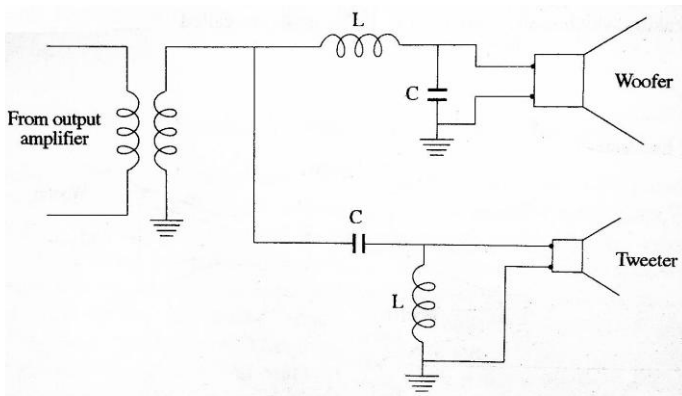
- The communication channel is the medium used for transmission of electronic signal from one place to the other.

Noise-

- Noise is an unwanted electrical signal which gets added to the transmitted signal when it is travelling towards the receiver. Due to noise, the quality of the transmitted information will degrade.
- Once added the noise cannot be separated out from the information. Hence noise is a big problem in the communication systems.

e) Draw the neat sketch of two way and three way cross over network?

Ans:- 2way cross over network: **02M**



3 way Cross Over Network: **02M**