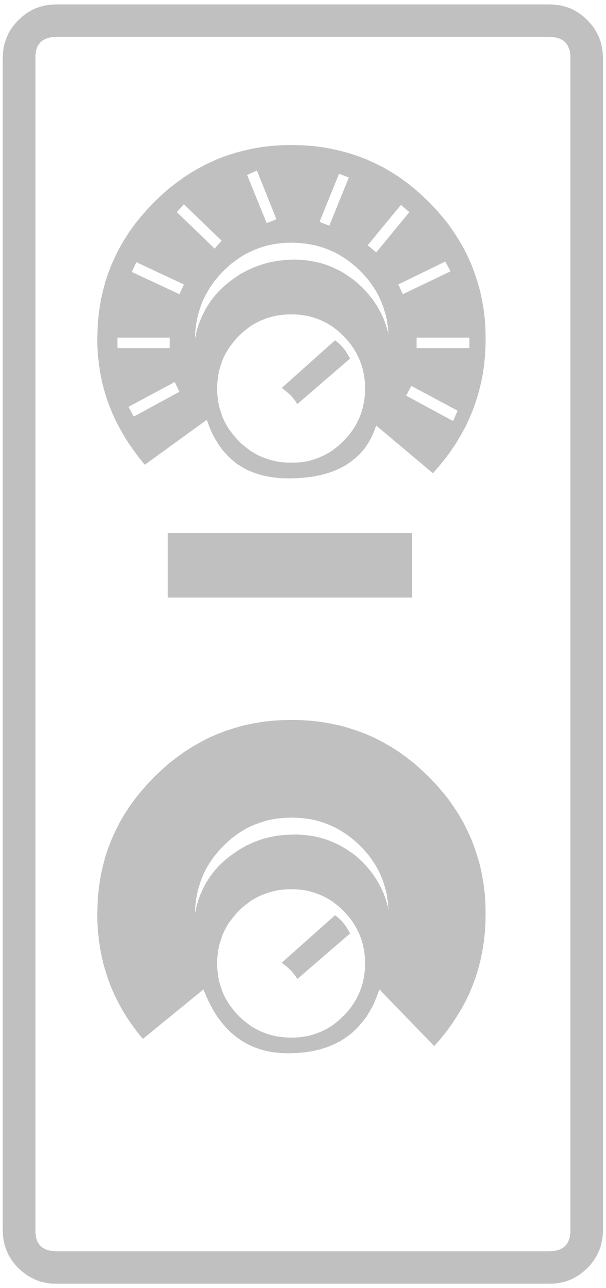
ACUTRON ELECTROACÚSTICA LDA.
✉ Urbanização Quinta Nova, Impasse 1, Lote 134
2685 Sacavém - Portugal
☎ 351-219401785 Fax 351-219401786
URL www.acutron.pt EMAIL info@acutron.pt

Acutron® digital audio
LARSEN®

SOUND SYSTEMS



An Acutron publication devoted to all electro-acoustic system users

Index

1 Who should read this manual

This manual is of multifaceted use. It is aimed to the designer of electro-acoustic reinforcement systems, to the system user and to the installation technician. It intends to provide a resumed yet practical vision of the subject to the reader, always presenting theoretical formulas and concepts where necessary, thus allowing the user to expand its knowledge to more advanced applications.

This is a work in constant update, because a work in this area can never be considered perfect or completed. We count on the reader to suggest the incorporation of pertinent sections and the correction of the existing ones. Such suggestions should be sent to techsupport@acutron.net.

2 Dimensioning a sound system

In order to properly dimension a sound system some prior steps are necessary:

1. Establishing the necessary system functions, taking into account the type of application and the customer specifications.

2. Analyze the characteristics of the environment to which the sound system will be applied to.

3. Choose the speakers, taking as a basis of the environment and dimension of the space, the type of message to transmit (voice, music or both) and the ambient noise level.

4. Choose the amplifier(s) that fulfill the requirements defined in the previous points, and those defined in points 5 and 6.

5. Define the sound sources (microphones, tuners, CD players, and so on).

6. Define the distribution model to be used in the system (low impedance, 100V) and the section of the connection cables to be used.

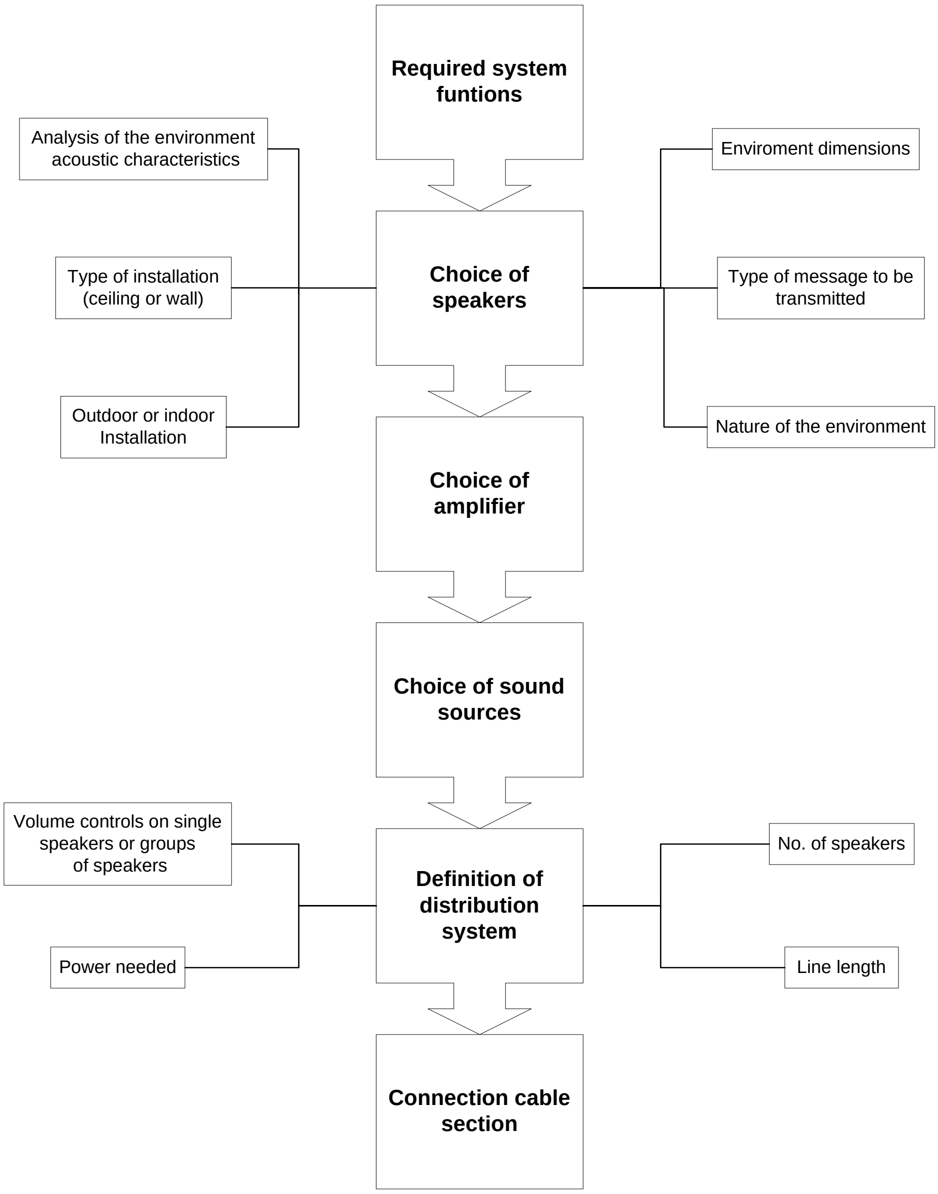


Fig.1- Analysis of a sound system

It is important to point out that the quality of a sound system depends on the quality of its components. The quality of a sound system will thus be the one corresponding to its weakest link. **Acutron Electroacustica** only supplies high quality components, carefully selected so that in the end the result is a high quality system.

Other limitation to the quality of the sound system will be the environment itself because in extreme cases, a high quality system is by itself not capable to satisfy in environments that cause high signal degradation. In these cases some structural space modifications will probably be necessary.

3 Components of a sound system

A sound system is built by components, and each component is responsible for a specific function within the system, but 3 components are present in the majority of sound systems. Those are the microphone, the amplifier and the loudspeaker.

3.1 Microphone

The microphone transforms the variations of sound pressure into electrical oscillatory voltage or current and is used to grab one or multiple sound sources, as voice, musical instruments, and so on.

There are two main microphone types, dynamic and condenser, along with the less used electromagnetic and ribbon types.

3.1.1 Dynamic microphone

Any type of microphone has a diaphragm that oscillates with the variation of sound pressure. In the case of the dynamic microphone the diaphragm is responsible for the simultaneous motion of a coil that is immerged in a constant magnetic flux. Variations in the area of the coil subjected to the magnetic flux results in an electrical current. This current variation can be transformed into a voltage variation if a resistor is used to load the coil terminals. This voltage variation will be proportional to sound pressure variation.

This is the type of microphone presently used in Acutron MIC2 terminal, which is fed and controlled using a simple 1-pair microphone cable.

3.1.2 Condenser Microphone

This type of microphone works on the principle of variation of a capacity proportional to the variation of sound pressure. For this the microphone uses two conductive plates separate by an insulator. One of the plates will be the diaphragm which, when oscillating modifies the distance between plates, thus effectively varying the capacity between them. When a direct voltage is applied to the plates, the capacity variation, provoke a variation in the current, directly proportional to the variation of the sound pressure.

Because the variations are very low in magnitude and of their inherent high impedance it is necessary that a low-noise pre-amplifier is used, fed in terms of power supply either by batteries or by the amplifier/mixer itself through the use of a *"phantom power"* arrangement. This fact may turn into a disadvantage in cases where this type of power supply is not available.

These microphones do deploy excellent specifications, but they have a higher cost than dynamic models and are considerably less robust, what generally limits its use in professional applications only.

Microphones in general are evaluated according to following technical specifications:

Quality- The quality of a microphone depends on the frequency range than it can pickup. A microphone of excellent quality is capable to pickup all the audible frequency range very linearly and with a low distortion. In some situations a microphone of superior quality may not be the best choice for a given application. Presence of significant disturbing noise in a spectral zone where a limited bandwidth microphone does not pick up significantly can potentially make it far superior to another of superlative quality, not needing electric filters that can strongly degrade dynamics in pre-amplifiers stages.

Directivity- The directivity of a microphone determines its capacity of picking up sound in a variety of directions. The most common type is the cardioid, but in some cases where the capacity of picking up sounds coming from different sources the omnidirectional will be the best choice. Use of this microphone type is subjected to a careful prior study of the system, due to some phenomena described under **5.3**. Still more directive microphones (hypercardioid or line types) are used in cases where increased side rejection is needed.

Sensitivity- The sensitivity of a microphone is expressed in mV/Pa (20mPa = 0dB). A higher sensitivity produces a higher voltage on the microphone terminals with the same sound pressure level, thus resulting in the capacity to pickup weak sound sources, with less noise added by signal amplification. Sensitivity also depends on directivity. Cardioid microphones are sensitive to sound sources with an angle of incidence in the microphone of 0º (front), but show very low sensitivity to sound sources with an angle of incidence of 180º (side), allowing for example the positioning of sound reproducers within the microphone zone without the risk of acoustic feedback.

3.2 Amplifier

Signal sources like microphones, CD players, cassette DAT and MiniDisc recorders, do not have the capacity to produce enough energy to directly feed a loudspeaker. It is therefore necessary to use an amplifier that transforms the low power signals of the signal sources in signals with sufficient power to feed the system loudspeakers.

The amplifiers amplify the standard 0.775V (0dBu=0dBm referred to 600 Ohm) signals, but when the signal source is lower than this, it may be necessary to use an pre-amplifier between the signal source and the amplifier, this pre-amplifier, can be present in the amplifier, transforming a Line input (*LINE IN*), into an microphone input (*MIC IN*), with an increased sensitivity. Acutron SLICE1154 amplifier already has all preamplifiers built-in and programmable, so they will only be present when needed. An example of a stand-alone pre-amplifier is the programmable-gain DA-28 manufactured by Acutron.

The amplifier can also feature an equalizer so as to allow a spectral modification of the signal to be amplified, like the SLICE 1154 of **Acutron Electroacustica**, or the stand-alone MIX8 mixer.

The important parameters in an amplifier are:

Intelligibility – Is the capacity of the device to spread out the sound in a perceivable mode to the listener. This is related with needed power relating itself with transducer efficiency and disturbing noise, with frequency response (only in the band of interest) and with absence of distortion.

Power - The power of the amplifier shall be greater to the total power of the loudspeakers in the system, if it is assumes that the system will never be driven into saturation. **Acutron Electroacustica** manufactures amplifiers with 2x75 W (*SLICE 2152*) and 2x150 W (*SLICE 2302*) for 4/8 systems, and 1x150W (*SLICE 1154* and *SLICE 1151*) and 1x300W (*SLICE 1301*), for 100V line systems. The two distribution systems (4/8 and 100V) do require different amplifiers which can not be exchanged.

Frequency response- The frequency response of an amplifier is a measure of the frequency range of signals that can be amplified linearly. **Acutron Electroacustica** amplifiers have a flat frequency response from 20Hz to 20 kHz except in the 100V line models where an internal filter limits the band to 150Hz in the lower end so as to accurately drive the transducers that are normally connected to them. This filter removes the inaudible components that would otherwise lower the level of audible sound in the system, due to the useless occupation of bandwidth.

Harmonic distortion and intermodulation distortion – Corresponds to the percentage of undesirable modification of the signal that the amplifier produces by itself, measured at powers ranging from the usual (10 times lesser than maximum) and maximum power. The **Acutron Electroacustica** SLICE amplifiers typically produce a distortion which is less than 0.2%, even in adverse conditions, due to absence of speaker coupling transformers, which substantially decreases the quality in terms of bandwidth and distortion, and to the use of high speed internal correction circuits which improve the intermodulation distortion, even the transient one (TIM, CCIF).

Number of inputs **-** The number of inputs will be the number of signal sources that the amplifier will be able to amplify. Of no concern in a large system because there are other devices to perform this function, but can be a limiting factor in a small system that uses just one amplifier.

Protection **-** The level of protection of an amplifier depends on the protection devices that the amplifier is fitted with so as to guarantee the integrity of the amplifier and the load in case of overload on the inputs, on the outputs, overheating or internal malfunction. The amplifiers that **Acutron Electroacustica** manufactures have protection against all of these events, controlled by an internal microprocessor. We are talking about devices that continuously measure the speaker load impedance without using high frequency carriers, thus not ruining the quality of the audible signal. The existence of circuits minimizing the connection transients according to IEC standards, to protect loudspeaker lines against atmospheric discharges and to detect the presence of off-band signals are a constant in the SLICE amplifiers.

Reliability - The reliability of the amplifier is related with the quality of the used components in the production and with the severity of the quality tests to which it is subjected to. **Acutron Electroacustica** amplifiers make use of excellent quality components and are tested to work in adverse conditions with great reliability. They present MTBF values of about 50 000 h, resulting from a tuned design, a fine selection of materials and fine production techniques such as burn-in in the production, by far exceeding the ones dictated by the ISO9001 standard.

Functionality - To possess an amplifier without the necessary number of inputs, the alarm signals needed in a compatible CE installation, gong signals needed input by input to be used for the calls, without programmable priorities in the case of multiple inputs, without the programmable input pre-amplifiers, long-distance volume control or the necessary series interfaces used for control and telemetry, can be an handicap difficult or onerous to bypass. SLICE amplifiers comply with all of those demands either as a serial built-in feature (alarm, gong, priorities, preamplifier programming), or optionally (/ 485- series interface, / RVC- remote volume control, necessary in large installations).

3.3 Loudspeakers

In spite several types of speaker exist, such as the electromagnetic, condenser, ribbon or even direct air ionization types, only electrodynamic, by far the most common will receive treatment here.

There are two categories of dynamic loudspeakers, depending on its air-coupling:

- Direct-coupling or baffle speakers
- horn speakers

Baffle speakers are used in the sound columns, ceiling speakers and projector speakers. They produce sound with great quality and limited directivity, but they are less efficient and sensitive than horn speakers.

Horn speakers are about 10 to 30 times more efficient (they produce more sound with the same electric power applied) than the cone speakers due to the use of what can be described as an “acoustic transformer”, but they reproduce sound with less quality (more limited bandwidth). They are more commonly used in places with great ambient noise and in the outdoors.

The use of horn speakers in systems where good quality is required is however possible if we divide the band to transmit in sub-bands, each one treated separately with a horn speaker and a dedicated amplifier (multi-amplification). This process is inherently expensive but it solves problems impossible to be dealt with in other ways.

The most important specifications of speakers are:

Dispersion, or Directivity- Is the dispersion of the sound that the loudspeaker produces. It is measured in degrees in relation to its center, on the horizontal and vertical planes or expressed by means of a directivity factor (Q). This is very similar to what we discussed before about microphones.

Directive speakers are useful in cases where specific areas need to be reached while minimizing reflections on nearby surfaces that may disturb the intelligibility through modification of transit time.

Type- The type of the transducer depends on the type of application and the necessary features. For example in false ceilings, recessed-mount speakers are normally used. Projectors are used in places of difficult speaker fixing and in the outdoors, and baffled speakers or column speakers may be used in other applications where flush mounting is needed.

3.4 Attenuators

These devices are introduced between the amplifier and the speaker or group of speakers, typically in 100V line systems. The power stepping must use inductive methods, in order to not lose power and to be robust (variable resistor attenuators are prone to mechanical wear and do waste power). Finally they must be specified using the sum of the powers of the speakers that are connected to it. The AT line of attenuators and the SAT line of attenuators/selectors manufactured by Acutron comply with these requirements, being optionally offered equipped with relays for emergency bypassing (R suffix models).

3.5 Zone selectors

They are meant to allow selective direction of the messages in a 100V line system to specific zones, at the same time enabling the general program to be kept on non-addressed zones. They can be controlled by several different terminals, with priority between them, so as to allow the operators in one physical location to exploit the system in message mode. The selectors can, in small systems switch over the amplifier output stage, needing only 2 amplifiers for all the zones, or to switch before the amplifiers, in large systems. In this last case they will need at least one amplifier per zone. Acutron manufactures the ZONE8 and ZONE16 zone selectors, which may be steered by means of multiple ZONE8R and ZONE16R terminals.

3.6 Noise Sensors

Those are devices used to measure the ambient noise so as to allow automatic up-down steering of the reproduction level in strict agreement to the disturbing noise. Acutron manufactures the NLCD model, digital and micro processed, capable of filtering, measuring and storing the noise value, which can be used to remotely control the volume level in amplifiers SLICE1XX1/485/RVC.

3.7 Automatic amplifier replacement controllers

In installations where uninterrupted operation is needed even in case of damage to the amplifiers, Acutron GateKeeper should be used. The device is capable of monitoring the state of up to 8 100V line amplifiers, replacing any damaged amplifier almost instantly.

3.8 Equalizers

In order to improve global frequency response, parametric equalizers should be used, flexibly allowing spectral manipulation. Acutron SHAPER has 2 independent channels, managed by software, features internal scenery memories and allows the optional insertion of delays between channels.

4 Types of connection

Acutron Electroacustica manufactures and supplies amplifiers for low impedance speakers (4 or 8), and amplifiers for of constant voltage at maximum output (100V), for use in this type of systems.

4.1 Low Impedance systems

This type of system is only used when a small number of loudspeakers are necessary (2-4), placed at a short distance of the amplifier. Initially of lower cost due to absence of line transformers, presents nowadays identical pricing in relation to 100V line systems. Today are normally used in high power systems (stage systems, for example).

This limitation exists because in this type of system speakers present low impedance to the amplifier, normally 4 or 8 , and this does not allow the connection on the same line of more than 2 or 3 loudspeakers in series or parallel. To connect more than 2 or 3 loudspeakers on the same line, it will be necessary to use a mixed series/parallel connection, in order for the resultant impedance of the line to be the same as the nominal load impedance of the amplifier (*normally between 4 and 8*).

When trying to dimension a low impedance sound system, it is recommended that the series/parallel connections use loudspeakers of the same internal impedance, in order both to ease the calculations and to evenly distribute the speaker power.
In this type of systems, the distance between the loudspeakers and the amplifier must be short, because power loss in the cables will be proportional to the square of the current. On high power low impedance systems, current values in the cables will be important and these will require special attention.

$$P = U * I = \frac{U^2}{Z} = I^2 * Z \qquad \textit{(Formula 1.1)}$$

To limit the loss of power in the cables, a cable of adequate section must be used in the connections. Any audio system, due to the fact that the working frequencies (20Hz-20 kHz) correspond to enormous wave lengths in relation to the cable dimensions, is a concentrated parameter system, not benefiting in any way with the use of optimized geometry cables designed for high frequency operation, in contrast to what many cable manufacturers try to make believe.

4.1.1 Parallel connection of the speakers

In a parallel connection, all the positive terminals of the speakers are connected to the positive terminal of the amplifier, the same happens with the negative terminal of the speakers and the one in amplifier. In this type of connection, all the loudspeakers will handle the same voltage at their terminals, thereby producing the same sound pressure if they are of identical.

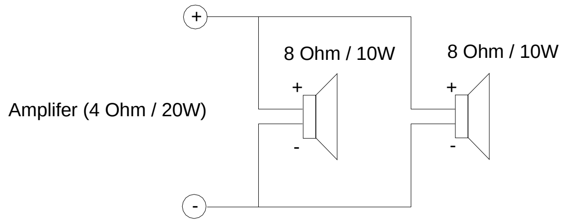


Fig.2- Parallel connection of the speakers

The total impedance of the line is calculated using the formula 1.2.

$$Zt = \frac{1}{\frac{1}{Z_1} + \frac{1}{Z_2} + \frac{1}{Z_3} + \ldots + \frac{1}{Z_n}} \qquad \textit{(Formula 1.2)}$$

Where Z_t is the total impedance and Z_1, Z_2...Z_n the impedance of each speaker

If all the speakers have the same impedance, the total impedance of the line can be calculated by the following formula:

$$Zt = \frac{Z_{speaker}}{n} \qquad \textit{(Formula 1.3)}$$

Where:

Z_t	*is total impedance*
$Z_{speaker}$	*is the impedance of each speaker*
n	*is the number of speakers*

4.1.2 Series connection of the speakers

The series connection of the speakers consists in connecting the negative terminal "-" of speaker 1 to the positive terminal "+" of the speaker 2, the negative terminal "-" of the speaker 2 to the positive terminal "+" of the speaker 3, until all the speakers in the system are connected The positive terminal of the first speaker will connect to the positive terminal of the amplifier, and the negative terminal of the last speaker will connect to the negative terminal of the amplifier.

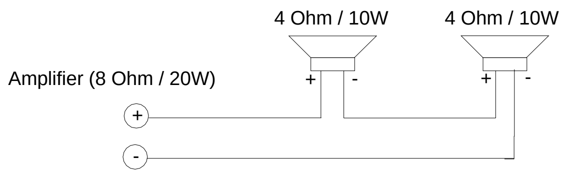


Fig.3- Series connection of the speakers

In this type of connection, the total impedance will be the sum, of the impedance of all the speakers.

$$Zt = Z_1 + Z_2 + Z_3 + \ldots + Z_n \qquad \textit{(Formula 1.4)}$$

The voltage in the speakers will be the same, if they have the same impedance.

4.1.3 Mixed connection

In this type of connection the two previously described types of connection are used. The calculation of the total impedance will take place using the formulas previously given, first calculating the total impedance in the series connections, and then calculating the total impedance in the parallel connections, replacing each serial connection by its equivalent impedance.

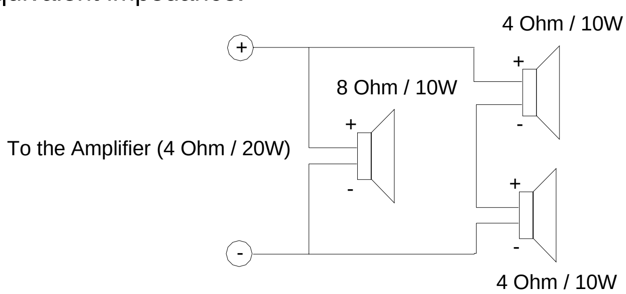


Fig.4- Mixed connection of the speakers

It is necessary to point out that in this type of systems it will always be necessary to proceed to new calculations each time the system configuration are modified (no. of speakers, power of the speakers, and so on), because any alteration induces a change in the total power of the system, and even more important, a change in the total impedance of the system. If new calculations are not made, the new total impedance or the new total power of the system may not be inside the working limits of the amplifier, resulting in serious damage to it.

Due to the limitations that this type of system imposes, in systems where a large number of speakers are used or a large distance between the amplifier and the speakers is needed, the use of constant voltage systems, to be described next, should be used.

4.2 Constant Voltage Systems

These types of systems have great advantages when compared to the low impedance systems, because they are based on the energy carrier systems where the value of the voltage used in the transport of energy is higher than the voltage used at the consumer. This allows for a lower current circulating on the cables, because the current is inversely proportional to the voltage for the same total power (*P=U*I*), resulting in fewer losses in the cables and the ability to use lower section cables. Each speaker has a transformer to adjust the impedance of the speaker (*normally low*) to the impedance of the line (variable with drawn power and inversely proportional to it).

Acutron Electroacustica has a large number of amplifiers and speakers of constant voltage (usually designed 100V line) and of high quality (mainly due to absence the transformer on the output), for any type of application. In fact the transformer is an expensive and problematic component when low distortion and large bandwidth are required, being easier to manufacture for small powers (cases of the ones used in the speakers) rather than for the large power ones normally used in the amplifiers.

In this type of systems, the speakers are all connected to the amplifier in parallel. The only limit to the number of on speakers connected to the same line consists on the elementary precaution dictating that the summed loudspeaker powers may not exceed the rated amplifier power.

This type of system still allows selecting the power of each speaker in the system in 3dB steps (half power), ideal for final installation fine tuning, so as to maximize acoustic comfort and intelligibility.

This is possible because the **Acutron Electroacustica** speakers have an internal transformer, with several taps (*5 terminals + 1 common*). Each terminal supplies to the speaker just a controlled amount of power, resulting in a determinate sound pressure level that the speaker will produce. As an example, for a speaker of 6W nominal rated power speaker, 6W, 3W, 1.5W, 0.75W and 0.25W power taps can be selected.

In this type of systems it is possible to use on the same line several different types of transducers such as baffled loudspeakers, sound columns or horn speakers, with no impairment on line functionality.

Another way to calculate the maximum number of speakers than can be connected to the same line is to use the following formula:

$$Z_{line} = \frac{U^2}{P} = \frac{P}{I^2} \qquad \textit{(Formula 1.5)}$$

Where:

Z_{line}	is the impedance of the line
U	is the voltage
P	is the power
I	is the current

We know that for one given power, the impedance of the line cannot be inferior to the calculated value of Z, this one being inversely proportional to the power P.

Example:

In a 100V system, equipped with a 150W amplifier, we have that:

$$Z_{line} = \frac{100^2}{150} = 66{,}66\Omega$$

The minimum impedance of the line must be 66,66 Ω .

The impedance of the line is the resulting value of the parallel of the speakers in the line.
The impedance of each speaker is the impedance seen from the transformer primary, for example:

- On a line we have two speakers with a nominal 6 Watt power rating, the impedance value over each transformer primary will be:

$$Z_{transformer} = \frac{100^2}{6} = 1666{,}66\Omega$$

Therefore, on the line, we will have an impedance of approximately:

$$Z_{line} = \frac{1666{,}66}{2} = 833{,}33\Omega$$

It is recommended that the line is measured prior to any amplifier connection, and to the comparison of the measured values versus the calculated values. If the line is in perfect conditions (*well connected cables, secure contacts, and so on*), the measured impedance value will come close to the calculated impedance value. These recommendations are described in **7**. If the recommendations are not followed that will represent a risk to the output stage of the amplifier. Amplifiers made by Acutron have protection against overloads but they can be damaged by ground leaks on line terminals, which can bypass the protection system, or by constant operation under overload conditions (visible by the intermittent activation of the output protection and consequent re-arming).

5 Electro-acoustic Considerations

5.1 Sound

The physical phenomena that we call sound consist in the transmission of vibrations generated by an oscillating object, through a physical media that propagates these oscillations. Our hearing organs pick up the oscillations of the physical mean.
The transmission media must be an "elastic" media that oscillates with the source object. D different types of material have different propagation speeds, for example the sound propagates in the water at a speed of 1520 m/s, while in the air that will be the media used in our study, is of 345 m/s.

The oscillations of the mean are as waves that form in a lake, when hit by a rock, and are characterized by differences of pressure of the media.
Being a repetitive phenomenon, the name period is given to the time difference between the repetitions. The number of repetitions per second is called the frequency, and it is calculated by the formula:

$$Freq_{Hz} = \frac{1}{Period} \qquad \textit{(Formula 2.1)}$$

The unit of the frequency is the Hz that corresponds to cycles per second. The human ear can detect sounds with a frequency between 20Hz and 20000Hz (20 kHz). A simple sound has a single frequency, while a complex sound is constituted by many simple sounds (several different frequencies) each one with its own characteristic amplitude.

5.2 Sound Pressure

Another concept that we need to study is the sound pressure level which a sound system can produce. This pressure as perceived by man is not a linear entity. In a sound system we will not double the pressure if we double the number of sound sources placed at the same distance, only if we multiply by a factor of 10 the number of sound sources. Perceived pressure is characterized by a logarithmic function. The unit in which we represent sound pressure is dB, or dB SPL (*Decibels of Sound Pressure Level*). The value of 0 dB corresponds to the minimum value of pressure that can be detected by the human ear.

In sound systems the oscillating object inducing the pressure variations is the loudspeaker cone or diaphragm, which in turn oscillates with the variation of the applied voltage. Speakers are characterized by their sensitivity in dB/1W/1m. The value of this sensitivity is obtained measuring the sound pressure produced by the speaker at 1 meter on its axis, when excited by pink noise with a RMS power of 1W. If a speaker produces a sound pressure of 70dB measured at 1 meter/1 Watt, this will be represented in the following way:

$$SPL_{1m/1W} = 70dB$$

The relation between the increase of the power in Watts, and the increase of the sound pressure that the speaker produces, is defined by the formula:

$$pressure\ rise_{dB} = 10 \log Power_{Watts} \qquad \textit{(Formula 2.2)}$$

An increase of 3dB of the sound pressure is the minimum detected increase in terms of auditory perception. This increase corresponds to doubling the power in Watts.

For example, if we increase a 10 Watt speaker to 20 Watt, we will have an increase of sound pressure of:

$$pressure\ rise_{dB} = 10 \log \frac{20\,W}{10\,W} = 3dB$$

We conclude that, if instead of duplicating the power of the speaker, we used two 10W speakers, the increase of the sound pressure would be equal (3dB if the speakers are apart from each other, otherwise other phenomena will take place because of increased directivity, inducing on-axis increases of up to 6dB).

The value of 3dB represents in a sound system where intelligible message broadcasting of messages is necessary the minimum value of sound pressure that the system will have to produce above of the value of the ambient sound pressure level (*ambient noise*), so that the transmitted messages can be perceivable by the receiver. To ensure intelligibility of messages and to accommodate variations in the disturbing noise level the system must be able to produce a sound pressure of 6 to 10dB above of the ambient noise.

To calculate the attenuation of sound pressure with distance, the following formula is used:

$$atenuation_{\mathrm{dB}} = 20 \log distance_{meters} \qquad \textit{(Formula 2.4)}$$

To calculate the sound pressure produced by a speaker with an $SPL_{1m/1W}$ of 70dB, at a distance of 10 meters, we have:

$$SPL_{10m/1W} = 70 - 20 \log 10 = 50dB$$

Two equal speakers placed side by side, would produce a sound pressure of 53dB.

In cases that, we need to know the increase of the sound pressure in a point, provoked by a speaker that produces a different pressure in this point, first we have to calculate the value of the pressure that each speaker produces in that point.

This situation corresponds to the following formula:

$$SPL_{total} = 20x\log_{10}\sqrt{10^{(-\frac{A}{10})}+1}$$ *(Formula 2.5)*

Where the:

SPL_{total} is the total SPL level produced by the 2 speakers
A is the difference between the more powerful source (to which the pressure increase will be referenced) and the weaker one. Note the negative sign before the fraction.

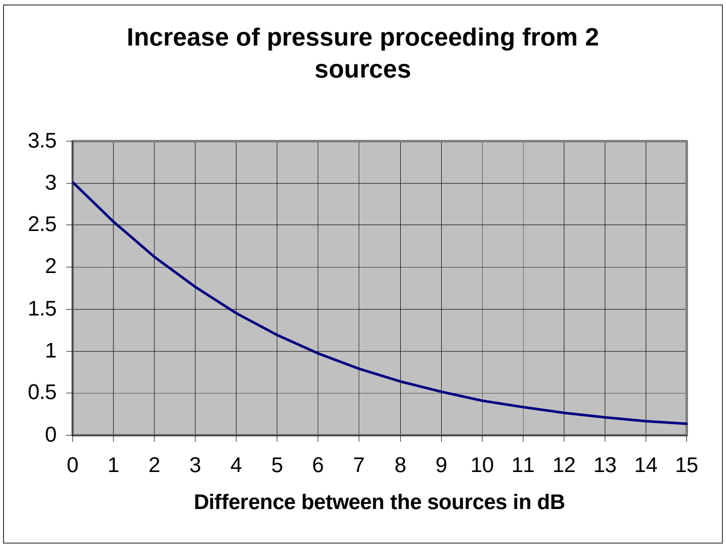


Fig. 5- Sound pressure calculation

Example:

We assume that we have 2 speakers, the first one produces a sound pressure of 70 dB in a point, while the other produces 77dB of sound pressure in the same point. The difference of pressure between the two speakers is 7 dB, which corresponds, according to the graph, to an increase of approximately 0.8 dB, therefore the sound pressure in the point will be:

$$77dB + 0.8dB = 77.8dB$$

Next there are a series of graphs that can serve as reference for the calculation of the sound pressure in determined spaces.

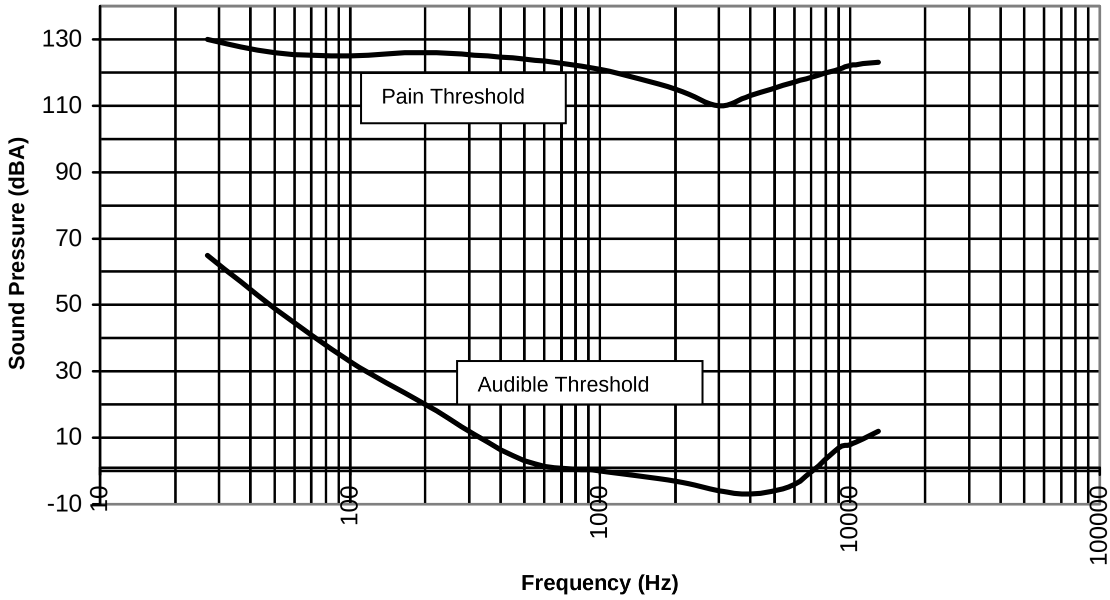


Fig. 6 – Human ear sensible area

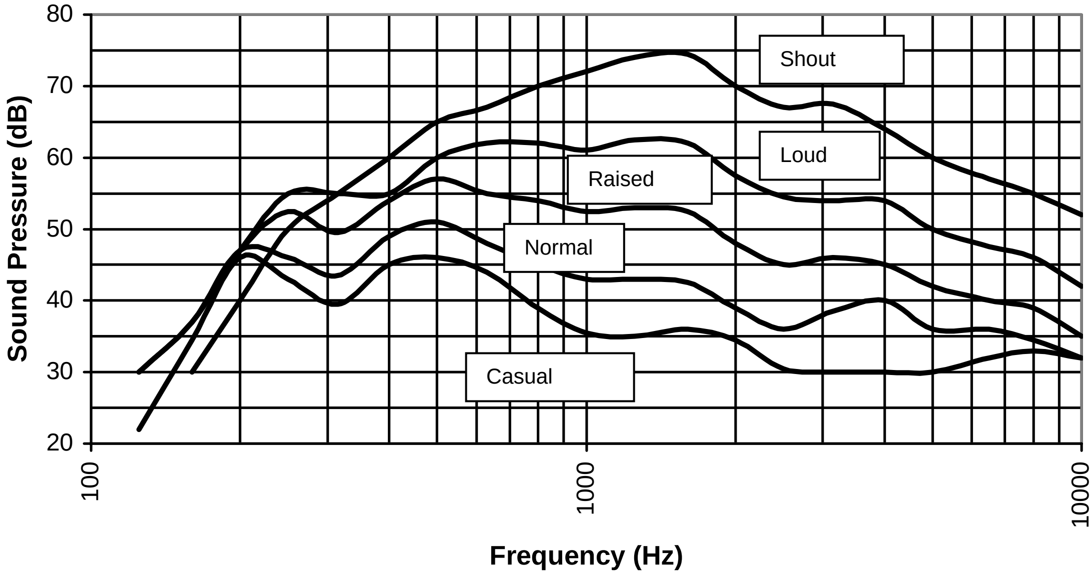


Fig. 7 – Spectrum of female voice

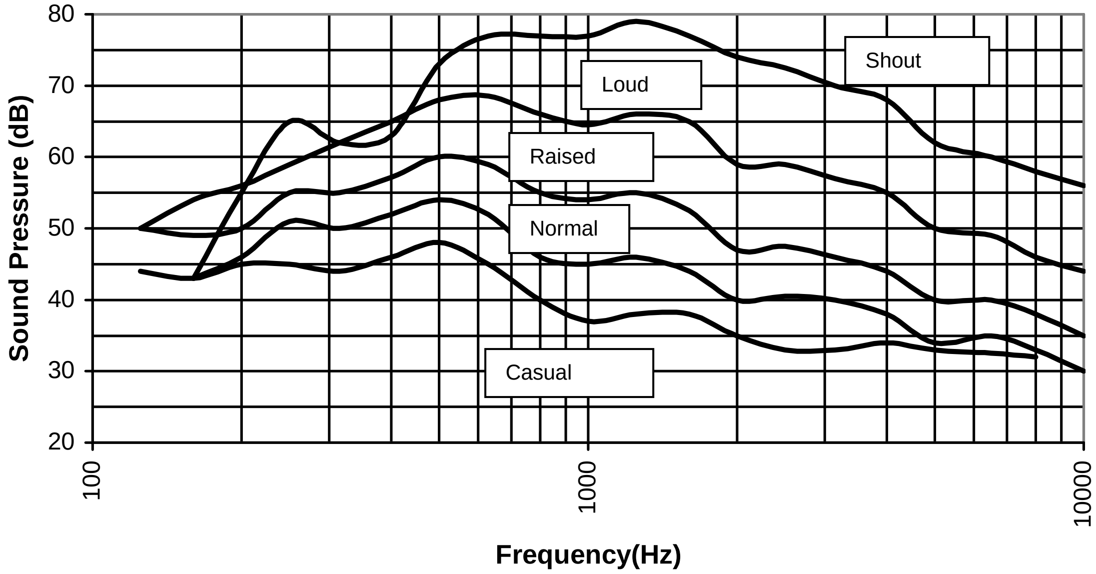


Fig. 8 – Spectrum of male voice

Location	Min (dBA)	Max
Inside Home	25	45
Inside Office	35	50
Inside Airplane Cabin	75	85
Inside Factory	65	100
Talking @ 1 m	55	65
Shouting @ 1 m	75	85
Clothes Dryer @ 1 m	55	65
Vacuum @ 1 m	65	80
Chain Saw @ 1 m	100	120
Clothes Washer @ 1 m	55	75
Car @ 8m @ 90 km/h	70	80
Airplane @ 300 m	95	110
Traffic @ 90 m	40	60
Rural Ambient	25	35

Fig. 9 – Sound Level Chart

5.3 Acoustic characteristics of the environment

There are two types of environments for the sound diffusion, open space and closed space. In the open space the sound waves spread without hitting any obstacle, therefore the sound reception is always direct. In closed spaces the sound reception can be direct or indirect. It will be indirect if the waves are reflected in obstacles (*walls, furniture, and so on*) that exist in closed spaces. These reflections can contribute to the degradation of the signal or to the impairing of its intelligibility. In extreme cases, a complete cancellation of the signal (destructive interferences) can occur.

The number of reflections and its intensity depend on the type of space, the covering materials of the walls and ceiling, the existing furniture in the space and the dimensions of the enclosure. In cases where the reflections degrade the signal, it will be necessary to proceed to modifications of one of these factors, even so, the most economic and with satisfactory results, will be the covering material of the walls and ceiling. Soft and irregular materials offer a considerable attenuation of the reflected waves at least at the medium-high working frequencies. On new spaces, care shall be taken so as to dimension the room so as the resonance wavelengths corresponding to each dimension are evenly spaced in bandwidth, no corresponding to harmonics of each other. As there are normally 3 dimensions on each room (length, width and height) those shall be related by the cubic root of 2, so as to spread resonances by consecutive thirds of an octave.

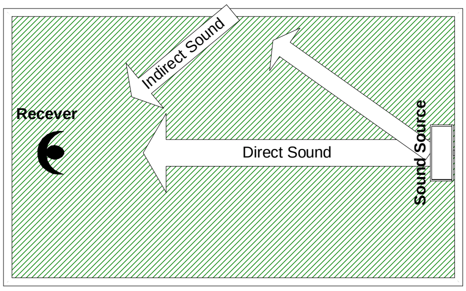


Fig.10– Environment with hard material in the walls

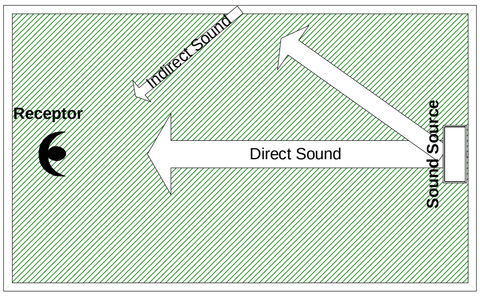


Fig.11 – Environment with soft materials in the walls

5.3.1 Reverberation

The reverberation, is the phenomena that the sound produces, when the sound does not reaches the receiver directly, but indirectly, after having been subject to innumerable reflections. The reverberation time is the time between the end of the emission of the original signal by the source, and the reception of the reflected signal by the receiver with an attenuation of 60dB.

The 60dB difference between the original signal and the reflected signal is the attenuation level required so that the reflected signal becomes imperceptible in relation to the original signal. The reverberation time depends on the absorption level that the space provides to the reflected signals. If the reverberation time of the system has values between 40 and 50ms, the indirect wave provokes a reinforcement of the original signal, between 50 and 80ms the original sound suffers a degradation, which can produce total cancellation of the signal. For reverberation times bigger than 80ms, the result will be an echo, which in certain cases also can contribute in the reduction of the intelligibility of the original sound.

To reduce the level of indirect sound, it will be necessary to reduce the dispersion of the signal, especially if the transducers are to work at high powers. This is possible with the use of sound columns or another type of directional transducers that limit dispersion to the areas considered necessary.

5.3.2 Feedback

The feedback phenomena, happens when a signal produced by a system is re-amplified by it.
Acoustic feedback is based on the same phenomena, where a signal produced by the speakers of the system, is picked up by a microphone a similar power to the original signal, originating a *"whistle"*, which can induce damages to the system. To prevent this phenomenon power of the signal caught by the microphone shall be reduced, and to achieve this, the electro-acoustic gain shall be kept to a minimum and sound diffusion must be kept away from the microphones. If possible, the use of lavalier microphones is recommended.

6 Installation and calculation of the speakers

6.1 Sound Columns

Sound columns must be installed on the wall at a height of 1.5 meters, for seated people and 1.7 meters, for standing listeners. The power to apply to the column depends on the intended sound pressure as was explained in 4.2.

For a better sound diffusion the column will have to be slightly pointed down, and it must not be installed on corners, because corners tend to intensify low frequencies. If it is not possible to avoid installation in corners, low frequency excess must be compensated using amplifier tone controls or an equalizer.

6.2 Ceiling Speakers

Ceiling speakers are used in places were a false ceiling permits their installation. This type of installation is generally more expensive owing to the number of speakers used, but it is by far the type of installation that allows smoother sound diffusion uniformity, and a minimum probability of reverberation, because working levels are lower than in other distribution schemes.

The choice of the speakers depends on the installation place and the sought sound pressure level. For example, in a WC, speakers with a humidity-resistant polypropylene cone are strongly recommended.
The typical distribution of this type of speakers is shown in *figure 12*.

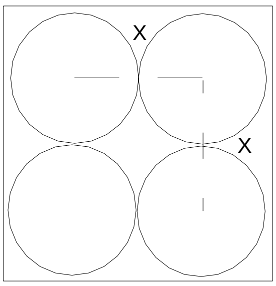


Fig.12- Typical distribution of the speakers

The circles represent the covering area of each speaker, and X represents in the distance between speakers.

For the calculation of the quantity of speakers to use it will be necessary to proceed to the following steps:

- ✓ Determine the ambient noise.

- ✓ Determine the sound pressure to be produced by the system.
- ✓ Determine the needed sound pressure uniformity.
- ✓ Calculate in the distance between speakers.
- ✓ Define the power to be applied to the speakers.

6.3 Determine the ambient noise

This point was described in **5.2**.

6.4 Determine the sound pressure to be produced by the system

With the ambient noise (*average*) already defined, the sound pressure that the system will have to produce will be the ambient noise plus 6 to 10dB.

6.5 Determine the sound pressure uniformity

The dispersion of the sound pressure will be, for the same power, the difference between the maximum sound pressure and the minimum sound pressure present in the local. For the majority of applications, a dispersion of 6 dB or ± 3dB is acceptable (*commercial airports, shopping centers, train stations, and so on*), but in places that require a higher pressure uniformity, for example in conference rooms, the use of 3dB dispersion is recommended.

6.6 Calculate the distance between speakers

The variation of the produced sound pressure along a linear aggregate of speakers in line is calculated using the following formulas, reasonably accurate for k>1 and related to *figure 13.*

$$\Delta P = 10\log_{10}\frac{8}{4+k^2}$$ *(Formula 3.1)*

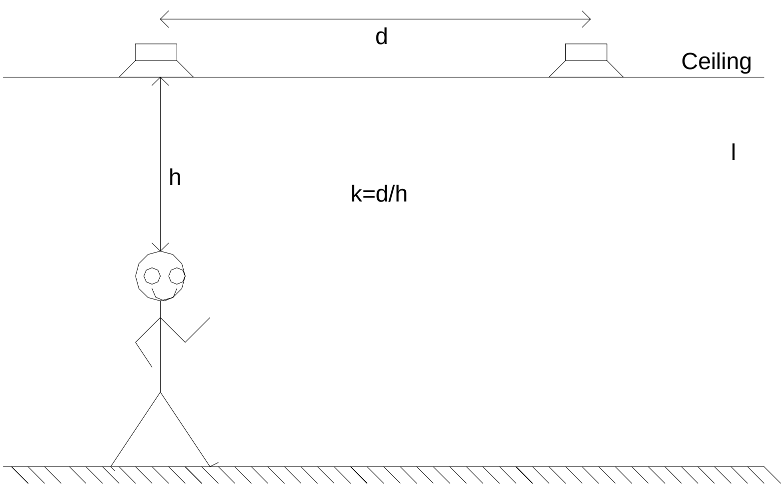


Fig. 13- Calculation of the distance between the speakers

The formula it 3.1 ignores the effect of reflections on the surrounding surfaces, and the contribution of adjacent lines of speakers. This formula leads to the following graph:

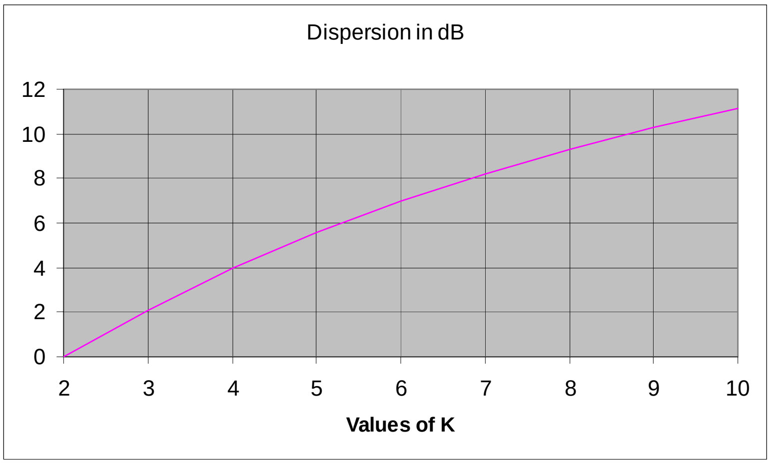


Fig. 14- Variation of sound pressure with distance between speakers/height factor

Example:

We want to calculate the distance between speakers in a room with 17 x 9 meters and a height of 2.7 meters. This room will be used for meetings and conferences.

We define as a maximum dispersion of 3 dB, and then the distance between speakers, it will be (considering that the average head height is 1.7, leading to an h of 1 :

$$d = 1{,}0x4{,}5 = 4{,}5$$

The total number of necessary speakers will be 8, as shown in *figure 15*.

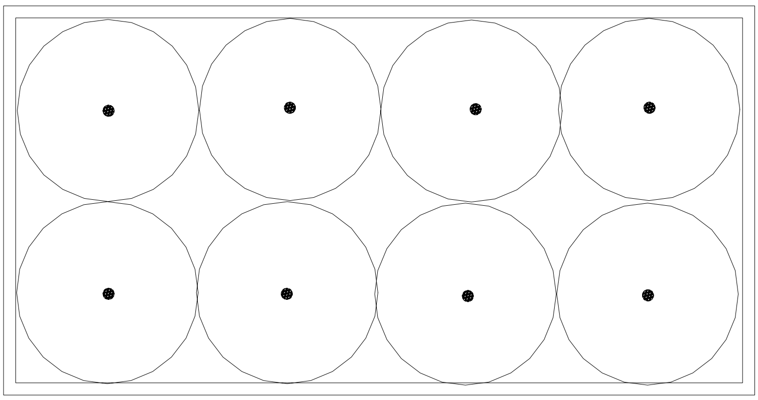


Fig.15 – Distribution of the speakers

6.7 Define the power of the speakers

To calculate the power of the speakers, we need to know the value of the sound pressure that the speaker must produce at 1 meter. Once the average value of the sound pressure that the system must deliver is defined, we have:

$$SPL_{\max} = SPL_{med} + \Delta P \qquad \textit{(Formula 3.2)}$$

The SPL_{max} will be obtained at the closest point to the speaker, as shown in *figure 13*.

Using the *figure 15* as an example, the speaker's SPL_{1meter} must be:

$$SPL_{1meter} = SPL_{máx} + 20\log_{10} h \qquad \textit{(Formula 3.3)}$$

Knowing the necessary SPL at 1 meter, we use the technical specifications of the speakers in order to choose an adequate speaker.

To calculate the power to be selected at the speaker, the following formula is used:

$$P_{Watts} = 10^{(SPL_{1metro} - SPL_{1W/1m})} \qquad \textit{(Formula 3.4)}$$

6.8 Cables to use in the installation

The following table will provide the maximum length of the cable for a given speaker or speaker group power given its cross-section. This table admits losses of 0.5dB (*10% in power*).

	Possible Amplifier			
	1151/1154			**1301**
	Power (W)			
Section (mm2)	50	100	150	300
0.25	325	163	108	54
0.5	650	325	300	108
0.75	975	488	325	163
1	1300	650	433	217
1.5	1950	975	650	325
2.5	3250	1625	1083	542

Fig.16 -Maximum length of the cables, in meters

As it was previously referred, in this type of system the speakers will be always connected in parallel.

This table reflects the following formula, which can be used for intermediate values:

$$L = \frac{10^{-\frac{q}{10}} V^2 S}{\rho P}$$

(Formula 3.5)

Where the:

L is the length of the cable in meters
q Is the admissible loss of power in dB
V voltage (100V)
S section of the cable in m^2
copper resistivity (1.673.10^{-8} .m)

7 Assembly verification of 100V line systems

Assembly verification of 100V line systems consists of:

- ✓ Cable verification
- ✓ Connection verification
- ✓ Speakers and projectors verification
- ✓ Line verification

7.1 Cable Verification

All the cables in the system must be secure and perfectly isolated.

7.2 Connection Verification

The connection between all the components must be secure and perfectly isolated. A simple wire of a cable leaned against another conductor can cause serious problems to a system. Isolate ALL the wires in the connections, even if not in use.

7.3 Speaker and Projector Verification

The speakers and projectors must be well secure and the selected power must be the correct.

Never switch the poles + and - between in the speakers connected in series/parallel. The swap will make that when a speaker pushes in the forward direction another one is pushing backwards, inducing acoustic pressures of opposite signal, leading to the cancellation of the emitted sound the speakers are close together. Observe polarities indicated in **3.1.1** and **3.1.2.**

7.4 Line Verification

The line verification will have to be made before connecting the amplifier, and consists of:

- ✓ Check the isolation
- ✓ Check the line impedance

7.4.1 Check the isolation

This verification intends to check the isolation of the line in relation to system ground. For this, an Ohmmeter must be used, with the scale set to MΩ. One of the terminals connects to system ground and the other one to the two shorted line conductors, as shown in *figure 17.* The Ohmmeter will have to indicate an infinite resistance. Because the working voltages are generally high, we do recommend the use of a high voltage Mega-ohmmeter (500V isolation test).

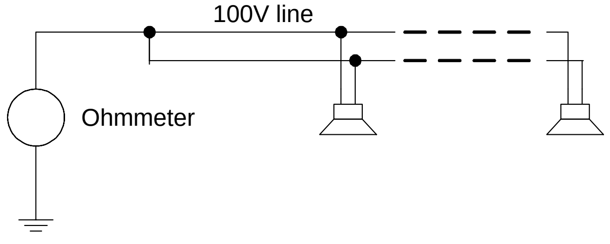


Fig. 17- Connection schematic to measure the isolation of the line

7.4.2 Check the line impedance

For this verification it is necessary to have:

- One signal generator, sinusoidal, 1V-10V, 1khz
- One AC Voltmeter
- One resistor (100 Ω - 10000 Ω)

The equipment is connected as shown in *figure 18*.

With the generator at 1kHz and the voltmeter with scale set to AC, the voltage at the terminals of the generator and at the terminals of the load are measured, and the values applied to the formula below:

$$Z_{line} = \frac{RVo}{Vi - Vo}$$ *(Formula 4.1)*

Where the:
Z_{line} is the impedance of the line
R is the value of the resistance
Vi is the voltage present at the output of the generator
Vo is the voltage measured at the line terminals

Fig. 18-Connection setup to measure the line impedance

This measurement must be done at 1 kHz, reason: because the impedance of the speakers changes strongly with the frequency, measurements taken at low frequencies (where the resonance effect predominates) or at high frequencies (where the inductive character and later the capacitive effect affect the measured values) may show up as erroneous ones.

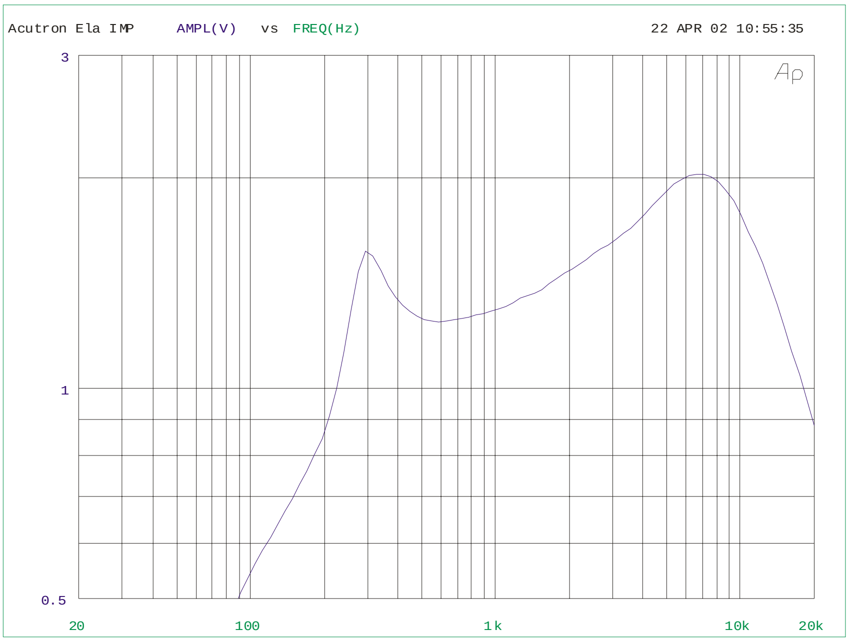


Fig. 19- Variation of the line impedance with the frequency

The calculated impedance is compared with the impedance of the system, obtained by means of formula 4.2, where the power will be the sum of the selected power in each one of the speakers connected to the line. The maximum value spread must be of 10%. If the value spread is higher than 10%, a new system checkup must be made before proceeding to a new measurement.

$$Z_{calculated} = \frac{10000}{Total\ Power} \qquad \text{(Formula 4.2)}$$