इरिसेट IRISET

TC2

PUBLIC ADDRESS SYSTEM

Indian Railways Institute of
Signal Engineering and Telecommunications
SECUNDERABAD - 500 017

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INDIAN RAILWAYS INSTITUTE OF SIGNAL ENGINEERING & TELECOMMUNICATIONS, SECUNDERABAD - 500 017

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PUBLIC ADDRESS SYSTEMS

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CHAPTER-1

APPLICATION OF P.A. SYSTEM AND ACOUSTICS

1.0. INTRODUCTION

P.A System is a setup, which is used to disseminate information to a limited public over a limited area. The basic function of an audio system is to deliver audible and recognizable sounds to the listeners.

PA system comprises all the devices and networks that exist between a source of sound (or its electrical equivalent) and its point of final reproduction.

1.1. Application of P.A. system in Railways:

1. **Passenger amenity**: For giving the detailed information about the train arrival, departure, late running if any, and location of trains and any other important information related to Railway users.

2. **Marshalling Yards**: Communication is being established between Yard Master and shunting staff for the formation, reception and dispatch of trains, through paging and talk-back systems.

3. **Breakdown train Emergency Equipment**: The P.A.System provided in ARTs (Accident Relief Trains) like megaphone must be kept in working condition so that if any accident occurs; it is to be installed to guide the passengers and staff in rescue operations at the site of accident.

4. **Special functions**: A quality P.A. System needs to be installed in some important functions such as Railway Week, felicitations, Scouts and Guides rally, some social work meetings, cultural programmes, etc. where Ministers, G.M., officials, VIPs, etc., may address.

5. **Railway Workshops**: To give the announcements pertaining to staff in Workshops when required. And also for entertainment music during lunch hours.

6. **Conferences**: For conducting seminars, special lectures, administrative meetings for a limited group of officials in conference halls. For this suitable conference system is installed in GM conference hall in zonal HQ and DRM conference hall at Divisional HQ.

1.2. Acoustics:

Acoustics is defined as the "Scientific Study" of Sound, especially of its generation, propagation, perception and interaction with materials and is further described as the "total effect of sound" especially as produced in an enclosed space.

In order to improve the audibility of an average listener, the sound and the acoustic are to be distributed uniformly throughout the area to be covered. For this, the sound from the source is converted to electrical energy by a microphone and the level of this electrical signal is increased up to a certain level by an amplifier. This amplified signal is then transmitted through the cables to the loudspeaker which converts the electrical signal back into the acoustic signal.

1.3. Terms related to the study of Acoustics:

Intensity: The **intensity**, I, in Watts/m², which is the sound power passing normally through a unit area of space. This relates to amplitude of acoustic force. The acoustic power is inversely proportional to the reverberation period for a given intensity level.

Loudness: It is the intensity of the sound stimulus as perceived by the human ear and chiefly a function of sound pressure. It depends upon the frequency and the complexity of the waveform.

Frequency: Frequency is the number of "Cycles per Second" of a wave and is expressed in "Hz".

Pitch: The property of a musical tone is determined by its frequency and intensity. Higher the frequency, higher is the pitch. This relates to the subjective effect of the frequency. Pitch is affected by intensity also. At low frequencies, pitch varies directly with intensity and at high frequencies it varies inversely with intensity.

Timbre: This denotes the harmonic content. Even though the instruments may strike the same note, their sounds can be distinguished by their differences in respective timbres. In musical parlance, low timbre sound is referred to as "light" and high timbre as "dark".

Brightness: This refers to the upper harmonic content of a tone. It is not synonymous with timbre since it varies with both intensity and frequency. It is also referred to as "Density".

Volume Range: The difference, expressed in decibels, between the maximum and minimum volumes of a complex audio-frequency signal occurring over a specified period of time is defined as the volume range of an audio system. Signal to noise ratio of the system determines the maximum volume of dynamic range of that system. For a live pickup, the ambient noise level of the recording studio and the system noise together limits the volume range.

Doppler Effect: The Doppler Effect can be described as the effect produced by a moving source of wave in which there is an apparent upward shift in frequency for observers towards whom the source is approaching and an apparent downward shift in frequency for observers from whom the source is receding. An example of the Doppler Effect is when a train is approaching an observer, blowing its whistle; the sound appears to increase in loudness and pitch. After passing the observer, the pitch and intensity drop quite rapidly until the sound fades out completely

1.4. Audibility:

Hearing is the end result of differential pressure changes applied to a membrane within the ear. If very little or no pressure changes are present, the ear hears nothing. Conversely, if extreme changes appear, the forced vibrations of the membrane may damage the ear. The lowest acoustic pressure that gives rise to a sensation of hearing is known as the "threshold of audibility", while the highest pressure to which the ear can respond without experiencing pain is called "threshold of pain". The ratio of acoustic power at these two limits is the dynamic range of the ear and is approximately 10^{12} (the threshold of pain represents 10^{12} times the power of the threshold of audibility) near the middle of the audio spectrum

1.5. Sound pressure and its units:

Sound at a particular point in air is the rapid variation in the air pressure around a steady state value. This sound pressure is measured in the same units is atmospheric pressure, and since it is an alternating quantity, the term "sound pressure" usually refers to its rms value. One μ bar equals one dyne/square centimetre or 0.1 Newton/square meter. A human speaker at a distance of 1-meter generates a sound pressure of 1μ bar.

The ratio of the loudest sound to the softest sound produced by the same source is referred to as the dynamic range of that source. A remarkable property of the human ear is its large dynamic range. It can hear sounds as small as about 2×10^{-4} μ bar, and as high as 200 μbars without becoming overloaded.

RMS sound pressure is commonly expressed in dB above 2×10^{-4} μ bar and referred to as "sound pressure level". Mathematically, if 'P' is rms sound pressure and 'L' is Sound Pressure Level, then

$$L = 20 \log_{10} P/P_{ref.} \text{ dB}$$

$$\text{Where, } P_{ref.} = 2 \times 10^{-4} \ \mu \text{ bar.}$$

Sound pressure and sound pressure level are analogous to voltage and voltage level in the field of electricity.

1.6. Acoustic impedance:

Acoustic impedance of a sound medium is the complex quotient of the sound pressure and the particle velocity multiplied by the unit of area (square centimetre, square meter, etc.) Mathematically

$$Z = P / VS$$

Where P is sound pressure, V is particle velocity and S is the unit area. The unit of acoustic impedance is the acoustic ohm.

1.7. Loudness and its units:

To determine whether one sound is louder, equally loud, or less loud, than another, a statistically significant number of people compare the sound and then average their opinions. Similarly, to determine how loud a sound is, a standard sound is chosen and a significant number of people compare the unknown with the standard.

In acoustics the accepted standard is 1 KHz tone or narrow-band noise centred at 1 KHz. The loudness level of any sound is defined as the sound pressure level of a standard sound, which appears to a significant number of observers to be as loud as the unknown.

The loudness is measured in sones and loudness level in phons in earlier days. But now, the most widely used measure of loudness is decibel (dB). The dB is a logarithmic unit used to describe a ratio.

Table below compares the loudness in sones, loudness levels in Phons and in dB of several common sounds.

Source of sound	Loudness Level (Phons)	Loudness (Sones)	Decibels (dB)
Threshold of pain	140	1024	130
Jet Aircraft	120	256	150
Truck	100	64	90
Orator	80	16	--
Low conversation	60	4	60
Quiet Room	40	1	30
Rustling of leaves	20	--	10
Hearing threshold	4	--	0

TABLE- 1.1

1.8. Sound pressure level (SPL):

In acoustics the ratings most commonly encountered are changes in power levels. First of all there exists a reference. It is 0.0002 dynes/cm^2 or 0.00002 Newton/m^2. These are identical pressures with different labels. Sound pressure levels are identified as dB SPL.

1 atmosphere = 101,300 newtons/m^2.

Therefore, 20 log 101,300/0.00002 = 194 dB = SPL.

Note that the SPL is analogous to voltage. So 20 is the multiplier used for calculating dB SPL

One microbar is equal to 1 dyne/cm^2
One microbar is equal to 74 dB – SPL

Sound pressure is also measured in Pascal - Pa.

1 Pascal is equal to 10 μ bars.

Objective:

1. Sound intensity is expressed in watts/cm^2 (T/ F)
2. The lowest acoustic pressure that gives rise to a sensation of hearing is called threshold of audibility (T/ F)
3. The highest pressure to which the ear can respond without experiencing pain is called threshold of pain. (T/ F)
4. Sound pressure and sound pressure level are analogous to voltage and voltage level in the field of electricity. (T/ F)
5. Acoustic impedance of a sound medium is the complex quotient of the sound pressure and the particle velocity multiplied by the unit of the area. (T/ F)
6. Threshold of pain is 140 db (T/ F)
7. Threshold of hearing is 20 db. (T/ F)

Subjective:

1. What are the applications of PA system on Indian Railways?

CHAPTER-2

MICROPHONES

2.0. INTRODUCTION:

Microphone is a transducer, which converts acoustic energy into electrical energy.

2.1. Microphones may be classified:

According to Mode of operation:

i) Pressure operated
ii) Velocity operated

1. Pressure operated:

Pressure operated microphones employ a diaphragm with only one surface exposed to the sound source. The displacement of the diaphragm is proportional to the instantaneous pressure of the sound wave. At lower frequencies such microphones generally cause a resonant response, giving rise to a peak that may reach 6 to 8 dB with reference to 1,000 Hz. The pressure-operated microphones are carbon, crystal, dynamic and capacitor microphones.

2. Velocity operated:

A velocity microphone is one in which the electrical output substantially corresponds to the instantaneous particle velocity in the addressed sound wave. A velocity microphone is also referred to as a gradient microphone. A gradient microphone is a microphone in which the output corresponds to the gradient of the sound pressure.

The velocity-operated microphones are ribbon microphones.

2.2. Types of microphones:

2.2.1. CARBON MICROPHONE:

In a carbon microphone, small carbon granules are held in close contact in a brass cup called a "button" which is attached to the centre of a metallic diaphragm. Sound waves, striking the surface of the diaphragm, disturb the carbon granules changing the contact resistance between their surfaces. The change in contact resistance causes a current from a battery connected in series with the carbon button and the primary of a transformer to vary in amplitude, resulting in a current waveform similar to the acoustic waveform striking the diaphragm. After leaving the secondary of the transformer, the minute changes of current through the transformer primary are amplified and reproduced in the conventional manner. The circuit diagram as well as construction of a single button carbon microphone is shown in Fig.2.1.The output voltage from a carbon or pressure microphone is proportional to the displacement of the diaphragm. The field pattern is circular.

One of the principle disadvantages of the carbon microphone is that it has continuous high frequency hiss caused by the changing contact resistance between the carbon granules. In addition, the frequency response is limited and the distortion is rather high.

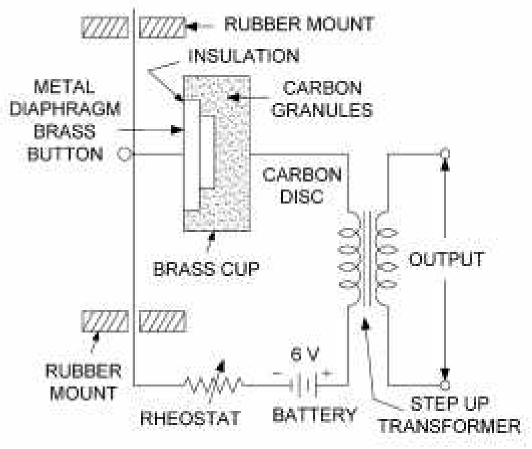


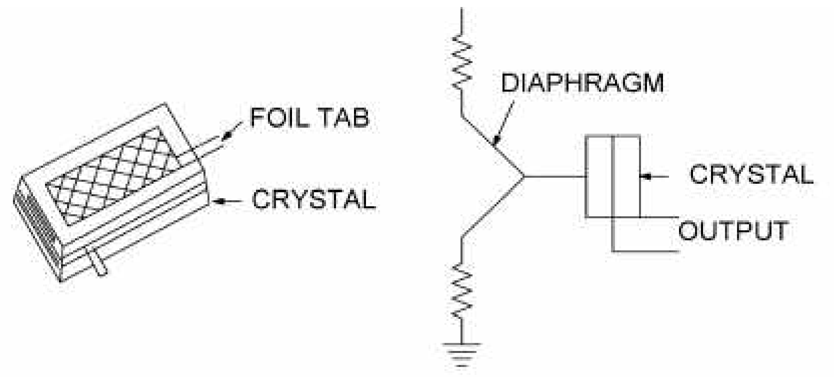
Fig. 2.1. Construction of Carbon Microphone

Carbon microphones used in communications i.e., telephone networks are designed to be moved, with the current flowing through them.

2.2.2. CRYSTAL MICROPHONE:

A crystal microphone employs one or more Rochelle salt crystals placed in such a manner that when their surfaces are struck by a pressure wave, they are bent or twisted out of shape. This action results in the generation of an electrical current because of the piezo-electric effect of such crystals.

When a crystal is subjected to strain, electrical polarization takes place. The polarization is proportional to the mechanical strain. The inverse effect is produced when electrical current is applied to the crystal. The mechanical movement in this case is proportional to the applied current.



A) Directly Actuated Type **B) Indirectly Actuated Type**

The crystal microphones are of two types, the direct actuated and the indirectly actuated.

In the former type, the sound waves strike the surfaces of the crystals creating mechanical strain (see Fig.2.2a). In the latter, the sound waves impinging on a diaphragm attached mechanically to the crystal elements (See Fig.2.2b).

Another type is the sound-cell crystal, which is shown in the Fig. 2.3. Numbers of crystal elements are stacked in a pile.

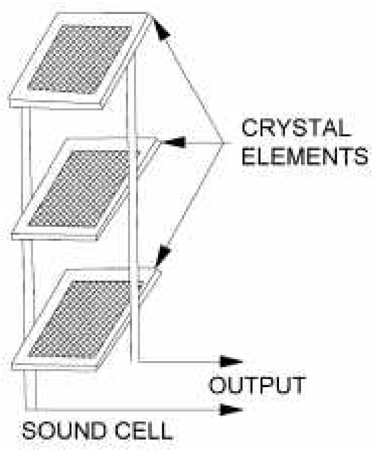


Fig. 2.3. Crystal Elements used in Microphone Construction.

2.2.3. DYNAMIC MICROPHONE:

The dynamic or moving coil microphone employs a voice coil attached to a diaphragm. The sound pressure on the diaphragm makes to move the coil in a strong magnetic field, which generates a voltage proportional to the sound pressure at the diaphragm. This microphone is also referred to as a pressure microphone.

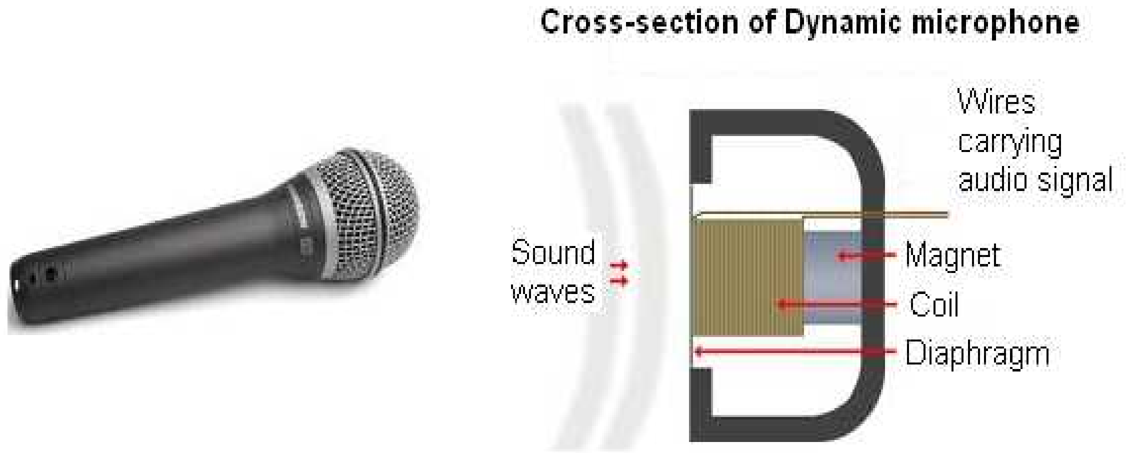


Fig. 2.4. Dynamic microphone

Microphones of this type do not require output transformers. The output voltage is taken directly from the voice coil. The output impedance is approximately 20 Ohms. A low impedance of this nature permits the microphone to be placed at a considerable distance from the preamplifier without affecting the microphone characteristics. This microphone, like other pressure-operated microphones has a circular field pattern. This is the most widely used microphone in these days.

2.2.4. CAPACITOR MICROPHONE:

This is based on the principle of variable capacitance. It is polarised by a battery. The appearance and basic functioning is shown in fig.2.5. The diaphragm of this microphone is a thin membrane of nickel, which is spaced about 0.001 inches (25 μmeters) from the fixed back plate keeping air as the insulating medium. As long as the diaphragm is not exposed to the sound pressure, the capacitance remains constant and the ac output voltage will be zero.

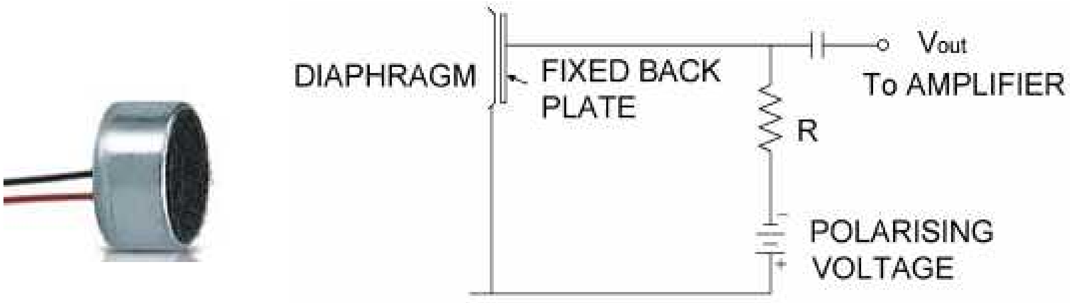


Fig. 2.5. Capacitor Microphone

Whenever the sound waves strike the diaphragm it undergoes compressions and ratification and the capacitance across the microphone varies. The capacitance is inversely proportional to the distance or space between the two plates. Q = CV. Since the voltage remains constant the variations in capacitance varies the charge. The current in the circuit varies making a variable voltage drop across ’R’, ac output voltage is taken through a capacitor and immediately fed to a pre-amplifier.

The function of a polarising voltage or its equivalent is to translate the diaphragm’s motion into a linearly related audio output voltage, which is amplified by a very high impedance FET which must be located close to the capacitor. Special measures must be taken to prevent the space or distance between the capacitor plates from changing because of temperature and humidity.

Use of foam windscreens for protection in damp or corrosive environments is recommended.

Frequency response is fair over the entire audio spectrum. It requires a polarising battery and a preamplifier. It is basically an omni directional microphone; it is best for indoor and outdoor use.

2.2.5. RIBBON MICROPHONE:

The ribbon (velocity) microphone is a microphone in which a very light metallic ribbon is suspended in a strong magnetic field. Pressure waves cause the ribbon to vibrate in the magnetic field generating voltage corresponding to the particle velocity of the pressure wave. Velocity microphones may be designed to have a wide frequency range, good sensitivity, low distortion, and low internal noise.

2.2.6. Wireless or cordless microphone:

Wireless microphones are classified as of two types. These are Hand held and collar type.

The specific application of handheld mic is that, a person giving a performance in between a programme can hold this mic in his hand and gives the announcements as and when desired.

Whereas the collar type mic can be used by a person gives any demonstration or lecture by permanently fixing the mic to his collar of his shirt, so that he can freely move. A typical example of a wireless microphone system consists of a lapel microphone concealed in the clothing of the user.

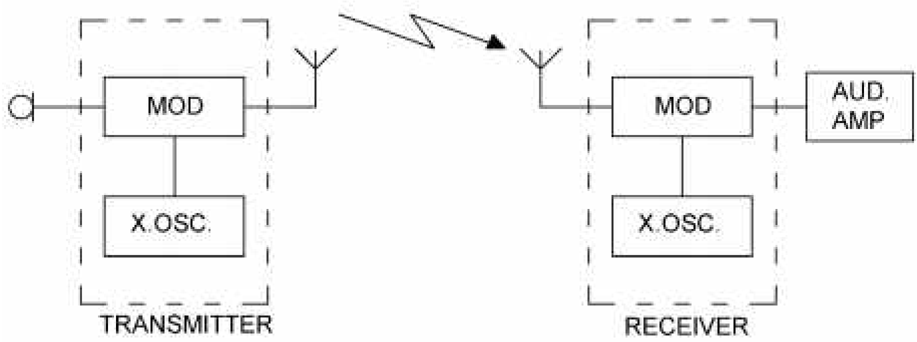


Fig. 2.6. Typical Block diagram of wireless microphone

Cordless microphone

Lapel microphone

Wireless mic works on the principle of VHF radio trans-receiver system. It consists of two units. The microphone along with transmitter is one unit and receiver the other unit. The microphone output will fed to input of the transmitter’s modulator circuit as an AF signal. A crystal oscillator will generate the necessary carrier frequency and fed to modulator. The AF generated by the mic is modulated with the carrier signal and transmitted as RF signal. The transmitter circuit requires an operating dc voltage of 9V, which will be provided by dry cell inside the casing of the microphone unit.

The receiver consists of a collapsible antenna and a demodulator circuit with a crystal oscillator. The transmitted signal is received by a receiver located at a remote point (near the audio amplifier) and converts back to audio signal. This AF signal is connected to the input of an audio amplifier through jacks. The receiver circuit requires operating voltage of 12V dc.

The maximum transmitting distance under normal conditions is about 200 feet. When several wireless microphones are used on the same set, each microphone transmitter must be operated on a different frequency to avoid interference.

2.3. Types of microphone stands:

The microphones are generally mounted on stands, and the common types are

i) Table Type
ii) Floor Type
iii) Dual head mic stand

Besides in studios, microphone booms are used. The details of the various types are shown in Fig.2.7. Generally the floor types are telescopic so that the heights are adjustable.

Table mic stand | Floor mic stand | Dual head mic stand

Fig. 2.7. Microphone Stands

2.4. A general comparison on performance of Microphones:

Microphone performance basically depends on the type of Microphone. It is not to say that which one type of microphone is better than the other, but which one is better suited for a particular application.

2.4.1. Moving Coil Microphone: These are well suited for high sound pressure level applications. They are rugged, robust and generate low self-noise. There will not be any problem with humidity and temperature variations. Comparatively less expensive and wide variety of models is available. It has got a good frequency response. Only disadvantage is that they are slower to respond to transients (i.e., Sounds that begin with quick attack and then quickly decay. (e.g., Drum hit).

In Railways, PA.Systems are installed mostly for the purpose of speeches and do not require considerable details of the entire sound spectrum. So, moving coil microphones are ideal in Railway applications. There are various types of dynamic mics available in the market, as an example AHUJA make SHM-1000 XLR model mic will be used as stage mic for large pickup area.

2.4.2. Ribbon Microphones: These are not used in Railways because they are very costly and careful handling is required. It is best suited for recording music and broadcast applications.

2.4.3. Capacitor Microphones: These are considered to be high performance instruments. They produce clear detailed sound. It has got excellent transient response and very sensitive making suitable for use at distance. It has got a wide signal to noise ratio. However, these microphones are expensive when compared to dynamic microphones.

In Railways these are used as lapel mic especially for the classrooms and lectures in Conference halls where a person has to make any explanation with demonstration (hands free).

2.5. Specifications of microphones:

1. Type
2. Sensitivity
3. Frequency Response
4. Max. Sound Pressure level.
5. Impedance
6. Minimum load impedance
7. Cable and Connectors
8. Front to Back Ratio
9. Polar Response.

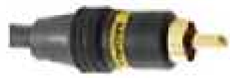
1. **Type**: This specifies the microphone whether it is a Dynamic, Ribbon, Capacitor or Crystal and also specifies whether it is a Pressure gradient or Pressure Operated.
2. **Sensitivity**: It is the amount of voltage developed or generated by the microphone for an applied sound pressure at a test frequency of 1 KHz. It is generally specified as mV/Microbar. One-microbar sound pressure is equal to 1 dyne/cm^2. It is also specified as mV/Pa where Pa is Pascal, which is equal to 10 microbars.
3. **Frequency Response**: It is the ability of a microphone to produce a proportionate output to the sound pressure applied for the specified range of frequencies.
   The frequency response is distorted when the microphone is kept too close to the mouth. It generates spherical sound waves with very high impact pressure when the distance from the mouth increases the spherical sound waves flatten and become plane waves. So the distortion diminishes with distance.
4. **Maximum Sound Pressure Level**: It is the maximum Sound Pressure level that can produce a proportional output with a total harmonic distortion limited to 1%.
5. **Impedance**: It is the impedance offered by the microphone at 1 KHz. There are low impedance and high impedance microphones.
   Low Impedance means less than 600 ohms
   High Impedance means more than 10K ohms.
6. **Minimum Load Impedance**: It is the minimum input impedance of the amplifier, which is used to utilize the microphone. The amplifier input impedance should not be less than the minimum load impedance of the microphone specified.
7. **Cables and Connectors**: It specifies the type and length of the cable with a particular connector.
8. **Front to Back Ratio**: It is specified in the case of unidirectional microphone, which gives the response of front sound to back sound. Generally it is 20 db.
9. **Polar response**: It specifies the type of directivity pattern that microphone responds. It is a graph of the microphone's directional sensitivity. It specifies whether it is an omni directional, Bi-directional or Uni-directional.

2.6. Microphone connectors:

Connectors are used to connect the microphones to the amplifiers. Widely used microphone connectors are 3 pin XLR type and jack type connectors. These connectors may be of mono or stereo type. The three pins connected are, ground (1), positive (2) and negative (3) respectively. The contacts are made of self cleaning type and non-reversible to avoid noise and interchanging of connections. RCA type connectors are widely used with music systems and they are for stereo type only.

1/4" Jack type connector

3 pin XLR male and female connector



RCA type connector

2.7. INSTALLATION PRACTICE:

Some of the important precautions to be observed in the operation of microphones are given below:

I. All microphones are delicate instruments; they must be handled carefully and never dropped, nor placed where there may be metal dust.

II. To avoid hum pick-up and especially in case of high impedance microphones, locate as far as possible away from electrical apparatus. Do not run microphone leads together with mains cable.

III. For public address, locate away from preferably to rear of the loudspeakers to prevent acoustic feedback 'howl'.

IV. Ribbon microphones should be placed at least 10 feet away from the speaker. One should speak into the microphone and on no account it should be tested by blowing into the microphone.

V. When using microphone with long twin core lead (i.e., in low impedance condition) in association with equipment having high impedance input, a step up transformer/matching transformer is required to be interposed in the lead near the equipment input socket. Microphones should not be mounted on vibrating surface, e.g., piano, amplifiers, radio, recorders, etc.

VI. Microphones must be protected from strong winds; otherwise 'roaring' noise will result. It is common practice to provide windscreens in such cases. A typical windscreen is shown in Fig.2.8. It consists of a wire framework covered with silk and designed to fit over the outside of a microphone to reduce the effects of wind noise. A wire frame is clamped over the end of the microphone housing. The microphone projects up to the centre of the screen, which contains a smaller screen shown by the dotted lines at the centre of the drawing. The larger frame is covered with a single layer of black silk. A windscreen is also called a "Wind Gag".

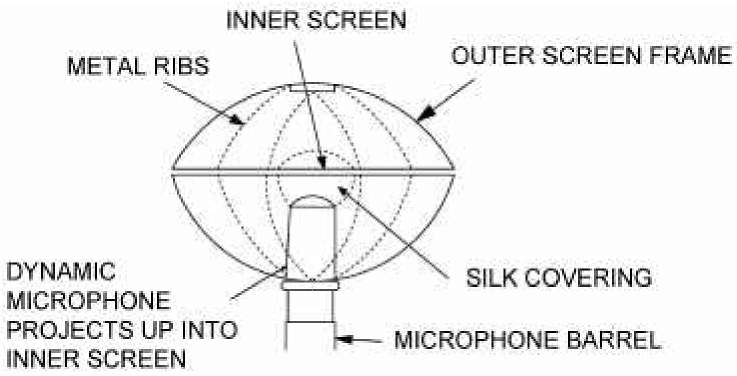


Fig. 2.8. Wind screen for Microphone

SUMMARY:

1. Microphones are transducers that convert sound energy into electric energy.
2. The three types of professionally used microphones are moving coil, ribbon and capacitor.
3. Microphones pick up sound from essentially three directions: all around Omni- directional, front and rear - bi-directional, and front - unidirectional.
4. Standard accessories used for professional microphones include the following: twin conductor cables called balanced lines, connectors, and various types of stands and clips for microphone mounting on a desk, floor, person, or musical instrument.
5. There are also multidirectional microphones - mics with more than one pickup pattern. Eg. Column microphones.
6. To help protect against loudness distortion, many capacitor microphones are equipped with a pad to reduce overloading the microphone's electronics.
7. Microphones have been developed for special purposes, the lavalier to be un-obstructive; the shotgun and parabolic mics for long distance pickup; the wireless mic for greater mobility and flexibility in plotting sound pickup.

Objective:

1. Pressure operated microphones employ a diaphragm with only one surface exposed to the sound source. (T/F)
2. A velocity microphone is one in which the electrical output substantially corresponds to the instantaneous particle velocity in the addressed sound wave. (T/F)
3. The velocity-operated microphones are ribbon microphones (T/F)
4. The pressure-operated microphones are carbon, crystal, dynamic and capacitor. (T/F)
5. Dynamic microphone do not employ output transformers (T/F)
6. The output impedance of a dynamic microphone is approximately 20 Ohms (T/F)
7. Capacitive microphones are high impedance microphones (T/F)
8. Capacitive microphones requires polarizing voltage (T/F)
9. Sensitivity is the amount of voltage developed or generated by the microphone for an applied sound pressure at a test frequency of 1000 Hz (T/F)
10. Frequency Response is the ability of a microphone to produce a proportionate output to the sound pressure applied for the specified range of frequencies. (T/ F)

Subjective:

1. Define pressure operated and velocity operated microphones?
2. What are the specifications to be followed while selecting a microphone?
3. Explain briefly the specifications used for microphone?
4. Explain the working principle of dynamic microphone?

CHAPTER-3

LOUDSPEAKERS

3.0. Introduction:

The function of the loudspeaker is to convert electrical energy into acoustic energy.

The loudspeaker should have the following qualities:-

i) Satisfactory sensitivity.
ii) Broad directivity.
iii) Low distortion over the audio spectrum.
iv) Smooth frequency response.
v) Balanced response.
vi) Good transient response
vii) Sufficient damping at base resonant frequency
viii) Adequate power handling capacity.

Loudspeakers may be divided into two main groups:-

i) **Cone type** - i.e., direct radiator, where cone or diaphragm is directly coupled to air.

ii) **Horn-type** – i.e., indirect radiator, where the diaphragm is coupled to the air by means of horn.

The horn increases the acoustical loading on the diaphragm and thereby increases the efficiency. It may be described as a device, which transforms acoustical energy at high pressure and low velocity to acoustical energy at low pressure and high velocity.

3.1. Dynamic Loudspeaker:

The most common type is the moving coil loudspeaker also known as dynamic loudspeaker. It consist a permanent magnet with strong magnetic field and a diaphragm, which acts like a piston. A voice coil is attached at the apex of the diaphragm. Audio frequencies are applied to the voice coil and cause it to react in the permanent field, resulting in motion of the cone. The motion of the cone sets up a varying air pressure, which is carried to the listener in the form of sound waves.

A cross-sectional view of a dynamic loudspeaker, with its essential parts indicated, is shown in Fig.3.1.Chasis is the support on which the cone diaphragm is attached. The voice coil is kept over a permanent magnet as shown and the limb on which coil is placed becomes North Pole and the other becomes South Pole. When the input signal is applied to the coil the coil moves back and forth produces vibration in the diaphragm and sound is produced.

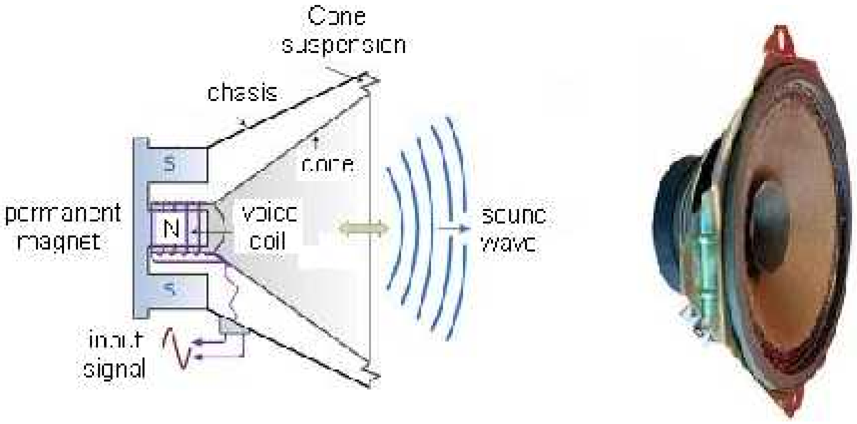


Fig.3.1. Cross Sectional construction of a Dynamic Loud speaker

The voice coil is connected to the output of an audio amplifier when audio frequency currents are applied to the winding of the voice coil, the voice coil will move either inward or outward, depending upon the instantaneous polarity of the signal. In other words, the voice coil is attracted or repelled by the magnetic field.

3.2. Shapes of cones:

Circular cones are generally used because the tools are easily and cheapest to make than elliptical types, which offer no rear advantage. The use of elliptical speakers is of special importance in television sets, where they can be fitted under the cathode ray tube to secure the largest possible cone area in the minimum available space.

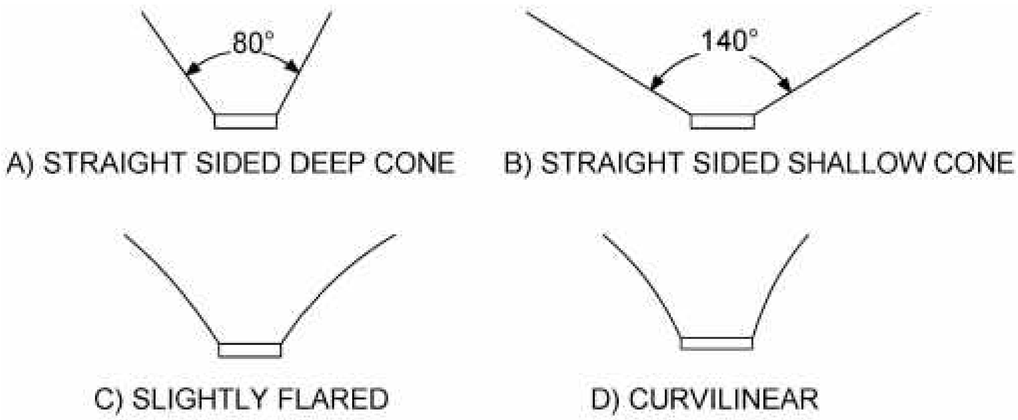


Fig.3.2. Cross Sections of Typical Cones

3.3. CABINET LOUD SPEAKERS:

The cabinet improves the acoustic response of the cone type speakers. The basic design consists of an enclosure with the loudspeaker unit set in the centre of a large box, which is completely air tight except for a port and the loudspeaker hole in the front panel. The port is so proportioned to the interior volume of the enclosure and to the loudspeaker characteristics that it functions acoustically as a low frequency loudspeaker. Thus, the low frequency response is increased, and distortion generally experienced with a non ported enclosure, is reduced.

The resonant frequency of a loudspeaker enclosure is damped by completely lining the interior surfaces of the enclosure with a highly absorbent material such as, rock wool. The resonant frequency of the panels may be damped to the use of diagonal braces and by filling unused spaces with sand.

3.4. Line source or column speakers: A system of several separate drive units mounted one above the other in a suitable enclosure.

Fig.3.3.Coulmn speaker

They form a line of sound source, or they could be a thought of as being in a column, hence they are called as column speakers.

It should be fairly obvious that on the axis of the system the sound waves from all the units are in phase and will therefore reinforce each other. Off this axis the different path lengths from the units will tends to cause cancellation. However it will show that phase cancellation can only occur if the wavelengths are comparable with or less than the length of loudspeaker column.

3.5. High fidelity (hi-fi) speakers: These are used to reproduce the generally audible frequency range of 50 Hz to 12 KHz (out of the entire audio range of 20 Hz to 20 KHz). The frequency response of ordinary speakers is irregular, with a number of resonant peaks and valleys, and has a range of about 60 Hz to 8 KHz only. By using a fairly large (30cm to 38 cm diameter) and heavy cone, the low frequency response of speakers can be extended downward to 45 or even 30 Hz but at the cost of high frequency response.

It is difficult to design a single speaker to cover the entire audio range. One can use separate speakers for different audio ranges or combine large and small speakers into a single unit, mounted in line or coaxially.

A coaxial speaker consists of two separate speakers mounted on the same axis. Alternatively, a single electromechanical driver may operate two different cones or diaphragms. A mechanical or electrical crossover network divides the low and high frequency bands, which are fed partly to each speaker unit.

Single speaker response of limited frequency can be prevented through a multiple speaker system comprising separate speakers with maximum efficiency. An example of this is a two-way system comprising a ’woofer’, a ’tweeter’ and a ’crossover network’.

3.6. Woofer: A woofer mainly reproduces lower notes in a musical programme. In some cases it handles all notes below 2 KHz. For a three-speaker system, it is designed to work up to 500 Hz. Woofers are cone speakers and the best of them are in a loose suspension with very high compliance. A woofer is operated in a closed box (baffle), so that air resistance limits the cone’s movement and avoids damage.

3.7. Tweeter: It is designed to reproduce higher notes in a musical programme and can be used with a woofer in a two-speaker system or with woofer and midrange speakers in a three-speaker system.

Tweeter in a two speaker system re-produces frequencies from 1 KHz onwards and in a three speaker system from 5 KHz onwards. Also, there is a super tweeter, which covers the range from 8 KHz onwards. A tweeter may be a small cone permanent magnet speaker or an electrostatic type.

3.8. Crossover network: The crossover networks are a frequency-dividing circuit that ensures that each drive unit is fed only with its correct frequency band. This circuit in a two-way system directs the high frequencies to a tweeter and low frequencies to a woofer.

The specific purpose of crossover network is:

1. To extend the frequency range by the use of two or more speakers of different size.
2. To avoid inter modulation distortion which may occur in a single unit.
3. To limit the input to the most useful frequency range in a given speaker.
4. To protect a delicate HF unit from LF input.
5. To facilitate suitable placing of bass and treble speakers for natural results.

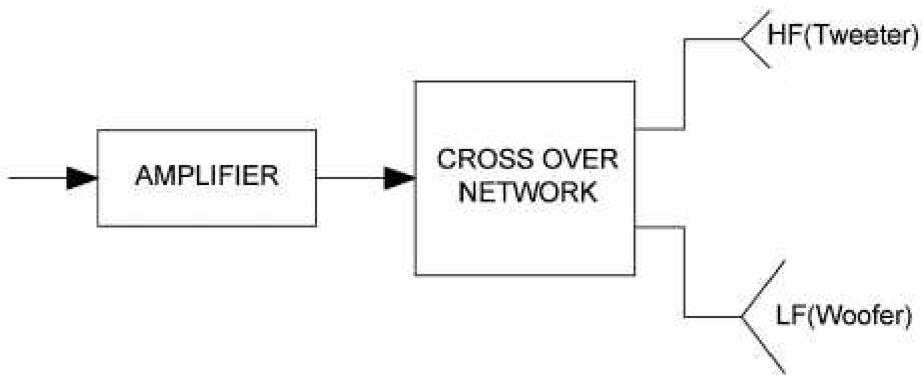


Fig.3.4. Multiple unit loudspeakers

The simplest form of crossover network is shown in Fig.3.5. It has a capacitor in series with the speaker. The capacitor offers very little impedance to high frequencies, which pass through the tweeter unimpeded. But the impedance offered by capacitor increases as the frequency decreases, and so very low power reaches the tweeter at lower frequencies.

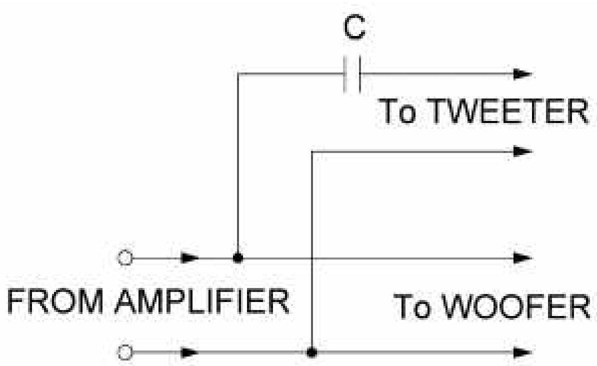


Fig.3.5. High pass crossover network.

Inductance of the woofer offers some impedance to high frequencies attempting to pass through it. The system is unsatisfactory for high fidelity. Impedances are not matched and there is no definite crossover point.

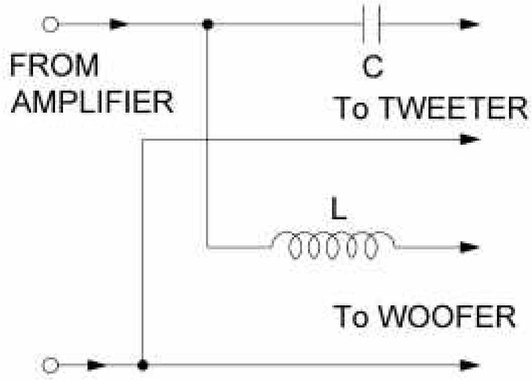


Fig.3.6. Constant impendence crossover network

A better type is the constant impedance network. The impedance offered to both amplifier and speaker is constant.

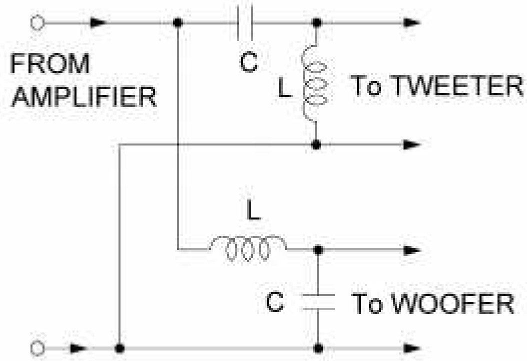


Fig.3.7. Filter type crossover network

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Ambitious approach to hi-fi reproduction is the three-way system. In addition to the former two-way system it contains a dual horn high frequency tweeter and an LC crossover network (filter type), which limits the woofer to audio frequencies below 600 Hz or less. The previous tweeter can be used here as a midrange speaker for frequencies of 600 Hz to approx. 4 KHz. The new tweeter here reproduces the range from 4 KHz to 20 KHz. More elaborate and expensive multiple speaker setups are used sometimes to achieve uniform response.

3.9. HORN TYPE LOUD SPEAKERS:

A typical example of the indirect radiator is the driver unit. It is a loudspeaker unit, which does not radiate sound waves directly from a vibrating surface but requires acoustic loading from a 'horn'. Horn loading is coupling a loudspeaker diaphragm to the air by means of a horn. Generally, the horn uses an exponential flare, starting with a small throat and expanding rapidly to a large bell.

A common type is the exponential horn. It is a horn with a constant rate of expansion or flare at an exponential rate (See Fig 3.8.A). The purpose of the horn is to provide an acoustical match between the diaphragm of the loudspeaker unit and the air in the throat of the horn.

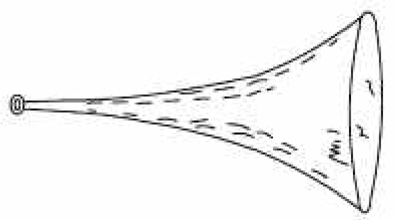


Fig. 3.8. A: Exponential Horn

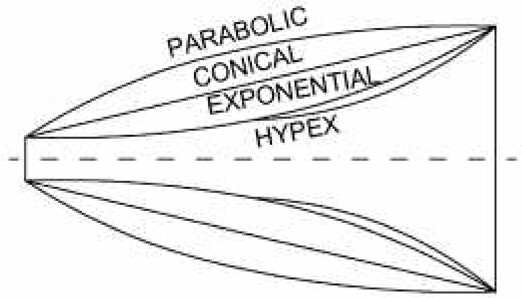


Fig. 3.8.B: Comparison of the different rates of flare used in the design of loudspeaker horns

A horn facilitates the transfer of electrical energy into acoustical energy and, if properly designed will do so with minimum distortion. The design of loudspeaker horn is complex and requires careful consideration to prevent reflection of the acoustical energy back into the horn bell.

The area of the throat determines the loading on the diaphragm. If the area of the throat is small compared to the area of the diaphragm, the efficiency is increased because of the heavier loading effect. However, small throats require a longer horn, which increases the frictional losses. A horn designed to use the Jonson Hypex flare has a considerably higher throat resistance as may be seen in figure 3.9 which indicates the comparative efficiencies of various types of flares.

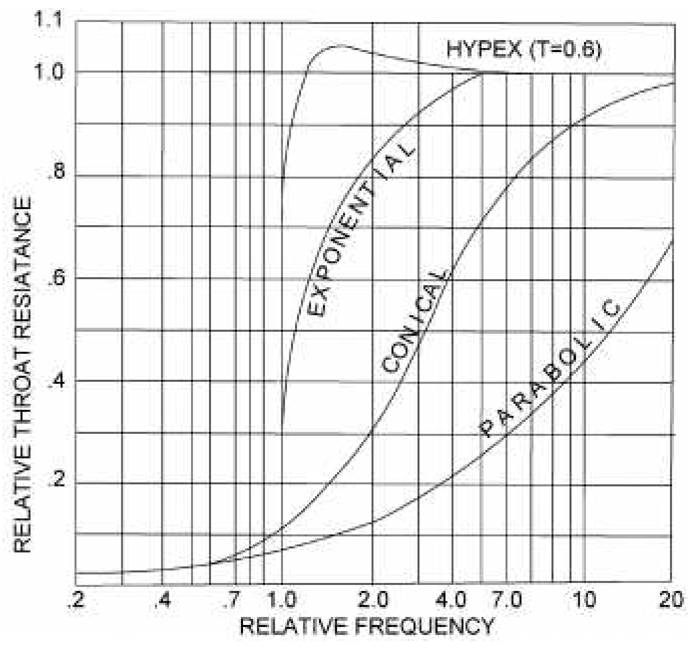


Fig.3.9. Comparative performance of the horn flare rates illustrated

The resonance is held to a minimum in a horn type loudspeaker by making the bell or mouth of the horn, a dimension that is two third or more of the lowest frequency to be reproduced. Horn resonance causes cancellation of certain frequencies and introduces distortion.

A variation of exponential horn is the re-entrant type. The folding of the horn permits it to be designed for a lower cut off frequency with a shorter physical length as shown in Fig. 3.10. It is an exponential horn folded within it to reduce its physical length as shown.

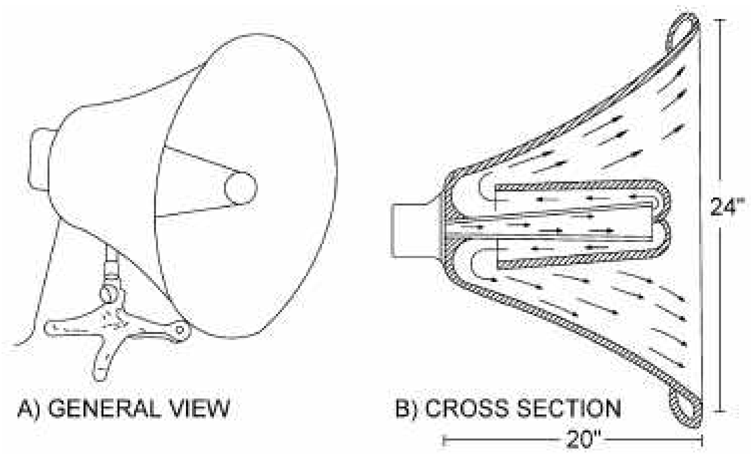


Fig. 3.10. Re-Entrant type Horn

3.10. Specifications of Loud Speakers:

Impedance: It is the impedance offered by a loud speaker at 400 Hz. The impedance changes with the frequency.

Power handling capacity (PHC): Loud Speaker manufacturers use this term to indicate the maximum volume of sound that the loud speaker will produce before it runs into distortion. Generally maximum 5% tolerance is allowed. It is also said that the voice coil of the loud speaker can handle the maximum radial power safely.

Frequency Response: It indicates the uniform sound pressure throw for the given band of frequencies and it is related with the enclosures that are used.

Sound pressure level: Loud speaker manufacturers indicate the accurate of sound pressure in DB Spl at 1meter distance when 1 watt of 1 KHz signal fed to the loud speaker. It is also related with the enclosures. Apart from the above specification some manufactures indicate the dimensions, weight and the size of the magnet used in the loud speaker.

Objective:

1. The function of the loudspeaker is to convert electrical energy into acoustic energy. (T/F)
2. Cone type of loud speaker is a direct radiator (T/F)
3. Horn-type loud speaker is an indirect radiator (T/F)
4. High fidelity (hi-fi) speakers **a**re used to reproduce the frequency range of 50 Hz to 12 KHz. (T/F)
5. limited frequency use can be prevented through a multiple speaker system comprising separate speakers (T/F)

Subjective:

1. Explain direct radiator and indirect radiator in terms of loud speaker?
2. Write short notes on Woofer and Tweeter?
3. What are the advantages of crossover networks?
4. Explain the working principle of a re-entrant horn?
5. What are the specifications to be followed while selecting a loud speaker?

CHAPTER-4

PA AMPLIFIERS AND MIXERS

4.0. Introduction:

An amplifier in PA equipment is a device, which takes low level input signal from microphones and amplifies it to a high level output signal at the desired output power, which will be delivered to the loud speakers at the output stage by suitable connection.

4.1. Features and facilities of an amplifier:

- Accommodates number of input devices
- Various controls of input devices
- Rated power output
- Output connectivity for loud speakers
- Main and Standby power supply connection

4.1.1. Front panel parts and their functions:

Fig. 4.1. Front view of a typical Amplifier

Microphone inputs: Amplifiers are having a minimum of 3 inputs and maximum up to 6 no. of input devices which includes auxiliary inputs (like tape recorder, mixer unit) can be connected. A typical amplifier’s front view has shown in fig.4.1. The jacks 1 to 5 are microphone input jacks for connecting low impedance microphones, with suitable male jacks.

Mic/Aux. Volume controls: The individual volume/gain control of each input is provided separately to adjust (increase/decrease) by the knobs.

Bass control:

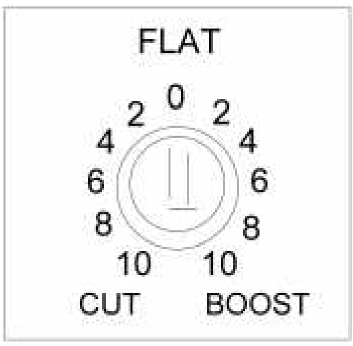


Fig. 4.2 Bass control

This is a low frequency control, which boosts and cuts the low frequencies. When moved from flat or 0 position to cut position the low frequencies are reduced which helps to eliminate feedback and howling on mic operation allowing opening of higher volume.

Treble control:

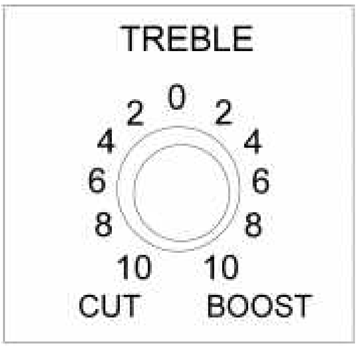


Fig. 4.3.Treble control

It is a high frequency control, which boosts and cuts the high frequencies. When moved from flat or 0 position to cut position the high frequencies are reduced. Too much boost of this control can cause oscillation and distortion of sound.

Master control: All mic/Aux. input signals are mixed and are finally controlled by the master control, which acts as a gate for overall opening of the volume level of amplifier.

Power ON/OFF switch: It is mains power ON/OFF switch for amplifier. The amplifier will have an inbuilt AC-DC converter, which converts 230V AC to 9/12/24V dc.according to the operating voltage requirement.

4.1.2. Rear panel parts and functions:

Fig.4.4. Rear view of Amplifier

Stand by battery connection: For uninterrupted operation of amplifier, connect 12V/24V battery in correct polarity. Changeover to battery operation is automatic when AC main supply fails. No battery current is consumed when the amplifier is working on AC mains.

High impedance speaker connection terminal strips: For the connection of loudspeakers in voltage matching method, three terminal strips are provided viz, com., 100V and 70V. The connection must be made only one at a time i.e., either com and 100V or com and 70V.

Low impedance speaker connection terminal strips: For the connection of loudspeakers in impedance matching method, four terminal strips are provided viz., com., 4Ω, 8 Ω and 16Ω. The connection must be made only one at a time i.e., either com and 4Ω or com and 8Ω or com and 16Ω according to the effective impedance of the load.

Pre-Amp out jacks: These jacks offer output for connecting tape recorder for recording the overall program or for feeding to Aux. Input of any amplifier for obtaining combined high-powered output.

Line output jacks: These jacks offer output for connecting to line input of a booster amplifier or any other amplifier.

4.2. Performance of amplifiers:

The following particulars are important to assess the performance of an Amplifier:

- Input Impedance.
- Input voltage to produce full rated output (volume control at maximum position)
- Output impedance.
- Rated output in watts.
- Percentage distortion at full output power expressed in relation to harmonics.
- Percentage of harmonics present at a quarter, half and three quarters of full rated output.
- Frequency response.
- Noise level at full output.
- Power Consumption.

The gain may be expressed in dB, but the input impedance should be specified to obtain the actual output in watts. The harmonic distortion is due to the transistor characteristics. Since harmonic content varies with load impedance, an increased load impedance results in decreased harmonic distortion, but this would result in decreased output wattage. Conversely, decreased load impedance will cause more harmonic distortion. Transformers caused harmonic distortion due to non-linearity of the magnetization curve of the iron. The output rating in watts is not a complete indication of the capabilities of that amplifier under operating conditions. Amplifiers are rated at some specified output in watts with a declared harmonic content, of say, 5%.

The power consumption of an amplifier is an important factor in determining the rating of power supplies, especially if they are to be operated from rotary converters or diesel generators. The efficiency is 20%, and in the event of class AB amplifier, the current consumption decreases slightly with decrease in signal output.

4.3. Measurements on amplifier:

The common measurements made to judge the performance of amplifier is: -

- Frequency Response
- Amplification
- Internal Noise Level
- Output Power
- Harmonic Content
- Output Regulation
- Power Consumption

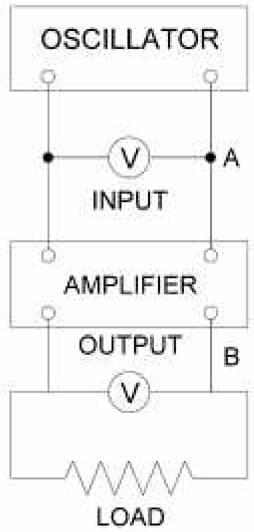


Fig.4.5. Circuit for frequency response measurement

4.3.1. Frequency response measurement:

The set up of equipment for determining the frequency response is shown in the Fig.4.5. Meters A and B measure the input and output voltages. Suppose the amplifier has a rated output of 15 watts for an input of 75 volts. If the input impedance is 500,000 Ohms and output impedance is 10 Ohms, voltmeters B should indicate 12.3 volts for .75 volts reading on meter A. The volume control should be at maximum. It is now turned to half the volume, so that meter B reads 8.7 volts corresponding to 7.5 watts output. Let these readings be obtained, at a frequency, say 1 KHz. Keeping this input voltage constant the output level is measured for frequencies ranging from 40 Hz to 10000 Hz. The results are tabulated as under:-

Frequency in Hz	Output in Volts	Voltage ratio relative to 1000 Hz.	DB level relative to 1000 Hz.

4.3.2. Output power:

For output power measurements, the arrangement of the equipments is similar to that given under Para above. The output volts is measured for three spot frequencies say 50, 1000 and 8000 Hz.

4.3.3. Harmonic content:

Harmonic distortion may be measured by means of an analyzer incorporating filters such that each harmonic can be isolated and measured.

4.3.4. Output regulation:

The output regulation is a very important criterion in the performance of an amplifier, since the loudspeakers are switched in and out frequently in actual practice. The method employed consists in driving the amplifier from an oscillator at 1 KHz, and connecting the output to a variable resistance with a voltmeter across it. With input volts and frequency constant, the load resistance is adjusted to various values and the voltage measured in volts and watts are calculated from these values. In practice a voltage increase of 40% can be permitted due to change in load, since the resulting change of 3 dB, will not be apparent in the loudspeaker output.

4.3.5. Power consumption:

The power consumption is a straightforward measurement with watt-meter in AC operated equipment, and the current is measured and wattage calculated in the case of DC equipment.

4.4. Audio mixer pre-Amplifier:

As the number of input devices can be connected to the main amplifier system is limited, there will be another device called audio mixer pre-amplifier, which accommodates more number of input devices with more no. of individual controls. The combined output of all individual channels will be connected to Aux. input of the main amplifier.

The designing aspect of mixer pre-amplifier will depends upon the requirement of no. of input channels i.e. 2, 4, 5, 8, 9, 12, 14 and 16.

4.5. Facilities of mixer pre-amplifier:

Input jacks: For connecting input devices like microphones, musical instruments, audio player, etc. Each input device will be provided with individual jacks.

Individual gain control: This is used to set optimum level for minimum noise and overload using the clip LED connectors.

Bass, Mid and Treble controls: Separate bass, mid and treble controls are provided for each individual channel.

Echo control: Each channel has an echo level control for deciding the level of that channel in the final echo mix.

PAN control: This routes the channel to either left or right output.

Channel ON/OFF switch; Each channel has a separate ON/OFF switch to allow switching OFF of the channel without disturbing the control setup.

Echo section: Echo, reverb and chorus effects can be added to all the input channels by different settings of the delay, reverb and level controls in the Echo section.

Tape section: A 4-way RCA connector allows connection of cassette recorder for recording/play back. An ON/OFF switch and gain control for play back allows adjustment of the level of cassette player in the overall programme mix.

Operating voltage: The operating voltage of mixer pre amplifier will be 12V/24V D.C.

4.6. Earthing of amplifier units:

Earthing of amplifier is necessary to provide greater safety for the person handling the amplifier and protection for the amplifier in case of faults. The connectivity of Earth to an amplifier is between the metal chassis of the amplifier and the Earth or ground. The chassis does not carry electric current under normal operation. But in the event of fault, when a live wire carrying current comes in contact with the chassis, then this current would flow to earth.

4.6.1. Methods of Earthing the Amplifier:

For Earthing of Amplifiers, use any one of the following methods.

i) Connect the earth terminal of the amplifier with a wire to a separate earth.
ii) Connect the earth terminal of the amplifier with a wire to any water pipe.

Objective:

1. An amplifier in PA equipment is a device, which takes low level input signal and amplifies to a high level output signal to the desired output power. (T/ F)
2. Bass is a low frequency control (T/ F)
3. Treble is a high frequency control (T/ F)
4. No battery current is consumed when the amplifier is working on AC mains. (T/ F)
5. For the connection of loudspeakers in voltage matching method, three terminal strips are provided viz, com. ,100V and 70V (T/ F)
6. For the connection of loudspeakers in impedance matching method, four terminal strips are provided viz., com., 4Ω, 8 Ω and 16Ω (T/ F)
7. Amplifiers are rated at some specified output in watts with a declared harmonic content, of say, 5%. (T/ F)
8. PAN control routes the channel to either left or right output (T/ F)

Subjective:

1. What are the features and facilities of an audio amplifier?
2. How do we access the performance of an audio amplifier?
3. What are the various tests to be conducted on an audio amplifier?
4. Explain frequency response test conducting on an audio amplifier?

CHAPTER-5

PLANNING OF SOUND REINFORCEMENT SYSTEM

5.0. Introduction:

A sound reinforcement system is governed mainly by the following factors:

- Size
- Type
- Location
- Audience to be covered
- Type of sound to be reproduced
- Desired psychological reaction on audience.

For the above purpose, a job analysis should be undertaken to cover the following: -

5.1. Indoor Installations:

1) Size of auditorium
2) Area to be covered
3) Dimensions
4) Approximate size of audience
5) Actual volume of the auditorium in cubic feet.
6) Reverberation time, if known
7) Seating Capacity
8) Type and distribution of absorbing materials.
9) Location of source of pick up.
10) Desired position of microphone
11) Desired position of loudspeaker
12) Ambient noise level.
13) Type of service:
    i) Voice or music reinforcement.
    ii) Remote pick up
    iii) Orchestra
14) Frequency characteristics of microphone or pick up.
15) Amplifier
16) Available Audio Power.
17) Desired Coverage
18) Permissible Cost.

5.2. Outdoor Installations:

1) Area to be covered, in sq. ft.
2) Dimensions
3) Approximate size of audience
4) Desired location of microphone
5) Desired location of loudspeaker.
6) Ambient noise level.
7) Loudest noise which the system should over-ride.

8) Type of Service.
   i) Voice or music reinforcement
   ii) Remote pick up.
   iii) Orchestra
9) Frequency Characteristics of Microphone
10) Amplifier
11) Audio power available
12) Desired coverage
13) Permissible Cost.

5.3. General requirements:

Composition of sound distribution system: Sound distribution system consists essentially of microphones, amplifiers and number of loud speakers, connecting cables, power supply units and voltage regulating devices. Standby equipments are also to be provided as per requirement.

Sound power and ambient noise levels: Taking into account such factors as audience absorption and maximum ambient level depends upon the area, but the mean variation in level of sound pressure shall be 5 to 15dB above the noise level. In quite places like waiting halls and refreshment rooms, sound level should be higher by 5dB; in closed auditoriums 5-10dB, in station premises, loco sheds, outdoor stadiums and similar noisy places 12-15dB.

A nomogram connecting the various design quantities; such as volume to be covered, required sound pressure level, reverberation time of the hall and the loud speaker efficiency to determine the required electrical power output of the amplifying system should be followed accordingly. However, care should be taken in using the nomogram as this is applicable only to hall having good acoustics and negligible feedback effect from the loudspeaker-microphone installation.

5.4. SPECIAL SOUND REINFORCEMENT SYSTEMS

Reproduction of sound is classified into two main categories, as per British Standard Code of Practice (CP/327.300)

i) ’A’ quality service, and
ii) ’B’ quality service

’A’ quality service is desirable where high quality sound reproduction (both intelligibility and fidelity) is aimed at. This necessitates planning of the acoustics of the auditorium also.

These are recommended for:-

a) Schools and other educational establishments
b) Certain Halls and Assembly Rooms.

’B’ quality service results are less realistic, but are adequate where fidelity is not a primary consideration, but intelligibility is the important criterion. These are recommended for:-

a) Hospitals and Hostels.
b) Factories and Workshops
c) Railway Stations.
d) Railway Marshalling Yards
e) Trains.

The basic functions of a sound reproducing system are:-

i) Reproduction of speech, music or stage performance
ii) Staff location paging
iii) Radio Broadcasting

5.5. Technical requirement for quality reproduction:

For "A" Category:

Frequency response: The frequency response for the entire system should be with in + 3 dB from 100 Hz to 10 KHz.

Harmonic distortion: The total harmonic distortion of the entire system shall not exceed 5% at the rated power output of the amplifier.

Signal to noise ratio: The signal to noise ratio under normal operating conditions of the amplifying system as a whole with flat operation of the tone control shall not be worse than 50 dB.

***Note*:** The normal operating conditions are those where sound pressure level of 70 to 80 dB is maintained.

Sensitivity: System should be capable of direct operation from input voltage rating 0.5mV to 1.5V.

For "B" Category:

Frequency response: The frequency response for the entire system should be with in + 3 dB from 100 Hz to 7.5 KHz.

Harmonic distortion: The total harmonic distortion of the entire system shall not exceed 5% at the rated power output of the amplifier.

Signal to noise ratio: The signal to noise ratio under normal operating conditions of the amplifying systems as a whole with flat operation of the tone control shall not be worse than 40 dB.

***Note*:** The normal operating conditions are those where sound pressure level of 70 to 80 dB is maintained.

Sensitivity: System should be capable of direct operation from input voltage rating 0.5mV to 1.5V.

5.6. ACOUSTIC SURVEY:

The object of an acoustic survey is to determine the acoustic defects in relation to the location of the sound sources so that necessary acoustic correction can be done to ensure an optimum, acceptable sound distribution**.**

5.6.1. ACOUSTIC DEFECTS: The acoustic defects are:

- Echo
- Flutter
- Reverberation

Echo: The sound reflection reaching a listener's ear at least 1/15th of a second after the original sound is termed as **echo**.

The effect of echo is aggravated by any focusing provided by any part of same building or near by building. This is chiefly confined to the frequency range above 1000 Hz and this is because of the highly directional nature of higher frequencies.

Flutter: Where parallel surfaces like side walls exists, there is a tendency for the sound energy to decay in series of steps, rather than a series of echoes of diminishing intensity, where the interval between successive steps is the time of sound to be reflected from one surface to the opposite surface. The effect is flutter.

Flutter is predominant for frequency above 1000 Hz and the aural effect is a hardness or harshness, particularly noticeable in speech.

Reverberation: It is an accumulation of echoes, one interfering with and masking the other, so that the individual echoes cannot be distinguished. It is the persistence of sound by reflection from surround surfaces after the source sound has ceased.

The effect of reverberation is garbling of speech and distortion of music. A certain reverberation is required for enhancing the effect of music or speech; too short a reverberation time produces a dead effect and in smaller rooms, this will affect the speech delivery of a speaker.

The reverberation time, a measure of reverberation, is defined as the time taken for a sound to decay in its intensity through a range of 60dB.

The total sound arriving at any one point in an auditorium consists of the original sound and successive reflected sounds at very short intervals, causing "reverberation". This effect is necessary to a small extent to improve intelligibility.

The reverberation time is calculated from the formula:

$$\text{Reverberation time in sec.} = \frac{0.015 \text{ x volume in cft.}}{\text{Total absorption}}$$

5.7. Acoustic corrections:

Requisite for acoustic corrections: The type and extent of acoustic defects are to be known, so that the acoustic correction can be applied.

Effect of parallel walls: Parallel walls must be avoided as far as possible.

Effect of side walls: Random reflections from the side walls will reduce the flutter and high efficiency absorbent material may be used on the side walls to reduce the flutter.

Areas opposite to sound source: Curvilinear surfaces and large areas of reflection opposite to the sound normally give rise to echo and they must be avoided.

5.8. Method of acoustic correction:

After calculating the reverberation time, the surface area requiring acoustic treatment may be calculated depending on the optimum reverberation time chosen.

While calculating the reverberation time, adequate allowance for the absorption by audience is to be given. Normally 50 to 60% occupation of the seats can be considered. One occupied seat will amount 33 absorption units, which can be defined as the product of area in square meters and absorption coefficient.

A common method of dealing with reverberation problems is to use absorbent material to absorb the sound and prevent reflection. Each material is rated by its ability to absorb sound and is expressed as "absorption co-efficient" of the material.

The amount of audio power for indoor locations depends upon size, reverberation period, floor plan, noise level to be overcome, nature of sound and efficiency of loudspeaker. An approximate formula is:

$$\text{Acoustic Watts} = \frac{\text{Vol. in Cu. ft.}}{10^5}$$

The efficiency of the loudspeaker should also be introduced in the calculations. For example, in an auditorium of 3 x 10^5 cft capacity, the required acoustic watt is 3 watts. Assuming an efficiency of 10% for the loudspeakers, the audio output required will be 30 watts. If a more accurate assessment is required, the monogram may be used.

For outdoor installation, a rough figure is 5 watts per thousand sq. ft. It should be remembered that sound pressure is reduced by 75% or 6 dB, each time the distance is doubled, or in other words to obtain the same sound level as obtained by a 10 watt speaker at 100 ft. for a distance of 1000 ft., 1000 watts will be required.

Assuming a reverberation period of 1 Sec., and loudspeaker efficiency of 10% output of amplifier should be 20 watts. Directional baffle type loudspeakers are employed to eliminate feedback.

The horns are tilted as because:-

For an outdoor installation, suitable for a crowd of 5000 people in quite surrounding with two projector type loudspeakers and a 20 watt amplifier is quite sufficient for speech. For musical reproduction, 4 loudspeakers with 40 watt amplifier will be required.

For a P.A. System arrangement in a playground or stadium, loudspeakers may be projector or directional baffle, with 5 watts input into each loudspeaker. The distance between each set of speakers should be 200 ft. or less. The announcer should be provided with a noise proof enclosure in such cases with a window from which the announcer can command a view of the activities. This prevents feedback.

5.9. PA System arrangement for VIP functions:

A typical P.A. System arrangement for VIP functions is as shown in the fig. 5.1 Depending on the function venue, whether indoor or outdoor or both, installing PA system for VIP functions, job analysis should be done according to the place of installation as mentioned in Para 5.1.and 5.2.

Selection of devices:

Microphones: Generally dynamic microphones are used for such occasions, as these are having unidirectional characteristics, which help in reducing feedback/howling, and also having low impedance permits long microphone leads.

Connecting microphones: Use of more number of microphones is generally essential in large stages. In such cases, output from several microphones should be connected to a mixing system and the common output will be connected to Aux. of main amplifier. Microphone cables carry low-level signal currents and are therefore susceptible for electrical interference.

Twisted pairs of conductors with sufficient insulation, screened continuously with close mesh of tinned copper braid shall be used.

The microphone cables shall be isolated from power, loudspeaker and telephone cables. Joints in the cables should be avoided as far as possible. The plugs and sockets used for microphone cables should have strong self-cleaning contacts so as to eliminate noise and they shall be non-reversible. Microphone cables should be laid without sharp bends as far as possible. Inside buildings, they may be laid on the floor along the walls or under the carpet to avoid any obstruction for the participants on the dais.

Numbering on microphone stands is necessary in case more no. of microphones is being used for easy identification.

Amplifying systems: The output power of the amplifying system should be so chosen as to be capable of establishing at any point amongst the audience, a desirable sound level of 80 dB during operation, the gain controls of the amplifying system should be set such that the signal reach each member of audience at comfortable listening level, i.e. during weak passage the signals are distinctly audible at each point, while loud passage these do not cause annoyance. The amplifying system should have a gain sufficient to deliver the required output power. However the level should not be less than 60 dB. The amplifiers should preferably be in multiples of 100W capacity, one for each group instead of using high power sets for entire installation.

100% standby amplifiers should be provided so that speech is not being held up due to defects in the working of amplifiers.

Easy changeover arrangement for switching from defective amplifiers to the standby amplifiers should be done by providing changeover switches.

Earthing should be done for all the amplifier systems properly.

Loud speakers: The number of loud speakers required, their location, direction and the power input to the loud speakers installed will have to be decided with the object of maintaining the intensity of reproduced sound above the local noise level.

The loud speakers used should have adequate power handling capacity and should normally of high efficiency type. For better reproduction, directional type of loud speakers (column speakers) should be used. The vertical directivity pattern of the system should be such as to feed the audience at uniform level. Column speakers are ideal for obtaining the vertical directivity pattern. The directivity of such speakers should be such as to provide sufficient intelligibility at all points of the seated area and avoid feedback to microphone. The spacing between two columns in a row should be approximately 8 meters apart.

Connecting loudspeakers:

a) All the loud speakers in each group should be connected in parallel and in phase across the output line.

b) The pair of wires from each group should be terminated on the announcer's panel at the amplifiers end, so that the line could be isolated from the output of the amplifier in case of any line fault or changeover to a standby amplifier.

When a number of loudspeakers are connected to the same output circuit, matching transformers shall be used with each loud speaker and connect to the 100V output line, so that it consumes the rated power.

Power supply:

- It shall be ensured that reliable mains power supply is available near the proposed location of equipment.
- The installation should be normally operated from 230V single-phase 50Hz AC mains supply with voltage regulating device.
- Diesel generator of sufficient rating should be provided for supply as and when main AC supply fails.

The system should also connect to 12/24V battery, which provides continuous operation without any interruption. Changeover to battery operation is automatic when AC main supply fails. No battery current is consumed when the amplifier is working on AC mains.

Installation:

1. All equipment should be robustly made and designed for continuous operation. Equipment should be securely installed in such a manner as to have convenient access to all sides of it.
2. When the number of equipment is not large, they may be placed on a table and wired. The positioning of the equipment is such that the lengths of the interconnecting cable are kept at minimum for convenience.
3. In case the number of equipment is large, it is desirable to mount them in racks of suitable dimensions. The height of the rack depends upon the number of equipment to be mounted and accommodation available, ensuring that all manual controls are within easy reach.
4. The patch cords if used should be tested and neatly arranged to avoid obstruction and should be easily identified.
5. The items should be installed at appropriate time after other arrangements like decoration, seating etc. are completed. This will minimize the risk of damage or loss. The wiring for the loud speakers and microphones may be laid just sufficiently in advance of the appropriate time for completing the installation so that preliminary tests that may be necessary to decide on the type and position of loud speakers could be made.

Testing after completing the installation:

A conditioning test of the installation is to be done to ensure safe and reliable operation free from faults and damages due to overheating and other possible defects that might develop in the initial period. The pre test should be ensured for at least 30 minutes before, on each and every equipment and 2 hours before on entire system.

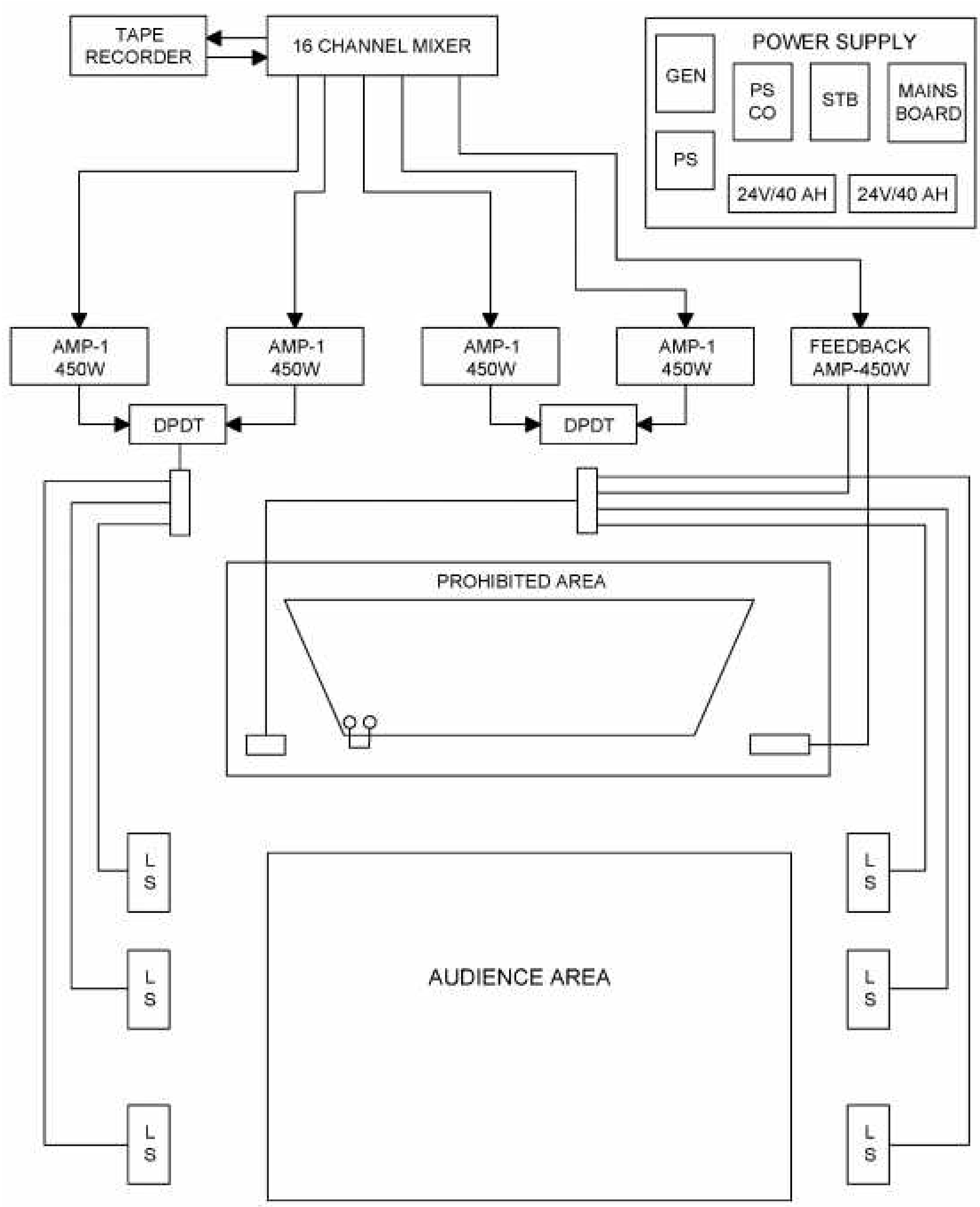


Fig.5.1. Typical PA System arrangement for VIP Function

5.10. Special types of sound reinforcement systems:

Some special types of sound reinforcement systems other than the conventional public address system used in Railways are:

i) Power Megaphone
ii) Paging and talkback System
iii) Centralized Sound redistribution system
iv) Intercommunications System
v) Integrated Sound system
vi) Train entertainment system
vii) Portable P.A. System.

5.10.1. Power Megaphone:

A typical example is the power megaphone. It is used where human voice has to be reinforced, and portability is required. Typical applications are in fire fighting, marine services and in crowd control. It consists of a sensitive carbon microphone, an efficient horn type speaker, a battery made up of flash light cells, the entire unit combined in a trigger operated assembly resembling a megaphone. A typical commercial unit is illustrated in the attached Figure 5.2. The sound will be heard at a distance of 400 yards, and if the equipment is used on the average, 10 seconds every minute, the life of the cell is 6 months. The latest versions use moving coil microphones, transistorised amplifiers and incorporate printed circuitry.

Fig.5.2. Power Megaphone

5.10.2. PAGING AND TALKBACK:

The system of calling or summoning the individuals or the general public is called “paging”. In railway stations the announcements are made to the passengers regarding the details of the trains, is an example for paging. It is a one-way communication. The system, which facilitates to talk back to the caller by the individual, is called paging and talk back system.

In railways ‘paging’ and ‘paging and talk back’ systems are very much required. Formation of the train by attaching and detaching the compartments or goods wagons is done in Railway marshalling yards. This work is executed in the supervision of yard master (YM) with the assistance of shunting staff. Hump shunting may be done at different points of the yard. Paging and talk back system facilitates yardmaster to communicate with the shunting staff or the points man regarding the formation of train.

The paging and talk back system utilises the horn type loud speaker as a loud speaker and as a microphone, which is similar to the same antenna used for both transmitting and receiving the radio frequency signals. Moving coil dynamic loudspeaker characteristics are similar to that of a dynamic microphone. This system is a simplex communication. The simple block diagram is as shown in the Figure 5.3. Yardmaster is provided with an amplifier, one microphone and a loud speaker. Audiovisual indications are provided on the Yard Master’s control console. All the speakers are connected in voltage matching system.

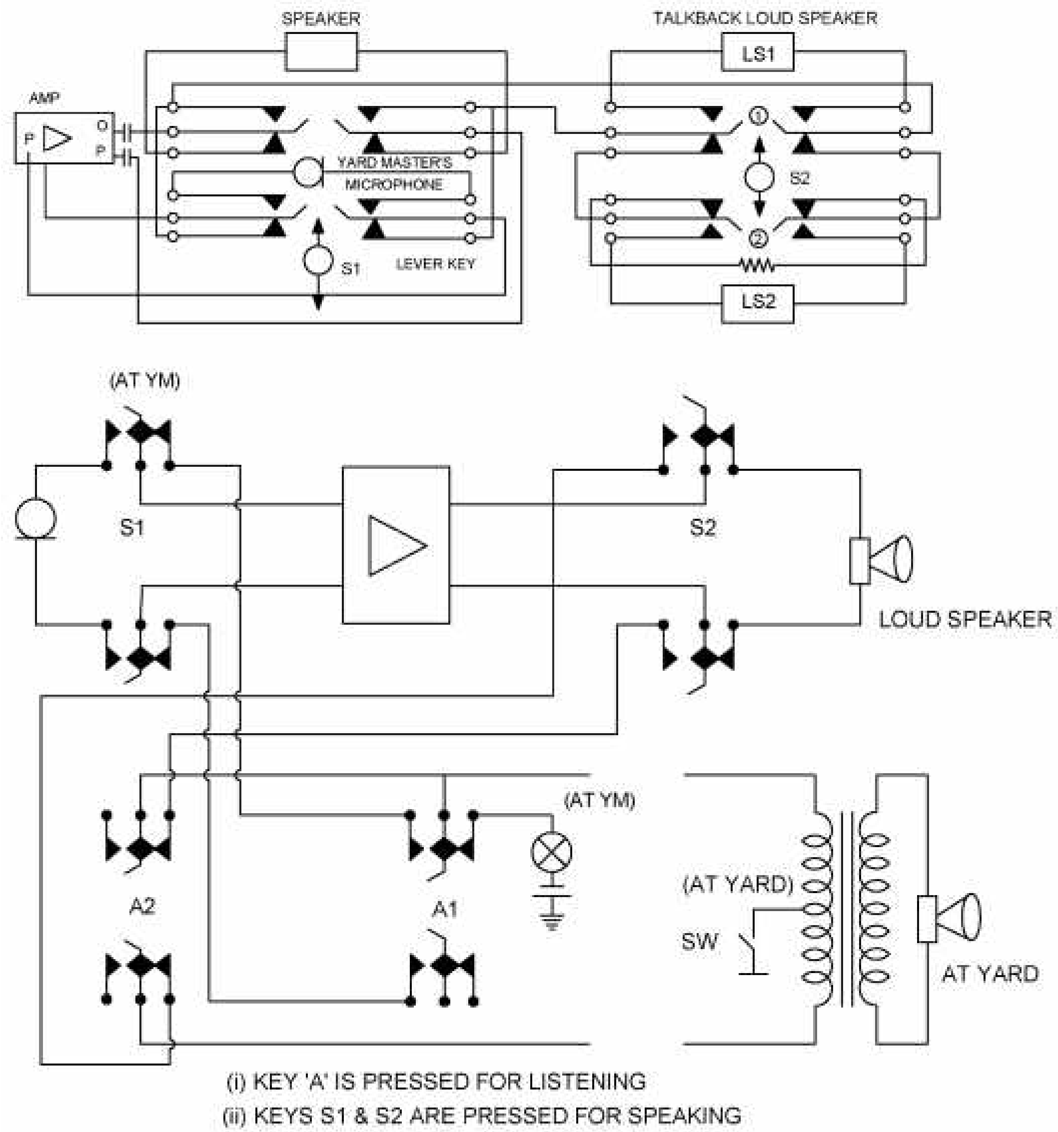


Fig. 5.3.Paging and talk back system

In the normal position local speaker is in o/p of the amplifier. S2 key selects the loudspeaker 1 or loudspeaker 2. When no speaker is connected, a dummy load resistance is connected to input of the amplifier.

1. When YM wants to speak with yard staff he selects the location by lever keys 'S2' and 'S1' keys are operated, the microphone is connected to the amplifier input and yard loudspeaker is connected to the amplifier output. Local loudspeaker is disconnected in the circuit. After the YM's instructions he changes the lever key S1 to the other direction. The local speaker is at the amplifier output and yard speaker is connected to the amplifier input so that the yard staff can speak to the yardmaster. Control of the keys is provided with yardmaster only.

2. When yard staff wants to call the yardmaster, a press to operate non-locking type switch is provided on the loudspeaker post. This is generally achieved by an earth return circuit which will activate the audio visual indication on the YM's control console to draw the attention of the yard master. Yard master will now operate the required key and the call is made.

In RE area underground cable is used and in non-RE area overhead lines are used.

RDSO standards TC 1-66 specifies the mechanical and electrical requirements of the indoor and outdoor ‘paging and talk back’ equipments including amplifiers, microphones, loudspeakers, control panels and talk back units forming part of the complete equipment.

Paging and talk back loudspeaker equipments for use in Railway Marshalling yards broadly consist of microphones, amplifiers, loudspeakers with associated line matching transformers switching equipments and control panels, the loudspeakers being connected to the amplifiers through switching equipments to appropriate aerial wires/cables and wiring, protectors and terminals. The clause that follows lay down the facilities that should be incorporated in the system. A common control panel shall be provided to control announcements on the paging system and intercommunication on the talk back system.

The paging system shall comprise a paging amplifier with associated loudspeakers and a microphone with provision on the control panel for the master to select any particular group of loudspeakers for making announcements by the operation of a key. Paging loudspeakers will be combined into suitable groups in as many numbers as possible in parallel, consistent with the rated capacity of the paging amplifiers, each group being served by a pair of aerial line wires/cable conductors.

A visual indication to monitor the level of speech on the paging system shall be provided on the control panel in the form of a VU meter.

Each Talkback point shall be provided with a loudspeaker capable of being used as a microphone as well, with the necessary line-matching transformer. Each Talk back point shall be provided with a push button of non-locking type to call attention of the master.

The Talk back system should be designed so as to require not more than two conductors between each Talk back unit and the master unit. In areas provided with either A.C. or D.C. electric traction, three conductors between each Talk back unit and the master unit are admissible, one conductor out of the three being used for signalling.

Operation of the push button associated with each Talk back unit should give a visual and audible indication to the master operating the control panel, the visual indication giving the identity of the Talk back unit. The indicators shall form part of the control panel and shall continue until acknowledged by the operator by the operation of a key relevant to the Talk back unit, which shall connect the Talk back to the master unit to enable inter communication. Only one amplifier shall be provided for the Talk back systems for inter communication between the Talk back unit and the master, the control resting with the master through a Talk/Listen key.

For talk back units located very far away from the master unit, transistorised pre amplifiers shall be provided at such talk back units controlled by a relay actuated by the Talk/Listen key on the control panel. A visual indication to monitor the level of out going speech on the Talk back system shall be provided on the control panel in the form of a VU meter.

The microphones provided for the paging amplifier shall also be capable of being used as the microphone for talk back purpose by the master. A volume control shall be provided on the control panel to adjust the level of incoming speech.

Hundred percent stand by shall be provided for the paging and Talk back amplifier and the power supply unit for the operation of the signalling relays, with facility to change over to the standby units with ease.

5.10.3. Platform announcement systems:

Another special system for Railways is the station announcing equipment. It consists of an announcing booth at a convenient location (preferably the Station Master's Office or enquiry cabin) from where the information regarding train movements could be obtained. With a network of loudspeakers at different locations, the cavernous structures of station normally present serious echo problems and also excessive reverberation time. Acoustical treatment is an extremely costly proposition so that considerable ingenuity has to be exercised in locating the loudspeakers.

The location of speakers may be broadly divided into:

i) Platforms ii) Circulating area
iii) Waiting Rooms iv) Restaurants.

The distribution lines are divided into groups according to the above classification and switching arrangement for selecting different groups are provided. The amplifier as well as microphone should be duplicated for maintaining continuous operation. Horn type reflex or bi-directional has to be used for (i) and (ii), and cabinet types for (iii) and (iv). Individual volume controls have to be provided for cabinet type speakers. The arrangement of a typical layout is shown in the Figure 5.4.

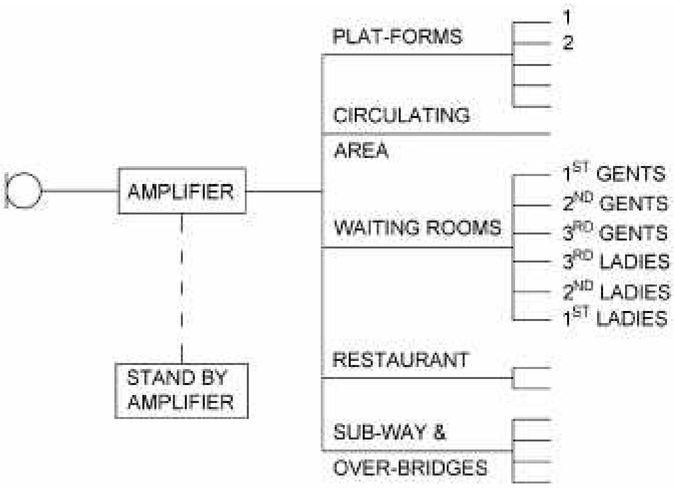


Fig.5.4. Typical layout of a station announcing system

The announcing booth should be noise proof and acoustically treated, with a glass window or windows on sides to have a commanding view of the Station Yard, if so situated. The control panel consists of switches for connecting the different distribution lines with a volume level indicator and pilot lamp indication in the centre. Mains and standby switches for the amplifiers and "Transpose" switches for rapid changeover from one amplifier to another are also provided. A record player or tape recorder is provided in one deck, and the amplifiers are housed in the double deck on one side. Wind netting is provided for ventilation.

5.10.4. Train entertainment system:

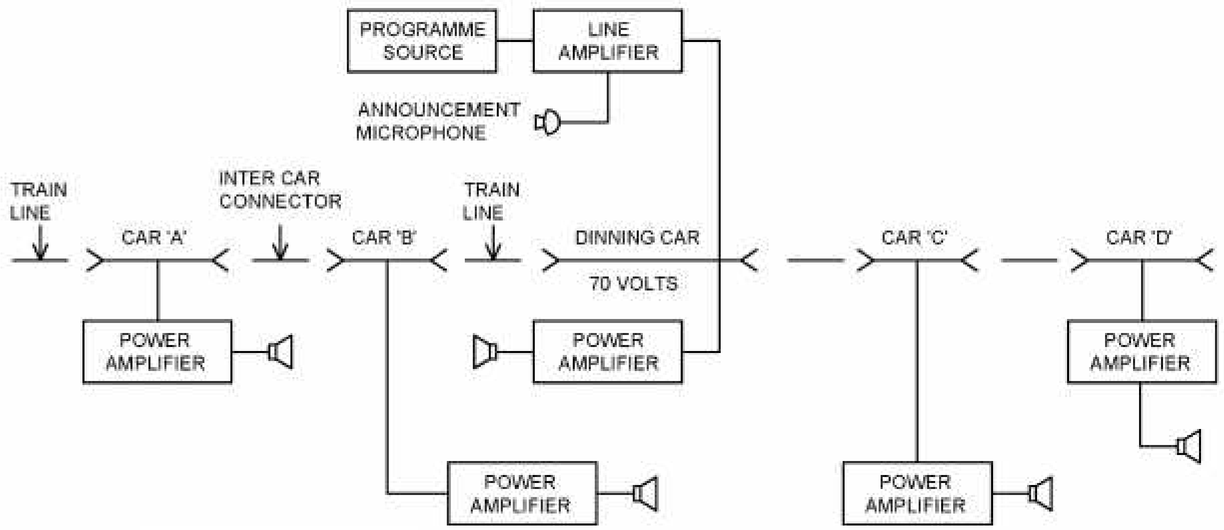
P.A. Systems are also installed in trains. These consist of two basic systems:

i) Low Level System.
ii) High Level System.

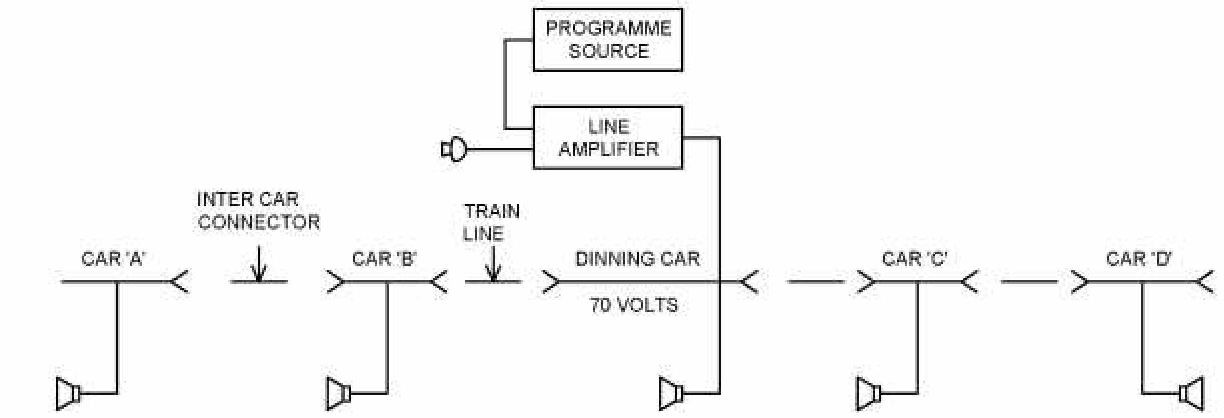
In type (i), each coach has its own amplifier, and in (ii) the main amplifier is in the dining car or guards cabin and high voltage distribution line (100V) is taken to each coach. For this arrangement, the entire rake of carriages have to be wired internally and fitted with external coupling units, so that if the formation is distributed or a new coach introduced in between these coaches the wiring will get disconnected. Hence, in some cases, the arrangement is restricted to the dining car only. Railways have evolved methods of loudspeaker announcements for intercity expresses, of which a desired type is to be chosen according to the passenger reactions of the particular route.

The three methods are:

i) 'Live' voice announcements by the guard i.e., manual system.
ii) Pre-recorded announcements operated by the guard i.e., semi automatic system.
iii) Pre-recorded announcements operated automatically during the progress of the train i.e., automatic system.



A) Low Level System



B) High Level System

Fig. 5.5.Train Entertainment System

The announcements will be a welcome to the passengers, when the train departs and details of its scheduled arrival and calling points. The location of refreshment car is also announced with details of sittings for means and service of light refreshments. Warnings of the approach to the next stop are also broadcasted, well in advance, for passengers to pack up their belongings and be ready to alight when the train arrives. Short intermissions of background music are also transmitted. In the automatic system, pre-recorded announcements are initiated at appropriate points in the journey by special devices, which measures accurately the distance travelled. The carriage lighting circuits are used for transmission of the audio signals.

As per the Railway Board's commercial circular 35/2012, Hindustani or Carnatic music shall be played in trains like Rajdhani, Shatabdi and Duronto or where the facility is available. It is desired that when the music is playing, time of the day and region through which the train is passing may be kept in view.

5.10.5. Conference system:

It provides the much-required sound reinforcement for eliminating problems of poor intelligibility while controlling acoustic feedback. The system is versatile and easy to install in a variety of applications in Railways. The main usage of this system in Railways is GM'S conference room at Zonal head quarters, DRM'S conference room and Institutes like IRISET etc. for conducting conventions and seminars on technical topics.

Conference system mainly consists of one Chairman Unit, one secretary unit, several delegate units as desired and central amplifier with connecting cable and loudspeaker system.

Chairman unit: This unit is specially designed for chairperson, who is chairing the meeting. It consists of a built-in loudspeaker and highly sensitive electret condenser microphone mounted on flexible gooseneck arrangement.

Fig.5.6. Chairman unit

The microphone of the unit is equipped with a lockable switch and a ring LED indicator for speaking and also visual identification of the speaker. It will also have a non-lockable PRIORITY switch, which interrupts and mutes the delegates' unit microphones. The built-in speaker has recessed volume control for setting desired volume levels.

Secretary unit: This unit enables proceedings to be recorded through a cassette recorder, for a stenographer present to take notes and to relay pre-recorded messages if any, to delegates.

Fig.5.7. Secretary unit

Delegate units: These units are similar to chairman unit with the exception of the priority switch not being provided.

Central Amplifier: It is provided for connecting conference units consisting of chairman, secretary and delegate units.

Inter connection of conference system:

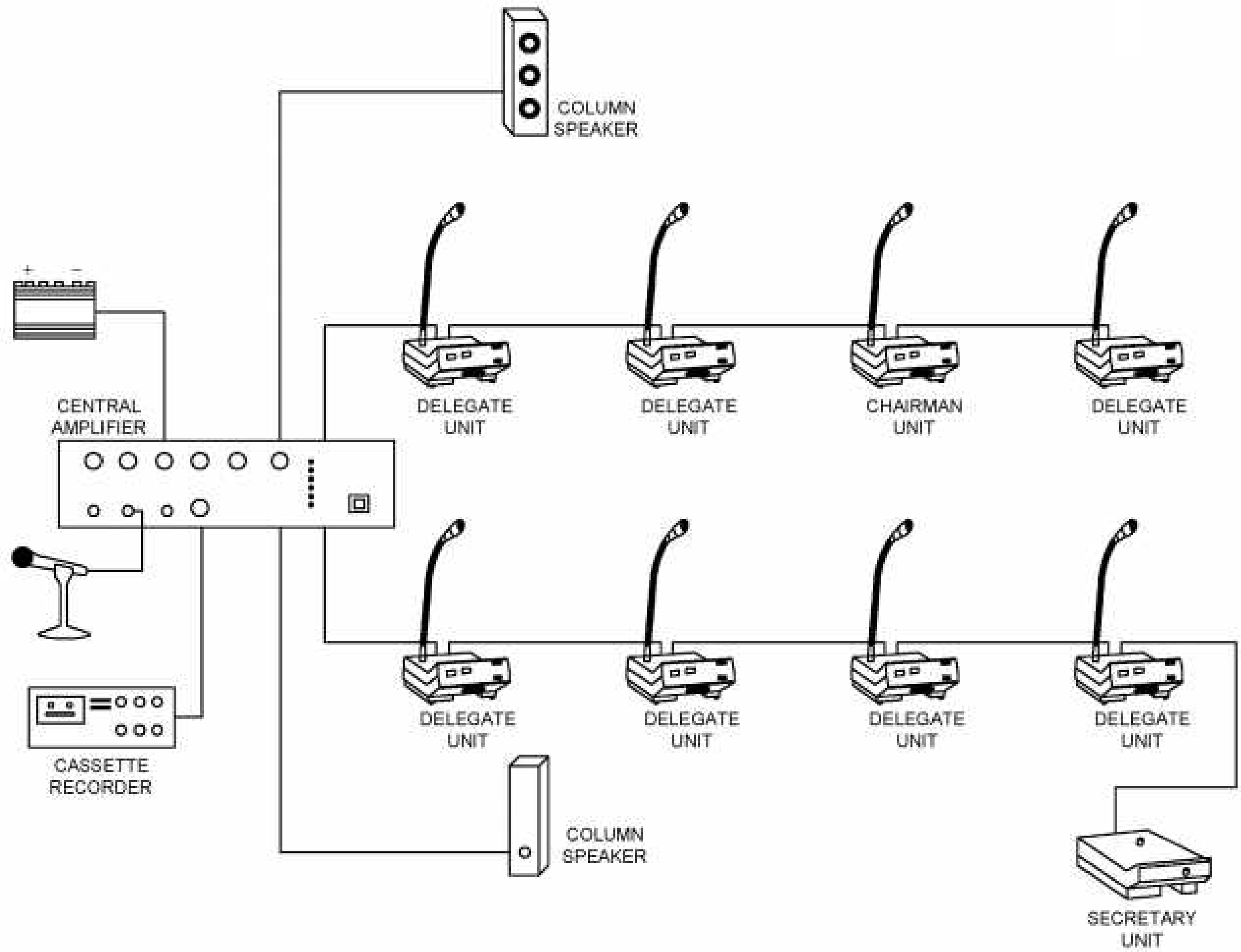


Fig. 5.8.Typical Interconnection of conference system

Objective:

1. The mean level of sound pressure shall be 5 to 15dB above the noise level. (T/ F)

2. The frequency response for the entire system should be within + 3 dB from 100 Hz to 10 KHz. (T/ F)

3. The total harmonic distortion of the entire system shall not exceed 5% at the rated power output of the amplifier. (T/ F)

4. The signal to noise ratio under normal operating conditions of the amplifying systems shall not be worse than 50 dB. (T/ F)

5. In the normal operating conditions sound pressure level is 70 to 80 dB (T/ F)

6. The sound reflection reaching a listener ear at least 1/ 15 th of a second after the original sound is termed as echo (T/ F)

7. Reverberation is an accumulation of echoes (T/ F)

8. The system of calling or summoning the individuals or the general public is called “paging”. (T/ F)

9. The system, which facilitates to talk back to the caller by the individual, is called, paging and talk back. (T/ F)

Subjective:

1. What are the factors to be considered for planning of sound reinforcement?

2. What are the general requirements for a sound distribution system?

3. What are acoustic defects and define them briefly?

4. What are various types of special types of sound reinforcement system used in railways?

CHAPTER-6

PHASING AND MATCHING METHODS IN SOUND REINFORCEMENT SYSTEMS

6.0. Phasing of loud speakers:

In a multiple loudspeaker system, when two or more driver units/loud speakers are facing in the same direction and are installed in same area, it is essential that their diaphragms/cones act in unison. For this action, the loud speakers must be connected in phase with each other, is called as phasing of loud speakers.

In any system, where the acoustic waves generated by individual loudspeakers should be "parallel" (loudspeakers in same plane) their phasing is extremely important. Where the sound comes from overhead or in front of the audience, reversal of phase of any one loudspeaker will result in a serious "hole" in coverage.

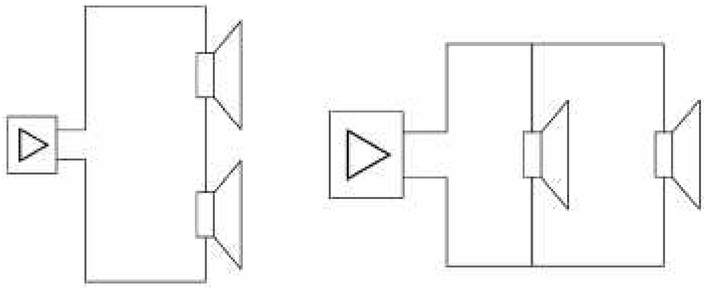
When the intensity from the reversal loudspeaker is equal to that from others in the system, the combined effect is a transverse wave movement that create the impression that the local loudspeakers are not working at all, and actually what the listener “thinks" that he hears, is confused sound from more distant loudspeakers. This will be the complaint usually received under these circumstances and it should be checked

(a) By verifying that the speakers are connected, and
(b) That their phasing is correct.

When setting up an installation, such as a large hall, where both parallel and successive waves occur one must work successively. First connect only the first group and check for correctness of phasing. Then connect only the second group and so on, checking each group for phasing within the group, separately from other groups. Next check for the best phasing between successive groups, first and second, then with both these connected, find the best phase for the third group, and so on, until the entire system is connected.

6.1. Connectivity of speakers in phasing:

Some loudspeaker manufacturers mark on winding terminal of the speaker for phasing purposes, so that when a positive voltage is connected to the marked terminal (usually a red dot), and a negative voltage to the other one the diaphragm moves forward. Where wiring is identified in polarity, by black/red, red/white or black/white insulation colouring, it makes the ensuring of correct electrical phasing simple. If all speakers are connected in parallel to the amplifier, it is a simple matter of connecting all marked terminals to one colour of wire and all unmarked terminals to the other colour.



Series connection **Parallel connection**

Fig.6.1. Phasing of Loud speakers in series and parallel connection.

The connections to the voice coil whether in series or in parallel, must be made in such a manner that in any one instant all diaphragms must be moving outward or inward in unison.

When the voice coil terminals are not marked, the simplest way of determining the correct phasing is by doing as follows. Take 1.5V or 3V batteries (dry cell) with the polarity marking of each observed, connect them momentarily to the voice coil of the speaker. The cones or diaphragms should move in the same direction for correct phasing. The voice coil connections of those speaker cones that move in opposite directions should be changed.

Phasing is of least importance where two loud speakers are at a good distance apart (as a thumb rule if the distance is more than 5 meters) or pointing in opposite directions.

6.2. Matching methods in sound distribution system:

Any system of connecting a number of speakers to an amplifier has to serve two purposes:

a) Deliver the full available output of the amplifier, in correct proportions, to the individual units of the system.
b) Do so efficiently.

The "correct proportions" is a matter of distribution, the realization of full available output, efficiently, is a matter of matching by impedance.

6.2.1. Impedance matching: For correct transfer of power from the amplifier to the speaker group, we require correct impedance matching.

The effective impedance of the load should be matched with the output impedance of the amplifier. For this, the distribution of number of speakers connected to amplifier may be either in series, parallel or combination of series parallel.

For the purpose of impedance matching, the amplifier output tapings will be given as Com., 4Ω, 8Ω and 16Ω.

For connecting the speakers in impedance matching method to an amplifier the following procedure must be followed-

Calculation of effective impedance:

When speakers are connected in series, the total impedance is the sum of the individual impedance of speakers.

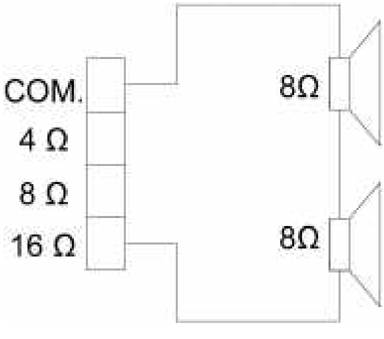


Fig. 6.2.impedance matching in series connection

As shown in fig. 6.2, each speaker will have an impedance of 8Ω; the resultant impedance will be 16 Ω. So for the perfect impedance matching we will select the 16 Ω tapping of the amplifier.

When the speakers are connected in parallel and the impedance of the speakers connected is same, then the effective impedance will be

$$\frac{\text{Impedance of each speaker}}{\text{No. of speakers}}$$

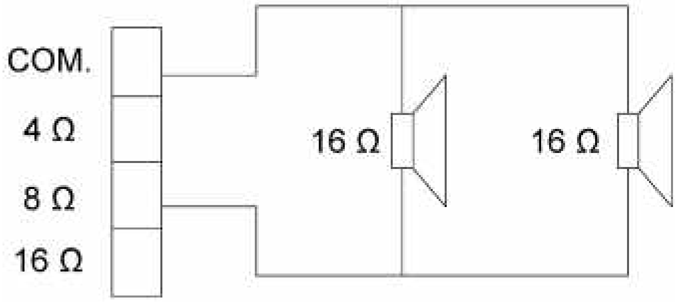


Fig. 6.3.impedance matching in parallel connection

As shown in fig.6.3 each speaker will have an impedance of 16 Ω each; the resultant impedance will be 8 Ω. So for the perfect impedance matching we will select the 8 Ω tapping of the amplifier.

When speakers are connected in series-parallel connection, the resultant impedance will be calculated as follows.

Four speakers connected in two groups as shown in fig. 6.4 having 16 Ω each,
Impedance of group A = $R_1 + R_2$ = 16+16 = 32 Ω.
Impedance of group B = $R_3 + R_4$ = 16+16 = 32 Ω.

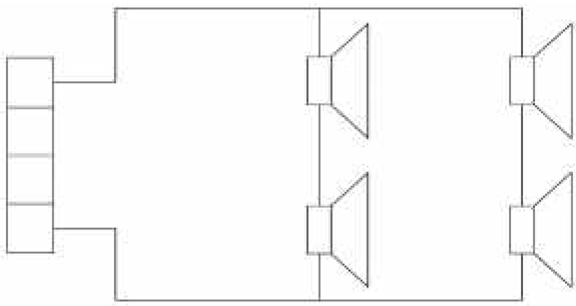


Fig. 6.4. Impedance matching in Series parallel combination

Total impedance = $\frac{R_A \times R_B}{R_A + R_B} = \frac{32 \times 32}{32 + 32} = \frac{1024}{64} = 16\ \Omega.$

However, in some cases it may not be possible to get exact resultant impedance as required to connect the suitable taping available on the amplifier output. In such cases, if (R_T) resultant impedance is more than the impedance tap of amplifier it is connected, the speaker group will draw less current, so power delivered from amplifier will be less and the sound output from the speakers will be less. But amplifier will run safe. If R_T is less than the impedance tap of the amplifier, the speaker group will draw more current from the amplifier and over heat the amplifier and damage output stage of amplifier. In such cases, wherever the perfect matching is not possible, it is advisable to adopt higher value of impedance of the speaker group will be connected to lower value of impedance of the amplifier.

Any individual unit may develop open circuits or may develop a short circuit. In loudspeakers, open circuits are the more common fault. In parallel connection, a short circuit would 'kill' the whole line, while an open circuit 'kills' only the defective unit. In series connection, an open circuit 'kills' the whole line, while a short circuit only 'kills' the defective unit. Thus, as opens are more common, the parallel connection has a slight edge from the safety factor viewpoint.

6.2.2. Voltage matching:

In this system, loudspeakers are connecting to the amplifier through a line matching transformer to either 70V or 100V tapping.

This method is based on "constant voltage" system, which simplifies to a great degree, the computation of the proper transformer taps. It also permits the addition of speaker to an existing system, without the recalculation of the load and source impedances so long as the total power consumed by the loud speaker is less than or equal to the amplifier rating. It permits loudspeakers to be connected across a transmission line with the same fashion as electric lights are loaded on a power line up to the capacity of that circuit. To use this method requires that a power amplifier incorporate an output transformer tap, which will deliver 70.7 volts at the rated output of the Amplifier.

The 70 volts is the maximum voltage on a sine wave test signal for a given amount of distortion. This expresses the standard for rating amplifier power. Standardizing this voltage means that the voltage is the same for a low power amplifier as for a high power amplifier. The condition of 70 volts at the output will exist when the amplifier is terminated in its rated load impedance and is supplying power to the load equal to the rating of the amplifier. However, to simplify calculations, it is considered at the rated output.

How to use the Constant Voltage System:

In a large installation requires a large number of speakers to be connected at a distance. As the distance increases the length of the cable increases so the strength of the sound signal decreases. Hence in such installations where the length of the cable is substantial, line matching transformers (LMT) are being used in voltage matching method, between the output of the amplifier and each individual speaker to be connected.

LMT is having multiple high impedance tapings on primary side and standard impedance tapings of com. 4 Ω, 8 Ω and 16 Ω on secondary side. The primary side will be connected to output of amplifier to 100V line and the secondary will be connected to loudspeaker. It will act as a step-down transformer to the speaker.

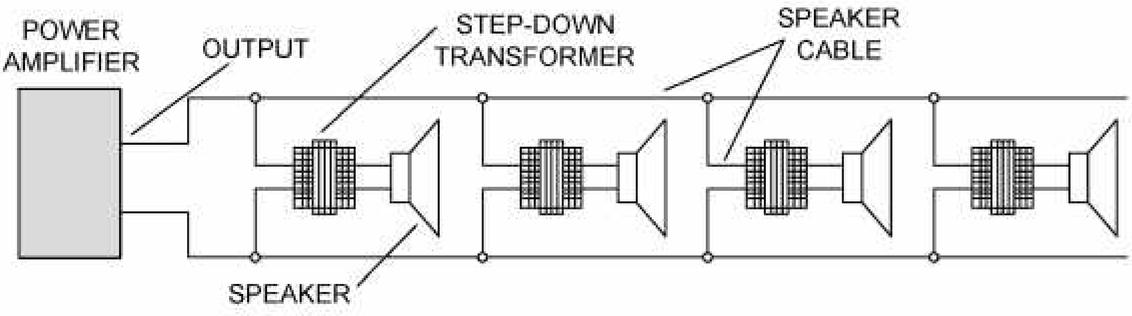


Fig.6.5. Typical connection of voltage matching

When transformer taps are marked directly in "watts" no mathematic calculations are required. Simply choose a transformer with the correct power tap and connect to the desired terminals.

For transformers marked in impedance, follow the procedure below. Actual calculations are simple. One basic formula is used:

$$\text{Required impedance (Z)} = E^2/P = \frac{(\text{Output Voltage})^2}{\text{Desired power}}$$

So that, an amplifier employing a 100-volt output tap, the formula is reduced to

$$\text{Required impedance} = \frac{10000}{\text{Desired power}}$$

Although for convenience in application, amplifiers, speakers and matching transformers are given a voltage and wattage rating, rather than their operating impedance, they are still designed on an impedance basis.

Example:

Problem: In a factory to connect 8 speakers of different wattages to an amplifier. Two speakers in the factory yard require 25 watts each. Two speakers in the factory work shop 20watts each. Two speakers in the canteen to require 10 watts each.Two speakers in the supervisors' room to receive 5 watts each.

Solution: For the above installation the required material is

1. Amplifier 1 No.
2. Loudspeaker 8 Nos.
3. Line matching transformers 8 Nos.

The amplifier can be capable of delivering at least the required wattage i.e.,

25 w x 2	=	50 w
20 w x 2	=	40 w
10 w x 2	=	20 w
5 w x 2	=	10 w
Total :		120 watts

CONDITION: The power handling capacity of the loud speakers must be greater than or equal to the required wattages if they are with out line matching transformer if the loud speakers used are with line matching transformers the wattage rating must be equal to the required wattage. The line-matching transformer should be capable of handling the required power.

Calculation for primary impedance of the LMT.

Amplifier Line Voltage is 100V.

$$\text{Primary impedance of LMT } Z_P = \frac{V^2}{P}$$

Where V is the line voltage and 'P' is the wattage of speaker.

$$\text{Therefore for 25 watts ZP} = \frac{100 \times 100}{25} = 400 \text{ ohms.}$$

$$\text{For 20 watts ZP} = \frac{100 \times 100}{20} = 500 \text{ ohms.}$$

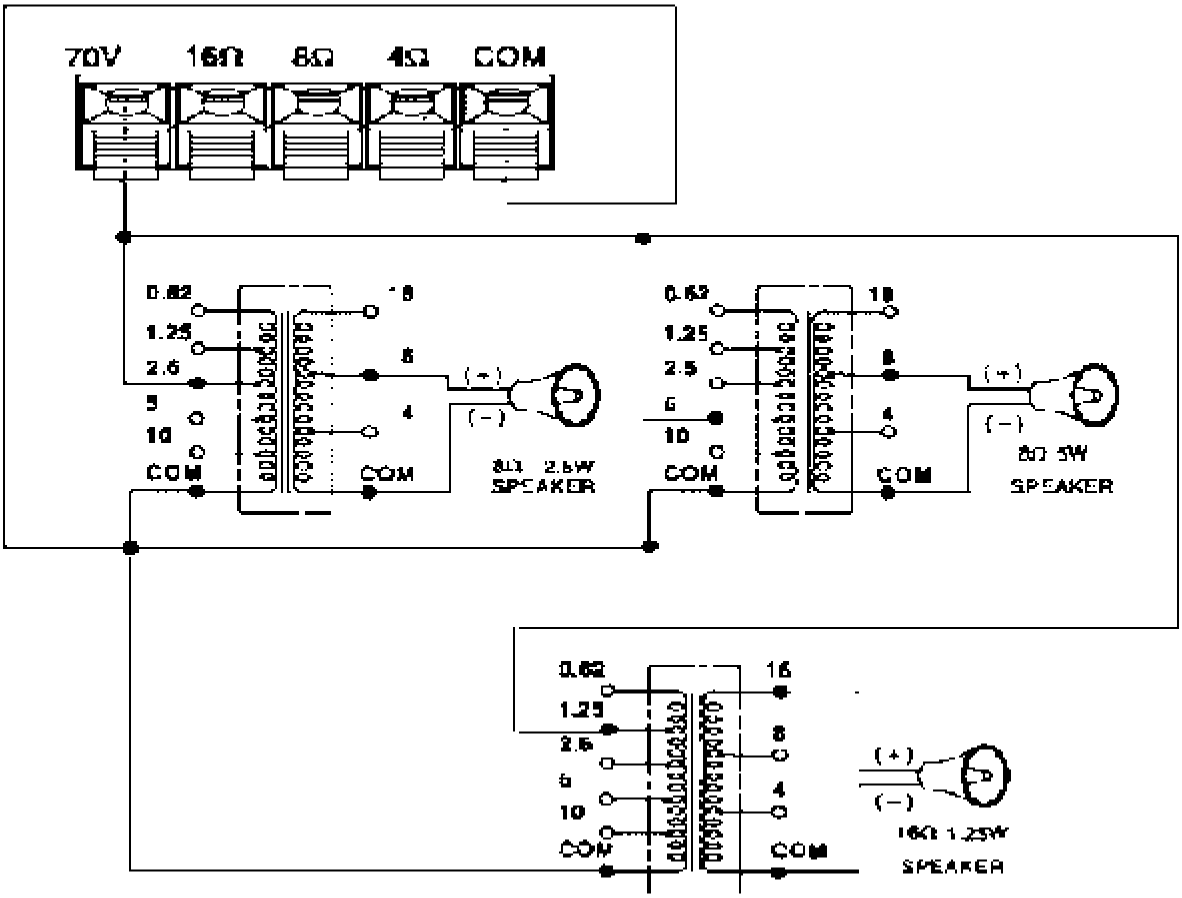
$$\text{For 10 watts ZP} = \frac{100 \times 100}{10} = 1000 \text{ Ohms.}$$

$$\text{For 5 watts ZP} = \frac{100 \times 100}{5} = 2000 \text{ Ohms.}$$

Directly loud speakers of required wattages could be connected, if the loudspeakers are with built in line matching transformers.

NOTE: The total sum of wattages of the speaker connected be less or same as the amplifier wattage. If it is more than the amplifier wattage, the amplifier will be overloaded and the load will draw more current, so that amplifier gets overheated.

Construction of Line Matching Transformer:



Objective:

1. The effective impedance of the load should be matched with the output impedance of the amplifier (T/F)
2. Line matching transformers (LMT) are being used in voltage matching method. (T/F)
3. The power transfer is maximum in impedance matching. (T/F)

Subjective:

1. What is the purpose of matching techniques used in PA systems?
2. Define impedance matching method and voltage matching method?
3. Explain impedance matching method?
4. Explain voltage matching method?