MAINTENANCE HANDBOOK
ON
PUBLIC ADDRESS SYSTEM

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Indian Railways
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ISSUE OF CORRECTION SLIPS

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PUBLIC ADDRESS SYSTEM

1. Introduction

Public Address System (PA system) is an electronic sound amplification and distribution system with a microphone, amplifier and loudspeakers, used to allow a person to address a large public, for example for announcements of movements at large and noisy air and rail terminals.

The simplest PA system consist of a microphone, an amplifier, and one or more loudspeakers is shown in fig 1. A sound source such as compact disc player or radio may be connected to a PA system so that music can be played through the system.

The process begins with a sound source (such as a human voice), which creates waves of sound (acoustical energy). These waves are detected by a microphone, which converts them to electrical energy. This signal is amplified in an amplifier up to a required level. The loudspeaker converts the electrical signal back into sound waves, which are heard by human ears.

A block diagram of PA system containing microphone, mixer, limiter, equalizer, amplifier and speaker is shown below in figure No.2:

Fig No. 1: Simple PA System

Fig No. 2: Block Diagram of PA system
1.1 Application of P.A. system in Railways

Passenger Amenity

For giving the detailed information about the train arrivals, departures, late running if any, and location of trains and any other important information related to Railway users.

Marshalling Yards

For communication between Yard Master and Shunting men through paging and talk-back system regarding formation and reception or dispatch of trains.

Breakdown train Emergency Equipment

The P.A. System in Accident Relief Train must be kept in working condition for guide the passengers and staff in rescue operations at the site of accident.

Special functions

Local Minister, G.M., etc., officials and VIPs may address some important functions such as Railway Week, felicitations, Scouts and Guides rally, some social work meetings, cultural programmes etc. a quality P.A. System needs to be installed.

Railway Workshops

Providing announcements to workshops staff when required and also for entertainment music during lunch hours.

Conferences

For conducting seminars, special lectures, administrative meetings for a limited group of officials in conference halls. In every zone, a G.M. Conference hall is available. In these suitable conference systems were permanently installed.

2. Acoustic

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.

2.1 Intensity

This relates to amplitude of acoustic force. It is expressed in watts per square centimeter. The acoustic power is inversely proportional to the reverberation period for a given intensity level.
2.2 Sensitivity

A speaker’s sensitivity is the on-axis loudness (dB SPL) measured at a specific distance that results from applying a specific amount of power (1 Watt @ 1 meter). The output level of the speaker at different power levels and distances can be calculated from this figure. For example: If a speaker’s sensitivity is rated at 96 dB SPL with a 1 W input measured at 1 mt from the speaker, then doubling the power to 2 W raises the output 3 dB to 99 dB SPL at 1 mt. Doubling the power again to 4 W produces 102 dB SPL.

2.3 Sound Pressure Level

Sound Pressure Level (SPL) is the acoustic pressure reference for the dB. The minimum threshold of undamaged human hearing is considered to be 0 dB SPL. The threshold of pain for undamaged human hearing is 120 dB SPL.

2.4 Loudness

It is the intensity of the sound stimulus as perceived by the human ear and chiefly a function of sound pressure. However, it is also dependent upon the frequency and the complexity of the waveform. The extra high-frequency content makes the sound seem louder. Our ears do not hear all frequencies equally. They are most sensitive at around 3-4 kHz, and much less sensitive at the extremes of frequency. Effectively, ears will turn down the volume and mute the damaging highest frequencies if the concert is too loud. However, any sound system can damage hearing if you get your ears close enough and drive it hard enough for long enough.

2.5 Attenuation over Distance (Inverse Square Law)

The inverse square law describes how sound attenuates over distance. It states that volume (SPL) decreases 6 dB each time the distance from the sound source is doubled. This is due to the diffusion of sound radiating from the sound source over a spherical area. As the radius of a sphere is doubled, its surface area quadruples, effectively dividing the acoustical power by four.

2.6 Frequency:

Frequency is the number of "Cycles per Second" and is expressed as "Hz".

2.7 Decibel (dB)

The decibel (dB) is a logarithmic unit that indicates the ratio of a physical quantity (usually power or intensity) relative to a specified reference level. A ratio in decibels is ten times the logarithm to base 10 of the ratio of two power quantities.

The decibel is used for a wide variety of measurements in science and engineering, most prominently in acoustics, electronics, and control theory. In electronics, the
gains of amplifiers, attenuation of signals, and signal-to-noise ratios are often expressed in decibels. The decibel confers a number of advantages, such as the ability to conveniently represent very large or small numbers, and the ability to carry out multiplication of ratios by simple addition and subtraction.

A change in power ratio by a factor of 10 is a 10 dB change. A change in power ratio by a factor of two is approximately a 3 dB change.

For instance, suppose we have two loudspeakers, the first playing a sound with power P1, and another playing a louder version of the same sound with power P2. The difference in decibels between the two is defined to be \(10 \log_{10} \frac{P2}{P1}\) dB. If the second produces twice as much power than the first, the difference in dB = 10 \(\log_{10}\) \(\frac{P2}{P1}\) = 10 \(\log_{10}\) 2 = 3 dB.

If the second had 10 times the power of the first, the difference in dB would be 10 \(\log_{10}\) \(\frac{P2}{P1}\) = 10 \(\log_{10}\) 10 = 10 dB.

If the second had a million times the power of the first, the difference in dB would be 10 \(\log_{10}\) \(\frac{P2}{P1}\) = 10 \(\log_{10}\) 1,000,000 = 60 dB.

This example shows one feature of decibel scales that they can describe very big ratios using numbers of modest size. But note that the decibel describes a ratio of powers.

Response of human ear is also logarithmic i.e. if the sound level is increased by 10 times it feels one time and if power increased 100 & 1000 times it feels two & three times respectively. It is very difficult to draw a graphic presentation of this ratio. Therefore by taking the logarithm of 10, 100, 1000 etc we will have smaller figures as below:

\[
\log_{10} (10) = 1; \log_{10} (10^2) = 2; \quad \log_{10} (10^3) = 3; \quad \log_{10} (10^4) = 4
\]

\(\text{dB}_m\)

It is understood that dB is a unit of power ratios but not of power. It can be applied for power also by providing a reference level. This power level is widely taken as one milli-watt across 600 ohm resistance. This one milli watt is also called ‘Zero Power Level’. So a power of X milli watt w.r.t. 1 milli watt can be expressed as \(\text{dB}_m\) or \(\text{dB}_m\) = \(\log_{10}(X/1)\). So when a reference of 1 mW across 600 ohm is taken it is called \(\text{dB}_m\).

**Zero Voltage Level**

It is a voltage applied across a 600 ohm resistance which dissipates one milli watt power (Zero Power). Since \(P = V^2/R\), so \(V = \sqrt{P \times R} = \sqrt{1 \times 10^{-3} \times 600} = 0.775\) volts or 775 mV
Zero Current Level

It is equal to current flowing in a 600 ohm resistance which dissipates one milli watt power (Zero power). Since $I^2R = P$, or $I = \text{Root of } P/R = \text{Root of } 1\times10^{-3}/600 = 1.291 \text{ mA}$.

2.8 Masking

It refers to one sound being obscured by another so it is indistinguishable from the first sound. It is one of the main obstacles to speech intelligibility. Typical sound systems include a number of potential sources of masking effects. Background noise is the most obvious. Sound system’s normal operating level should be at least 15–25 dB above the background noise level. High distortion level in amplifiers, speakers, or other sound system components, is another possible source of masking.

2.9 Reverberation

Reverberation is another common source of masking-related intelligibility loss. Significant reverberation occurs in a large room (i.e., church, gymnasium, or auditorium) where repeated reflections merge into a seemingly continuous sound with a gradual rate of decay. Many installed sound systems are used in spaces where there is little or no significant reverberation.

2.10 Equalization

It is the process by which the amplitude of discrete frequency ranges is adjusted. In distributed systems, EQ is most often used to compensate for speaker and room characteristics but can also be used for aesthetic enhancement. Many applications do not require equalization. The benefits of using equalization include improved speech intelligibility, enhanced sound quality due to a better spectral balance, and increased gain without feedback.

2.11 Impedance

The unit of impedance is $\Omega$, and its abbreviation is Z. When an electric current runs through a coil, resistance occurs that prevents the flow of the electric current. The value of resistance when an alternating electric current runs is called impedance, where $Z = \Omega$. There is very thin coil wrapped around the magnet inside of a speaker. The impedance value of this speaker coil is designed either at 4Ω, 8Ω, or 16Ω. The impedance value of the output connector of an amplifier is also designed at 4Ω, 8Ω, and/or 16Ω.

2.12 Impedance matching

Interfacing between the amplifier and speakers is commonly done in one of two ways. Small systems with one or two speakers will typically use a direct connection
between the speakers and the amplifier. This is sometimes called low impedance operation, because the load impedance ranges from 4 Ω to 16 Ω nominal.

Systems with more than 2 speakers usually use transformers at the amplifier and at each speaker to simplify impedance matching and reduce line loss. These systems are commonly called distributed line systems, 100 volt systems, or constant voltage systems. In both cases, speakers should be wired in parallel (plus to plus and minus to minus).

These systems work by including transformers at the input to each speaker and directly after the amplifier output. The transformers are used to convert the impedance of each speaker to a higher value, and to convert the amplifier output impedance to a correspondingly high value.

High impedance (100 volt and 70 volt) systems have three major advantages over low-impedance systems:

1) System impedance-matching is made much easier. It is simply a matter of adding up speaker power taps and selecting an amplifier rated for at least that much power.
2) Line loss is greatly reduced, especially over long cable runs, resulting in better performance and reduced cost compared to long low impedance lines.
3) The amplifier output is electrically isolated from the speaker line by the output transformer, protecting the output stage against a grounded line and thus eliminating a potential source of system failure.

![Fig No. 3: High Impedance Matching](image)

Impedance matching is necessary to get optimum volume, avoid wasting power, avoid excess stress & prevent damage to amplifier & speakers, reduce distortion and noise & avoid uneven sound distribution.

### 2.13 Balanced Line

A balanced line is simply one that has three conductors wired separately. Two of these are signal wires which are wired out of phase with each other and the third
one is ground (usually this is the shield). The big advantage of a balanced cable is that it is designed to cancel many types of noise.

2.14 Feedback

Feedback is the loud squeal that is often heard in a P.A. system when a microphone is pointed too close to a speaker cabinet or the volume gets too loud. A squeal lasting less than a second is generally harmless. However, keep feeding back for very long duration tweeters and/or horns will get so hot that their coils burn and they stop working.

2.15 Dropping

Most P.A. speakers can take a degree of rough handling. However, if a cabinet takes a hard enough impact, it is possible that internal parts of the speaker can shift. Speakers have heavy magnets hanging off the back of them and momentum on a hard enough drop will cause the magnet to shift. Remember, the way a speaker creates sound is by vibrating hundreds and even thousands of times per second – it doesn’t take much of a shift to throw the alignment of the various parts of a speaker out enough that they will rub. When a moving part on a speaker rubs, the part receiving the friction eventually rubs through and causes the speaker to fail. Usually it is the wire in the coil that is rubbing and it eventually rubs so thin that it breaks or shorts, thereby causing the speaker to stop moving.

2.16 Bad Cables

Besides all of the other nasty things we have discussed about using improper cables, another problem they can cause is oscillations. Oscillations can occur in a P.A. system when the ground has come off in a cable. Everything may seem to be working alright but a missing ground can cause a high frequency (high pitched) sound that is so high that you cannot hear it, but it is, nonetheless, causing the tweeter to burn out. A high quality cable is much less likely to have a bad ground connection than a lower quality one and one blown tweeter can pay for a significant number of good cables.

3.0 Microphones

3.1 General Requirements (Para 3.1 is a extract of Telecom Manual Chapter 20)

3.1.1 Use of microphones

(a) Moving Coil Microphone -These are commonly used and have robust construction. Normally low impedance microphones are used as these permit the use of long microphone lines. These should have uni-directional characteristics which help in reducing acoustic feedback/howling specially in indoor sound system.

(b) Condenser Microphone -Due to smaller size condenser microphones are commonly used for lavallier application either as Tie Pin Type or as Neck Type. These
microphones require 1.5 V battery to power electret condenser cartridge. These microphones have higher sensitivity compared to moving coil microphones.

3.1.2 Sensitivity

Sensitivity of the microphone shall be of the order of -55dB relative to 0.0002 dyne/cm for an impedance of 50 ohms.

3.1.3 Frequency response

The microphone chosen should have uniform frequency response within ± 3.0 dB from 100 to 10000 Hz.

3.1.4 Connecting microphones

(a) Use of more than one microphone may be essential in large stages. In such cases, output from several microphones should be mixed in a mixing system and the common output fed to the amplifier, where the amplifier itself is capable of mixing individual microphone inputs, separate mixing system is not required.

(b) The microphone plugs and sockets should be of multi-contact (three or more) type and freely interchangeable.

3.1.5 Sitting of microphone and loudspeaker

(a) Sound distribution system, especially in a closed hall, has the risk of acoustic feedback from the loudspeaker to the microphone causing singing. Siting of microphones and loudspeakers should be such that there is good pick-up of speaker's voice without abnormal rise in bass and good distribution with uniform coverage without acoustic feed back. The microphone should be sited normally in an acoustic shadow.

(b) It is also desirable to create the illusion that sound is being heard directly. There should not be cases of sound from the loudspeaker, or reflected sound from the walls, reaching the audience after the sound from the speaker has reached directly.

(c) Microphone should be, as far as possible, behind the loudspeaker in order to minimize the acoustic feedback. The correct distance between microphone and source should be predetermined and arranged to be constant as far as possible. It is important to see that if the level of reverberant sound or the surrounding noise near the microphone is high, the distance between microphone and source shall be reduced. The sound source should be directly towards the microphone, as otherwise the high notes, which are highly directional, would not be satisfactorily picked up by the microphone and thereby the clarity of the speech sound reproduced by the system will be poor.

(d) In 25 KV AC electrified area, the microphone siting should aim at avoiding the electrostatic or electromagnetic induction either in the equipment or in the lead from the microphone to the amplifier.
3.1.6 Wind Shields

Microphones, when used outdoors, may have to be fitted with some means of protection against wind. However, it is desirable that performance of microphones should not be adversely affected by such wind shield.

3.2 Definition

A microphone is an example of a transducer, a device that changes information from one form to another. Sound information exists as patterns of air pressure; the microphone changes this information into patterns of electric current.

3.3 Types of microphones

3.3.1 Carbon Microphone

The carbon microphone is consisting of two metal plates separated by granules of carbon. One plate faces outward and acts as a diaphragm. When sound waves strike this plate, the pressure on the granules changes, which in turn changes the electrical resistance between the plates. (Higher pressure lowers the resistance as the granules are pushed closer together.) 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. One of the main disadvantages of the carbon microphone is that it has continuous high frequency hiss caused by the changing contact resistance between the carbon granules.

Fig No. 4: Carbon Microphone

3.3.2 Crystal Microphone

Crystals which demonstrate the piezoelectric effect produce voltages when they are deformed. The crystal microphone uses a thin strip of piezoelectric material attached to a diaphragm. The two sides of the crystal acquire opposite charges when the crystal is deflected by the diaphragm. The charges are proportional to the amount of deformation and disappear when the stress on the crystal disappears. Early crystal
microphones used Rochelle salt because of its high output, but it was sensitive to moisture and somewhat fragile. Later microphones used ceramic materials such as barium titanate and lead zirconate. The electric output of crystal microphones is comparatively large, but the frequency response is not comparable to a good dynamic microphone, so they are not serious contenders for the music market.

3.3.3 Dynamic Microphone

Dynamic microphones are versatile and ideal for general-purpose use. They use a simple design with few moving parts. When a magnet is moved near a coil of wire an electrical current is generated in the wire. Using this electromagnet principle, the dynamic microphone uses a wire coil and magnet to create the audio signal.

The diaphragm is attached to the coil. When the diaphragm vibrates in response to incoming sound waves, the coil moves backwards and forwards past the magnet. This creates a current in the coil which is channeled from the microphone along wires. A common configuration is shown below.
3.3.4 Condenser Microphone:

A capacitor has two plates with a voltage between them. In the condenser microphone, one of these plates is made of very light material and acts as the diaphragm. The diaphragm vibrates when struck by sound waves, changing the distance between the two plates and therefore changing the capacitance. Specifically, when the plates are closer together, capacitance increases and a charge current occurs. When the plates are further apart, capacitance decreases and a discharge current occurs.

The resulting audio signal is stronger signal than that from a dynamic. Condensers also tend to be more sensitive and responsive than dynamics, making them well-suited to capturing subtle nuances in a sound. They are not ideal for high-volume work, as their sensitivity makes them prone to distort. The required voltage across the capacitor is supplied either by a battery in the microphone or by external phantom power.

![Condenser Microphone Diagram]

*Fig. No. 7: Cross-Section of a Typical Condenser Microphone*

3.3.5 Ribbon Microphone

A ribbon (velocity) microphone is a type of microphone that uses a thin aluminum, duraluminum or nanofilm ribbon placed between the poles of a magnet to generate voltages by electromagnetic induction. Pressure waves cause the ribbon to vibrate in the magnetic field generating voltage corresponding to the particle velocity of the pressure wave.

Ribbon microphones are typically bidirectional, meaning they pick up sounds equally well from either side of the microphone. Designed to have a wide frequency range, good sensitivity, low distortion, and low internal noise. 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. Ribbon microphone is shown in figure No.8.
3.3.6 Wireless or Cordless Microphone

A wireless microphone is a microphone without a physical cable connecting it directly to the sound recording or amplifying equipment with which it is associated. It has a small, battery-powered radio transmitter in the microphone body, which transmits the audio signal from the microphone by radio waves to a nearby receiver unit, which recovers the audio. The other audio equipment is connected to the receiver unit by cable. Wireless microphones are widely used in the entertainment industry, television broadcasting, and public speaking to allow public speakers, interviewers, performers, and entertainers to move about freely while using a microphone to amplify their voices. These are Hand held and collar type as shown in figure 9.

Advantages
- Greater freedom of movement for the artist or speaker.
- Avoidance of cabling stressing problems common with wired microphones.
- Reduction of cable "trip hazards" in the performance space

Disadvantages
- Some wireless systems have a shorter range, while more expensive models can exceed that distance.
- Possible interference with or, more often, from other radio equipment or other radio microphones.
- Operation time is limited relative to battery life.
3.4 Specifications of microphones.

3.4.1 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.

3.4.2 Sensitivity:

It is the amount of voltage developed or generated by the microphone for an applied sound pressure at a test frequency of 1000 Hz. It is generally specified as mV/Microbar. One-microbar sound pressure is equal to 1 dyne/cm². It is also specified as mV/Pa where Pa is Pascal, which is equal to 10 microbars.

3.4.3 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.

3.4.4 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%.

3.4.5 Impedance:

It is the impedance offered by the microphone at 1000 c/s. There are low impedance and high impedance microphones. Low Impedance means less than 600 ohms High Impedance means more than 10K ohms.

3.4.6 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.

3.4.7 Cables and Connectors:

It specifies the type and length of the cable with a particular connector.
3.4.8 Front to Back Ratio:

It is specified in the case of unidirectional microphone, which gives the response of front sound and back sound. Generally it is 20 db.

3.4.9 Polar response:

It specifies the type of directivity pattern that microphone responds it is a graph of the microphones directional sensitivity. It specifies whether it is an omni directional, Bi-directional or Uni-directional.

3.5 Installation Practice:

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

- All microphones are delicate instruments; they must be carefully handled and never dropped, nor placed where there may be metal dust.
- To avoid hum pick-up and especially in case of high impedance microphones, locate as far as possible from electrical apparatus. Do not run microphone leads together with mains cable.
- For public address locate well away from preferably to rear of the loudspeakers to prevent acoustic feedback 'howl'.
- Ribbon microphones should be at least 10" from the speaker. One should not speak into the microphone and on no account it should be tested by blowing into the microphone.
- 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.
- Microphones must be protected from strong winds, otherwise 'roaring' noises will result. It is common practice to provide windscreens in such cases.
- A typical windscreen is shown below (Figure 10). 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.

![Fig. No. 10 Wind screen for Microphone](image-url)
3.6 Cleaning of Microphones

Regular cleaning of microphone will not only improve its performance, but is also good hygiene. Following are the simple yet effective techniques for cleaning microphones.

3.6.1 Dynamic Microphones

The best way to clean a microphone is to remove the grille. Most vocal microphone grilles simply unscrew. If the grille doesn't slide off easily, gently rock it back and forth while pulling it away from the cartridge. Do not pull sharply or with excessive force, since that could damage the cartridge or separate it from the microphone housing. Once the grille is removed, it can be thoroughly cleaned without damaging the microphone. Since most of the offensive material on the grille comes from the human body, plain water should be a sufficient cleanser. Adding a mild detergent (dishwashing liquid) to the water will act as a mild disinfectant and remove odors absorbed by the foam windscreen. To remove lipstick and other material stuck in the grille, use a toothbrush with soft bristles.

3.6.2 Condenser Microphones

Due to the more delicate nature of condenser microphones, never use water or any other liquid for cleaning purposes. Even a small amount of moisture may damage a condenser element. For microphones with removable grilles, the grille and foam windscreen may be washed as described above. Again, the grille and windscreen must be completely dry before reattaching it to the microphone. To clean a microphone with a permanently attached grille, use a dry, soft bristle toothbrush and gently scrub the grille. Keep the microphone upside down so that loosened particles fall away from it. Take care not to let stray bristles get caught in the grille. This technique also works well for lavaliere and miniature gooseneck microphones.

For condenser microphones that will be subject to harsh conditions, such as vocals and theater applications, it is advisable to use a removable external foam windscreen. This will protect the microphone from saliva and make-up, and can be removed and cleaned with soap and water after the performance. Remember; never get water near a condenser element.

3.6.3 Microphone Cord

Microphone cord is provided with each microphone. Length of this cord is 5 meter to 10 meter or as per our requirement. This cord is a three core cable connected with jack plug at one end and XLR plug at other end. Jack plug is inserted in amplifier’s microphone socket and XLR plug is connected with microphone.

Diagrams showing different parts of jack plug and XLR Plug are given in figure No. 11 & figure No.12.
Jack Plug

Jack plug is used at one end of the microphone cord. It is used for inserting in microphone input socket of amplifier. It is generally provided at the front panel of the amplifier. The jack plug has three parts Sleeve, Tip and Ring & the Microphone cord is a three core cable, Positive, Negative & Screen. The screen is connected with the sleeve, Positive is connected with Tip and Negative is connected with Ring of the jack plug as shown in following diagram:

![diagram of jack plug](image)

**Fig No. 11: Jack Plug**

**XLR PLUG**

It is Ahuja standard robust and relatively simple to install microphone connector. X denotes to Earth or Screen (Pin 1), L denotes to Live or Positive (Pin 2), R denotes to Return or Negative (Pin 3). Sketch of XLR plug is shown below:

![diagram of xlr plug](image)

**Fig. No. 12: XLR Plug**

4. **Loudspeaker**

4.1 **General Requirements (Para 4.1 is a extract of Telecom Manual Chapter 20)**

4.1.1 Criteria for determining the loudspeakers required -The number of loudspeakers, their location, height, direction and the power input to the loudspeakers installed will have to be decided with the object of maintaining the intensity of reproduced sound above the local prescribed noise level so that the masking effect of noise over the signal could be reduced considerably.
4.1.2 The loudspeakers used should have adequate power handling capacity and should normally be of high efficiency type.

4.1.3 Loudspeakers used for "A" Category reproduction should have effective frequency range of 100 to 10,000 Hz. (The response of the speaker system within the environment after installation should be considered as the effective frequency response). For this reproduction, directional type of loudspeakers (column) should be used.

The vertical directivity pattern of the system should be such as to feed the audience at uniform level, avoid harmful level, reverberant sound or echo, and feedback of energy to the microphones. In the horizontal plane, the directivity should be uniform across the width of the hall.

4.1.4 Column type loudspeakers

(a) Column loudspeakers are ideal for obtaining the vertical directivity pattern. The height of column and number of speakers in it determine the directivity. A wide range of high quality reproduction may be obtained by employing multiunit type, wherein the whole frequency range will be covered by two or three groups of speakers arranged in separate columns, but mounted close to each other and connected through a properly designed dividing network.

(b) The directivity pattern of such speakers should be such as to provide sufficient intelligibility at all points of the seated area and avoid feed back to microphone, dead spot and echo.

(c) For best results, the column loudspeakers shall be installed vertically at a height of 1.5 m above the platform level and inclined at an angle of 8 degree to 10 degree towards the ground.

4.1.5 For "B" Category reproduction, the loudspeaker should have useful response from 100 to 7,500 Hz. Cabinet/horn type loudspeakers should be adequate for such purposes.

4.1.6 Cone type loudspeakers with wooden/metal cabinets -Cone type loudspeakers of appropriate power output may be used in comparatively quiet covered areas like waiting rooms, retiring rooms, etc.

4.1.7 Horn type loudspeakers -Horn type loudspeakers are suited to open platform and large halls with high roofs. They shall be so placed and their size so chosen that their radiation may not be in opposition and also the reflections from the roof and walls are avoided. An electrical filter to cut off low frequencies may be used with the line matching transformer to avoid damage to the voice coil at low frequencies.
4.1.8 Connecting loudspeakers

(a) All the loudspeakers 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 announcers 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 changed over to a standby amplifier, if provided.

(c) When a number of loudspeakers are connected to the same output circuit, matching transformers shall be used with each loudspeaker so that it consumes the rated power.

(d) These transformers should have at least the minimum frequency characteristic required of the public address system. The power handling capacity of the transformer used with a loudspeaker should not be less than the power to be absorbed by the speaker. These should have several taps on primary and or secondary to give multiple turns ratio.

(e) These transformers enable the loudspeakers, through the selection of proper turns ratio, to take an input of predetermined value of audio load from the amplifier, at the same time, care being taken not to overload the loudspeaker. Where the constant voltage output line from the amplifier is used, the total wattage of loudspeaker load should not exceed the rated power of the amplifier.

(f) When a single loudspeaker unit is connected to the amplifying system, it's impedance should be matched to the source impedance so as to consume the rated power.

4.2 Definition

A loudspeaker (or "speaker") is an electro-acoustic transducer that produces sound in response to an electrical audio signal input. 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.
4.3. **Dynamic Loudspeaker:**

![Diagram of a dynamic loudspeaker]

*Fig. No. 13: Cross section & Construction of Dynamic Loudspeaker*

The most common type of driver, commonly called a dynamic loudspeaker. It has a light weight diaphragm, or cone, connected to a rigid basket, or frame, via a flexible suspension, commonly called a spider, that constrains a coil of fine tensile wire to move axially through a cylindrical magnetic gap.

When an electrical signal is applied to the voice coil, a magnetic field is created by the electric current in the voice coil, making it a variable electromagnet. The coil and the driver's magnetic system interact, generating a mechanical force that causes the coil (and thus, the attached cone) to move back and forth, thereby reproducing sound under the control of the applied electrical signal coming from the amplifier.

![Diagram of a dynamic loudspeaker]

*Fig. No. 14: Dynamic Loudspeaker*
4.4 Cabinet Loud Speaker

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 no 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.

4.5 Line Source or Column Speaker

Column Speakers use multiple speaker cones create a slim line column offering excellent vertical sound dispersion with a long 'throw', but limited horizontal coverage. For this reason, several column speakers can be mounted in a cluster and are often used around pillars for sound reinforcement.

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.
4.6. High Fidelity (Hi-Fi) Speaker

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.

4.7 Woofer

Woofer is designed to produce low frequency sounds, typically from around 40 hertz up to about a kilohertz or higher. The most common design for a woofer is the electro dynamic driver, which typically uses a stiff paper cone, driven by a voice coil which is surrounded by a magnetic field. The voice coil is attached by adhesives to the back of the speaker cone. The voice coil and magnet form a linear electric motor. When current flows through the voice coil, the coil moves in relation to the frame according to Fleming's left hand rule, causing the coil to push or pull on the driver cone in a piston-like way. The resulting motion of the cone creates sound waves as it moves in and out.
4.8 Tweeter

A tweeter is a loudspeaker designed to produce high audio frequencies, typically from around 2,000 Hz to 20,000 Hz (generally considered to be the upper limit of human hearing). Specialty tweeters can deliver high frequencies up to 100 kHz. Tweeter in a two speaker system re-produces frequencies from 1KHz 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.

Fig. No. 19 Tweeter & its exploded view

4.9. Crossover network:

Audio crossovers are a class of electronic filter used in audio applications. Most individual loudspeaker drivers are incapable of covering the entire audio spectrum from low frequencies to high frequencies with acceptable relative volume and lack of distortion so most hi-fi speaker systems use a combination of multiple loudspeakers drivers, each catering to a different frequency band. Crossovers split the audio signal into separate frequency bands that can be separately routed to loudspeakers optimized for those bands.

The specific purpose of crossover network is:

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

Fig. No. 20 Multiple unit loudspeakers
4.10 Horn Loud Speaker

A horn loudspeaker is a loudspeaker or loudspeaker element which uses a horn to increase the overall efficiency of the driving element, typically a diaphragm driven by an electromagnet. The horn itself is a passive component and does not amplify the sound from the driving element as such, but rather improves the coupling efficiency between the speaker driver and the air. The horn can be thought of as an "acoustic transformer" that provides impedance matching between the relatively dense diaphragm material and the air of low density. The result is greater acoustic output from a given driver.

Horns have been used to extend the low frequency limit of a speaker driver. When mated to a horn, a speaker driver is able to reproduce lower tones more strongly. The flare rate and the mouth size determine the low frequency limit. The throat size is more of a design choice. Horns have been known to extend the frequency range of a driver beyond five octaves.

![Fig. No. 21: (a) Horn Speaker](image1)
![Fig. No. 21: (b) Bull Horn Speaker](image2)

A horn facilitates the transfer of electrical energy into acoustical energy and, if properly designed will be so with a minimum of 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.

The reflex loudspeaker or bullhorn, a type of folded horn speaker used widely in public address systems as shown in figure 21(b). To reduce the size of the horn, the sound follows in zigzag path through exponentially-expanding concentric ducts in the central projection (b, c), emerging from the outer horn (d).
4.11 Specifications

Impedance:

It is the impedance offered by a loud speaker at 400 Hz. The impedance will be changed with the frequency.

Power handling capacity (PHC)

Term use to indicate the maximum volume of sound that the loud speaker will produce before it runs into distortion with maximum 5% tolerance. 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

At 1 watt power at a distance of 1 meter (SPL at 1W, 1M). Loud speaker manufacturers indicate the accurate sound pressure in DB SPL at 1 meter distance when 1 watt of 1Khz signal feed 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.

5.0 Amplifier

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

5.1 General Requirements (Para 5.1 is an extract of Telecom Manual Chapter 20)

5.1.1 Capacity of Amplifiers

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

5.1.2 Input

In addition to the required number of microphone input channels, the amplifier must have a tape recorder/ CD player input channel. It shall be possible to control the proportion of the levels of the signals mixed.
5.1.3 Sensitivity

As the input voltages required to be amplified may range from 0.5 mili volt to 1.5 V, the amplifying system should have sensitivity sufficient to operate directly from the lowest and highest input voltages to be met with.

5.1.4 Frequency response

The frequency response of the amplifiers should be within ± 3.0 dB from 100 to 10000 Hz for "A" Category reproduction and from 100 to 7500 Hz for "B" Category reproduction.

5.1.5 Matched impedance working

For matched impedance working, the output impedance of the amplifier should be such as to operate into the range of impedances presented by the load.

5.1.6 The output transformer of the amplifier should have impedance tapping of 4,8 & 16 ohms to enable operation with loudspeakers of these standard impedances. For constant voltage working, the transformer should be provided with 70 to 100 volts constant voltage tapings.

5.1.7 High power amplifiers should incorporate safeguard against excessive voltage or current rise in case of open circuit condition or short circuit conditions respectively, in output circuit.

5.1.8 Standby amplifiers

(a) Standby amplifiers shall be provided so that announcement is not held up due to defects in the working of amplifiers.

(b) Easy changeover arrangement for switching from the defective amplifiers to the standby amplifiers by the announcer without the aid of any technical staff is preferable.

(c) Provision should be available for easy localisation and rectification of faults in any part of the installation.

5.1.9 Installation

(a) 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. Access by unauthorised persons should be guarded against. Precautions should be taken to keep away dust from the equipment, especially if earth moving machines, concrete mixers etc. are working in the immediate vicinity of the accommodation provided.

(b) When the number of equipment is not large, they may be placed on a table and wired. The positioning of the equipment should be such that the lengths of the interconnecting cables are kept minimum for convenience.

(c) In case the number of equipment is large, it is desirable to mount them in racks of suitable dimensions. The racks may be of metal or wood and either in one piece having compartments or different sections of uniform width assembled together. Each compartment of section shall contain one item of equipment. The height of the
rack will depend on the number of equipment to be mounted and accommodation available, ensuring that all manual controls are within easy reach.

(d) Switches should be provided for isolating any faulty section of the equipment thereby facilitating operation and avoiding danger to the operating personnel. The arrangement made should enable the remaining part of the equipment to be available for use.

(e) The patch cords if used should be tested and neatly arranged to avoid obstruction and should be easily identifiable. Necessary safety measure should be adopted to avoid accidental contacts with high voltage points in the rack.

(f) Insulation required in 25 kV ac electrified area - The amplifier along with the cable and loudspeakers shall be such as to withstand a dielectric strength test voltage of 1000 AC rms for two seconds, when applied between the terminals of the speaker and the body.

5.1.10 Power Supply

(a) It shall be ensured that reliable mains power supply is available near the proposed location of the announcing equipment.

(b) The installation should be normally operated from 230 V single phase 50 Hz AC mains supply.

(c) A voltage regulating device will have to be provided, if the regulation of the power supply is poorer than ±5%.

(d) When only DC supply is available, if necessary, an inverter of required capacity should be provided to convert DC supply into AC supply.

(e) If no mains supply is available, petrol or oil engine driven generating set of required capacity giving 230 volts single phase 50 Hz, AC should be used. Such a generating set should be located away from central equipment and microphones and preferably in another building at some distance from the hall having sound distribution installation to avoid noise (Electrical & Mechanical) produced by the generating system.

(f) All amplifiers should preferably be capable to operate on 12 V/24 VDC Car Battery besides on 230 V, 50 Hz AC supply.

5.2 Controls & Features

5.2.1 Front Panel Controls

A photograph of typical PA Amplifier Ahuja Model SSA-350 & a diagram showing front panel controls is shown in figure 22 (a & b).

![Fig. No. 22(a): Amplifier-Ahuja, Model SSA 350](image-url)
### Legends

<table>
<thead>
<tr>
<th>No.</th>
<th>Name of Control</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1, 2, 3, 4, 5, 7 &amp; 8</td>
<td>Mic Input Jack Sockets</td>
<td>For accepting unbalanced signal from a low impedance microphone.</td>
</tr>
<tr>
<td>6, 9, 21, 22, 23, 24, &amp; 25</td>
<td>Mic Volume Control Switch</td>
<td>For adjusting the volume level of individual mic</td>
</tr>
<tr>
<td>19 &amp; 20</td>
<td>AUX Volume Control Switch</td>
<td>For controlling volume of auxiliary circuits.</td>
</tr>
<tr>
<td>10</td>
<td>Box Speaker/Driver Unit Selector Switch</td>
<td>For selecting the type of loud speakers being used</td>
</tr>
<tr>
<td>11</td>
<td>Bass Control Switch</td>
<td>For increasing or decreasing the signal level of low frequencies</td>
</tr>
<tr>
<td>12</td>
<td>Treble Control Switch</td>
<td>For increasing or decreasing the signal level of high frequencies</td>
</tr>
<tr>
<td>13</td>
<td>Power Switch</td>
<td>For make the amplifier ON or OFF</td>
</tr>
<tr>
<td>14</td>
<td>Power LED</td>
<td>LED glows when the Amplifier is switched ON</td>
</tr>
<tr>
<td>15</td>
<td>Overload LED</td>
<td>LED glows when the circuit protector trip</td>
</tr>
<tr>
<td>16</td>
<td>Reset Button</td>
<td>This button pops out when the circuit protector trips. Rectify the cause and press the ( \text{RESET} ) button for resetting normal operation of the amplifier</td>
</tr>
<tr>
<td>17</td>
<td>LED Array</td>
<td>These LEDs indicates the output level of the amplifier</td>
</tr>
<tr>
<td>18</td>
<td>Master Switch</td>
<td>For adjustment of overall volume level of the Amplifier</td>
</tr>
</tbody>
</table>

### 5.2.2 Rear Panel Controls

A diagram showing rear panel control of PA Amplifier Model Ahuja SSA-350 is given below in figure 23:
Fig. No. 23: Rear Panel controls of Amplifier

<table>
<thead>
<tr>
<th>Legends</th>
<th>Name of Control</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>26</td>
<td>AC Mains Cable with Plug</td>
<td>For connecting Amplifier with 230 Volt AC Supply</td>
</tr>
<tr>
<td>27</td>
<td>Battery Terminal Block</td>
<td>For connecting DC battery as standby power source</td>
</tr>
<tr>
<td>28</td>
<td>Speaker Terminal Block (70V/100V)</td>
<td>For connecting speakers with 100V/70V line matching transformer</td>
</tr>
<tr>
<td>29</td>
<td>Speaker Terminal Block (4Ω /8Ω)</td>
<td>For connecting low impedance speakers</td>
</tr>
<tr>
<td>30</td>
<td>Line Input Jack Socket</td>
<td>For connecting inputs such as a CD player. Also for connecting an external mixer to enhance the number of inputs.</td>
</tr>
<tr>
<td>31,32</td>
<td>AUX Input Jack Socket</td>
<td>For accepting an unbalanced signal from an auxiliary source like a tuner, cassette player, echo or audio mixer etc</td>
</tr>
<tr>
<td>33</td>
<td>Preamplifier Output Jack Socket</td>
<td>For connecting to the auxiliary input of another amplifier or a cassette recorder for recording purpose</td>
</tr>
<tr>
<td>34</td>
<td>Line Output Jack Socket</td>
<td>For connecting to a booster amplifier to obtain combined higher power output</td>
</tr>
<tr>
<td>35</td>
<td>Earth Terminal</td>
<td>For connecting electric earth</td>
</tr>
<tr>
<td>36</td>
<td>Fuse</td>
<td>AC main fuse</td>
</tr>
</tbody>
</table>

5.3 Interconnections

- The amplifier can be placed as a table top unit. The is so installed that its location or position does not interfere with its proper ventilation.
- Amplifier must be powered through an AC earthed mains outlet.
- All connections must only be carried out or change after switching off the amplifier.
- To avoid loud switching noise, always switch ON the amplifier after switched ON the connected audio input with it.
- After operation first switched OFF the amplifier before the other units.
- Diagrams of a typical amplifier connections on front and rear panel are shown below in figure 24 & figure 25:
5.3 Connections of Front Panel

Interconnections on front panel of a typical amplifier, make Ahuja, Model SSA-350 is shown below in figure 24.

Jack plugs of microphones of various audio sources are inserted in microphone sockets. The Volume level of each microphone’s audio can be increased or decreased by the respected microphone volume controls. The over all volume of all microphones can be control by master control.

Fig. No. 24: Connections on front Panel of the Amplifier
5.4 Connections of Rear Panel

Interconnections of rear panel of a typical amplifier make-Ahuja, model SSA-350 is shown below in figure 25.

Fig. No. 25: Connections at Rear Panel of the Amplifier
5.6 Speaker Connection

Installed sound systems commonly use either a direct connection (also called low impedance) or constant voltage (also called high impedance) amplifier/speaker interface.

5.6.1 Low Impedance

- Box type speakers can be connected to COM & 4Ω or 8 Ω terminal strip directly according the impedance of the speaker.
- The box speaker / Driver unit switch must be kept at Box speaker position. If by mistake the switch remains in Driver unit position the quality of sound will not be rich and natural.
- No driver unit / Horn speaker / Column speaker (with 100V LMT) should be connected at this terminal strip.

![Diagram of terminal strip for connecting load in impedance matching](image)

*Fig. No. 26: Terminal strip for connecting load in impedance matching*

5.6.2 Connecting Four Speakers of 8Ω Impedance

Four speakers of 100 watt each are connected in parallel-series combination as shown. Two groups of two speakers are connected in parallel and then these groups are connected in series. The resulting impedance would be 8 Ω. Thus they can be connected to the 8 Ω tapping, as shown in figure 27.

![Diagram of connecting speakers in impedance matching](image)

*Resultant Impedance = (8 Ohm/2) x 2 = 8 Ohm*

*Fig. No. 27: Connecting speakers in impedance matching*
5.6.3 Connecting two Speakers of 8Ω Impedance

Two speakers of 200 watt shall be wired in parallel combination as shown. The resulting impedance will be 4 Ω. This parallel combination of the speakers can be connected on 4 Ω tapping.

![Diagram of connecting two speakers in parallel](image)

\[ \text{Resultant impedance} = \frac{(8 \text{ Ohm})}{2} = 4 \text{ Ohm} \]

*Fig. No. 28: Connecting speakers in impedance matching*

5.6.4 High Impedance

- Only Driver unit/Horn/Column speakers with 100V line matching transformer are to be connected to Com-70V/100V terminal strip.
- The box speaker/Driver unit switch of amplifier shall be kept at Driver Unit position.
- If any speaker is connected on Com-70V/100V terminal strip than no box speaker shall be connected on Com-4Ω/8 Ω terminal strip.

![Terminal strip for connecting speakers in Voltage Matching](image)

*Fig. No. 29: Terminal strip for connecting speakers in Voltage Matching*

5.6.5 Connecting 35 Driver Unit with 100V LMT

35 driver units with 100V line matching Transformer are connected in parallel as shown. It shall be noticed that the total wattage shall not exceed by the amplifier wattage capacity. If amplifier is 350 watt than speaker tapping of 10 watt can be selected.

![Diagram of connecting 35 driver units in parallel](image)

*Fig. No. 30: Connecting speakers in Voltage Matching*
5.6.6 Connecting combination of Driver Unit and Column speakers with 100V LMT

25 driver units with LMT at 10 watt and 20 column speakers with 100 LMT at 5 watt can be connected with 350 watt amplifier as shown in figure given below:

![Figure No.31: Connecting speakers in Voltage Matching](image)

5.7 Use of 70V Line

A loudspeaker / Driver unit with its LMT adjusted to 15W tap, when connected to Amplifier’s COM & 100V terminal, it will draw 15W from the amplifier but if same is connected to COM & 70V it will draw only half power i.e. 7.5W. The 70V line tapping of amplifier is useful where large number of speaker/driver unit are to be installed for more even distribution of line.

5.8 Correct Phasing of Loudspeakers

When two or more speakers/units are installed in the same area and are facing to same direction, it is essential that their cone/diaphragm act in unison. Otherwise the sound levels of both speakers will be cancelled out. To avoid any mistake, the terminals of box speakers and the driver unit are marked ‘+’ & ‘-’. Always connect the Com tape of amplifier to ‘-’ of speaker and 4 Ω/8 Ω tape of the amplifier to ‘+’ of the speaker. In case of LMTs the Com of all the LMTs shall be connected to the Com of the amplifier and power tape to 100V/70V line.

![Figure No.32: Driver Unit](image)
5.9 Connecting Amplifier 350 watt & Booster Amplifier 500Watt

- Connect microphones and other program sources to the input jacks of 350 Watt amplifier.
- Connect the Pre-output jack of this amplifier to the input jack of Booster amplifier 500Watt. The sensitivity switch of Booster shall be at 200mV. Now system will behave as 350+500=850 watt.
- LS connections to both the amplifier shall be independently. Each Amplifier shall be connected either box speakers or speakers using LMT but never to both together.
- Setting of both amplifiers shall be according to type of load speakers connected.
- If speakers using LMT are connected the total power shall not be exceed 500 watt for Booster & 350 watt for Amplifier.
- Bass & Treble controls of each amplifier can be adjusted to give optimum tonal quality of sound.
- This kind of system is ideal where both high and low impedance speakers are being used.
- The connection diagram is shown in the figure 33 below:

![Diagram](image)

**Fig. No.33: Master Amplifier connected with one Booster Amplifiers**
5.10 Connecting Master Amplifier 350 Watt and four Booster Amplifiers 500 Watt

- Connect microphones and other sources to the input jacks of Master Amplifier.

- Connect Line-Out of the Master Amplifier to the input jack of the first Booster A using patch-cord with ¼ inch phone plugs at both ends. The sensitivity switch shall be towards 1V.

- Connect the output jack of Booster A to the input jack of second Booster B. The sensitivity switch of this second Booster B shall be at 1V.

- Connect the third and forth Boosters (Booster C & Booster D) in the same manners. A maximum of four boosters shall be connected in the way otherwise the line output of the Master Amplifier will got loaded.

- Now the inputs connected to the master amplifier will feed all the five amplifiers creating 350+500+500+500+500=2350 Watt output system.

- Loudspeaker connections to each of the five amplifiers shall be done independantly. Each amplifier can be connected to either box type speakers or to speakers using LMT but never to both together.

- Speaker system impedance shall be matched to the output impedance of the amplifier and thus shall be connected to the corresponding tape of the amplifier.

- The box speakers / driver unit switch should be as per type of load connected.

- When speakers with 100V LMT are used, total power drawn should not exceed 500 watt in case of any of the booster amplifiers & 350 watt in case of master amplifier respectively.

- During operation of the system any adjustment in the total quality of the sound if required can be made from the Master Amplifier.

- The connection diagram is shown in figure 34.
Fig No. 34: Master Amplifier connected with 4 Booster Amplifiers
6. **Audio Mixer Pre-Amplifier**

The main amplifier system has limitation of accommodating more number of input devices therefore there is a 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 to the main amplifier section. 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.

In this chapter we will describe a typical PA Audio Mixer Amplifier, make- Ahuja Model MX-15. The front panel of mixer is given below in figure 35:

6.1 **Front Panel Controls & Features**

![Fig No. 34: Front Panel Controls](image)

<table>
<thead>
<tr>
<th>Legend</th>
<th>Name of Control</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>MIC-1 Volume Control</td>
<td>For increasing or decreasing the volume of microphone-1. Similar volume controls are also provided for MIC-2 to MIC-9 input channels</td>
</tr>
<tr>
<td>2.</td>
<td>MIC-1 ON/OFF Switch with LED</td>
<td>Switch to make OFF or ON the MIC-1 channel. Switches with similar functions are also provided for MIC-2 to MIC-9 input channels.</td>
</tr>
<tr>
<td>3.</td>
<td>MIC-12/AUX-3 Volume Control</td>
<td>For controlling the volume of microphone 12 / auxiliary-3. Similar volume controls are also provided for MIC-10/AUX-1 and MIC-11/AUX-2.</td>
</tr>
<tr>
<td>4.</td>
<td>MIC-12/AUX-3 Switch with Bi-colour LED indication</td>
<td>For selecting either MIC-12 or AUX-3. Similar switches with bi-colour LED are also provided for MIC-10 / AUX-1 &amp; MIC-11/AUX-2</td>
</tr>
<tr>
<td>5.</td>
<td>BASS Control</td>
<td>This control boosts or cuts the low frequencies. Musical instruments like Tabla, Drum respond to boost the low freq. when switch move from 0 to 10 positions the low freq are reduced.</td>
</tr>
<tr>
<td>6.</td>
<td>TREBLE Control</td>
<td>It boosts or cuts the higher frequencies. It improves the brilliance of sound for string instruments like Violin, Sarangi, Sitar and to some extent human voice.</td>
</tr>
<tr>
<td>7.</td>
<td>LED Array</td>
<td>The 10 numbers of LED array calibrated in dB indicates the Line Output level.</td>
</tr>
<tr>
<td>8.</td>
<td>Power Switch</td>
<td>For Power make ON or OFF.</td>
</tr>
<tr>
<td>9.</td>
<td>Power ON LED</td>
<td>It glows when the set is switched ON.</td>
</tr>
<tr>
<td>10.</td>
<td>Master Control</td>
<td>All MICs and AUXs input signals are mixable and these mixed signals are finally controlled by the Master Control for overall opening of the volume from the mixer. For best performance it shall be kept at 6 to 8 position.</td>
</tr>
</tbody>
</table>
6.2 Rear Panel Controls & Features

Various rear panel controls of a typical mixer make-Ahuja, model MX-15 is given in figure 35 below:

![Rear Panel Controls Diagram]

**Fig No. 35: Rear Panel Controls**

<table>
<thead>
<tr>
<th>Legend</th>
<th>Name of Control</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>11.</td>
<td>Battery Terminal Strip</td>
<td>12V car battery is connected on this terminal strip. Battery operation is automatic in case of AC failure</td>
</tr>
<tr>
<td>12.</td>
<td>AUX-1 &amp; AUX-2 Input Jacks</td>
<td>For connecting AUX input sources such as a cassette player, additional mixer etc.</td>
</tr>
<tr>
<td>13.</td>
<td>AUX-3 (CD/TAPE) Stereo Input</td>
<td>For connecting higher output level sources such as a CD player, Stereo cassette player, key board etc.</td>
</tr>
<tr>
<td>14.</td>
<td>MIC-1 to MIC-12 Input Jacks</td>
<td>These jacks are for connecting Low impedance microphones.</td>
</tr>
<tr>
<td>15.</td>
<td>Line Output Jacks (1V)</td>
<td>These four jacks are connected in such a way that they do not load each other. This output is for connecting the Mixer to the Line input of an amplifier.</td>
</tr>
<tr>
<td>16.</td>
<td>Pre Output Jacks (200mV)</td>
<td>These two jacks are for connecting AUX input of any amplifier or cassette recorder for recording the program.</td>
</tr>
<tr>
<td>17.</td>
<td>Headphones Output</td>
<td>Audio Output for headphone monitoring</td>
</tr>
<tr>
<td>18.</td>
<td>Earth Terminal</td>
<td>For connecting electrical earth</td>
</tr>
<tr>
<td>19.</td>
<td>AC Mains Cable</td>
<td>For connecting 230 V0lt AC to equipment</td>
</tr>
<tr>
<td>20.</td>
<td>AC Fuse</td>
<td>For securing the equipment by higher input AC voltage</td>
</tr>
</tbody>
</table>
6.3 Typical Applications

6.3.1 Open Conference / Convention

Fig No. 36: Use of mixer in open conference

6.3.2 Musical Program

Fig No. 37: Use of mixer in musical programme
7. Conference System

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

7.1 Chairman unit

![Chairman Unit Diagram]

This unit specially designed for chairperson. It consists of built-in loudspeaker and highly sensitive electrets condenser microphone mounted on flexible gooseneck arrangement. The various parts/controls of the unit as shown above are given below:

<table>
<thead>
<tr>
<th>Legend</th>
<th>Part/Control</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Ring LED</td>
<td>It glows when pressing TALK or PRIORITY Switch</td>
</tr>
<tr>
<td>2</td>
<td>Microphone</td>
<td>Transmit voice when speak after pressing TALK key</td>
</tr>
<tr>
<td>3</td>
<td>Level LED</td>
<td>It indicates the speech level through the microphone</td>
</tr>
<tr>
<td>4</td>
<td>Priority LED</td>
<td>It glows when mic switched ON by pressing priority switch</td>
</tr>
<tr>
<td>5</td>
<td>Priority SW</td>
<td>By pressing it, chairman can suppress voice of all delegate units</td>
</tr>
<tr>
<td>6</td>
<td>Talk SW</td>
<td>It makes microphone ON by pressing once. Mic switched off on repressing and built-in speaker gets active</td>
</tr>
<tr>
<td>7</td>
<td>Talk LED</td>
<td>It glows when pressing talk switch.</td>
</tr>
<tr>
<td>8</td>
<td>Headphone Vol.Control</td>
<td>For adjusting the output level from the headphone jack socket</td>
</tr>
<tr>
<td>9</td>
<td>Headphone Jack Socket</td>
<td>For connecting stereo headphone plug</td>
</tr>
<tr>
<td>10</td>
<td>8 pin line connector</td>
<td>For connecting unit to other units through 8 core shielded cable</td>
</tr>
<tr>
<td>11</td>
<td>Vol. Control</td>
<td>For adjusting the output level of built-in speaker</td>
</tr>
<tr>
<td>12</td>
<td>Speaker</td>
<td>Built-in speaker for receiving speech of other units</td>
</tr>
</tbody>
</table>
7.2 Delegate units:

These units are similar to chairman unit with the exception of the priority switch not being provided. Diagram is shown below in figure 39:

Fig No. 39: Delegate Unit

7.3 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 No. 40: Secretary Unit

<table>
<thead>
<tr>
<th>Legend</th>
<th>Name and Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Record/Playback Connector: It is a 4 pin male socket for connecting to the ‘LINE IN’ (Record) and ‘LINE OUT’ (Play back) of a cassette recorder. A 4-pin female plug suitable for this socket is supplied along with this unit.</td>
</tr>
<tr>
<td>2</td>
<td>Speaker Volume Control: For adjusting the output level of the built in speaker</td>
</tr>
</tbody>
</table>
7.4 Conference Expansion Unit

![Conference Expansion Unit Diagram]

**Fig No. 41: Conference Expansion Unit (Front & Rear Controls)**

<table>
<thead>
<tr>
<th>Legend</th>
<th>Control / Part</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Power LED</td>
<td>It glows when the expansion unit is switched ON</td>
</tr>
<tr>
<td>2</td>
<td>Power Switch</td>
<td>For Switched ON the unit.</td>
</tr>
<tr>
<td>3</td>
<td>Battery Terminal Block</td>
<td>For connecting stand by batteries.</td>
</tr>
<tr>
<td>4</td>
<td>IEC main Socket with fuse</td>
<td>For connecting 23V AC Supply.</td>
</tr>
<tr>
<td>5</td>
<td>Earth Terminal</td>
<td>For connecting electric earth</td>
</tr>
<tr>
<td>6 &amp; 7</td>
<td>8-Pin Line Connector</td>
<td>For connecting the conference units in a closed loop</td>
</tr>
</tbody>
</table>

7.5 Cables & Connectors

**Connection Cable CC-46**

Connection cable terminated with 8-pin female plugs on both end

6.3mm (1/4 inch) Stereo Phone plug for Send/Return application

**Expansion Cable CC-47**

Extension cable terminated with 8-pin female at one end and 8-pin male plug at other end

[As viewed from soldering side]

8-pin Female Plug:
- Pin 1: +Ve Supply
- Pin 2: -Ve Supply
- Pin 3: Mic Auto Off
- Pin 4: Signal Out

8-pin Male Socket:
- Pin 5: Signal Out
- Pin 6: Signal IN
- Pin 7: Signal IN
- Pin 8: Priority
7.6 Central Amplifier:

It is provided for connecting conference system consisting of chairman, secretary and delegate units. It operates on AC mains and standby DC voltage and designed for both tabletop & rack mounting. A photograph of typical central amplifier model Ahuja CM-4000 is shown in figure 42. It has a 50W PA amplifier for sound reinforcement.

Fig No. 42: Central Amplifier

7.6.1 Front Panel Control:

Controls on front panel of Central Amplifier are shown below:

Fig No. 43: Front Panel Controls of Central Amplifier

<table>
<thead>
<tr>
<th>Legends</th>
<th>Part Name</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1, 2, &amp; 3</td>
<td>MIC Volume Control</td>
<td>For increasing and decreasing the volume of individual microphone.</td>
</tr>
<tr>
<td>4</td>
<td>AUX Volume Control</td>
<td>For increasing and decreasing the volume of Auxiliary Audio</td>
</tr>
<tr>
<td>5</td>
<td>BASS Control</td>
<td>For attenuating the signal level of low frequencies</td>
</tr>
<tr>
<td>6</td>
<td>TREBLE Control</td>
<td>For attenuating the signal level of high frequencies</td>
</tr>
<tr>
<td>7</td>
<td>Master Switch</td>
<td>For adjustment of the overall volume of the Amplifier</td>
</tr>
<tr>
<td>8</td>
<td>LINE Switch</td>
<td>For adjustment of the overall volume level from the conference units.</td>
</tr>
<tr>
<td>9</td>
<td>LED Array</td>
<td>Indicates the output level of the Amplifier</td>
</tr>
<tr>
<td>10</td>
<td>Power LED</td>
<td>This LED glows when amplifier tuned ON</td>
</tr>
<tr>
<td>11</td>
<td>Power Switch</td>
<td>For switched ON &amp; switched Off the Amplifier</td>
</tr>
</tbody>
</table>
### 7.6.2 Rear Panel Controls

Controls on rear panel of Central Amplifier are shown below:

![Fig No. 44: Rear Panel Controls of Central Amplifier](image)

<table>
<thead>
<tr>
<th>Legends</th>
<th>Part Name</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>12</td>
<td>Battery Terminal Block</td>
<td>For connecting Batteries + &amp; - terminal as standby power source</td>
</tr>
<tr>
<td>13</td>
<td>Main Socket with Fuse</td>
<td>For connecting 230 V AC supply through 3 pin plug cable</td>
</tr>
<tr>
<td>14</td>
<td>Earth Terminal</td>
<td>For connecting electric earth</td>
</tr>
<tr>
<td>15</td>
<td>Speaker Terminal Block (70V / 100V)</td>
<td>For connecting speakers with 100V/70V line matching transformer</td>
</tr>
<tr>
<td>16</td>
<td>8-pin Line Connectors</td>
<td>For connecting conference units in a closed loop</td>
</tr>
<tr>
<td>17</td>
<td>Speaker Terminal Block (4/8/16 ohm)</td>
<td>For connecting low impedance loud speakers of 4 ohm/ 8 ohm/ 16 ohm</td>
</tr>
<tr>
<td>18</td>
<td>Auto MIC Switch OFF</td>
<td>For selecting the facility of automatic “Switch OFF” for delegate microphones.</td>
</tr>
<tr>
<td>19</td>
<td>Send/Return stereo jack Socket</td>
<td>For connecting an external signal processor like a feedback destroyer, compressor limiter etc</td>
</tr>
<tr>
<td>20</td>
<td>LINE output jack socket</td>
<td>For connecting to a booster amplifier to obtain combined higher output</td>
</tr>
<tr>
<td>21</td>
<td>PRE-AMP/ RECORD output RCA Socket</td>
<td>For connecting to the AUX input of another amplifier or a cassette recorder for recording purpose.</td>
</tr>
<tr>
<td>22</td>
<td>PRE-AMP/ RECORD output jack Socket</td>
<td>For connecting PA amplifier</td>
</tr>
<tr>
<td>23</td>
<td>AUX Input Jack Socket</td>
<td>For accepting an unbalanced signal from an auxiliary source like a tuner, cassette player, Echo or Audio Mixer etc</td>
</tr>
<tr>
<td>24</td>
<td>MIC-3 Input Jack Socket</td>
<td>For accepting unbalanced signal from a low impedance microphone</td>
</tr>
<tr>
<td>25</td>
<td>MIC-2 Input Jack Socket</td>
<td>For accepting unbalanced signal from a low impedance microphone</td>
</tr>
<tr>
<td>26</td>
<td>MIC-1 Input COMBO Socket</td>
<td>For accepting balanced signal from a low impedance microphone through a male XLR plug as well as a quarter inch stereo plug</td>
</tr>
</tbody>
</table>
7.7 Interconnections

Figure 45 given below shows the various connections through central amplifier.
7.8 Typical Application

An example of a typical small and medium size venue such as Conference room, Meeting room, and Board room is shown below:

![Diagram of Connections in Conference Room](image)

*Fig No. 46: Connections in Conference Room*

7.9 Special Features

Auto Microphone Off

If a delegate unit keeps ON without making speech in to it, the microphone is automatically got switched off after approximately 75 seconds. This function can be disabled by placing Auto MIC Switch Off Switch at rear panel of central amplifier in OFF position.

Priority Function

A priority switch is provided on the chairman unit for seeking attention of all delegates during conference. When priority switch is pressed, all active microphones get muted and a single tone chime is relayed through all the in built speakers to get attention. In this period only chairman can speak.

Built-in PA Amplifier

In certain situations additional sound reinforcement through external speakers may be required. The central amplifier has been provided with a built-in 59W PA amplifier for powering these external speakers.
8. **Maintenance** ((Para 8 is a extract of Telecom Manual Chapter 20))

8.1 General conditions of wiring and components of the entire system to be checked once every year.

8.2 The frequency response and the noise level of the amplifier shall be checked annually and relevant parameters are to be recorded in tables as shown below.

<table>
<thead>
<tr>
<th>Sr.</th>
<th>Input Signal Level</th>
<th>Frequency(Hz)</th>
<th>Output Voltage</th>
<th>Load Resistance</th>
<th>SPL</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.5 mV &amp; 1.5 mV</td>
<td>100Hz</td>
<td>500Hz</td>
<td>1000 Hz</td>
<td>1500 Hz</td>
<td>2000 Hz</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sr.</th>
<th>Frequency(Hz)</th>
<th>Input Signal Level</th>
<th>SPL in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>1000 Hz</td>
<td>.5 mV</td>
<td>100 mV</td>
<td>500 mV</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sr.</th>
<th>Location Number</th>
<th>Sound Pressure Level in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sr.</th>
<th>Location No. as Test III</th>
<th>Sound Pressure Level in dB</th>
<th>Noise Level in dB</th>
<th>S/N Ratio</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sr.</th>
<th>Position of Measurement</th>
<th>SPL With Tone ON</th>
<th>Time Taken to decay the level by 60 dB</th>
</tr>
</thead>
</table>
8.3  The earthing arrangement shall be maintained properly and inspected once in a quarter. Alternating current induction hum at 50 Hz and its harmonics also should be checked.

8.4  Inspection and Testing

8.4.1  The completed permanent installation should be inspected and tested by the ADSTE/DSTE to ensure that the work is being carried out in a satisfactory manner and that the materials and components used conform with the standard practice. Temporary installation may be similarly tested and inspected by a senior inspector.

8.4.2  Routine inspection of the installation shall be carried out at intervals in accordance with the manufacturer’s instructions or as specified from time to time.

8.4.3  A log book shall be kept in which details of all routine attention, faults and tests should be carefully recorded for scrutiny.

8.5  Selection of Announcers

8.5.1  Choice of announcers

When DSTE/ADSTE assists in the choice of announcers, the following essential characteristics should be sought for in the prospective announcers.

(a) They should be possess a pleasing voice so as to reflect their helpful, polite and friendly attitude.
(b) They should be free of any defect or impediment in speech, including marked accents.
(c) They should be able to speak habitually with smoothness of diction and clarity of pronunciation.

8.5.2  Quality of Announcements

The following precautions will enhance the quality of announcements:

(a) Keeping a distance of 4" to 6" from the microphone and maintaining it.
(b) Checking up the correct volume range required.
(c) Speaking slowly and deliberately and preferably with a slight pause between words.
(d) Being particularly careful in the pronunciation of names. Names which can not be readily understood or which are inherently difficult to pronounce, may be spelt.
(e) Avoiding undue raising of voice. A relatively quiet, natural conversational tone is best.
(f) Modulate the voice according to the meaning -avoid monotones.
(g) Avoiding a sing-song style of delivery (usually characterised by a nasal tone of voice).
(h) Being careful about the use of words containing sibilants, especially if the announcer’s voice is of such a type as to emphasis unduly hissing or lisping tendencies.
(i) Avoid coughing, sneezing or clearing of throat while the sound distribution system is kept energised.
(j) Avoid the habit of blowing into the microphone.
(k) Avoid chewing of "pan" or tobacco while making announcements.
(l) Lay emphasis on key/important words and operating part of sentences.

8.6 Room for Announcer

8.6.1 Requirements for the announcer’s room

(a) The announcer should be seated in an enclosure having good acoustic characteristics, i.e. acceptable reverberation time without any specific acoustic defects.
(b) The ambient noise level in the enclosure should not be more than 40 dB referred to 0.0002 dyne per cm2.
(c) Air-conditioning may be extended to this enclosure, if any nearby room is air-conditioned, otherwise a noiseless exhaust fan shall be provided.
(d) The window panes should preferably be of double glass.
(e) Minimum size of the room shall be 2.5 m X 3.5 m.
(f) As far as possible, the room should be made dustproof, else, the dust should be absorbed and not swept.
(g) The announcer shall be suitably connected by telephone with the section controllers at big junction stations. Such telephones shall have only visual indicators for calling the announcer's attention.
(h) Where train describing system exists, it may be extended to the announcer also, if necessary.

9. Wiring and Cabling (As per telecom Manual Chapter 20)

9.1 Microphone Cables

(a) These 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 copper braiding should be sheathed with an insulated covering. The microphone cables shall be isolated from power, loudspeaker and telephone cables.
(b) Joints in the cables should be avoided as far as possible.
(c) The plugs and sockets used for microphones cables should have strong self-cleaning contacts so as to eliminate noise and they shall be non-reversible and have sufficient number of pins to connect not only the main conductors but also the cable shields.
(d) 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 damage due to any heavy object falling on them and cutting them.
(e) In 25 kV ac electrified areas, cables with their shields earthed must be used, if electromagnetic induction is anticipated.
(f) The plugs and sockets for loudspeaker connections should be of a type that cannot be easily or accidentally inserted in electric or power circuits.
(g) The speaker cable should be twin core rubber or PVC insulated lead-covered cables. These should be rated for 250 volts insulation and should be isolated from microphone cable and power cable.

9.2 Distribution and Connecting cables

The cables chosen for distribution and connections should be such that the line losses do not exceed the values specified in table given below:

**Loudspeaker Cable Sizes And Lengths For Specified Line Losses**

**Low Impedance Lines = 15% Power loss.**

<table>
<thead>
<tr>
<th>Wire Size (mm)</th>
<th>2 Meters</th>
<th>4 Meters</th>
<th>8 Meters</th>
<th>16 Meters</th>
<th>32 Meters</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.06</td>
<td>30</td>
<td>60</td>
<td>120</td>
<td>240</td>
<td>480</td>
</tr>
<tr>
<td>1.60</td>
<td>20</td>
<td>40</td>
<td>80</td>
<td>160</td>
<td>320</td>
</tr>
<tr>
<td>1.32</td>
<td>12.5</td>
<td>25</td>
<td>50</td>
<td>100</td>
<td>200</td>
</tr>
<tr>
<td>1.00</td>
<td>7.5</td>
<td>15</td>
<td>30</td>
<td>60</td>
<td>120</td>
</tr>
<tr>
<td>0.80</td>
<td>4.5</td>
<td>9</td>
<td>18</td>
<td>36</td>
<td>72</td>
</tr>
</tbody>
</table>

**High Impedance Lines = 5% power loss.**

<table>
<thead>
<tr>
<th>Wire Size (mm)</th>
<th>100 Meters</th>
<th>250 Meters</th>
<th>500 Meters</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.60</td>
<td>300</td>
<td>750</td>
<td>1500</td>
</tr>
<tr>
<td>1.32</td>
<td>200</td>
<td>500</td>
<td>1000</td>
</tr>
<tr>
<td>1.00</td>
<td>120</td>
<td>300</td>
<td>600</td>
</tr>
<tr>
<td>0.80</td>
<td>75</td>
<td>187.5</td>
<td>375</td>
</tr>
<tr>
<td>0.63</td>
<td>50</td>
<td>125</td>
<td>250</td>
</tr>
</tbody>
</table>

**Maximum length to transmit upper frequency.**

<table>
<thead>
<tr>
<th>Wire Size (mm)</th>
<th>300 Meters</th>
<th>5000 Meters</th>
<th>7000 Meters</th>
<th>10000 Meters</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.65</td>
<td>1150</td>
<td>1150</td>
<td>900</td>
<td>820</td>
</tr>
<tr>
<td>2.06</td>
<td>1200</td>
<td>950</td>
<td>750</td>
<td>600</td>
</tr>
<tr>
<td>1.60</td>
<td>900</td>
<td>730</td>
<td>600</td>
<td>520</td>
</tr>
<tr>
<td>1.32</td>
<td>750</td>
<td>580</td>
<td>460</td>
<td>410</td>
</tr>
<tr>
<td>1.00</td>
<td>600</td>
<td>460</td>
<td>380</td>
<td>320</td>
</tr>
<tr>
<td>0.80</td>
<td>...</td>
<td>380</td>
<td>300</td>
<td>260</td>
</tr>
<tr>
<td>0.63</td>
<td>...</td>
<td>...</td>
<td>...</td>
<td>200</td>
</tr>
</tbody>
</table>
Standard Cabling

For small public address systems up to a total of 340 Watts load, a 2 core cable with the following lengths dependant on the length of the speaker cable is required.

<table>
<thead>
<tr>
<th>Cable Distance</th>
<th>Cable Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>Less than 600m</td>
<td>1.0 mm²</td>
</tr>
<tr>
<td>600m to 1200m</td>
<td>1.5 mm²</td>
</tr>
<tr>
<td>More than 1200m</td>
<td>2.5 mm²</td>
</tr>
</tbody>
</table>

If these guidelines are adhered to then the maximum signal drop at the last speaker should be in the order of 1dB at an average speech signal level and in the order of 3dB at maximum output, both of which are negligible. It should be noted that whilst we recommend conductor sizes the actual type of cable used should be appropriate to the individual site and installation.

10. Earthing and other Safety Precautions (As per Telecom manual chapter 20)

10.1 General

The layout and wiring of cables for loudspeakers shall, as far as possible, be so done as to ensure safety and avoidance of the obstructions in the normal functioning of the installation. The Indian Electricity Rules 1956 and the Indian Standard Code of Practice for Electrical Wiring and Fittings in Buildings (IS:732-1958) shall be followed, as far as they are applicable for wiring loudspeaker installations.

10.2 Earthing

(a) Proper earthing of the entire installation (with appropriate earthing of the individual equipment also) is absolutely essential to avoid danger from any possible shocks to the users of the equipments, the operating personnel or the audience.

(b) Earthing connections to the nearby water mains are of usually lower resistance than any form of buried earth electrode system. An equipment of installation may be satisfactorily earthed by means of connection to the nearest water supply mains by a good soldered joints. In the absence of a suitable water main, the earthing may be done by connection to other efficiently earthed object.

(c) The use of two or more separate earthing connections at different points on the system is inadvisable due to risk of trouble from circulating currents. The earth connection from water supply mains or other earthed electrode should be brought to an earth-busbar in the equipment room. The earthing connection to the installation equipments should be drawn from this earth-bus bar.

The screening leads of a microphone should have a separate insulated lead run direct to the earth connection. It shall not be connected to the electrical earthing of the installation which gives rise to hum.
10.3 Earthing leads

A fairly heavy cable (such as 7/0.75 mm VIR in conduit) or bare copper wire (4.0 mm dia) is normally satisfactory as the earthing connection lead. If bare copper wire is used, care should be taken to run it insulated from other metallic objects all along its length.

10.4 Earthing in 25 kV AC electrified area

(a) All the sheaths/screens of wires/cables and metallic conduits must be earthed at both ends.
(b) In electrified areas, the metal mounting of the cone type loudspeakers and the body of the horn type loudspeakers shall be earthed. The resistance of this earthing shall not exceed 10 ohms.

10.5 Additional requirements in 25 kV ac electrified area

a) In the electrified areas, the length of parallelism between the loudspeaker circuit and catenary system should be limited to 1.2 Km. where this length is exceeded; suitable sectionalising transformers shall be provided.
b) A minimum separation of 5 meters between the nearest wiring of the loudspeaker and the catenary system shall be kept.
c) Screened cables & wires or cables & wires in metallic conduit should be used in electrified areas, so as to eliminate the effects of induction both electromagnetic and electrostatic. The cable screening conduit shall be effectively earthed at both the ends. If sectionalising transformers are provided, earthing of the cable screen should be done on the two sides of the transformers. The value of this earthing resistance should be as low as possible and shall not exceed 5 ohms.
d) The screened cables used for working the loudspeakers shall have a screening factor of 0.5 within the field intensity of 50V/Km to 450 V/Km at 50 Hz.
e) It is desirable to run a main cable for the loudspeaker circuits as far away as possible from the catenary system and connect the loudspeaker at different points by distribution cables run at right angles to the catenary system.
f) Wiring for equipment should also be screened (sheathed).

10.6 Safety requirements for amplifiers operated from electric mains

The amplifier or amplifiers operating from mains supply shall conform to the requirement specified in IS: 9302 -Mains operated audio amplifiers.

10.7 Fire and Explosion risk

The installation of the system in a situation, where there may be a possibility of an explosion, or in an inflammable atmosphere, should as far as possible, be avoided. However, if it becomes necessary to have the installation in such a situation, it should not be vulnerable to fire explosion risks.
11.0 PA System at Railway Stations

On Railway stations the PA system is provided for auto announcement through IPIS (Integrated Passenger Information System) and manual announcement through microphone for transmitting informations regarding train position/other informations to passengers and Railway staff working on Railway platforms/station premises.

11.1 Auto Announcement System (PC Based Announcement System)
[Para 11.1 is As per telecom manual chapter 21.6]

1. It is an integrated system to work as Auto Announcement PA System, Display system & Coach Guidance announcing system. The system shall be capable of automatic announcement with pre-recorded voice prompt, which shall be stored in the hard disk of the system.

2. The Data is entered by the data entry operator / Station Master by entering Train number, arrival / departure time and status of the train in the screen format.

3. The selected massage is scrolled on the monitor so that the operator can know the announcement / Display being made on platform PA system and display boards.

4. PA Systems are to be provided at the Stations covering Concourse, Platform area and different locations. The type of Speakers, Mikes, Acoustical environment, Type of Loudspeaker, Wiring and Cabling, Earthing and other Safety precaution should of standard make and be as per RDSO specification.

11.1.1. Features

1. It shall be possible to choose any of the system to keep on or any of the system remaining idle.

2. Messages announced on the Platform PA system shall synchronize with the information shown on the display board.

3. It shall be possible to add, modify and delete timings of the trains in the master database, which is password, protected.

4. Hot key shall be used for transferring data from PC to various systems.

11.1.2 System Configuration

System mainly consists of Standard PC in which the Operational software is loaded to run the systems i.e. PA System on platforms, Display system, Coach Guidance Announcing System.
11.2 System Description

Fig No. 47 Interconnection of PA System with IPIS
A simple sketch for inter connections of PA system with IPIS for auto announcement and also for manual announcement is shown in above figure. In this Diagram four amplifiers are connected in such a way as given below:

PRE AMP OUT (Pre amplifier out) of amplifier No.1 is connected to AUX (auxiliary) of amplifier No.2. [Both are generally provided at rear panel of amplifier]

PRE AMP OUT of amplifier No.2 is connected to AUX of amplifier No.3. [Both are generally provided at rear panel of amplifier]

PRE AMP OUT of amplifier No.3 is connected to AUX of amplifier No.4. [Both are generally provided at rear panel of amplifier]

If the amplifiers are more then four than they will also connect in similar fashion.

Here only four amplifiers have shown but the numbers and wattage of amplifier will increase or decreased as per load (Wattage) requirement.

Speakers installed at different platforms as per requirement are connected with 100 Volt line of each amplifier’s speaker terminal strip which is generally provided at rear panel of the amplifier. A microphone is connected with any microphone input jack socket provided at front panel of Amplifier No.1 for manual announcement to entire speakers. Volume of manual announcement can be control by Master Switch/ concerned Microphone Volume Control Switch. Audio out from IPIS system is fed to AUX socket of Amplifier No.1 provided mostly at rear panel of the amplifier so that the auto announcement from PC will be heard in all speakers. Volume of announcement can be control by Master control/ Microphone volume control / AUX volume control of Amplifier 1.

12. Fault Finding

12.1 Noise

Noise is a very common, and sometimes a very difficult problem to solve. Noise comes from many sources. Some of the most common are:

- Poor wiring in the location where the P.A. is being used
- Fluorescent lights
- Dimmers
- Poor design (particularly shielding and location of the mixer’s power transformer)
- Magnetic fields induced by other nearby components (particularly power amplifiers)
- Radio stations or other transmitters in the area
- Large motors near by
- Grounding problems (particularly when using equipment manufactured by different companies)
- Noise in the input signal (particularly guitar pickups)
- Poor cables (particularly on the inputs)
Most of these can be reduced or even eliminated by using high quality cables and balanced lines. To check the source of the noise, unplug all inputs from the mixer/amplifier input jacks and observe noise. If noise is disappeared then it is clear that the noise was definitely coming from something that are plugging into the input of the mixer/amplifier and not from the P.A. itself. Now start plugging the sources back into the mixer/amplifier, one by one. Observe for noisy input. Take steps to replace the noisy input such as microphone, cassette recorder, guitar etc.

If noise is still there than follow following few suggestions:

- Change your cables to better quality ones with a high degree of shielding.
- Wherever possible, use balanced sources (some manufacturers have both balanced and unbalanced outputs on their products. If they do, change to balanced.)
- Use of a balanced cable will even improve noise in unbalanced sources as long as the mixer inputs are balanced.
- Use direct boxes where possible for instruments being plugged into the mixer. This isolates the signal, converts any high impedance instruments to be balanced low impedance and allows you to "lift" the grounds on these devices. We will able to reduce or eliminate many radio signals, buzzes and ground hums with this method.
- Reduce the length of the cables.
- Make sure the input cables are not lying too close to a transformer, motor, amplifier or other source of magnetic radiation.
- Plugging the components of a P.A. system into different electrical outlets can sometimes cause problems. Wherever possible, try to connect all parts of the P.A. into the same circuit, even if you have to run extension cords to accomplish this.
- Turn any lights on dimmers off, or if this is not possible, turn them fully on.
- Reduce the volume output on the Amplifiers an increase the output on your mixer. This sometimes solves the problem.

12.2 Fault-Finding in Live Productions

Sometimes problems are during set-up or sound check, and occasionally during the performance itself. Even in fairly basic PA systems there are plenty of possibilities for something to go wrong such as: leads can get pulled loose, batteries (even new ones!) can die unexpectedly, and amplifiers can overheat and shut down. Our first reactions shall be as given below:

i. Stay calm! Panicking or losing your temper with people or equipment will not make the problem solve.
ii. Focus your attention in understanding that the signal is reaching to its destination or not. If not, why?
iii. Be methodical. Don't change connections, leads and components at random. Start diagnoses the problem along the signal path. Eliminate the possible causes of failure.. If a microphone isn't working, replace it with one you know to be OK. If it still doesn't work, replace the lead with one you know to be OK.
iv. Be safe. Electricity kills. Never take the covers off live equipment (always disconnect from the mains before you take the cover off anything). Never replace fuses with any other conductor (wire, nails, and screws). If fuses keep blowing, the equipment is faulty and potentially dangerous. Never attempt repairs that need both hands while you are hanging off a ladder.

v. Mark faulty equipment. If you find a lead is faulty, don't just put it back in the bag. Mark it faulty in some obvious way & put it to one side, and don't take it out again until it has been repaired and tested. The same goes for everything else, from microphone to mixer to monitor.

vi. Most of actual system faults are caused by faulty cables. Cables may fail at the solder points (bad soldering and poor strain-relief are the most common causes of this: cheap cables and connectors are a false economy), and although this normally results in loss of continuity, sometimes a loose end can make contact with one of the other conductors, causing short-circuit conditions. In the worst case, this can cause failure of the preamplifier or amplifier stage driving it.

12.3 Mains hum

Mains hum is a low frequency hum or buzz which increases in volume as the PA system is turned up louder. The most common reasons for mains hum is when two pieces of mains powered equipment are connected via an audio cable (e.g. jack plug). For example, mains powered monitors connected to the mixer.

If there is a mains hum problem then first identify which piece of equipment is causing the problem. Disconnect all the equipment from the mains and also all the connecting signal cables between the equipment. Now start introducing the equipment piece by piece until the problem is notice. Never, ever remove an earth wire from a mains plug to avoid the problem.

12.4 Feedback

Acoustic feedback will occur when the microphone picks up audio from the loudspeaker and transmits it back into the amplifier. A "loop" of sound occurs which builds in intensity to a piercing scream. This is usually due to the volume being too high or the microphone too near to the speaker. The scream won't stop until the loop of sound is not break physically. Every venue has a point where feedback cannot be avoided. However following points to be noted:

- Is the microphone is being use correctly.
- If microphone is far away from the mouth.
- Are using microphones are cheap as they may more prone to feedback.
- Are your speakers positioned far enough in front of the line of your microphones. Try moving them forward a little at a time.
- Change the angle of PA speakers. Even slightly changing the angle of the speakers can affect feedback enormously.
- Feedback occurs at varying frequencies. You may be able to adjust the EQ on the mixer to cut down the feedback.
13.0 Precautions

13.1 During Installation

- Do not expose the unit to rain or an environment where it may be splashed by water or other liquids, as doing so may result in fire or electric shock.
- Use the unit only with the voltage specified on the unit. Using a voltage higher than that which is specified may result in fire or electric shock.
- Do not cut, kink, or modify the power supply cord. Never use damaged power supply cord. Avoid using the power cord in close proximity to heaters, and never place heavy objects including the unit itself on the power cord, as doing so may result in fire or electric shock.
- Be sure to replace the unit's terminal cover after completion of connections. Because high voltage is applied to the speaker terminals, never touch these terminals to avoid electric shock.
- Be sure to ground to the safety ground (earth) terminal to avoid electric shock. Never ground to a gas pipe as a catastrophic disaster may result.
- Avoid installing or mounting the unit in unstable locations, such as on a rickety table or a slanted surface. Doing so may result in the unit falling down, causing personal injury and/or property damage.
- Never plug in or remove the power supply plug with wet hands, as doing so may cause electric shock.
- When unplugging the power supply cord, be sure to grasp the power supply plug; never pull on the cord itself. Operating the unit with a damaged power supply cord may cause a fire or electric shock.
- When moving the unit, be sure to remove its power supply cord from the wall outlet. Moving the unit with the power cord connected to the outlet may cause damage to the power cord, resulting in fire or electric shock. When removing the power cord, be sure to hold its plug to pull.
- Do not block the ventilation slots in the unit's cover. Doing so may cause heat to build up inside the unit and result in fire.
- Avoid installing the unit in humid or dusty locations, in locations exposed to the direct sunlight, near the heaters, or in locations generating sooty smoke or steam as doing otherwise may result in fire or electric shock.
- Always use thick cables like cable type 40/36 for low impedance speaker connections to avoid power losses in the cable.
- Do not exceed 90% of the amplifier’s output power when using 100V line (Speech only).
- Do not exceed 70% of the amplifier’s output power when using 100V line (High level music or voice).
- Speaker Connections: Low Impedance Speaker Impedance taps of 4 Ω, 8 Ω and 16 Ω have been provided for direct connection of speakers to the amplifier when the distance between the amplifier and speakers is less than 50 meters. Use thicker cable for connections. Be sure that the total impedance of the speakers is equal to or more than the impedance specified on the terminal strip.
Phasing Of Loudspeakers: When two or more speakers/units are installed in the same area and are facing in the same direction, it is essential that their cones/diaphragms act in unison. Otherwise, the sound level of one speaker will be canceling the sound level of the other. To avoid any mistake the terminals of all driver units are marked L1 and L2.

Loudspeaker Connections: The loudspeaker connections must be made to only one selected impedance i.e. either to "COM" and 4 Ω or to "COM" and 8 Ω "COM" and 16Ω terminals on the amplifier. When speakers are connected to any one of the above impedances, make sure that no speaker is connected to either 70 V or 100V line output terminals. Making connections to two impedance tape simultaneously will cause overloading and damage to the amplifier.

Selection of Loudspeakers: Proper selection of column speakers/driver units-horns and their appropriate impedance matching to the amplifier, avoiding over loading and short circuit while making connections are of utmost importance in speaker installations. Any mistake made would not only result in inferior sound quality but would also become a cause for damage to the amplifier.

Never connect speaker units having LMT in series. They shall be connected in parallel as shown below:
13.2 During Operation

- If found following irregularity during operation, immediately switch off the power, disconnect the power supply plug from the AC outlet. Make no further attempt to operate the unit in this condition as this may cause fire or electric shock:
  
  - If you detect smoke or a strange smell coming from the unit.
  - If water or any metallic object gets into the unit.
  - If the unit falls, or the unit case breaks.
  - If the power supply cord is damaged (exposure of the core, disconnection).
  - If it is malfunctioning (no tone sounds.)

- To prevent a fire or electric shock, never open or remove the unit case as there are high voltage components inside the unit.

- Do not place cups, bowls, or other containers of liquid or metallic objects on top of the unit. If they accidentally spill into the unit, this may cause a fire or electric shock.

- Do not insert or drop metallic objects or flammable materials in the ventilation slots of the unit’s cover, as this may result in fire or electric shock.

- Do not place heavy objects on the unit as this may cause it to fall or break which may result in personal injury and/or property damage. In addition, the object itself may fall off and cause injury and/or damage.

- Make sure that the volume control is set to minimum position before power is switched on. Loud noise produced at high volume when power is switched on cans the sound distorting. This is an indication of a malfunction, which in turn can cause heat to generate and result in a fire.

- Remove the dust accumulated in the unit over a long period of time, a fire or damage to the unit may result.

- If dust accumulates on the power supply plug or in the wall AC outlet, a fire may result. Clean it periodically. In addition, insert the plug in the wall outlet securely.

- Switch off the power, and unplug the power supply plug from the AC outlet for safety purposes when cleaning or leaving the unit unused for 10 days or more. Doing otherwise may cause a fire or electric shock.

- Avoid jointing of microphone cables. If it is un-avoidable, make sure a good screened connector is used.

- Ensure that all loudspeakers are in phase there is no short circuit on the loudspeaker line before connecting to the amplifier.

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