Design and implementation aspects of a small anechoic room and sound-actuation system.

Joel García Rodríguez

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Promotor:
Dr. Ir. Abdellah Touhafi
Tutor from Spain:
Dr. In. Armando Malanda

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ABSTRACT

Looking for future projects, the possibility to have optimum conditions to perform microphone tests, and any noise measurements of different small devices which will be constructed in the university, the need to construct an anechoic chamber was considered, so in this thesis a research about anechoic chambers was made.

An acoustic anechoic chamber is a shielded room designed for performing sound measurements under conditions close to free space.

Due to the difficulties that might entail the construction of a large anechoic room, we addressed the design of a chamber with smaller dimensions. This way we try to have an enclosure with conditions close to free field and to be able to perform the necessary tests with a low-cost construction. The research group wants to build an environmental emulator to test their sensor networks that are used for environmental noise study. By using multiple small rooms they want to emulate a bigger space.

This report summaries the design and construction of a small anechoic chamber, with a focus on microphone testing. Finally a low cost prototype of that small anechoic room will be built, in which we could perform the tests.

The prototype has been built with low cost materials that are MDF for the walls, and fiber glass for the absorbent material.

A sound actuation system has been designed too. The latter being made up of an amplifier and two speakers. The first speakers is a coaxial speaker for generating pink or white noise or sweeps. The second speaker is the tweeter to generate pure tones. For the coaxial speaker it was necessary to build a baffle. A baffle was designed in order to achieve the best performance of the loudspeaker.

Finally, the design was made for a small anechoic room. We took into account the Eigen-modes, the absorption, the insulation and predominant axis of measurement as major factors of consideration. So insulation, absorption and construction issues are detailed and addressed, creating a base for the experimental evaluation for a following project.

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Chapter 1. Report

1.1 Aims

1.1.1 Main Aim

The main aim of this bachelor thesis is to design and implement a small anechoic chamber and sound-actuation system, to tackle small electronic devices.

1.1.2 Secondary aims

– Small dimensions with a good dispersion of Eigen-modes, and easy to handle.
– Choice of an absorbent material to have a low cut-off frequency, and a wide frequency range to work.
– Look for an absorbent material with a low cut-off frequency.
– Assemble a good insulation solution to fulfill a NC10 (see ANNEX A) inside the anechoic chamber.
– Design and implement the sound-actuation system:
    – Choose a flat response loudspeakers and amplifier.
    – Design the baffle for the loudspeakers.

1.2 Introduction

An anechoic chamber is a room designed to stop depending on the purpose, sound or electromagnetic waves reflections. It also requires a high insulation from outside sources. These two conditions simulate an open-space of infinite dimension with almost no external influence. The anechoic chamber were originally used in the context of acoustics, but the radio frequency counterparts have also been in use to test antennas, radars, electromagnetic interference...

Anechoic chambers range from small compartments to ones as large as aircraft hangars. The size of the chamber depends on the size of the objects to be tested and the frequency range of the signals used, although scale models can sometimes be used by testing at shorter wavelengths.

In acoustics the anechoic chambers are commonly used to conduct experiments in "free field" conditions, being all the sound energy traveling away from the source without any reflection. In general, the interior of an anechoic chamber is very quiet, with typical noise levels in the 10–20 dBA range.

To achieve the "free field" conditions special shape absorbent material is used, being it wedges of a determinate length. The length of the wedges will determinate the frequency range in which we will have "free field" conditions. Those more usual materials to build those wedges are, mineral wool, polyurethane or melamine

The anechoic rooms which have that absorbent material in all the walls are called *full anechoic chamber (Fig. 1.2-1)* or most often *anechoic chamber*, as it absorbs the sound in all the directions, these have a metal grid over the floor wedges to hold all the devices to measure. When heavy items will be measured a different chamber is used, these are called *semi-anechoic chambers (Fig. 1.2-2)*, which don't have wedges in the floor.

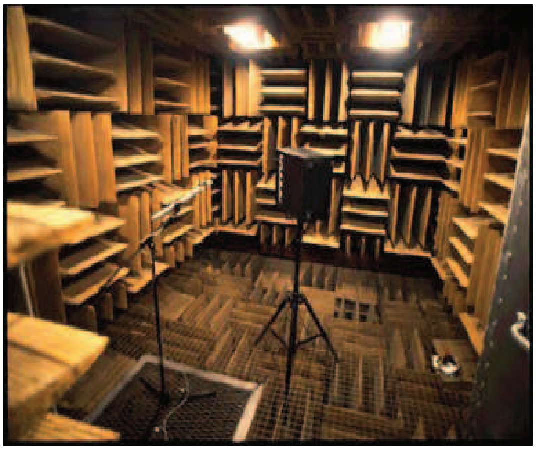


Fig. 1.2-1: Full Anechoic Chamber

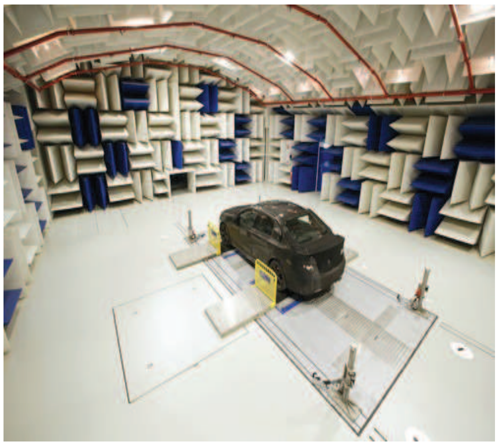


Fig. 1.2-2: Semi-Anechoic Chamber

1.3 General Characteristics

We have calculated several anechoic room sizes and considered the implementation of one specific size. A first anechoic chamber that was designed is a box of (inner dimensions) 2100 x 1500 x 1300 mm (width x length x height), with walls of 30 mm MDF boards. The absorbent materials are 30 mm depth wedges which give a cut-off frequency of 297 Hz approximately. One part of the box is open to engage the baffles by four locks. It contains a system of 8 spring, each hooked to the corners of the box, holding in that way the microphones in the middle of the chamber, at the axis of the loudspeaker.

Two loudspeakers, 15" coaxial driver and a plated diaphragm tweeter, will be mounted in two baffles.

The 15" Coaxial (15XA38 Nd) is a loudspeaker designed for compact bass reflex cabinet. The low frequency unit is a 350 W with a rated 8 ohms and 99 dB sensitivity at 1 m, with a frequency range response of 25 to 3500 Hz. The high frequency unit is a 90 W with a rated impedance of 16 ohms and a frequency range of 500 Hz to 20 Khz.

A filter (FD2XA) is necessary for the 15XA38 Nd loudspeaker, so a two way filter was chosen. Is a filter specially designed for the loudspeaker which we will work, coaxial 15XA38 Nd. It has a crossover frequency of 1.8 Khz, a low/high frequency impedance of 8/16 ohms and a power capacity of 600 W.

The tweeter chosen is a Plated Diaphragm with horn superior sound reproduction quality loudspeaker, with a extremely linear frequency response, and low distortion and a rated impedance of 8 ohms of 230 x 230 x 148 mm (the horn). Has a power of 80 W and sensitivity of 102 dB (1w at 1m). The frequency range of work is from 700 Hz to 23 Khz, but the cutoff frequency is 800 Hz.

The baffle for the coaxial is a 700 x 850 x 300 mm (width x height x depth) with a Bass reflex configuration, having a vent of 170 mm diameter and 140 mm length. Having a volume of 142.8 L the box tuning frequency (f_b) is 43.9 Hz, and the frequency where the SPL increase 3 dB (f_3) 49.5 Hz.

The tweeter being a high frequency loudspeaker do not need a baffle, neither a reinforcement on low frequencies by a Bass reflex configuration, as even is close in its back part, so there had not been problems with the rear wave. Despite this to simplify the operation method it was assembled in a close baffle of 700 x 850 x 300 mm (width x height x depth).

The amplifier is a 8 channel amplifier, with 90 watts per channel at 8 Ohms, and 130 watts per channel at 4 Ohms. With DIP switch control for clip limiters, high-pass filters to protect speakers (33 Hz or 70 Hz), bridge, mono and parallel operation configuration and high-performance Class AB+B circuitry. Low typical distortion from 20 Hz to 20 Khz (10 dB below rated power) less than 0.1% THD.

To make the tests Class 1 Gras microphone will be used as reference (microphones from the sound level meters Cesva and Norsonic) and the microphones to test will be which were designed in the laboratory.

1.4 Shape and dimensions.

The dimensions of the chamber are always a critical design factor, and more if it is a small chamber. It is a problem because in a room with parallel walls we can find constructive and destructive interference caused by the interference of the incident and reflected waves (Standing-waves).

To mitigate that problem, which in smaller chambers is not negligible like in the big ones, the best solution could be a non-regular shape, in spite of this recommendation a rectangular squared was chosen as the shape of the design, as it simplifies the construction of it, and, although this set-up is prone to standing waves, the fundamental frequency of resonance is usually low enough to be disregarded.

1.4.1 Dimensions due to the Eigen-modes.

A room acts as the sounding of a musical instrument. A speaker producing sound pressure variations will create vibrations according to the resonance frequencies of the room, producing some standing waves creating sound pressure levels depending on the position in the room. Standing in such a room will as such influence the standing waves and will change the measurements. Knowing that the number of Eigen-modes is unlimited and having a discrete distribution, the density increases with the frequency.

Those standing-waves can appear in three different forms:

- Axial: The mode is generated between 2 surfaces. Two values are zero.
- Tangential: The mode is generated between 4 surfaces. One values are zero. About 3 dB less of energy than the Axial
- Obliques: The mode is generated between 2 surfaces. Have not zero value. Approximately 6 dB less than the Axial

The values of the Eigen-modes depends on the geometry and the dimensions of the chamber, and its determination is very complex. The study is done using ondulatory acoustic model. When the chamber is parallelepiped-champed with totally reflective surfaces, we can simply calculate the Eigen-modes using the Rayleigh formula:

$$f_{k,m,n} = \sqrt{\left(\frac{k}{L_x}\right)^2 + \left(\frac{m}{L_y}\right)^2 + \left(\frac{n}{L_z}\right)^2} \qquad (1.4.1\text{-}1)$$

Where:

L_x, L_y, L_z are the dimensions of the chamber (m)

k, m, n can take integer values (0,1,2,3,4...)

Then a combination of k, m, n values gives one frequency and its Eigen-mode, which name is Eigen-mode k, m, n, i.e for k=1, m=0, n=1 the Eigen-mode is the 1,0,1.

Then looking at the formula, can deduce why the dimensions of the chamber are so important, if we choose dimensions that are the same or one the multiple of another the Eigen-mode could coincide and therefore could be a big concentration of energy in one frequency, with the corresponding sound intense "coloration". So is important to choose dimensions which give a uniform distribution of Eigen-modes, avoiding energy concentrations.

In the Fig. 1.4-1 we see a shadow part, which represent the possible relations between length and width for a rectangular room (supposing 1 meter high) for which the distribution of Eigen-modes could be as uniform as possible. But this does not mean that the distribution will be perfect and will not be problems with a possible coloration of the signal, because those can appear, having for example different Eigen-modes very close together, even no coincident, can also present problems.

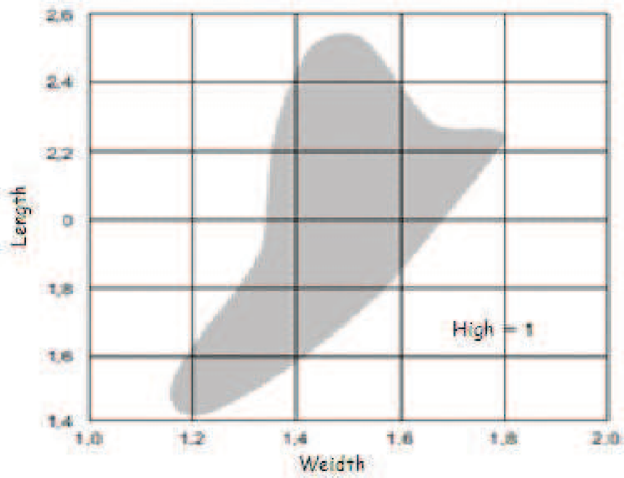


Fig. 1.4-1: Recomended relations between dimensions to have a uniform distribution of Eigen modes.

Moreover, and as mentioned above, the density of modes increases with the frequency. This means that from a certain frequency the concept of coloration does not make sense, as the high density of Eigen-modes is equivalent to an absence of it, by the fact that there is no longer discrete energy levels. In the (1.4.1-2) is shown the formula to calculate the upper limit frequency for which the density of Eigen-modes is so great that its influence is almost negligible.

$$f_{max} = 1849\sqrt{\frac{RTmid}{V}} \qquad (1.4.1\text{-}2)$$

Where:

RTmid is the average value of the reverberation times for the octave bands at 500 Hz and 1 KHz .

V is the volume of the room (m^3).

In the above expression can be understood why the effect of the Eigen-modes has a bigger effect as small is the room into consideration, as in our case. So as greater the Volume and greater the absorption these frequency will be lower. A possibility to minimize the effect of coloration is by equalization system, or if it is possible assembling resonator elements at problematic frequencies. Generally the Eigen-mode which effects have to be minimize are below 200 Hz.

Well, known that, as was said above the best option could have been to build a small box of a non-regular shape, but a rectangular cuboid was chosen, as it simplifies the construction, and the resonance frequency can be low enough to be disregarded So the shape is a rectangular box a good choice of the dimensions had to be made. So according to the necessity of a small volume, the Fig. 1.4-1 could not be used, because the volumes which it gives were a little bit big

Then, the longest distance was taken for the length, as measures in the length will be undertaken, and taking for the width and height values differentiated among itself, which would not give many problems in the modal distribution. So the dimensions chosen are 2100 x 1500 x 1300 mm (length x width x height). In the Fig. 1.4-2 can see the distribution of the axial Eigen-modes for our room. This simple approach of studying the axial-mode distribution has the advantage of paying attention to the dominant axial modes knowing that the weaker tangential and oblique modes can only help by filling the more widely spaced axial modes.

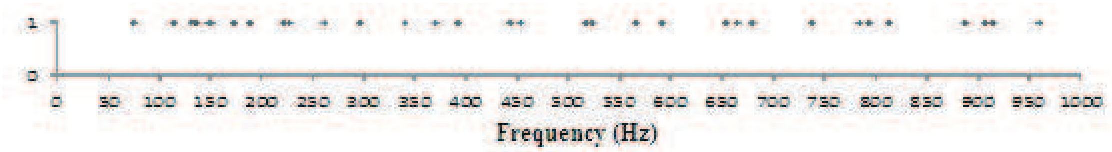


Fig. 1.4-2: Axial Eigen-Mode distribution for our small room

Why to take the largest dimension for the length? One of the reasons is the larger the length, the lower the resonant frequency will be. In this way the resonant frequency of the length will be so low that it will be disregarded. So looking to formula (1.4.1-3) the resonant frequency will be 74 Hz, which is low enough for the frequency range in which we are going to work.

$$f_c = n\frac{c}{2d} \qquad (1.4.1\text{-}3)$$

The other reason is because due to the microphones placement, which is explained in chapter 1.9.3Microphones page. 37.

1.4.2 Dimension due to the volume of the devices to measure.

The volume of the chamber (V_c), according to the ISO 3745 standard, must be at least 200 times the volume of the sound source which will be measured. Given that the devices to measure are going to be small, we are not going to have any problem with that restriction.

1.5 Absorbent material

When a sound wave hits a wall covered by an absorbent material, first of all a small part returns to the air from the surface of the acoustical material. Also there is a heat loss in the air, but that is just appreciable in the high frequencies. The part of the sound which penetrates in the acoustical material is going to experiment a heat lost by the frictional resistance which the acoustical material offers to the vibration of air particles. When the sound strikes against the surface of the wall, some part can enter in the wall being also dissipated by the frictional resistance of the wall, and the other is reflected, still staying in friction with the absorbent material which covered the wall. In that process the sound wave is getting weaker and some part of it can returns. That last part is what we want to avoid, having a material which some characteristics such that the sound wave never returns, at least for some frequencies (Fig. 1.5-1). That means to have a 1 in the absorption coefficient.

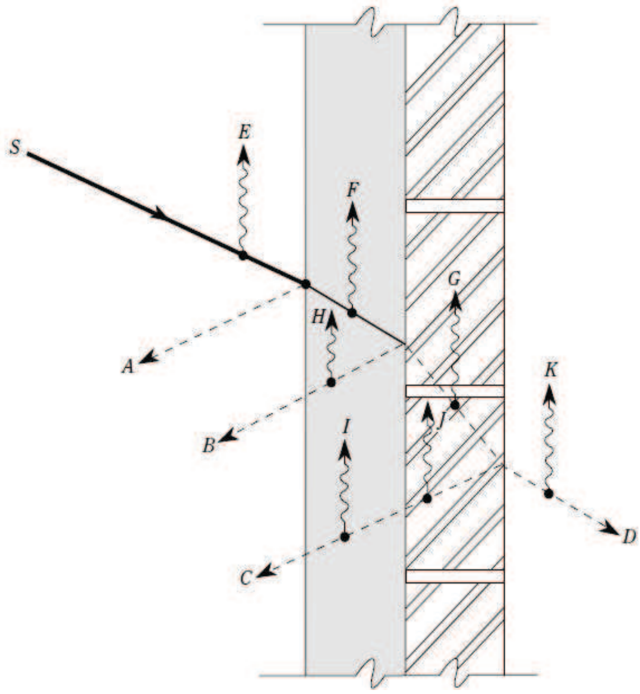


Fig. 1.5-1: A sound ray impinging on an acoustical material

1.5.1 Reverberation Chamber Method

The ISO 345:2003 standard "*Acoustics*: *Measurement of sound absorption in a reverberation room*", describes one of the methods to calculate the absorption coefficients of a material, based on diffuse field condition. It consists in introduce the material of which want to calculate the absorption coefficients in a reverberation chamber which reverberation time is known. Has to be a large room with highly reflective walls, ceiling, and floor, having in that way a very long reverberation time (the longer it is, the more accurate the measurement). With the material placed in the floor the new reverberation time is measured and with the area of the material, the absorption coefficients are obtained. Then with that method is usual to get values greater than 1. The diffraction of sound from the edges of the sample makes the sample appear acoustically, of greater area than it really is. That does not mean that the material absorbs more energy that incise to it, neither that it will absorb 100% in all conditions. In fact, it is likely that almost never does. The absorption coefficient of a material depends on, and much, the incidence angle. This is understandable seeing that for horizontal incidence, the wave travels parallel to the material, the coefficient is 0. Normally, as the angle of incidence increases the absorption increases. And the ISO 354 gives a global value (integral from 0 to 2π) called diffuse field.

Another way would be to measure in an impedance tube, the absorption at different angles, but that is another way to measure the coefficients and material traders do not provide that information.

Other question is, at what frequency will absorb 100%? Well, probably in anyone, but we have to be aware that a band of frequencies (the data is in octaves) contains many individual frequencies, so sure, some are absorbed much more than others, even very contiguous. Due to the fact that we measure in octave bands some absorbed frequencies are not detected as such.

We are now concerned about the interpretation of the absorption values of the materials that we use. Since we don't have the conditions of a diffuse field in our chamber, the best option is to calculate the reverberation time. This by emitting a logarithmic sweep and measuring the reverberation time which allows us to estimate the frequency from which up we could have free field conditions.

To make a good choice of the absorbent material, we have to know how the absorption coefficient can change depending on some characteristics or assemblies:

- The porosity.
- The density.
- The thickness.
- The distance between the material and the wall.

1.5.2 Variation of the absorption depending of the porosity

The key word in this discussion of porous sound absorbers is interstices. It is simply the space between two things. If a sound wave strikes a porous material, the sound energy sets the material fibers vibrating. The fiber amplitude will never be as great as the air particle amplitude of the sound wave because of frictional resistance. Some sound energy is changed to frictional heat as fibers are set in motion, restricted as this motion is. So for a good absorption a very porous material is better, as the easier penetration of the sound wave will be greater, increasing the absorption coefficient of a it based on it porosity, that is why the best absorbents are the fiber glass or the rock wool.

Thus the best absorbents are the fiber glass and the rock wool, due to them high porosity, but it has some inconvenient due to that high porosity, like the delicacy and the particles than detach with the time, in the case of the fiber glass really dangerous particles. That is why, and also some esthetic characteristics, never is used over a wall, being used behind some protection, like and a less delicate material.

1.5.3 Variation of the absorption depending of the density

Know that the high porosity help to the incidence of the sound wave, now have to know how the density of it will help to the absorption of the sound.

If the density of a material is low, the frictional resistance which impose the material, and accordingly the heat lost will be low, therefore the absorption of the material will be low. Otherwise if the density goes up the heat lost imposed by the friction with the material particles will be increase with it, but just till one value, where the absorption can decrease, because the material is so dense that could be a less penetration of the sound wave, being it reflected. It is worth mentioning that the recommended densities are between 40 Kg/m^3 and 100 Kg/m^3, never being higher than 100 Kg/m^3.

Have to say, that some materials are made by layers of the same material but of different densities, increasing as we get closer to the wall. With that the impedance of the room will be better, helping to the incidence of the sound wave within the material having a low density in at the beginning of the material and increasing it as it going deeper, increasing in that way also the absorption (typical in the anechoic wedges).

1.5.4 Variation of the absorption depending of the thickness.

Another factor to consider when choosing an absorbent material is its thickness, as the absorption depends on it, may be higher or lower for certain frequencies.

First of all have to take into account that the absorption is low for all those frequencies where the thickness of the material is much more lower than the wavelength of that frequency $D \ll \lambda$

We assume that the material is attached to a rigid wall, so that force to the air particles near the wall being with almost null velocity, increasing gradually as we move away from the wall, but being almost null nearby it (Fig. 1.5-2). As the speed of the sound is so low within the absorbent material, the friction with the material particles is minimal, producing a minimum transformation in heat energetic.

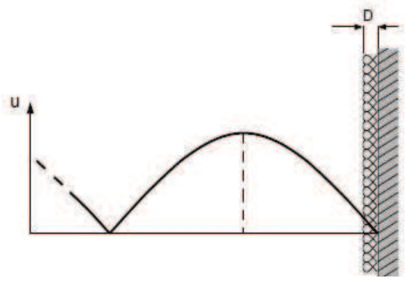


Fig. 1.5-2: Evolution of the amplitude of speed "u" particles of air depending on the distance to the rigid wall (D<< λ)

Instead, if the frequency increases the wavelength λ decreases, so the thickness D will be higher in comparison with the λ, increasing with it the velocity within the material, and the dissipated energy, increasing in this way the absorption for that frequency; being the maximum values of absorption for the frequencies where D is equal to λ/4 or an odd multiple of λ/4. So if we want to have maximum absorption for a wide range of frequencies, we must choose a thickness for which the minimum frequency meets:

$$D = k\frac{\lambda}{4} = k\frac{c}{4f} \qquad (1.5.4\text{-}1)$$

In the Fig. 1.5-3 is clear how the velocity of the sound wave increases within the absorbent material. So if the thickness is small the condition of high absorption be fulfilled for high frequencies, being necessary a high thickness to have a good absorption for low frequencies. Other way to justify the incrementation of the absorption with the thickness, is that the sound-wave travels more distance within the absorbent material and its velocity has a higher values within it.

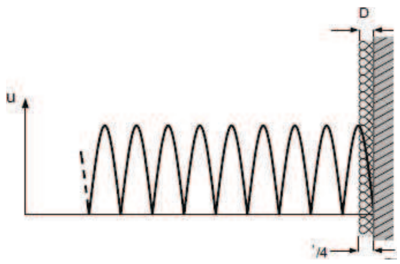


Fig. 1.5-3: Evolution of the amplitude of the speed "u" particles of air depending on the distance to the rigid wall (D= λ/4)

1.5.5 Variation of the absorption depending of the distance between the material and the wall.

As explained in the previous section to achieve a high absorption at low frequencies high thickness are assumed to be necessary. Despite this another way to assemble the material can be. This is to assemble the material at some distance from the wall, know that the maximum absorption will be for the frequency, for which the distance "d" is the same that λ/4, matching the distance with the highest velocity of the sound-wave at λ/4 (Fig. 1.5-4).

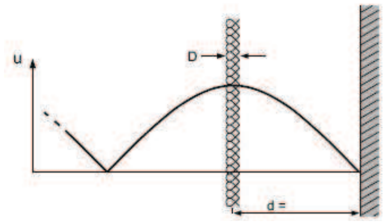


Fig. 1.5-4: Amplitude of "u" speed particles of air within a absoprtion material separated λ/4 form the wall

As bigger "d", less will be the frequency for which the absorption will be higher. So to increase the absorption at low frequencies is necessary to increase the distance between the material and the wall, having unfortunately a decrease at high frequencies.

On the other hand, in the practice is recommended to place a material in zigzag in order to have a variable distance between the material and the wall, and thus, smooth out the possible big differences that could appear between the absorbent coefficients.

1.5.6 Anechoic Wedges

Now should focus on a special material, which is used specially in anechoic chambers, and which has all the characteristics explained above, the anechoic wedges.

Those are usually built-up by some "hard" material, at least harder than the wools (glass and rock), these are foams like melamine, polyurethane, graphite. The reasons to do not build the wedges in fiber wool, despite being a material with high acoustic properties, because its high porosity and density, are the delicacy (even more in wedge shape), dirt and danger of its particles. So that is why other materials are used, even having worse acoustic features, which are not going to be a problem how we will look below.

Why the shape of wedge? This structure behaves like a sound snare. The sound incises on the surface of a wedge and it is reflected several times in the wedge and in the next (Fig. 1.5-5). The result is an increase in the effective area of three or more times, increasing in that way the absorption of a material that in flat shape could be worse like would be the fiber glass.

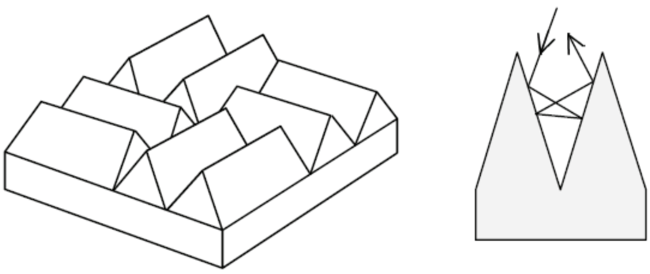


Fig. 1.5-5: Anechoic wedges, and the way of how it absorbs the sound.

Sometimes these wedges are built-up by different density materials, being the less dense in the peak of the wedges (helping to the incidence of the wave) and the densest in the base (to improve with it the absorption). Also the variability between the distance of the material to the wall, the form of wedge, helps to have a smooth absorption coefficients, avoiding the possible high values for some frequencies and low values for other frequencies which could give a flat material. To improve them absorption at low frequency a small space of air is left between the wedges and the wall.

The absorption of the wedges is proportional to its length, so as longer wedge higher will be the absorption for low frequencies, being the 100% absorption for the frequency, which the length is equal to a quarter of it wavelength (1.6.1-1). So a compromise have to be taken between the length of the wedges and the cut off frequency which we want, having the less cut off frequency with a length which can fix inside our box, not being them so long. All that is explained in the next chapter (1.6.1Due to the length of the wedges).

1.6 Low Cut Off Frequency.

Inside a chamber, can have two different cut off frequencies, the first one is which the anechoic wedges gives, due that to the length of the material, and the other one due to the geometry of the chamber. That last one is defined in the ISO 3745, and is directly related to the operational method for measuring sound pressure levels within large chambers. Our method will not be the same, but we can use the formulas, adapting them to our method and calculating the cut off frequency which will have in our small room.

1.6.1 Due to the length of the wedges

According to the ISO 3745 the depth of absorbent material (wedge + air) have to be quarter of the wavelength corresponding it to the center frequency of the lowest frequency band of interest (1.6.1-1). The small separation from the wall is to have a small improvement at the absorption in low frequencies.

$$f=\frac{c}{\lambda}=\frac{c}{4\cdot lc} \qquad l_c \geq \frac{\lambda}{4} \qquad (1.6.1\text{-}1)$$

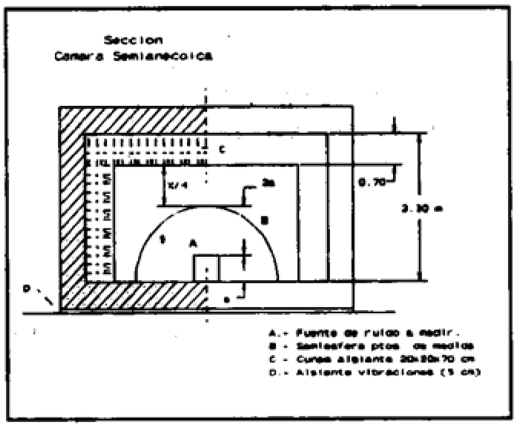
Then looking at the dimensions which we managed and to some restrictions that the standard imposes (are explained in the next chapter), wedges of some depth had to be chosen, having a good relation between the size (could not be so big, because the dimensions of the box) and the cut off frequency which had to be as lower as possible to have a wide range of frequencies to work. So having that in account wedges of 30 cm were chosen, which theoretically should give an absorption equal or greater of 0.99 in a range frequency from 287.5 Hz up to 20 Khz.

1.6.2 Due to the geometry of the chamber

To determinate the acoustic power of a source in accordance with the ISO 3745 the NPS of this have to be measured in a region such that:

- Enough away so that, that is not within the near field
- Enough away from the walls so that the partial absorption which have the wedges does not disturb the field.

This is expressed according to the following constraints:



- The distance between the microphone and the sound source must be greater than 2a, being "*a*" the largest dimension of the source, and never less than 1 meter.
- The distance between the microphone and the chamber walls must be greater than $\frac{\lambda}{4}$, being λ the wavelength of the lowest frequency to be measured.

Is true that the main purpose of our chamber is not to measure the NPS in a region to determinate the power of a source, although with the features of the chamber may perform that measure in a different way. But we will test microphones which also have to be in some area, separated enough from the wall and the loudspeaker to have a good results. So having those remarks in account will know which will be the cut-off frequency of our configuration.

.According to these constraints we take the formula to calculate the geometric cut off frequency, which for the example of the image for the height the formula is:

$$a+2a+\frac{\lambda}{4}+l_c=h \qquad (1.6.2\text{-}1)$$

For the width and the length the formula is:

$$a+4a+\frac{\lambda}{2}+2\cdot l_c=l\,,w \qquad (1.6.2\text{-}2)$$

These are the formulas for the operating method inside big anechoic chamber of the standard ISO 3745. In our we will undertake our tests in the length of the chamber, which give us the larger dimensions to have a good separation between devices (1 meter) and from the walls. So our formula will be different.

According to this, the formula to take our lowest cut-off frequency is like a mixture of the previous two:

$$a+2a+\frac{\lambda}{4}+2\cdot l_c=l \qquad (1.6.2\text{-}3)$$

- $2a$ Instead $4a$ because we are not going to take the distance forward and backward, only forward.
- $\frac{\lambda}{4}$Remains from the first because we need that separation from the wall only in the opposite wall (he wall near the microphone), not behind the source.
- l_c Remains from the first because the loudspeaker will be attached to the wall, being round of the wedges. In the case of the measurement of a sound source, the formula will change to $2l_c$, because the source will be in front of the wedges.

This chamber is 2100 x 1500 x 1300 mm (length x width x height), with an absorbent wedges of 30 cm long, so the dimension of the useful part of the box are 1500 x 900 x 700 mm (width x length x height).

- The wedges have a 287.5 Hz cut off frequency.
- The geometric cut off frequency would be calculated for the possible operative method to calculated the power of a sound source, to test the distance will be greater so will be even lower. So the frequency will be for a source or (10cm length) 61,25 Hz. (Not important because the limit is the cut off frequency of the absorbent material, that is higher).
- The real cut off frequency of the chamber would be the higher, 287.5 Hz.
- The distance between the loudspeaker and the microphone will be 1.7 m, because the microphone has to be $\frac{\lambda}{4}$ from the wall separated, in this case (30 cm), but on the other side the loudspeaker is not in front of the wedges, but attached to the wall.

  For the case of a measurement of a sound source power, the distance will be 1.4, also a distance which will be good enough to have a wide frequency range to measure the power of a source.

1.7 Airborne Insulation

The sound insulation is to prevent its spread by reflective and absorbent materials. It consists in get a big reflection factor. To achieve this, have to stand in the sound way a material which impedance (Z) has to be very different to the impedance of the medium through the sound is traveling g.e. airborne (low impedance), walls (high impedance).

For our purpose we need to insulate the box to fulfill a NC10 curve inside it. To achieve that we will study two ways to insulate, the first one, with a single wall insulation, which could provide an enough insulation for middle-high frequencies, and a double wall insulation with what wanted to achieve and improvement in low frequencies insulation. Without have to put a really thick single wall.

1.7.1 Measurement Indices

There are several standard indices to quantify the airborne sound insulation. Consider the most commonly used:

- Sound insulation (D): The difference in sound pressure levels between the local noise level where the source (L_1) and the place where sound is received (L_2).
- It is estimated by the expression:

$$D = L_1 - L_2 \qquad (1.7.1\text{-}1)$$

- Sound insulation standard (Dn): The difference in sound pressure levels between the local transmitter and receiver, but taking into account the influence on the level, holds the reverberation. In the receiver local, if there is a high reverberation, the value of sound level is greater than would be expected due to the insulation caused by the wall, so that the sound insulation is reduced. Otherwise occur in the case of high local absorption in the receiver local: low reverberation.

Therefore, the standard sound insulation for a given frequency between two rooms is calculated using the expression:

$$D = L_1 - L_2 + 10\log\left(\frac{T}{0.5}\right) = L_1 - L_2 + 10\log\left(\frac{10}{A}\right) \qquad (1.7.1\text{-}2)$$

Where:

T = reverberation time frequency receiver for local consideration (s).
A = equivalent absorption area of the local receiver to a frequency (m^2).

- Transmission Loss (TL): This index is generally used for laboratory measurements (cold transmission) and is defined as:

$$TL = 10\log\left(\frac{W_1}{W_2}\right) \qquad (1.7.1\text{-}3)$$

W1 and W2 being the incidence power on the sample and transmitted it. In conditions of diffuse sound field, can be evaluated by the formula:

$$TL = L_1 - L_2 + 10\log\left(\frac{S}{A}\right) \qquad (1.7.1\text{-}4)$$

Where:

S = Surface of the test sample (m^2).

A = Equivalent absorption area of the receiver local for a frequency (m^2).

How we do not fulfill diffuse field condition, we cannot use that last parameter. So the calculations will be done with the D (Sound Insulation) and Dn (Sound Insulation Standard).

1.7.2 Back-Ground Noise

To know how high the insulation have to be, first of all a background noise measurement was made. The measurements were made with the Sound Level Meter & Spectrum Analyzer CESVA SC310; two measurements in real time analyzer (1 in octaves and other in third octaves), and the other in sound level meter to see how the background noise changes with the time.

The octave band measurement was realized to know which NC curve the background noise of the lab fulfilled. The other one, the third octave measure was made, due that the insulation calculations are always made in third octaves, so will be convenient to have the energy which we have in the lab to know how much have to insulate per third octaves to fulfill the NC10 inside the small room.

As was expected, a noise with a high level in low frequencies was measure, probably because the computer ventilators and the heaters, and going down as the frequency increase, due to the absorption at high frequencies is always higher, main reason due to the absorption of the air.

Looking at the Fig. 1.7-1 we see that the background noise fulfill a NC25, can go to a NC30. These variations are due to that the measure was made when the people were working in the laboratory. All the impacts were small, with a high frequency components, so will not be a big problem for the insulation, unlike if they were noise with many low frequency components, which would be more difficult to insulate. Despite this minor impacts, during all the time as we can look on the figure Fig. 1.7-2 the noise is always around 30 dB.

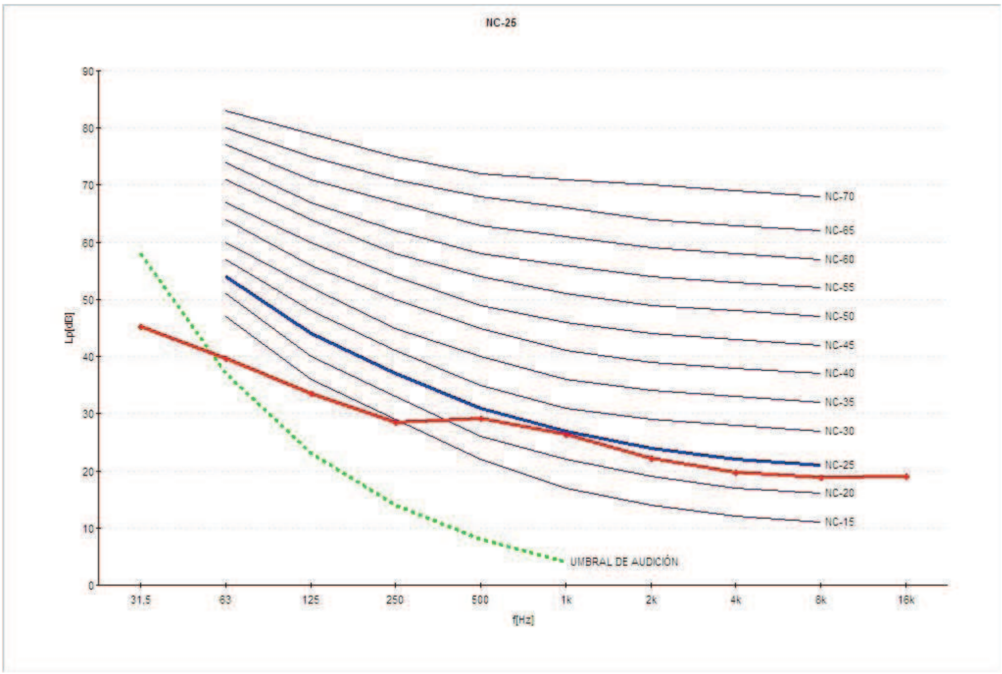


Fig. 1.7-1: NC25 Curve of the background noise of the laboratory

These punctual noises are also reflected in the graphic of the temporal variation of the noise shown in the Fig. 1.7-2, as said looking how the noise is always around 30 dB, being some increments in some points due to the punctual impacts of the people works in the laboratory. Can see how the impacts are formed of high frequency components, looking that for the high frequency lines, the increase is higher than for the lower ones. Also is possible to see how in low frequencies the energy is higher than in highers, as was said before (blue line). The thick red line represent the overall equivalent continuous sound level with integration time T (1 s) and A frequency weighting.

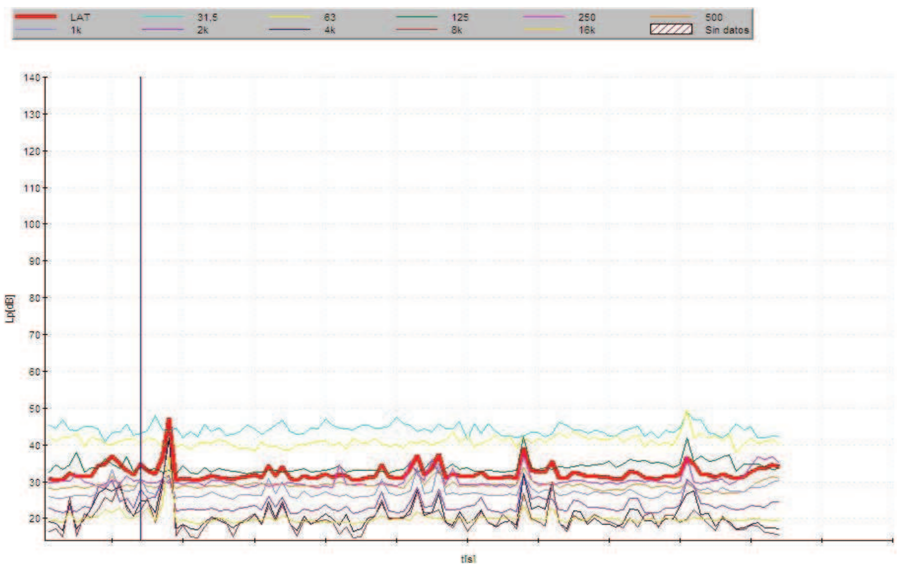


Fig. 1.7-2: Variation of the background noise of the laboratory with the time.

Finally in the Fig. 1.7-3 we can see how the energy is distributed for each third octave frequency band, looking again how in low frequencies the energy is bigger decreasing as the frequency increase. Could said that till 63 Hz the energy is around 40 dB, and from 63 Hz up 30 dB decreasing the energy with the frequency till be around 20 dB.

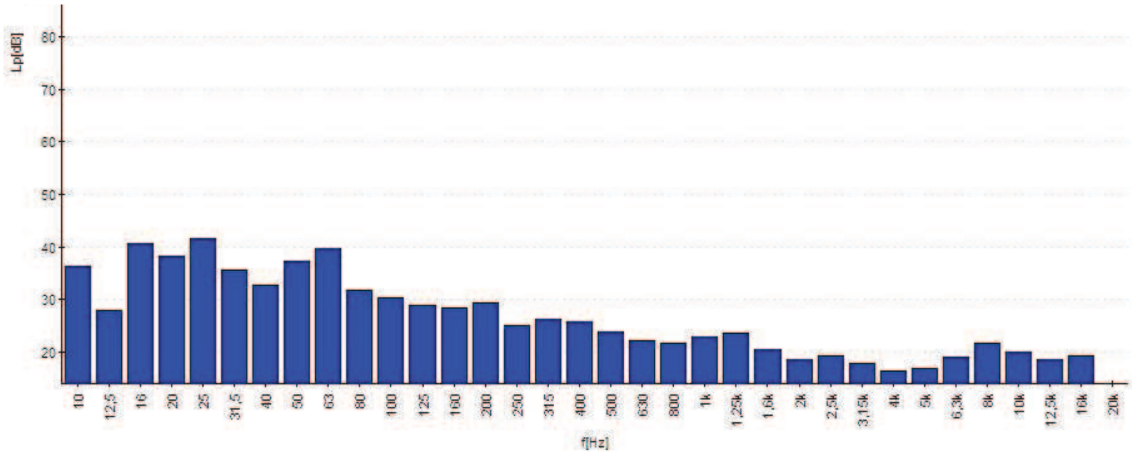


Fig. 1.7-3: Energy distribution for each third octave band of laboratory background noise

1.7.3 Single Wall Insulation

Single wall means which is formed by several independent walls, i.e. need not be a homogeneous wall (single material), but must meet the same points on a normal does not change their mutual distance when the wall vibrations do.

1.7.3.1 Law of mass and frequency

For a single wall, the law of mass and frequency indicates that the sound insulation is greater the larger the surface mass (mass per unit area), that is, heavier, and also higher for high frequencies.

The expression of this law is:

$$D = 20\log\left(\frac{w \cdot m}{2 \cdot Z}\right) \qquad (1.7.3\text{-}1)$$

Where:

w = Pulse (w=2πf) (Hz).
m = Surface mass (Kg/m^2).
Z = Air acoustic impedance (Kg/m^2).

Theoretically, this law tells us that doubling the mass is an improvement of 6 dB in isolation.

This experimental law is therefore not absolute, but approximate, although widely used to give a first idea of the acoustic behavior of a wall.

1.7.3.2 Single wall real insulation

The mass law is only true in a frequency range that is determined by two characteristic frequencies of a real wall and the environment which are not law-abiding masses, with a noticeable reduction in sound insulation.

- The system's natural frequency (f_0) which depends on the mass of the wall and on the perimeter mounts.

$$f_0 = \frac{1}{2\pi}\sqrt{\frac{k}{m}} \qquad (1.7.3\text{-}2)$$

- The coincidence or critical frequency (f_C), where the incident waves coincide in frequency with the longitudinal waves of bending of the wall.
- This frequency depends solely on the wall material and thickness, in words

$$f_c = \frac{c^2}{2\pi \cdot d}\sqrt{\frac{12\rho}{E}(1-\mu^2)} \qquad (1.7.3\text{-}3)$$

Where:

c= Airborne sound speed (m/s).
d= Wall thickness (m).
ρ= Wall material density (Kg/m^3).
M= Poisson elasticity coefficient of the material.
E= Young elasticity modulus of the material (N/m^2).

- To avoid the phenomenon of the coincidence frequency, two sheets of different thickness and density can be used, having then two different coincidence frequencies, so the loss of sound insulation in each of them will compensate the other, because the coincidence frequency will be different.

It is noted that there are three areas where sound insulation is governed by different factors, such as shown schematically in Figure.

- The area of "domain of elasticity" ($f < f_0$), which generally corresponds to very low frequencies and isolation down to "f_0", which is almost zero.
- The area of "domination of the mass", which itself is governed by the law of mass, characterized by $f_0 < f < f_C$ approximately.
- The area of “damping domain", which corresponds to $f > f_c$, where the insulation decrease considerable to f_c and considerably increases from this value in a progressive manner. In this area, the factor that governs the insulation variations is the damping.

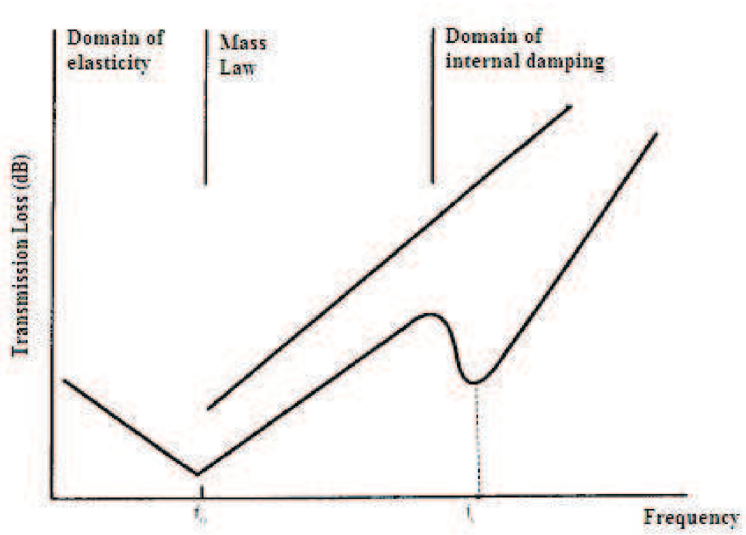


Fig. 1.7-4: Acoustic Insulation of a single wall

1.7.3.3 *Theoretical* Calculations for Single wall insulation

Due to the lack of information about the boards which we use an approximate calculation of a single wall insulation value was made. Know that the graphic represent just and ideal insulation of a wall. But have to take in account that in the low frequencies will be the system's natural frequency and in high frequencies the phenomenon of the coincidence. The calculations are made for boards of 30 mm thickness with a density of 690 Kg/m^3. But to build the box, two different sheets of 18 and 12 mm and 720 and 730 Kg/m^3 respectively have been used, being the 18 mm sheet the outer sheet of the wall, because that has a higher surface density so the insulation will be a little bit higher in a first step. Know that the insulation of two sheets will be the approximately the same because the surface mass of both is equal to the surface mass of the 30 mm board, the difference in dB can be insignificant (1 dB).

<table>
<tr><th>Thickness (mm)</th><th>Density (Kg/m^3)</th><th colspan="2">Surface Mass (Kg/m^2)</th></tr>
<tr><td>30</td><td>690</td><td colspan="2">20,7</td></tr>
<tr><td>18</td><td>720</td><td>12,96</td><td rowspan="2">21,72</td></tr>
<tr><td>12</td><td>730</td><td>8,76</td></tr>
</table>

Fig. 1.7-5: Demonstration of the similarity of the surface mass of a board of 30 or a composite of two 18 and 12 mm

So in the Fig. 1.7-6 can see the insulation values for 30 mm thickness wall.

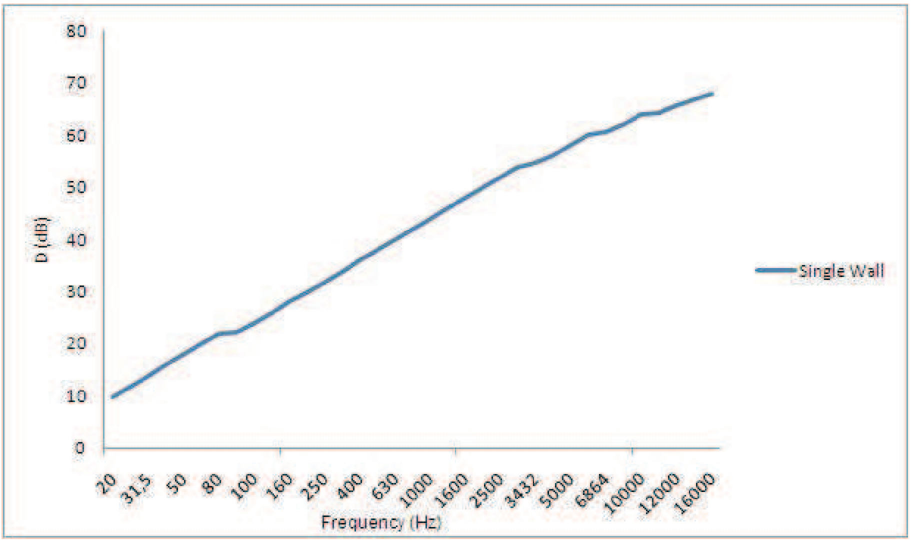


Fig. 1.7-6: Ideal Insulation of 30 mm board stimated by the law of mass and frequency

Can be looked that the insulation for very low frequencies is not high and maybe not enough for the noise which we have in the laboratory. Despite this, for the frequency range which we will work (still have to determine, but probably the lower limit higher than 63 Hz), and the background noise that there is in the laboratory for that frequency range, theoretically one wall insulation will be enough. Having 30dB till 63 Hz, frequencies for which we could have and insulation around 10-20 dB, would have inside of the box a level of approximately of 20 dB, a value which fulfill the NC10. For higher frequencies the energy per third octave band goes from 30 dB down till 20 dB, so with and insulation value of 40 dB the level of the background noise inside the small room will be really low, around 0 dB, which is perfect for our purpose. Anyway those are theoretical calculations so to improve the insulation, a study of double wall insulation was made, in case it is necessary.

1.7.4 Double Wall Insulation

To improve the insulation characteristics, specially at low frequencies, can divided “m” mass single wall in two sheets of mass $m_1 + m_2 = m$ separated a “d” distance, the set offers sound insulation than single wall of equivalent mass.

The analysis of insulation, in this case leads to the apparition of frequencies around which there is a strong reducing insulation. In this case it is the "natural frequency of the system” and “cavity frequencies", which give rise to areas dominated by different factors of influence.

1.7.4.1 The system's natural frequency (f_0)

Refers to a set of masses m_1 and m_2, attached by a spring of stiffness K. This system of mass-spring-mass, with the ability to vibrate, has its own resonance frequency which is defined by the following expression:

$$f_o = \frac{1}{2\pi}\sqrt{K\frac{m_1+m_2}{m_1 \cdot m_2}} \qquad (1.7.4\text{-}1)$$

Where:

K= Rigidity of the separating medium (N/m^3)
m_1, m_2= Surface mass (Kg/m^2).

The separating medium can be formed by air, a certain material or a mechanic system. In the case of be formed by air, the resonance frequency is given by the expression:

$$f_o = 60\sqrt{\frac{1}{d}\frac{(m_1+m_2)}{m_1 \cdot m_2}} \qquad (1.7.4\text{-}2)$$

Where:

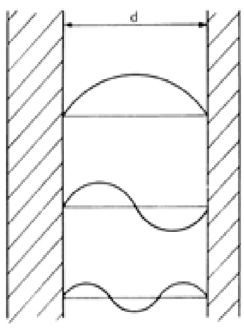
d= Thickness of the air gap (m).

In lower frequencies than f_0 the wall works like a single wall with a surface mass equal to the sum of the masses of both plates. To address the loss of insulation caused by the natural frequency, it is best to design a double-wall so that f_0 is a very low frequency cannot perceive, or it is in a frequency range that will not interfere with the activity undertaken. Normally it is intended that this frequency is below the measurement range (100 Hz).

1.7.4.2 The Air Separation Between Double Wall Layers

The air that separates the double wall creates a series of stand-waves that will generate a high loss of sound insulation of the enclosure. The location of these frequencies depends on the separation between the two sheets. According to the expression:

$$d = n\cdot\left(\frac{\lambda}{2}\right) = n\cdot\left(\frac{c}{2f}\right) \qquad (1.7.4\text{-}3) \qquad\qquad f_c = n\frac{c}{2d} \qquad (1.7.4\text{-}4)$$



Where:

n= 1, 2, 3…
c = Airborne sound speed (m/s).
d= Thickness of the air gap (m).

For these frequencies, the system behaves as a single mass $m_T = m_1 + m_2$, as the two sheets couple acoustically disappearing the insulating effect of the double wall.

To solve this problem we must fill the air gap with a sound absorbent material (i.e. rock wool or fiber glass). This absorbent material eliminates the resonances of the separation, converting sound energy into heat and hence increasing the insulation of the system. Obviously the higher sound absorption coefficient of the material introduced the higher sound insulation improvements to the partition.

1.7.4.3 The phenomenon of coincidence

Is a typical effect of simple walls. This phenomenon can also occur in double walls, as a double wall is composed of 2 simple walls. The area of coincidence is an area of low insulation, that occurs because the the incident waves coincide in frequency with the longitudinal waves of bending of the wall.

If we have a double wall with 2 identical sheets, the frequency of coincidence is a problem, because both partitions have the same f_c and therefore none of them prevent the passage of sound through this frequency.

To solve the problem just have to use 2 different partitions, with different mass and different surface bending rigidity, i.e. partitions use different materials or use partitions of the same material but different thicknesses. Thus, the frequency of coincidence of a partition (f_{C1}) will be different from the other partition (f_{C2}), so the loss of sound insulation in each of them will compensate the other.

1.7.4.4 Theoretical Calculations for double wall insulation

The calculations were made for two sheets of 30 mm, but in the same way that for the single wall each wall is composed of two sheets of 18 and 12 mm.

In the Fig. 1.7-7 can see that the f_0 was obtained in contrast to the single wall insulation case, being it 83 Hz, not known the value of insulation which could had in that frequency but being that almost zero (a value of 7 is shown just to indicate that in that frequency the insulation will be really low).

Also the frequencies which will generated inside the air gap has been obtained. In that case the values in the graphic are calculated for an air cavity not filled of any absorbent material, value (with absorbent material) which could be obtained in a experimentally way. The coincidence frequency also has been tried to avoid with two sheets of 18 and 12 mm but those frequencies are not shown in the graphic.

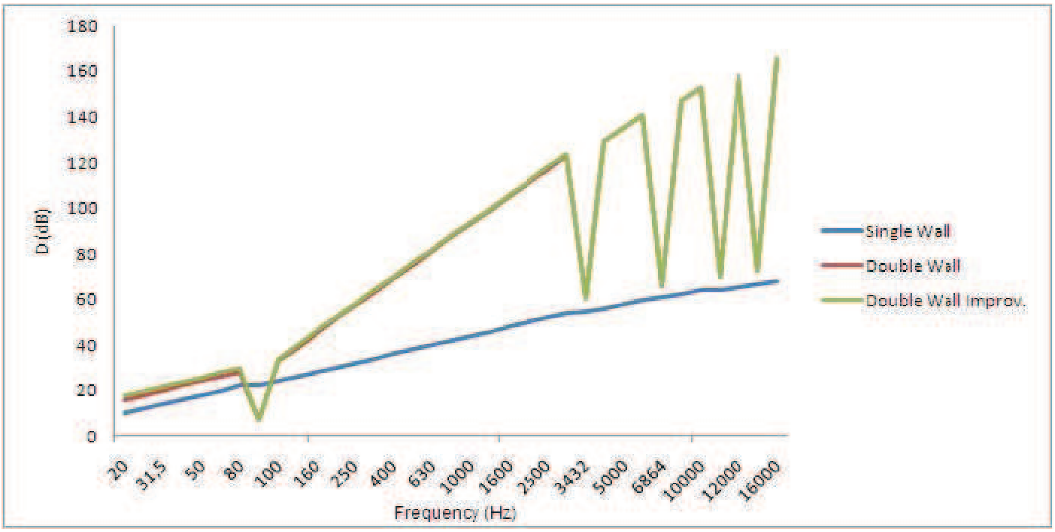


Fig. 1.7-7: Double wall Insulation of 30 mm boards (18 + 12 mm)

So in the graphic the important frequencies which can be shown are:

f_0 = 83 Hz (approx 0 dB insulation).

f_{c1}= 3432 Hz (60 dB).

f_{c2}= 6864 Hz (66 dB).

f_{c3}= 10296 Hz (70 dB).

f_{c4}= 13728 Hz (72 dB).

As is shown the insulation for the frequencies which create stand waves inside the air cavity, decrease substantially the insulation, despite this would be enough insulation to fulfill the NC10 inside the small anechoic room. Anyway to avoid these stand waves just with fulfill the cavity with a high density rock wool or fiber glass will be enough. Doing that the insulation of the partition could be improve in all the aspects as the absorbent material will insulate more than the air.

For the system's natural frequency nothing can be done, but for the purpose which we will manage with, and the frequency range in which we will work the system's natural frequency obtained is low enough to do not be a problem.

1.8 Vibration Insulation

Another important insulation that has to be carried out is the insulation to the vibrations. Those often are vibrations of a very low frequency which are in the building. These are spread through the building structure, so if the chamber is in direct contact with the building these will also transmit to the chamber.

To avoid that, special amounts are assembled, depending on the weight of the chamber different technologies are used, being common the rubber technologies, and the amounts built-up by springs.

To choose the amount have to know the weight and from which frequency up the insulation is necessary With some vibration measuring equipment the frequencies of the building can be measure, and having the lowest fundamental frequency, the necessary amount. Have to take in account that the low cut off frequency for the amount have to be $F_0/1.5$ being F_0 the lowest resonance frequency to insulate.

The usual values of frequencies to insulate are about 2 to 20 Hz. Being the most common frequencies 4 and 5 Hz, but it depends on the building.

1.9 Sound Actuation System

The sound actuation system for the chamber will be composed by two loudspeakers, one coaxial, to emit sweeps, pink or white noise or pure tones in low frequencies, and one plated diaphragm tweeter to emit pure tones in high frequencies Also a 8 channel amplifier which has 90 watts per channel, the class 1 microphones and the microphones to test, and some equipment to calculate all the necessary parameters.

1.9.1 Loudspeaker

Due to the test that will be carried out, a really flat frequency response loudspeakers were needed, because speakers whose answer will vary considerably depending on the frequency, will change the emitted signal, making in this way that the values which were obtained by the microphones would not be worth as they would be largely modified by the speaker. So that is way the loudspeakers had to be as flat as possible and it then certainly would know that the values obtained are due to the microphones response and not because a bad response of the loudspeakers.

Know that the best option could have been to get three different loudspeakers (sub-woofer, woofer, tweeter), so each would have had a very flat response for each frequency range in that them would have had to work. And probably that would have been the option chosen if the dimensions of the chamber would have been greater But with the dimensions which we are working we needed small size loudspeakers, to can assembled them in a three way baffle which could enter in the anechoic room. And for small size, yes for the tweeters, but for the low and mid frequencies, there are not really flat response loudspeaker.

Another reason to do not build a three way baffle, was that in the chamber we will have not a big separation between microphones and loudspeakers, so for the loudspeakers which would have been more separated from the center (in the baffle sides, Fig. 1.9-1), the frequencies which these would have emitted, would have incised in the microphones with a big angle, then it would have been to obtain values for certain frequencies that were not under the same conditions as for other.

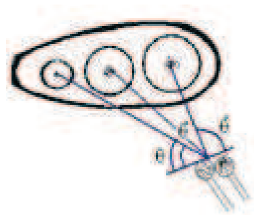


Fig. 1.9-1: Angle between microphones and three way baffle

So we looked for a loudspeaker which could emit in a big range of frequencies. The best option would have been a full-range loudspeaker, but those loudspeakers are not able to emit sound with a really flat response, as they are designed for music purposes

So finally a class A coaxial loudspeaker[1] was chosen It is a 15” bass loudspeaker and 2” compression driver combination, with a neodymium magnet, which has high magnetic features,

1 See the Annex B. , page X for more information about Class A coaxial loudspeakers

that allow to use light weight and reduced volume magnet, improving the flux obtained with ferrite magnets. It could give an enough flat response for a wide frequency range (35 to 20000 Hz), can see that in Fig. 1.9-4, so not having large variations, and using equalizers we can make it very flat. Because is a coaxial, two way filter was necessary, so a filter specially designed for our coaxial loudspeaker was chosen, a filter assemble with high power components.

The Characteristics of the coaxial are:

For the low frequency unit:

Nominal Diameter	380 mm 15 inch
Rated Impedance	8 ohms
Power Capacity	350 w AES[2]
Sensitivity	99 dB 2.83 V at 1 m
Frequency range	25 – 3500 Hz

Fig. 1.9-2: Characteristics of the 15XA 38 Nd coaxial low frequency unit characteristics

For the high frequency unit:

Rated Impedance	16 ohms
Power Capacity	90 w AES above 1 KHz
Sensitivity	105 dB at 1 m
Frequency range	500 – 20000 Hz

Fig. 1.9-3: Characteristics of the 15XA 38 Nd coaxial high frequency unit characteristics

In the Fig. 1.9-4 is also possible to see the lines of the harmonic distortion, second and third harmonic (H2 & H3). Is convenient to have these curves as far as possible, to guaranteed that the loudspeaker will not emit high level harmonics when you introduce any signal in it. Problems can appear if these curves are less than 10 dB from the signal, because some distortion could appear changing the real signal, but these are 20 dB from the signal or even more. So the harmonics are of such low level that will not be a problem.

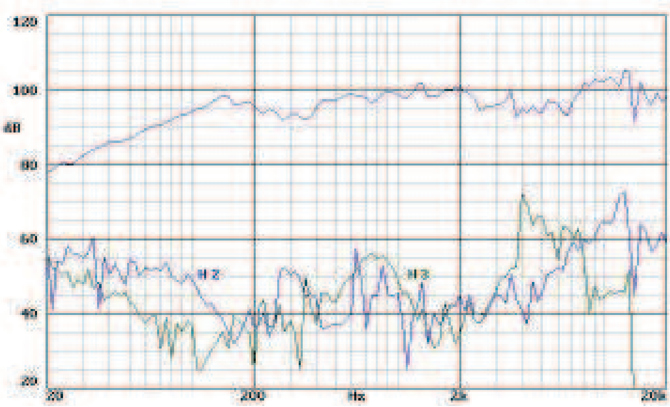


Fig. 1.9-4: Frequency Response and lines of harmonic distorsion for coaxial loudspekar

The two way filter characteristics are:

2 The power capacity is determined according to AES2-1984(r2003) standard.

Type	2 ways
Crossover Frequency	1.8 KHz
Power Capacity	600 w
Low/High Frequency Impedance	8 16 ohms
Attenuation Slop	12 dB/oct.
High Frequency Attenuation	0 to -1.5 dB

Fig. 1.9-5: F2XA Two way filter characteristics

Having a device which we could emit noise or sweeps, a specific loudspeaker to emit pure tone was required. As also tests with pure tones will be made.

So a loudspeaker with a flater frequency response was found, with which be able to emit pure tones in a wide frequency range.

A plated diaphragm tweeter with horn was chosen, which has a extended frequency response, high sensitivity, low distortion and a extremely linear frequency response. It characteristics are shown in the next table:

Overall Dimensions (WxHxD)	230 x 230 x 148 mm
Rated Impedance	8 ohms
Power Capacity	80 w AES at 1 Khz
Sensitivity	102 dB 1 w at 1 m
Frequency range	700 – 23000 Hz
Cutoff frequency	800 Hz

Fig. 1.9-6: TPL150H characteristics

In the Fig. 1.9-7 can see the extremely flat response which we obtained with that tweeter, from 1KHz to 20 KHz, and with a equalization may extend it down till 800 Hz. Also is possible to see the low distortion of the loudspeaker being the curves of the second and the third harmonic separated for the frequency range of work 40 dB.

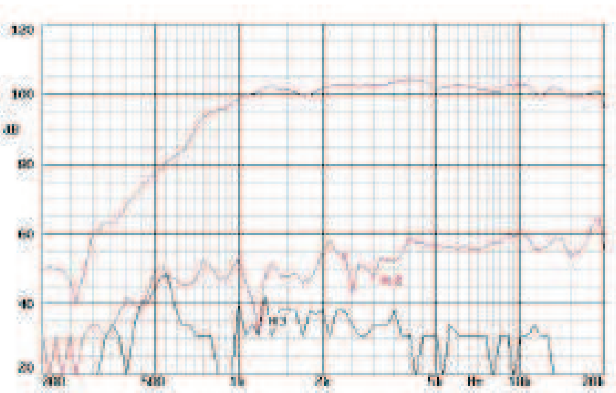


Fig. 1.9-7: Frequency Response and lines of harmonic distorsion for the tweeter

1.9.2 Loudspeaker Enclosure

A loudspeaker enclosure is a purpose-engineered cabinet in which speaker drivers and associated electronic hardware, such as crossover circuits and amplifiers, are mounted. Enclosures may range in design from simple, rectangular particle-board boxes to very complex cabinets that incorporate composite materials, internal baffles, ports and acoustic insulation.

First of all, the loudspeaker are designed to be used in cabinet, if not, if are mounted freely, in absence of some air pressure, the speakers will easily enter in uncontrollable excursion and thus distort (with the tweeter that do not happen because the frequencies in which it work are less prone to this phenomenon).

The other reason why the loudspeakers are mounted in baffle, are to avoid the sound waves generated by the rearward-facing surface of the diaphragm of an open driver interacting with sound waves generated at the front of the driver, thus producing a cancellation of the sound. This phenomenon is particularly noticeable at low frequencies where the wavelengths are large enough that interference will affect the entire listening area.

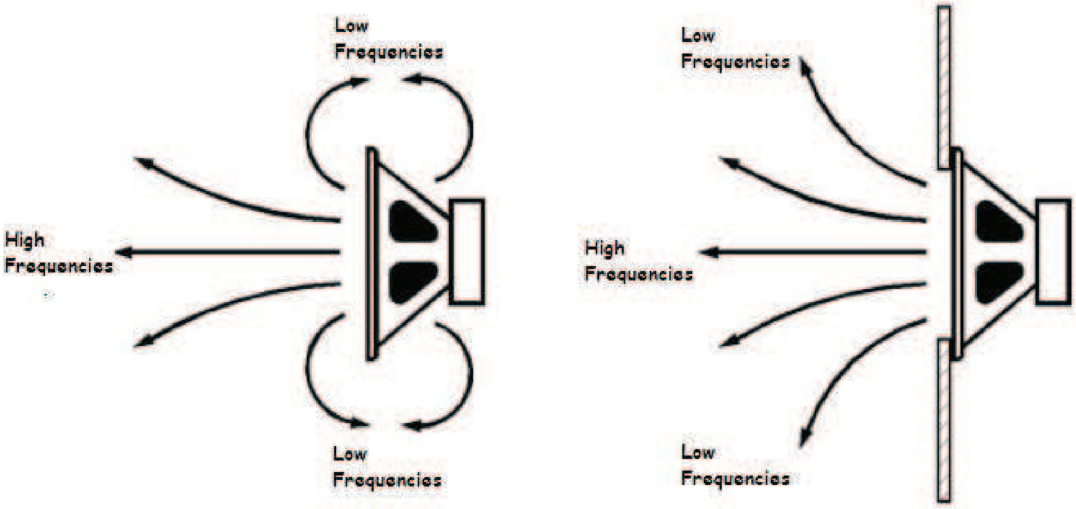


Fig. 1.9-8: Cancellation of the Sound in Low Frequencies and how to avoid it with the loudspeaker enclosure

Additionally, because they would travel different paths through the listening space, the sound waves would arrive at the listener's position at slightly different times, introducing echo and reverberation effects which are not part of the original sound.

Enclosures can have a significant effect beyond what was intended, with panel resonances, diffraction from cabinet edges and standing wave energy from internal reflection/reinforcement modes being among the possible problems. Bothersome resonances can be reduced by increasing enclosure mass or rigidity, by increasing the damping of enclosure walls or wall/surface treatment combinations, by adding stiff cross bracing, or by adding internal absorption.

Looking to our loudspeakers two loudspeaker cabinet were designed, paying all the attention to the cabinet for the coaxial loudspeaker, as the cabinet for the tweeter TPL150H did not need to have any special features like a vented or a specific volume, for a reinforcement in low frequencies or a specific resonance frequency at low frequencies, because it works in a range of frequencies that are high enough to do not need any reinforcement

For the design of the cabinets the software winISD beta was used, which help to make fast calculations with good results. Finally a Vented cabinet was designed.

1.9.2.1 Bass-Reflex or Vented Box

A Bass reflex system (also known as a ported, vented box or reflex port, Fig. 1.9-9) is a type of loudspeaker enclosure that uses the sound from the rear side of the diaphragm to increase the efficiency of the system at low frequencies as compared to a typical closed box loudspeaker or an infinite baffle mounting, and that is the main reason because a Bass reflex was assembled instead a closed box.

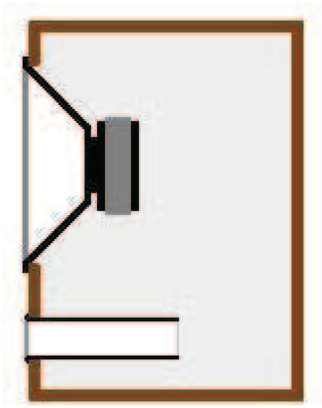


Fig. 1.9-9: Bass Reflex cross section

In contrast to closed box loudspeakers, which are substantially airtight, a bass reflex system has an opening called port or vent which consists of an opening with a pipe or duct of circular or rectangular cross section. The air mass in this opening resonates with the "springiness" of the air inside the enclosure in exactly the same fashion as the air in a bottle resonates when a current of air is directed across the opening. The frequency at which the box or port system resonates, known as the Helmholtz resonance, is dependent on the effective length and cross sectional area of the duct, the internal volume of the enclosure, and the speed of sound in air.

The main advantage of the Bass reflex is that the resonant system augments the bass response of the driver, and if designed properly, can extend the frequency response of the driver enclosure combination to below the range the driver could reproduce in a sealed box. The enclosure resonance has a secondary benefit in that it limits cone movement in a band of frequencies centered around the tuning frequency, reducing distortion in that frequency range.

The trade-off for this augmentation is that, at frequencies below tuning frequency, the port unloads the cone and allows it to move much as if the speaker were not in an enclosure at all. This means the speaker can be driven past safe limits at frequencies below the tuning frequency with much less power than in an equivalently sized sealed enclosure. For this reason, high-powered systems using a bass reflex design are often protected by a filter, which attenuates the signal in those low frequencies, to do not drive the loudspeaker with the same power that is necessary for frequencies over the tuning frecuency, because the diaghpram can be broken.

Also have to be careful with the resonant system, because is possible to extend the response of the driver to such a low frequency for which the loudspeaker can distort or it can even be broken, not to be the diaphragm or cone is not able to work is such frequency.

By their own nature, resonant systems cannot start and stop instantly. Vented speakers stagger two resonances, one from the driver and boxed air and another from the boxed air and port, in order to achieve their bass output, a more complex case than an equivalent sealed box. This causes increased time delay (increased group delay imposed by the twin resonances), both in the commencement of bass output and in its cessation. Therefore a flat steady-state bass response does not occur at the same time as the rest of the sonic output; rather, it starts later (lags) and accumulates over time as a longish resonant *tail*. Because of this complex, frequency-dependent loading, ported enclosures generally result in poorer transient response at low frequencies than in well-designed sealed box systems.

1.9.2.2 WinISD beta

Is a freeware speaker designing software for Windows 9x/NT environment. You can design Closed, Vented and Bandpass boxes with this program. It also allows you to calculate few different types of filters.

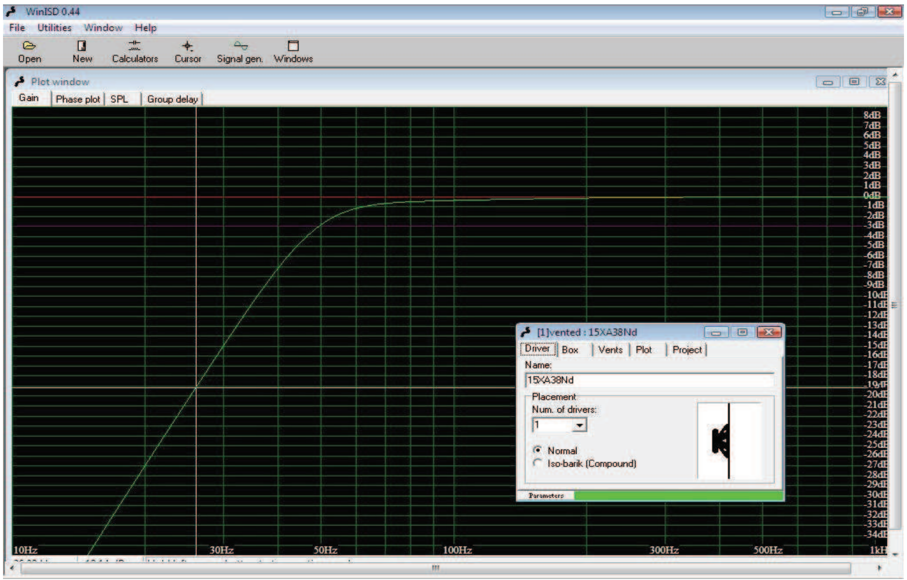


Fig. 1.9-10: WinISD freeware interface

To start a new project just press *New* and a window like that will appear:

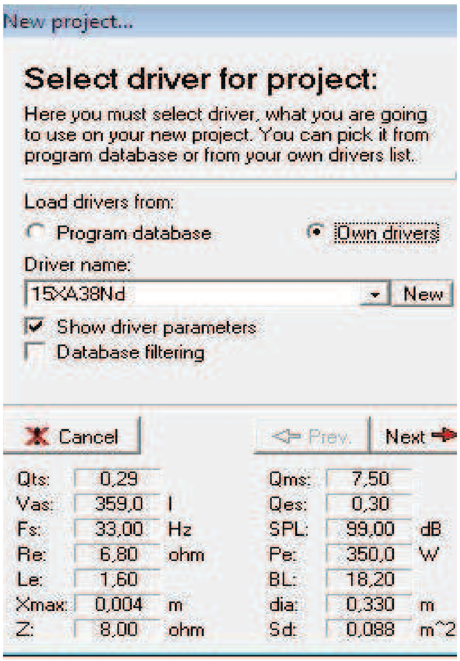


Fig. 1.9-11: Window to choose a loudspeaker or introduce the parameters of the loudspeaker for which the enclosure will be designed

Then chosen the option of “Own drivers”, because exits the possibility we can design a cabinet for loudspeakers which the freeware has in its database, press *New* and introduce all the values of the Thiele/Small parameters (Annex B. , page. 73) required, in that way the freeware will know how is the loudspeaker for which we are going to build the enclosure.

After that pressing *Next*, the software will ask how many loudspeakers will form the cabinet (from 1 to 100), and if it is a normal loudspeaker or it is Iso-barik.

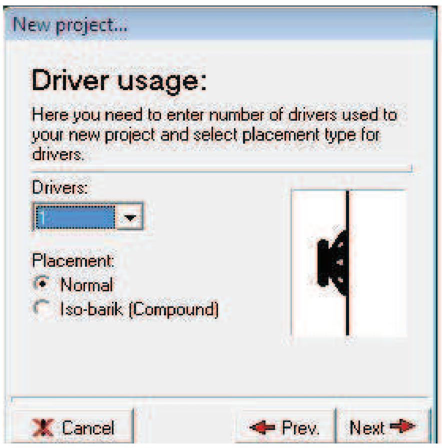


Fig. 1.9-12: Window to choose the quantity and the Placement of the Loudspeakers

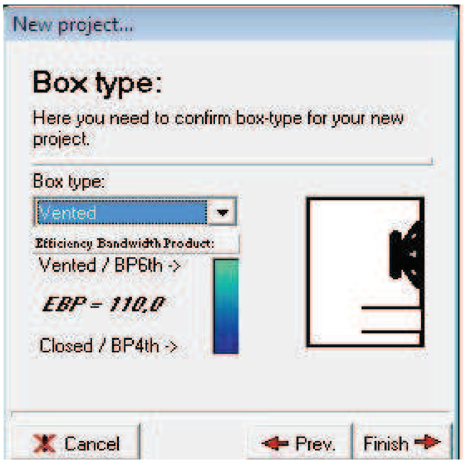


Fig. 1.9-13: Window to choose the type of the enclosure

Chosen the number of loudspeakers that in our case is one and normal, in the next step it ask for the type of the box, if we are going to build a Close, Vented (Bass Reflex), 4th order bandpass, or 6th other bandpass. Also make a remark of the value EBP (Efficiency Bandwidth Product), which value specified in our case that our loudspeaker will be best used in a vented enclosure . In our case a Vented cabinet.

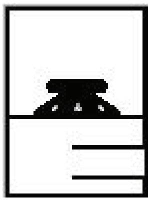


Fig. 1.9-14: 4th order Bandpass

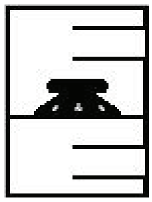


Fig. 1.9-15: 6th order Bandpass

Then press *Finnish* and the WinISD beta will give all the parameters of the enclosure for the loudspeaker.

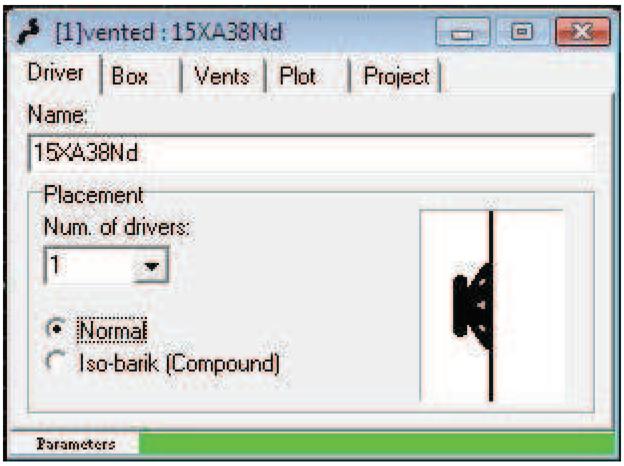


Fig. 1.9-16: Driver Characteristics window

In that window, can check all the parameters that the WinISD beta calculated for the enclosure. In that first window we can again change if now we want to check another loudspeaker. But have to take in account that the parameters which will appear are calculated for one loudspeakers, just the value of NPS will change, because with 2 loudspeakers the NPS will be higher.

In the tab *Box* as we can look in Fig. 1.9-17 the WinISD provides the Volume of the enclosure and the tuning frequency. Changing the volume also the Tuning frequency will change, adapting if needed the box to the new volume.

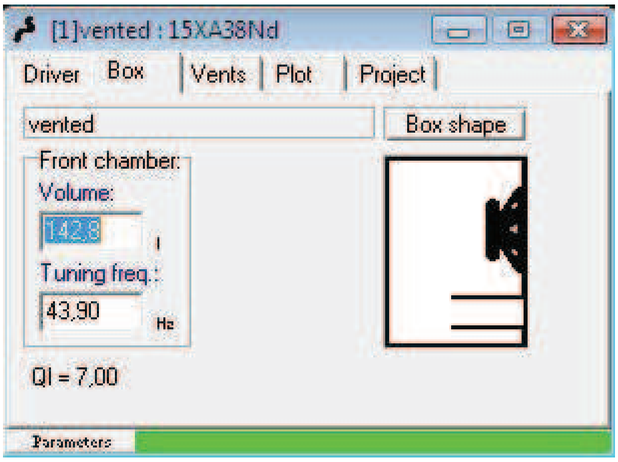


Fig. 1.9-17: Box characteristics (Volume and Tuning frequency) window

Pressing the *Box Shape* bottom the Fig. 1.9-18 window appears. Where in the part of the box shape normal or sloped box, which is a box with the top part narrower than the bottom, can be chosen Also indicating the board thickness which the box is going to be made the freeware gives the outer dimensions of the possible box, these can be changed, fixing one by yourself and changing the other the program will give the other dimension of the box.

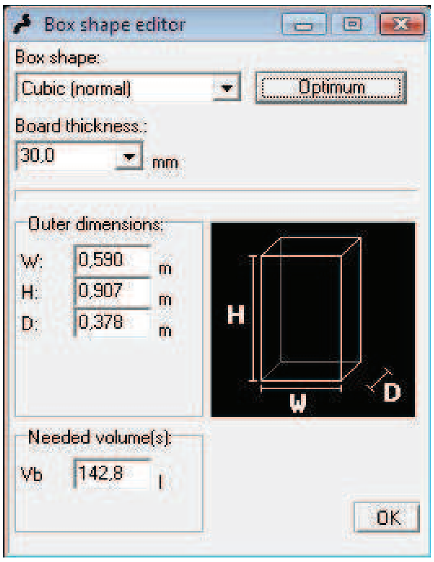


Fig. 1.9-18: Box shape editor window

In the tab *Vents* is everything related to the vents, how many of them we will built, the diameter and the shape and the mach. In the number of them we can put from 1 to 8 vents, and the diameter we can change it looking which diameter will mach better with vent mach what is the speed of the air as it goes back and forth in the port. Want to keep generally below 5% of the speed of sound (~17 m/s) at the higher power level you want to use in order to avoid objectionable noises. It helps a great deal to use flared ends on the port - allowing you to bend the 5% rule a bit. If you click in the shape it change to a square vent from appearing instead the diameter the dimension for the two sides of the square. The vent length change automatically when you change the other values.

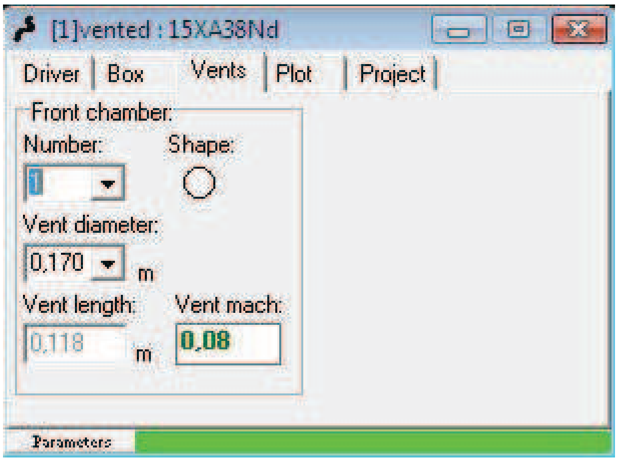


Fig. 1.9-19: Vent features window

In the tap *Plot* just appear the Power which will emit the loudspeaker to simulate the NPS graphic and in which distance the measurement will be make. Also some characteristics to change for the graphic like the width of the line and the color which it is going to be painted

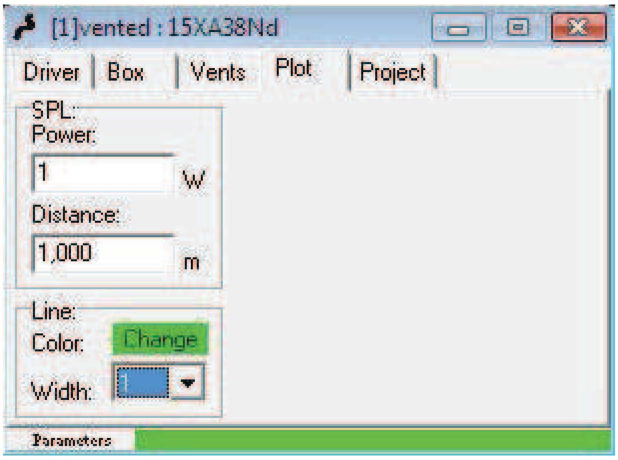


Fig. 1.9-20: Variables to have in account at the time to plot the values

Finally in the *Project* tap we see just some specifications of the document, but what is really important is the window which appears when the button *Show as text* is pressed Fig. 1.9-21.

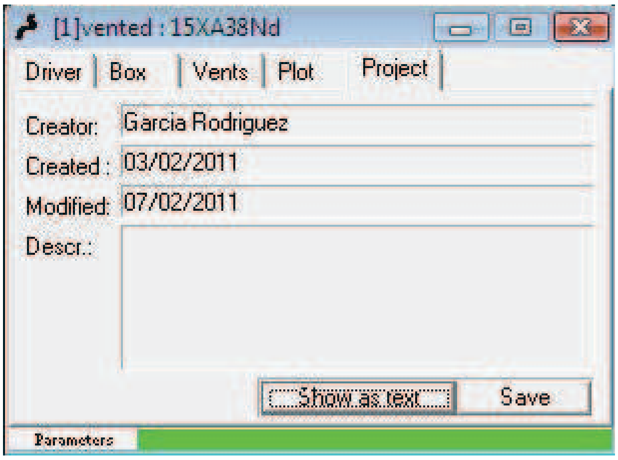


Fig. 1.9-21: Project data window

Can see a document in which information about the loudspeaker and and the creator of the document.

Is possible to choose from which to which frequency you want the values, also to specified the value of the step between one frequency and the following. There is available the possibility to have the phase of the sound in the point where the values will be measure, just clicking in the case the values will appear per frequency in the text document. Clicking the *Driver data* the next specifications of the loudspeaker appears:

- The name of the loudspeaker
- The Thiele/Small parameter:
    - V_{as} Equivalent Compliance Volume
    - Q_{ts} Total *Q* of the driver at F_s
    - F_s Resonance frequency of the driver
    - SPL Sound pressure level at 1 m.

After specified all the parameters which are wanted, in one text document appears all the values per frequency for the SPL and the relative gain which the loudspeaker box provides at 1 m emitting 1 watt (values specified before in the tap *Plot*), then we can save or just check the values an close the window.

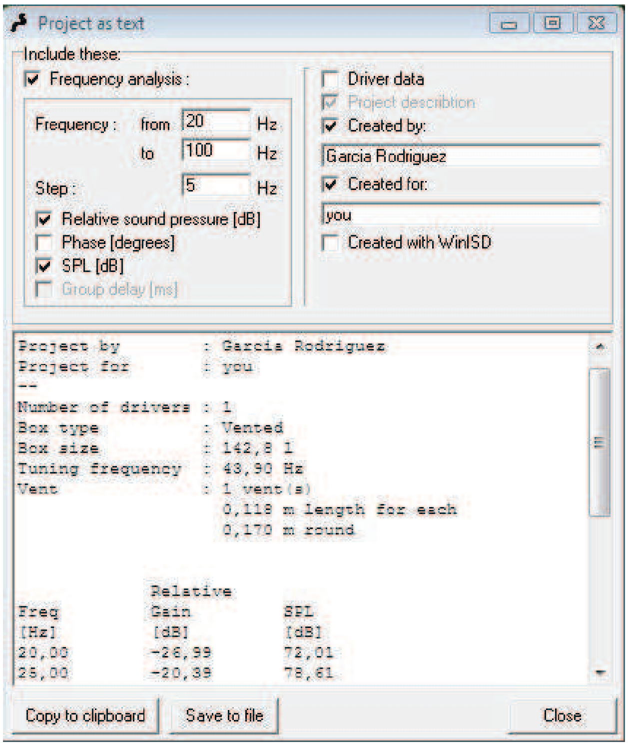


Fig. 1.9-22: Variables to include in a text file and the text file

1.9.2.3 Results obtained with WinISD

For a the coaxial loudspeaker 15XA38 Nd with these specifications:

Coaxial 15XA38 Nd			
Qts	0,29	Qms	7,5
Vas (l)	359	Qes	0.3
Fs (Hz)	33	SPL (dB)	99
Re (Ω)	6,8	Pe (W)	350
Le	1,6	BL	18,2
Xmax (mm)	4	dia (m)	0,33
Z (Ω)	8	Sd (m^2)	0,088

Fig. 1.9-23: Characteristic of the Coaxial 15XA38 Nd

We obtained a Bass reflex cabinet built-up with 30 mm MDF boards with one vent:

Volume (l)	142,8
Outer Dimensions (W x H x D; mm)	760 x 910 x 300
Vent Length (mm)	118
Vent diameter (mm)	170
Tuning Frecuency fb (Hz)	43,9
3dB decay frecuency f3 (Hz)	49,5
Vent mach	0.08

Fig. 1.9-24: Characteristics of the Bass Reflex

In the Fig. 1.9-25 can look how the Bass reflex will work (high pass filter). Introducing a slow attenuation as the frequency goes down, being 43.9 Hz (cross of purple and blue lines) the frequency for which the power will decrease to the half (-3 dB). The cross indicates the frequency for which the loudspeaker would work without risk of break, because the loudspeaker has a cut off frequency of 30 Hz approximately so the tuning frequency of the Bass reflex is higher. Lower than 43.9 Hz, have to be careful because the loudspeaker will work like freely and the power which we will give to it has to be lower because the displacement of the diaphragm will be bigger due to the lack off pressure (the filter which the amplifier has will fix really well).

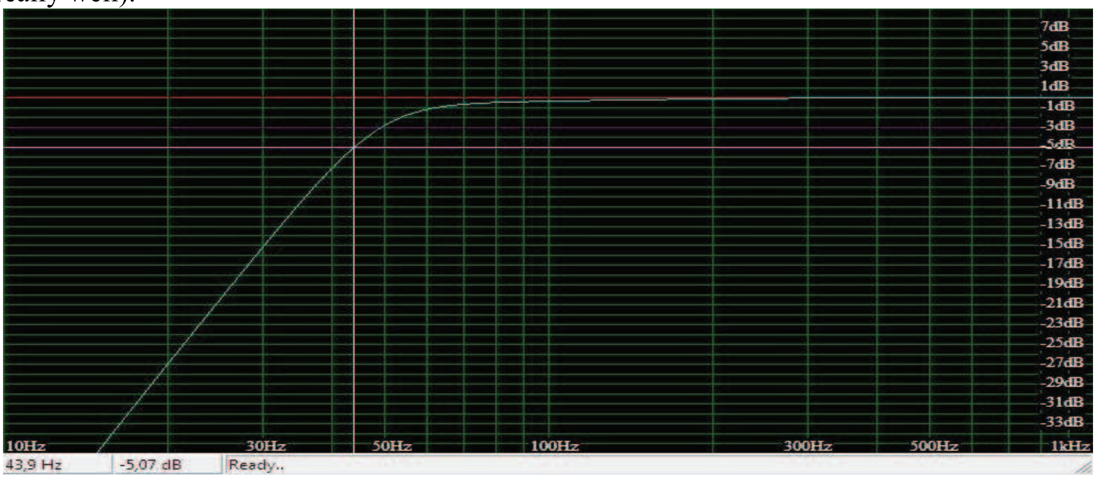


Fig. 1.9-25: Gain of the Bass Reflex

As already is known the resonant systems cannot start and stop instantly so an interesting graphic can be which one where we could see how the delay that the bass reflex experiment when it emits sound, change with the frequency, in the Fig. 1.9-26. Can look how the delay increases as the frequency decreases, just being more difficult to stop the radiation as the frequency is lower. The relative maximum is in the tuning frequency of the enclosure (43,9 Hz). Anyway these values are really small to be bad for the tests. If the values would be highers, these could appreciated badly in the results obtained in the tests, because the baffle would stay radiating at some frequencies of the signal more than for others, so the energy in those frequencies for which the radiation would stay longer, would be higher.

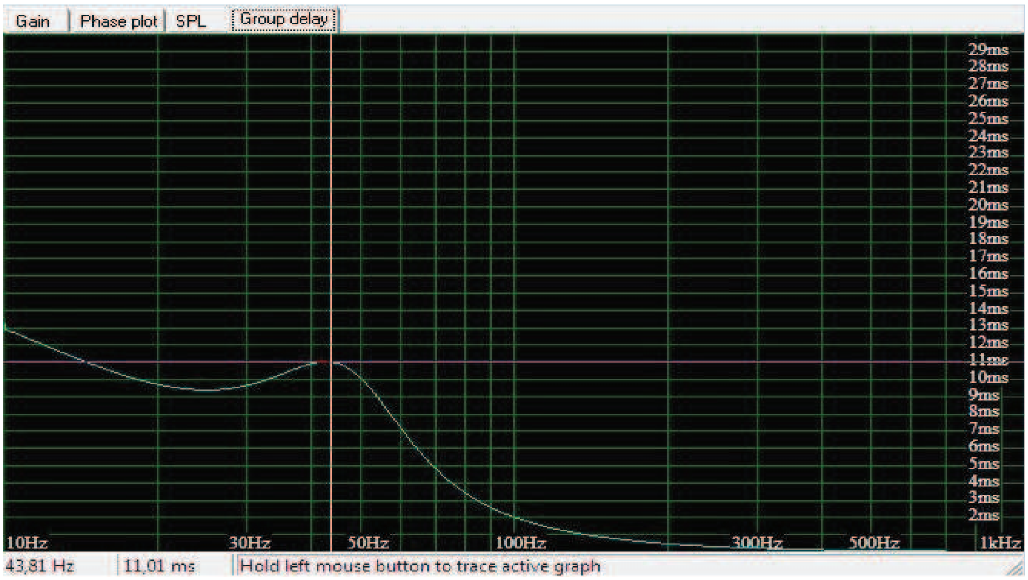


Fig. 1.9-26: Variation of the Delay for the Bass reflex

1.9.3 Microphones

As said above, two microphones will be used. One of them a Class 1 microphone, which will be used like reference. Know that this kind of microphones are the best quality microphones that there are, high sensitivity, high SNR, omnidirectional. So the values which are obtained with them are the real ones, not being modified as could happen with a less category microphone, due to it directionality or low SNR or sensitivity could measure invalid values.

Now, having the reference, the other microphone will be one of the microphones which are built in the laboratory. The quality of them will be evaluated, comparing the results obtained by this microphone with the values of the reference microphone.

The other reason to take the longest dimension, for the dimension where the measures will be undertaken (length) is because the two microphones will be placed one next the other so these are going to be displaced some degrees from the axis of the loudspeaker, which could affect badly to the values obtained, being these measured which a certain angle of incidence, not being perpendicular as has to be.

So farther away from the loudspeaker, smaller will be the angle which the microphones will be displaced. It is true that the coaxial, is at least for the low frequency unit big enough, so the microphones should were far one from the other. But for the high frequency unit, that angle could affect badly, because the tweeter is smaller and the high frequencies are more directional, so the angle referring to the loudspeaker will be apparently bigger.

1.9.4 Amplifier

To emit the signals required in the tests, a professional amplifier was chosen Many possibilities were studied, but the in initial requirement just a really flat frequency response amplifier was needed with also a low distortion. Finally a professional amplifier of 8 channel was chosen, having in mind a future project of sound source detection, for which several channel are going to be needed. Some different amplifiers were assessed but finally the CX 168 amplifier was chosen because the quantity of different features that it has, which will be really useful.

1.9.4.1 Switches Configuration & Setting the Modes

By using the mode switches and connecting to the amplifier properly, the CX168 amplifier can be configured as a 1 to 8 channel amplifier. This flexibility enables the CX168 to be used for most any multichannel application. Each of the four amplifier sections may be operated in Stereo, Parallel, or Bridge mode independently of the others. For example, channels 1 & 2 could be set for stereo operation while channels 3 & 4 are set for parallel operation; all while channels 5 & 6 and channels 7 & 8 are set for bridge mode.

The channels are grouped as follows:

Channels 1 and 2
Channels 3 and 4
Channels 5 and 6
Channels 7 and 8

You can set the operating mode for each group by setting that group's configuration DIP switches in the desired positions. It is not possible to group the channels differently. For instance, you can not bridge channels 3 & 5 because they are not in the same channel group.
Before setting the configuration DIP switches, you must first decide how the amplifier will connect to the speaker.

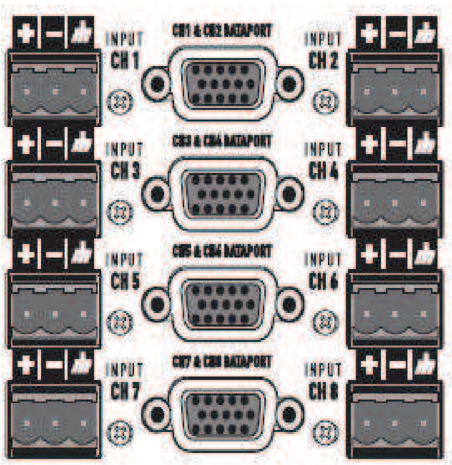


Fig. 1.9-27: Channels inputs

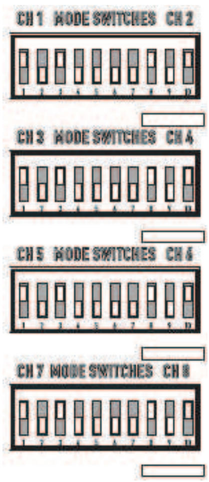


Fig. 1.9-28: Mode configuration DIP switches

First of all some remarks about the modes to connect the loudspeaker have to made, as the characteristics like the power will change in dependence in which mode we will work.

The following description apply to a channel pair, such as Ch. 1&2 or Ch. 7&8.

- Stereo Mode: Stereo mode supports two completely separate audio channels, usually referred to as 'left' and 'right'. Stereo configurations have two separate input signals and two separate output signals.

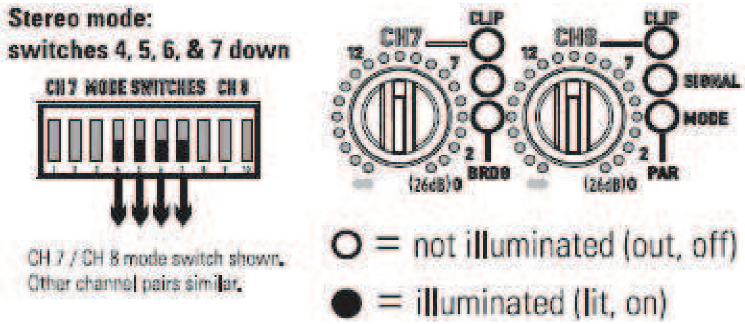


Fig. 1.9-29: Mode switch settings and LED indicattion for stereo mode

- Parallel Mode: Parallel mode applies one input signal to both channels. Both inputs of a channel pair are connected in parallel when the mode switch is set for parallel, therefore, connect only one input per channel pair when in parallel mode. The outputs are connected the same as stereo mode. Each speaker will be supplied the same signal, which is still controlled by that channel's gain control.

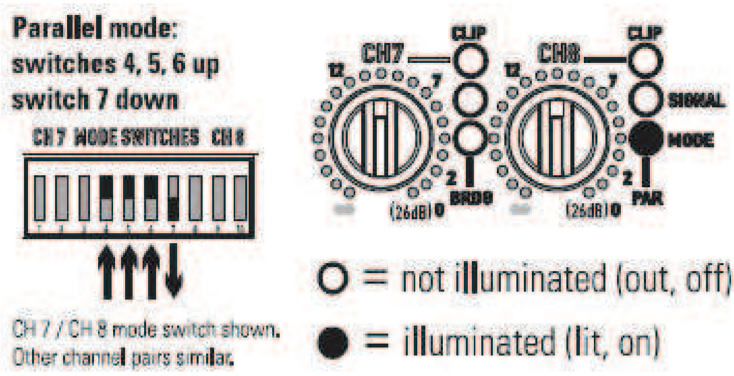


Fig. 1.9-30: Mode switch settings and LED indicattion for parallel mode

- Bridge Mode: Bridge mode combines the two channels of a pair (such as Ch. 3 & Ch. 4) into one higher powered channel. Like parallel mode, both inputs of a channel pair are connected in parallel when the mode switch is set for parallel, therefore, connect only one input per channel pair when in bridge mode. There are 4 possible channel pairs that can be bridged on the CX168. They are Ch.1-2 Bridge, Ch. 3-4 Bridge, Ch. 5-6 Bridge, and Ch. 7-8 Bridge. Any pair or number of pairs can be independently configured in bridge mode. Use the first channel's input and gain control when in bridge mode. The second channel's gain control is non-operative and should be turned all the way down.

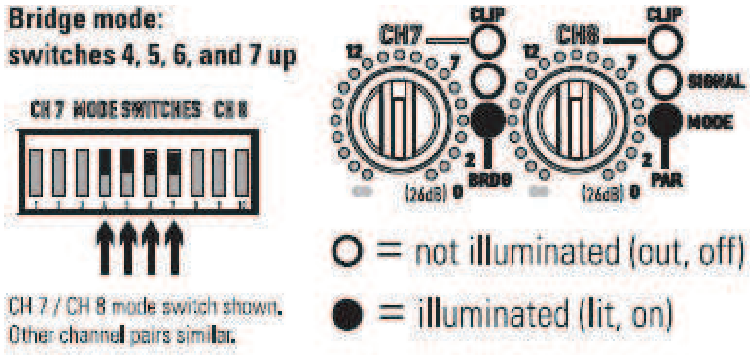


Fig. 1.9-31: Mode switch settings and LED indicattion for bridge mode

1.9.4.2 Specifications

The datasheet of the CX 168 is attached in the Annex C, anyway the most important features are shown.

	CX168	
Stereo Mode (all channels driven)	Continuous average output power per channel	
8Ω / 20 Hz – 20 kHz / 0.05% THD	90 W	
4Ω / 20 Hz – 20 kHz / 0.1% THD	130 W	
Midband Ratings	All channels driven	Single channel
8Ω / 1 kHz / 0.1% THD	100 W	120 W
4Ω / 1 kHz / 0.1% THD	140 W	180 W
70V / 20 Hz – 20 kHz / 0.2% THD		
Bridge-Mono Mode	Bridge-mono mode operation	
16Ω / 20 Hz – 20 kHz / 0.1% THD	180 W	
8Ω / 20 Hz – 20 kHz / 0.1% THD	260 W	
140V / 20 Hz – 20 kHz / 0.2% THD		
Signal to Noise (20 Hz – 20 kHz)	-107 dB	
Input Sensitivity	1.35 Vrms at 8Ω	
Voltage Gain	20x (26 dB)	
Input Clipping	6 Vrms (+18 dBu)	
Output Circuitry	Class AB+B	
Frequency Response	20 Hz – 20 kHz, + 0.2 dB | 8 Hz – 50 kHz, +0/-3 dB	
Damping Factor	> 200 (5 kHz and below)	
Input Impedance	6k ohms unbalanced, 22k ohms balanced	
Distortion (SMPTE-IM)	< 0.02%	
Distortion (typical)		
20 Hz – 20 kHz: 10 dB below rated power	< 0.1% THD	
1.0 kHz and below: full rated power	< 0.03% THD	
Cooling	Variable-speed fan / rear-to-front air flow through tunnel heat sink	
Connectors	Input: 3-pin Euro-style detachable terminal blocks (one per channel) DataPort: HD-15 connector (Ch. 1+2, 3+4, 5+6, 7+8) Output: two 8-pin Euro-style detachable terminal blocks	
Controls	Front: AC switch, Ch. 1, 2, 3, 4, 5, 6, 7 & 8 gain knobs Rear: DIP switches for Ch. 1 - 8, clip limiter on/off, LF filter on/off, LF filter freq select 33 or 70 Hz for CX168 LF filter freq select 50 or 75 Hz for CX108V, inputs parallel or stereo; bridge mode	
Indicators	Power-On: Green LED / Parallel inputs: Orange LED (1 per channel pair) / Signal -35 dB: Green LED (1 per channel) Bridged: Yellow LED (1 per channel pair) / Clip: Red LED (1 per channel)	
Amplifier Protection	Full short circuit, open circuit, thermal, ultrasonic, and RF protection. Stable into reactive or mismatched loads	
Load Protection	On/off muting, individual channel DC fault blocking	

Dimensions (HWD)		3.5" (8.9 cm) 2 RU x 19" (48.3 cm) rack mounting x 14" (35.6 cm) from front mounting rails
Weight - Net / Shipping		21 lb (9.5 kg) / 27 lb (12.3 kg)
Power Requirements		100, 120, 230 VAC, 50 – 60 Hz (configured at factory)
120V Current Consumption*	Idle	0.6 A
1/8 power pink noise (typical of program material at maximum unclipped power)	8Ω	6.2 A
	4Ω	9.2 A
1/3 power pink noise (typical of program material with severe clipping)	8Ω	9.2 A
	4Ω	14.2 A

** Multiply currents by 0.5 for 230V units*

Have to said that the Signal to Noise ratio, is a negative value, not because the noise is higher than the signal, is just because the method to measure. The signal noise ratio is measured in absence of any signal, being that value, just the value of the noise in dB

1.9.4.3 Clip Limiter

The CX168 amplifier has separate clip limiters for each of the 8 channels. These clip limiters respond only to actual amplifier clipping. Amplifier clipping generates internal error signals which cause the clip limiter to quickly reduce gain and minimize the overdrive. To preserve as much of the program dynamics as possible, limiting occurs only during actual clipping. Each channel's clip limiter can be switched on or off individually.

The clip limiter is internally set to respond as fast a possible after clipping is detected. For program material that is primarily "full-range", the effect on the overall audio quality should be imperceptible. It is recommend using the clip limiters for almost all applications, especially full-range audio applications. Clipping can cause high-frequency artifacts to be output to the speakers, potentially damaging fragile high-frequency drivers.

- Switches 1 & 10 up = Clip Limited ON

- Switches 1 & 10 down = Clip Limiter OFF

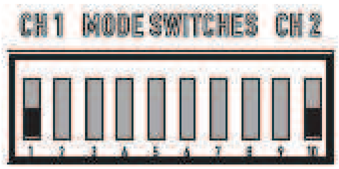


Fig. 1.9-32: Switch to set up the clip Limiter.

For program material that is primarily low-frequency in nature (low-frequency or sub-woofer drive) this may be perceived as a "rubbery" effect on the audio. If this is the case, it may be preferable to turn the clip limiters off and let the amplifier clip occasionally. With robust, low-frequency drivers, the occasional clipping should cause no problems.

CAUTION! *Clip limiting reduces extreme overdrive peaks, allowing a higher average signal level without distortion. Increasing the gain with the clip limiter engaged until clipping is again audible, can double the average output power. Be careful not to exceed the power rating of the speakers!*

1.9.4.4 Low frequency filter

When driving speakers with limited low frequency response, it is important to limit the low frequency response of the amplifier. Doing so can result in more usable bass response since the speaker is not being overloaded by very low frequencies it can't handle.

Low frequency sound waves require much more speaker cone motion to produce the same apparent loudness as higher frequencies. Properly designed speaker enclosures help the speaker to move more air with less motion using techniques like porting. Such enclosures only benefit from porting down to a certain frequency. Below this frequency as already know, the speaker is unloaded and is basically free to move around uncontrollably without producing much bass. Limiting the frequency range of the low frequency content enables the speaker to behave the way it was designed to. If frequencies lower than designed are supplied to the speaker, performance will degrade.

All low frequency filters will change the character of low frequency transients. For best results, the cabinet design, speaker capabilities, and program material must be taken into account when configuring low frequency filtering.

As the channel modes the low frequency filter is set per channel by the switches:

Switch 3 up: first channel's filter OFF.
Switch 3 down: first channel's filter ON .
Switch 8 up: second channel's filter OFF.
Switch 8 up: second channel's filter ON.
Switch 2 up: first channel's filter frequency is 33 Hz.
Switch 2 down: first channel's filter frequency is 70 Hz.
Switch 9 up: second channel's filter frequency is 33 Hz.
Switch 9 up: second channel's filter frequency is 70 Hz.

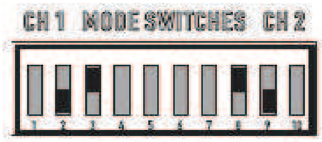


Fig. 1.9-33: Switch to set up the low frequency filter.

Some tips about the loudspeaker will give for the correct use of it:

– The OFF position should be used only for sub-woofer systems with rated frequency response below 33 Hz. or if low frequency filtering is provided by other devices.
– Know the specification of the speaker cabinet you are driving. Match the low frequency roll off setting to the specified low frequency capability of the speaker cabinet. Do not drive the speaker with frequencies below its rating.
– Unless you have low frequency filtering before the amplifier, use the low frequency filter to protect your loudspeakers from cone over-excursion caused by frequencies below the speaker's limits.
– The 33 Hz. roll off is a good all purpose setting. This setting is a good starting point for most large, full-range cabinets.
– The 70 Hz. roll off is a good setting to use with smaller, compact speaker cabinets having limited bass capability
– The frequency selection (33 or 70 Hertz) has no effect unless the filter is set to the ON position.

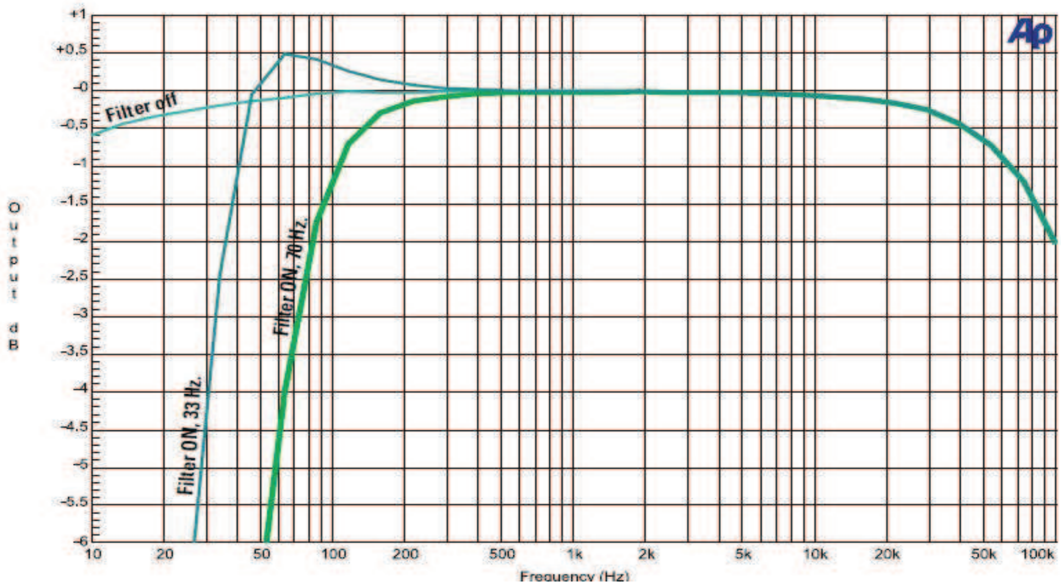


Fig. 1.9-34: Low frequency filter response (Off - 33 Hz - 70 Hz)

1.9.4.5 Output Connection Modes

As the input channel are set up in one way (stereo, parallel or bridge), the outputs have to be configured in the same way. Important to notice that the polarity change from channel to channel and is different for bridge mode. Be certain of polarity of connections before applying power. Reversed polarity may degrade audio frequency response.

- <u>Stereo and Parallel Mode</u>: In stereo or parallel mode, each speaker is connected to its own individual channel of the amplifier. ***Use 4 ohm minimum impedance in stereo or parallel mode.***

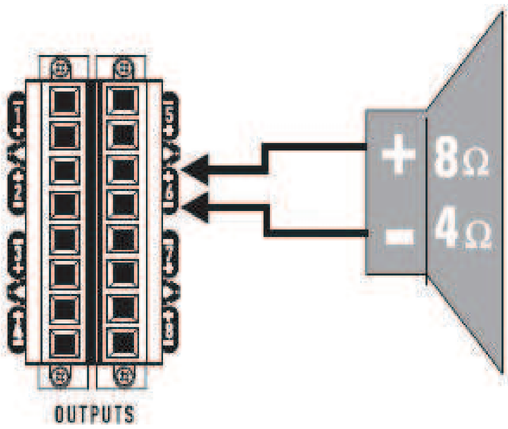


Fig. 1.9-35: Stereo or parallel connection- single speaker shown connected to amplifier channel 6 output. Ensure that all speakerconnections maintain proper polarity (+ to +, - to -).

– Bridge Mode: In bridge mode, each speaker is connected to a bridged-pair of outputs. The channels must first be *bridged* by setting the mode configuration switches to the bridge settings. Then connect the speakers as shown in the Fig. 1.9-36. ***Use 8 ohm minimum impedance in bridged mode. Do not use 2 ohm or 4 ohm loads in bridge mode! This mode puts a high demand on the amplifier and speaker. Excessive clipping may cause protective muting or speaker damage. Ensure the speaker has a sufficient power rating.***

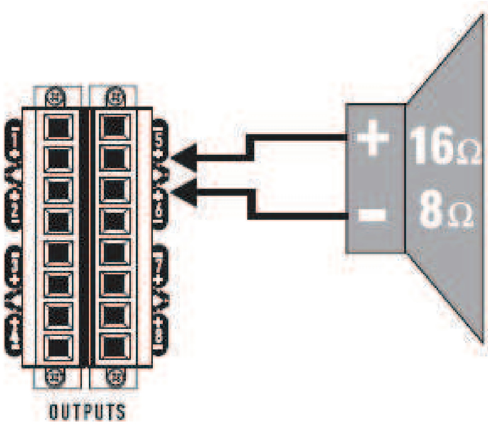


Fig. 1.9-36: Bridge connection- single speaker shown connected to amplifierchannels 5 and 6 (bridged with mode switch settings).Ensure that all speaker connections maintain proper polarity(+ to +, - to -).

1.10 Prototype

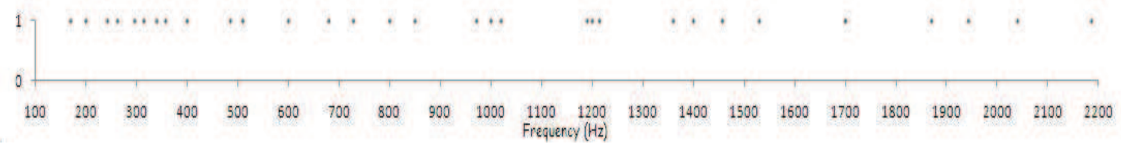
Finally before build-up the proposed small anechoic room, a prototype will be built to check how it is going to work and after decide if build the bigger one or not, depending on the necessities of the acoustic laboratory.

1.10.1 Dimensions

To choose the dimensions of that prototype the same criteria as for the bigger box was followed. Due to the small dimensions the graphic of the relation between dimensions to have a uniform distribution of Eigen-modes, anyway we were careful to do not have the same value for any dimensions and not to be a multiple between them.

The chosen dimensions are 700 x 1000 x 850 mm (width x length x high), plus the walls which are the same as the bigger box (30 mm) the outer dimensions are 760 x 1060 x 910 mm (width x length x high).

The distribution of Eigen-modes which these dimensions give is:



The resonant frequency which we obtained for the room will be higher, because the smaller size of the room, being it 171 Hz. When the Tmid will be calculated the maximum frequency should be estimated to know from which frequency up the Eigen-modes can be negligible. We saw also that the axial modes are distributed in uniform way, that is what was expected.

1.10.2 Absorbent Material

In the prototype anechoic wedges were not used. We were looking for some 100 mm thick material, so different materials were studied like a melamine pyramid foam and flat fiber or rock wools boards. Finally the fiber wool was chosen because it has a better acoustic properties and we found a thicker one.

The melamine pyramid foam was of 50 mm thick with and absorbent coefficient shown in the Fig. 1.10-1. Having a high absorption for high frequencies but being a little bit poor in low frequencies.

F (Hz)	125	250	500	1000	2000	4000
Pyra Méla (50 mm)	0.13	0.25	0.5	0.75	0.88	0.94

Fig. 1.10-1: Absorption Coefficients for 50 mm melamine with pyramid shape

A proposal which was studied was to increase the thickness with a 50 mm thick flat melamine board, as know that with the thickness the absorption increases. But these materials, will not absorb more because it porosity, like can read in the datasheet (Non-fibrous and non-volatile (unlike mineral wool)). So increasing the thickness we will not have a certain difference. On the other hand it has a better esthetic characteristics that is why it is used in room where a fiber glass can be ugly and dirtier.

So know that we did not want a material with a good esthetic characteristics we choose the fiber glass, material which because it porosity has a big acoustics features, having in account that it is a very dangerous material because it particles and really dirt. So we solve that covering the fiber glass with a cloth, and protecting it.

So for the fiber glass we choose a 120 mm thick material. Was not possible to find its absorbent coefficients so in Fig. 1.10-2, for 100 mm thick absorbent coefficients are shown, being for the 120 mm fiber glass a little bit highers.

F (Hz)	125	250	500	1000	2000	4000
RollisolPlus (100 mm)	0.41	0.81	1.07	1.07	1.01	1.00

Fig. 1.10-2: Absorption Coefficients for 120 mm fiber glass

1.10.3 Cut off frequency

The absorbent material used as can see in the Fig. 1.10-2 gives an absorbent coefficients greater than one, due to the reverberation chamber method. So do not know which value is referred to 100% absorption, which is necessary to achieve free field conditions.

Regards to the geometric cut-off frequency in the prototype anechoic wedges will not be used, so do not have sense to use the formulas used for the small anechoic room.

So the best method will be to emit sweeps and measure the reverberation time. And dependence on the reverberation time which we will obtain determinate a cut off frequency from which up we will work, with certainty that the reverberations will not affect to the values.

1.10.4 Airborne Insulation

The insulation of the prototype is exactly the same of the bigger room, having the same wall (30 mm) built-up of sheets of 18 and 12 mm thick.

1.10.5 Vibration insulation

The anti vibration amounts to use in the prototype will be the same that in the big box, but taking in account that this box is smaller so the amount less weight should hold. That is why the model of the vibration amount will change, choosing one which will fix with the weight of the prototype.

1.11 Construction

Finally the size of both boxes, the wall insulation and the materials which will be used to absorb is clear. Also how many microphones we will use and how the baffle will be. But is not clear how we will assemble all these devices.

1.11.1 Prototype

We will start explaining the prototype as that is which we will build first. Have to say that the construction of both "boxes" will be similar, changing just the material and the doors, which will be really difference from the prototype to the small anechoic room.

1.11.1.1 Walls and Baffle

The prototype outer dimensions are 760 x 1060 x 910 mm (width x length x high). And it is going to build with 30 mm thick wall, made of two sheets of 18 and 12 mm.

All the boards are designed with tabs**,** how is possible to see in the drawings.

So in that way we guaranteed a good join between boards, avoiding nails which are rigid joins, and for the vibration transmission are bad. So having the tabs and with glue, which being a plastic material could help to vibration insulation, all the boards will be glued. Sealing by inside with some silicon, to improve and guaranteed that all the boards are really well glued.

To build the baffles, the same method will be followed, having the boards the same tabs than the walls, so with glue these will be glued.

1.11.1.2 Doors

To access inside the box just one door will be assembled. Leaving the other side open.

That door is as is 760 x 910 mm and 30 mm thick, and it is in the part where the microphones will be placed. To can open to change the microphones if necessary. That door is composed by two hinges and one look to be able to open and close. Also a sealant tape of 3 mm thick will be place between the door and the walls to make sure that the door close well so the insulation will not be affected.

1.11.1.3 Absorbent Material

The fiber glass, because its properties, was impossible to fix it just gluing it to the walls, because due to be a really delicate material with the time it will be unstuck from the wall, or even due to it weight (in the top of the box) it could broke up. So with some strips it was fix to the wall. The high of the strips have to be the thick of the fiber glass and distance between strips 2 o 3 cm less than the width of the fiber glass, so the material will be fixed by the compression. After to protect our self from the material it will be cover by a cloth. To work with that material, gloves, work clothes and mask have to be used, because the particles of the material, are so bad for the healthy if are breathed and the skin will be itching.

1.11.1.4 Microphones

The access to the microphone cables, is performed by making two small holes in the door just to pass the cables which are connected to the microphones, and after passing the cables, close the hole with a silicon, fixing the cable really tight.

Inside the box the microphones will be hold by 8 springs, which can they can hold the board at a distance of approximately 50 cm from the corner of the walls.

These springs in one side are attached to some small wood blocks in the box corners and the other side hold the corner of a small piece of wood were the microphones are placed. As the microphones have some length 4 springs are needed for the front of the microphones and another 4 for the back part to hold it straight.

The wood board is a small piece of wood with two holds of the microphone diameters These boards are placed one in the front part and the other in the back part of the microphone to guaranteed that the microphones are fixed tight.

1.11.1.5 Baffle

The baffle how it was reported previously has been built-up gluing the boards and sealing after.

In the rear part, a hold will be made to put the speakon connector to connect in one part the baffles to the amplifiers, and in the other to connect with the 2 way filter. Which is placed inside the baffle. Also the back part, by inside, will be cover by fiber glass to absorb the rear wave that the loudspeaker creates. That fiber glass will be cover by a cloth to protect the loudspeaker and the filter of the particles.

As being a Bass Reflex a vent or port is necessary, so a tube of PVC (other material also could fix), of 17 cm diameter and 11,8 cm length.

These will go attached to the left part of the box which is open. The attachment will be, by 4 locks, being two in one left side and the other two in the right side of the baffle. In that way we guaranteed that the baffle is well attached to the box.

Like in the door, the same sealant tape will be glue in the walls to which the baffle will be attached, having in that way also a locking which will help to the insulation.

1.11.2 Small anechoic room

1.11.2.1 Walls and Baffle

As in the insulation chapter was reported, the walls are the same both for the prototype to the box, so the way to assemble the walls is also exactly the same the for the prototype, fixing the walls by the tabs and gluing them.

1.11.2.2 Doors

That part for the small anechoic room has not been determined, so several options are going to expose and when constructed should be chosen the easiest to build and which best suits to the needs of the moment.

The first option would be the same that the prototype assembling instead one, two doors, as in the small room is possible to measure small sound sources, so for that purpose should be closed by the two sides. One of the doors will be attached instead of with hinges with 4 locks, so then can be removed to attach the baffle. The easiest one but it has the disadvantage that will be difficult to access to the middle part of the box. So for that will be possible to put another small door, in the middle part, to have an access there.

That door will be as bigger as wanted. Having, as bigger the door, more access to the middle part. That door will be fixed instead of hinges with 4 screws and nuts, and opening it with two hand grips Making 4 holes in the door and introducing the door holes on the screws, which are fixed to the box, and with the nuts closing hard the door.

The other option is with hydraulic suspensions, making as door the roof or a lateral part, having in that way all the length able to operated in it.

And the third one is to open the box in a way that the box will split into two triangles, holding the top by hydraulic suspensions. The problem of that option is that will be really heavy, and so difficult to open because its weight. In the Fig. 1.11-1 some sketch is shown.

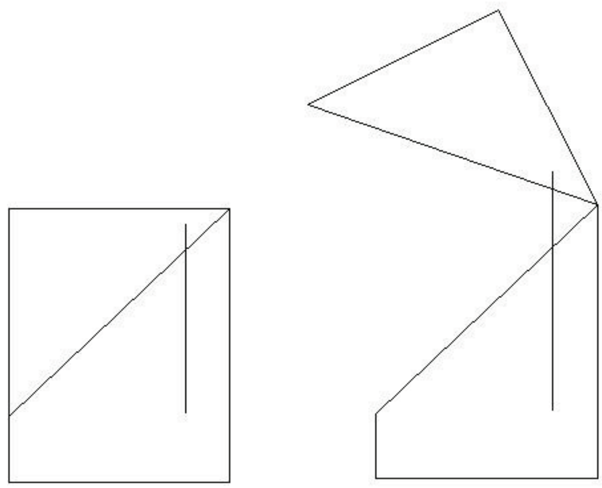


Fig. 1.11-1: Third option for the possible doors

1.11.2.3 Absorbent Material

The absorbent material in the small room will be the anechoic wedges. The way to assembled these to the room, could be by a metallic grid which is fix to the box some cm separated from the walls and in that grid the wedges will go fixed. That is the most common way to assemble anechoic wedges, because the air gap between the wedges and the wall improve the absorption in low frequencies. Despite this being the more common way, it can changes in dependence of the company which provides the wedges. So the way to assemble them will be specified by the company which provides the wedges.

Sometimes another metallic grid is assembled in front of the wedges to protect the anechoic wedges from possible hits in the actions inside the box. So that should be considered. Know that one grid over the floor wedges will enabled all the floor of the box to move the devices over it.

1.11.2.4 Microphones

The access to the microphone cables, will be performed by the same way that for the prototype, if the door is the same than in it. If not other option which suits to the new door will be research.

Inside the box, dependence on if it is decided to assembled the grid, the microphones will be places in one way or in another different.

Assembling just one grid over the floor anechoic wedges, the microphones will be placed on a stand, being sure that those are placed in the axis of the loudspeaker.

If not the same way of the prototype will be use, choosing a spring with a length which suits to place the microphone in the loudspeaker axis.

1.11.2.5 Baffle

The baffle will be the same baffle that for the prototype. Just change that the baffle does not fix with the width and height of the box. So in the side where the baffle will be attached, the board will have a hole of the baffle size of the baffle, and then it will be attached in the same way that in the prototype, with 4 locks.

References

1.- Bibliographic references

[1] Antoni Carrión Isbert ,1998, Diseño acústico de espacios arquitectónicos*, 1st edition*

[2]F. Alton Everest ,2001, The Master Handbook of Acoustics, *4th edition*, McGraw-Hill

[3]Ian R.Sinclair ,1998, Audio and Hi-Fi Handbook, *3rd edition*, Newness

[4]Lawrence E. Kinsler, Austin R.Frey, Alan B.Coppens, James V.Sandres ,2000, Fundamental of Acoustics, *4th edition*, John Willey & Sons, Madrid ISBN: 84-368-0840-1.

2.- Documented references

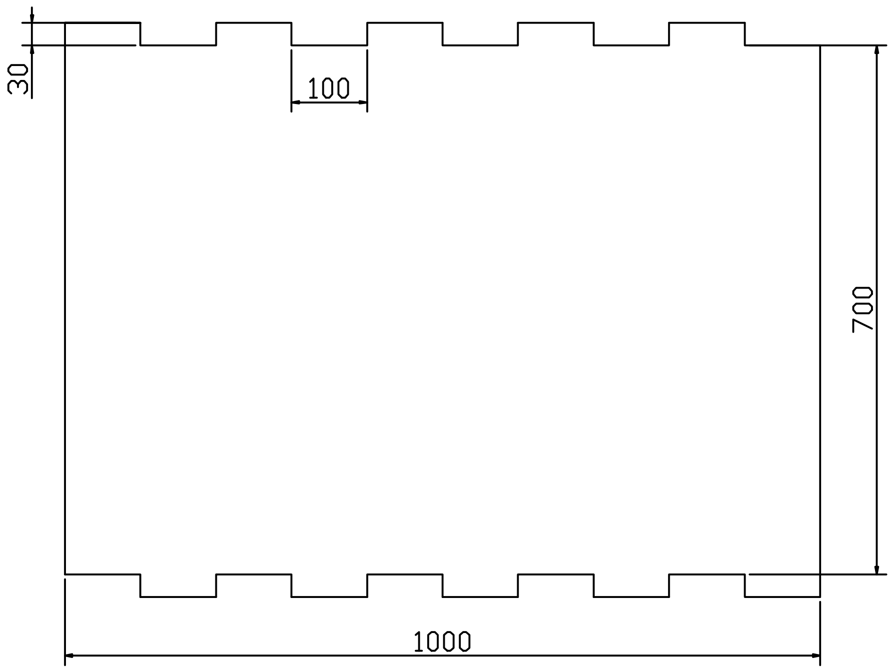
Internet:

- http://en.wikipedia.org/wiki/Main_Page
- http://profesional.beyma.com/ingles/recursos.php
- http://www.isover.net/asesoria/manuales/edificacion/Acustica_absorbentes.pdf
- http://ajdesigner.com/

Professional Verbal Sources:

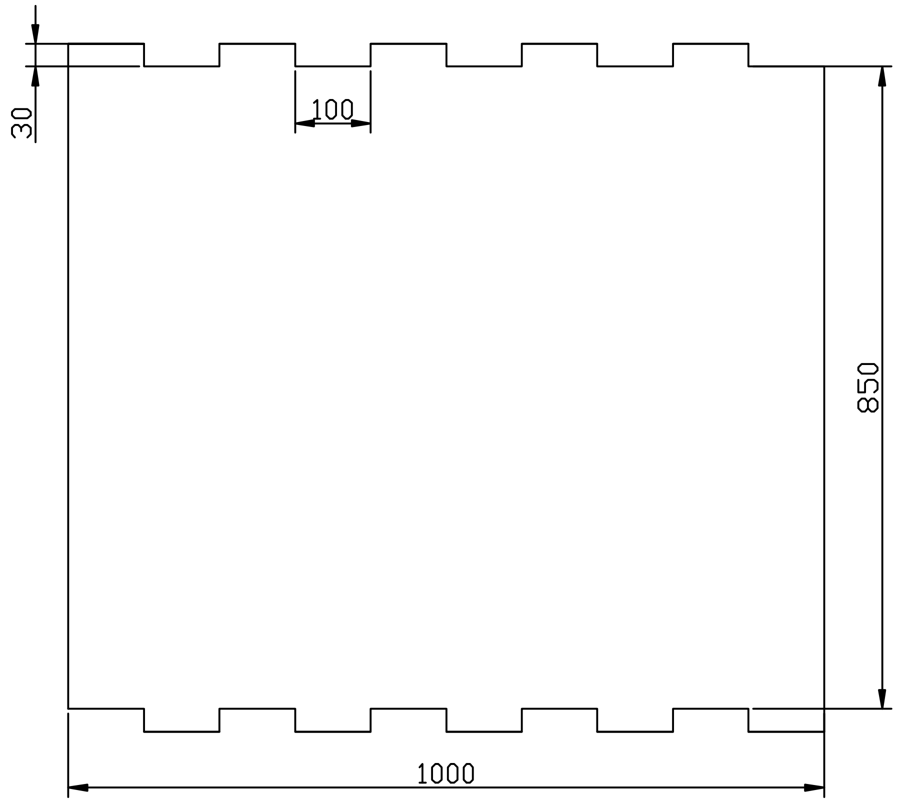
- Ricardo San Martín, Physics department of the Public University of Pamplona
- Rudy Gaspard, AED distribution product specialist.

Chapter 2. Drawings



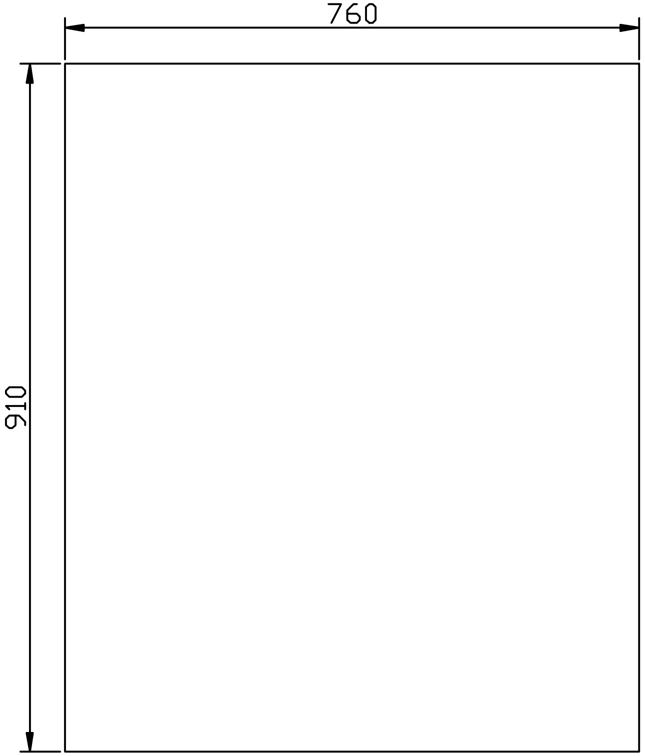
2 MDF Board 30 mm (12 + 18 mm)
(1 Top/1 Floor)

Number: 1/12	Name: Top & Floor Boards
Erasmus Hogheschool Brussel - Anecholc Chamber	



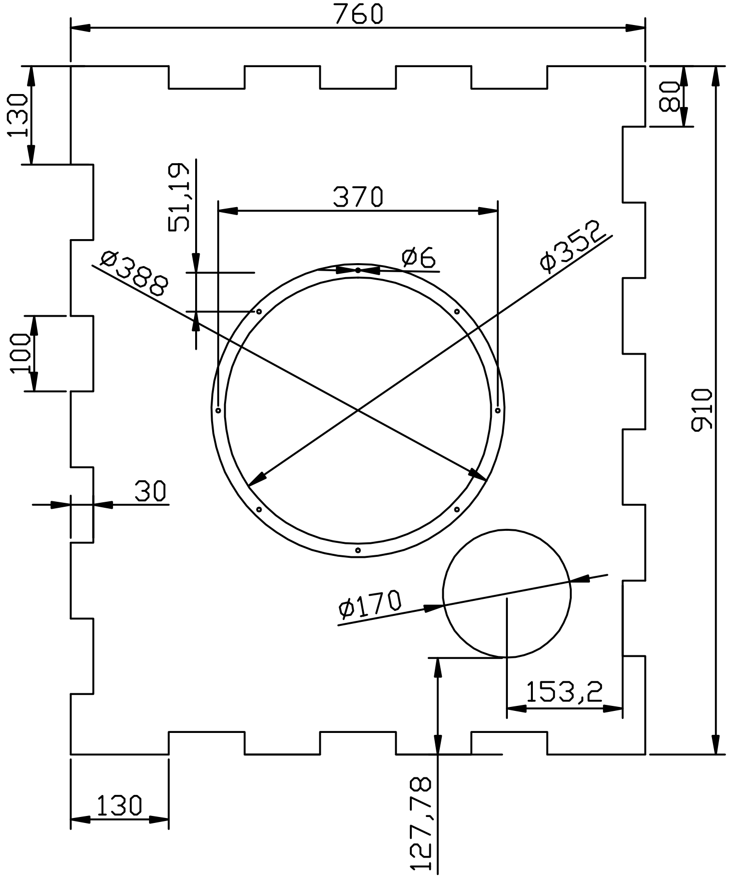
2 MDF Board 30 mm (12 + 18 mm)
(1 Right Lateral/1 Left Lateral)

Number:	Name:
2/12	Lateral Boards
Erasmus Hogheschool Brussel - Anecholc Chamber	



1 MDF Board 30 mm (12 + 18 mm)
for the Back

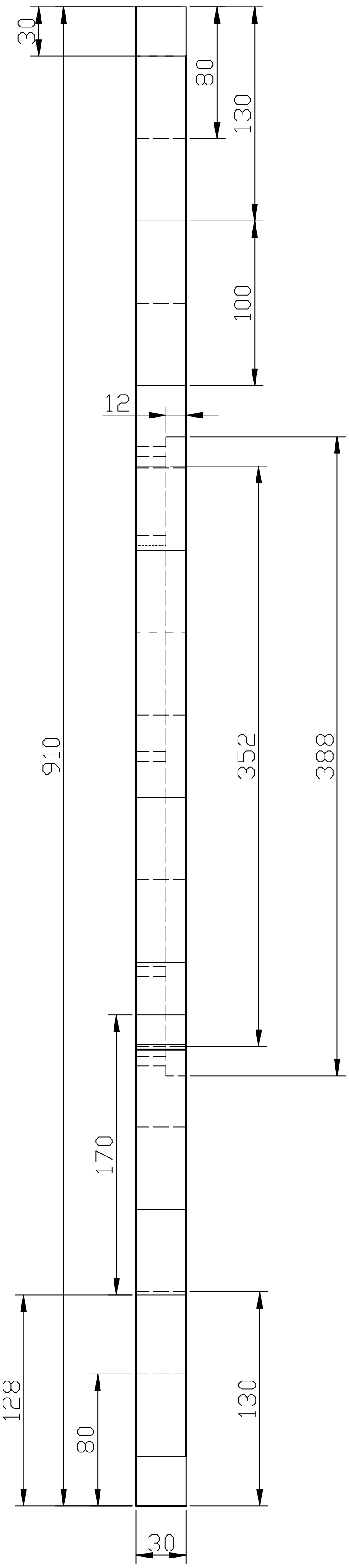
Number:	Name:
3/12	Back Board
Erasmus Hogheschool Brussel - Anecholc Chamber	



1 MDF Board 30 mm (12 + 18 mm)

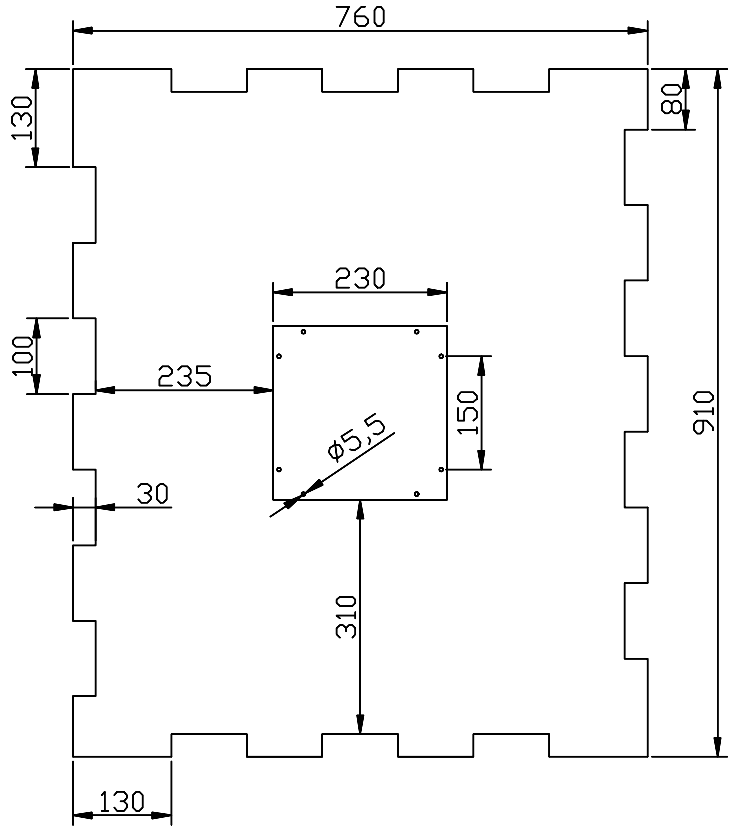
Number: 4/12	Name: Front Board - Coaxial
Erasmus Hogheschool Brussel - Baffle	

1 MDF Board 30 mm (12 + 18 mm)



Number:	Name:
5/12	Side of Front Board (Coaxial)

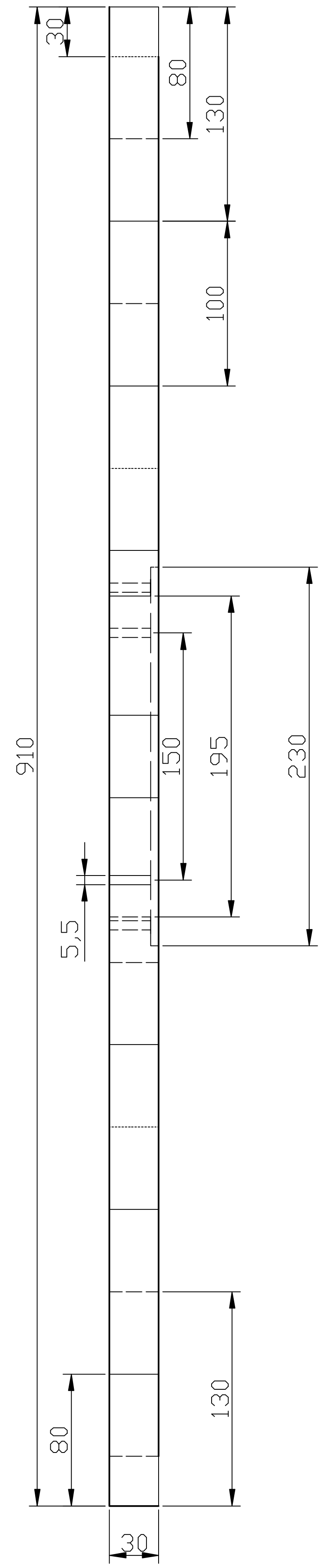
Erasmus Hogheschool Brussel
–
Baffle



1 MDF Board 30 mm (12 + 18 mm)

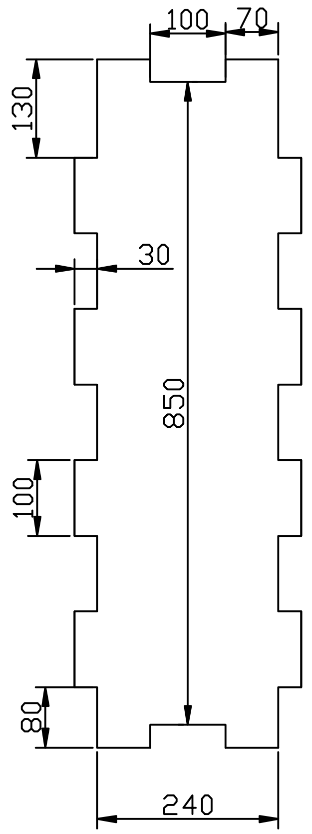
Number:	Name:
6/12	Front Board - Tweeter
Erasmus Hogheschool Brussel - Baffle	

1 MDF Board 30 mm (12 + 18 mm)



Number:	Name:
7/12	Side of Front Board (Tweeter)

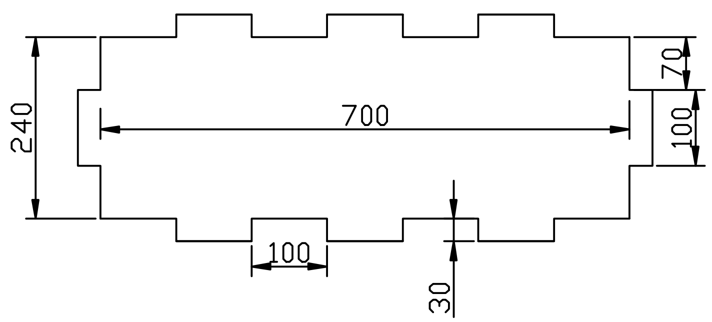
Erasmus Hogheschool Brussel
–
Baffle



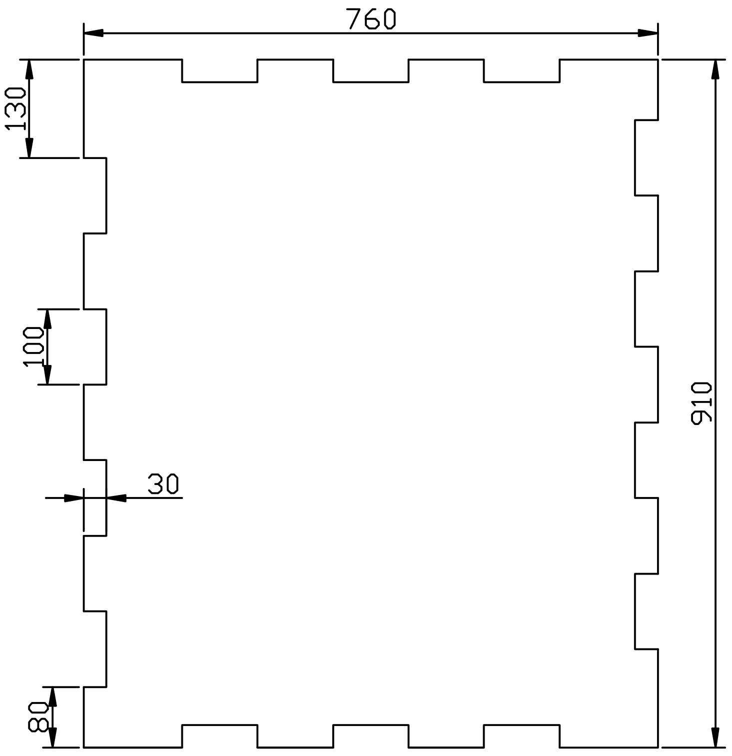
4 MDF Board 30 mm (12 + 18 mm) for the Laterals
(2 for Coaxial Baffle/2 for Tweeter Baffle)

Number: 8/12	Name: Lateral Boards
Erasmus Hogheschool Brussel - Baffle	

4 MDF Board 30 mm (12 + 18 mm) for the Laterals
(2 for Coaxial Baffle/2 for Tweeter Baffle)

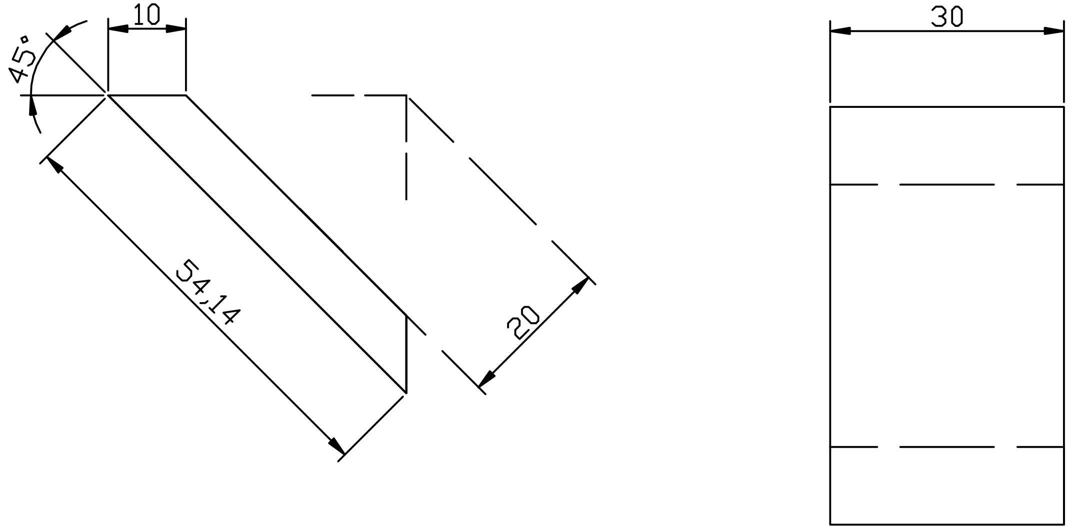


Number:	Name:
9/12	Top & Floor Boards
Erasmus Hogheschool Brussel - Baffle	



2 MDF Board 30 mm (12 + 18 mm) for the
Back
(1 for Coaxial Baffle/1 for Tweeter Baffle)

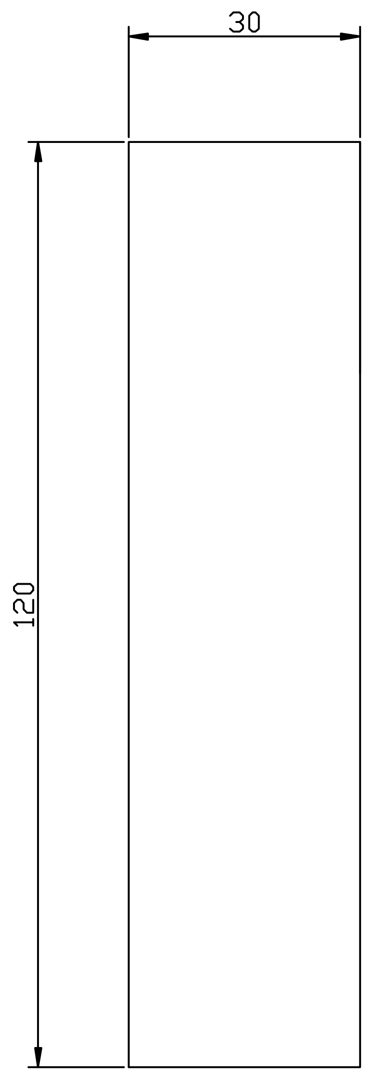
Number:	Name:
10/12	Back Boards
Erasmus Hogheschool Brussel - Baffle	



8 Blocks of 10 mm to hold the springs

Number:	Name:
11/12	Block For Springs

Erasmus Hogheschool Brussel
-
Anechoic Chamber



10 meters of 120x30 mm block
to hold the Fiber Glass

Number:	Name:
12/12	Block to Hold the Fiber Glass
Erasmus Hogheschool Brussel – Anechoic Chamber	

Chapter 3. Conclusions

Finally the conclusion is that a small anechoic room is possible to make with low cost materials.

- The dimensions chosen give a distribution of Eigen-modes which are not going to influence the acoustic characteristics of the room, avoiding energy concentrations at specific frequencies. The chosen chamber sizes are 2,10 m x 1,50 m x 1,30 m (width x length x height) for the small anechoic box, and 0,70 m x 1,00 m x 0,85 m for the prototype.

- The use of MDF boards (30 mm (18 + 12 mm sheets)), making a single wall system, is theoretically enough for the prototype insulation and also for the small anechoic room, due to the high insulation that the boards have.

- For the small anechoic room the anechoic wedges which could fit are 30 cm wedges which have a cut off frequency of 287.9 Hz.

- Since we're not in need for an esthetic absorbent material, we have used low cost fiber glass which is protected by clod. This material is very efficient even in small thickness. As such we've used 120 mm fiber glass as absorbent material.

- The vibration insulation is not necessary for the prototype as the frequency range in which we will work will be high enough that those frequencies will not affect the measurements.

- The choice of a coaxial loudspeaker is the best for our purpose. Due to the small dimensions of the room the sound emission will be perpendicular to the measuring object. This improves the relation between emitted and measured value since the amount of indirect waves will be small.

- The Bass reflex cabinet is the best solution for our purpose known that we do not have to much space, and an improvement at low frequencies is needed.

- The tweeter will provide the needed flat frequency response which is needed to make the tests with pure tones.

- The 8 channel amplifier will help to developed the future project in which several channels will be needed to emit different signals. The used system contains also a low-pass protection filter to avoid breaking the loudspeaker when sending too much energy in the lower frequency band.

Annex A. General definitions

This annex has a series of definitions of basic concepts of acoustics, if in doubt, can be consulted and then continue reading the report, being understood everything written.

1. Basics

1.1 Acoustics

It is the science that studies the various aspects of the sound. Particularly the phenomenon of generation, propagation and reception of sound waves in various media, as well as transduction, perception and its various technological applications.

1.2 Sound

Vibrations transmitted trough and elastic solid or a liquid or gas, with frequencies in the approximate range of 20 to 20.000 Hz, capable of being detected by human organs of hearing.

1.3 Sound insulation

The ability of a given structure to resist the passage of sound, using really dense materials.

1.4 Sound Proofing

It consist in reduce the pressure of the sound inside a room, or in improve some characteristics of the enclosure by absorbent materials, diffusers, or resonators

1.5 Sound Pressure

Symbol: P

Unit: Pascal Pa (1 Pa = 1 N/m^2).

Difference between total instantaneous pressure at a given point in the presence of an acoustic wave, and the static pressure at the same point.

1.6 Decibel (dB)

Symbol: dB

Decibel is the a relative unit used in acoustics to express the relationship between two acoustic variables, the magnitude is studied and a magnitude of reference. Is a logarithmic unit.

1.7 Amplitude

Symbol: A

Unit: Decibels dB.

The motion of air molecules in a wave, the pressure value at a point about the origin. As bigger motion. The larger the amplitude of the wave, the molecules hit harder in the eardrum and louder is the perceived sound.

1.8 Particle velocity

Symbol: v

Unit: m/s

Is the speed of a particle leaving the balance.

1.9 Propagation velocity

Symbol: c

Unit: m/s

Is the speed with which the wave propagates through the medium. For the sound in the air 343 m/s.

1.10 Acoustic impedance

Symbol: Z

Unit: N.s/m^3

The acoustic impedance (Z) is an intensive state property. Is the resistance that a medium impose to the waves that propagate over this, i.e. a form of energy dissipation of the waves traveling in a medium. Is defined as the ratio of sound pressure (p) and particle velocity (v) a material medium.

$$Z=\frac{P}{v}$$

1.11 Characteristic acoustic impedance

Symbol: Z_0

Unit: rayls

It can be calculated by the product of the density (ρ) of the material and the sound velocity (c) in the material. For the air Z_0=415 rayls. .

$$Z_0=\rho\cdot c$$

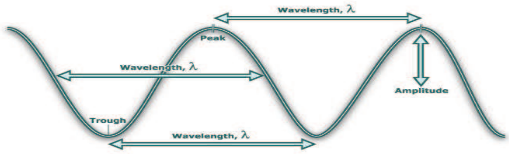
1.12 Wavelength

Symbol: λ

Unit: meter m

Distance between two consecutive points are at the same state of vibration. It determines the spatial period of the wave. Varies depending on the medium.

$$\lambda = \frac{c}{f}$$



1.13 Frequency

Symbol: f

Unit: Hertz Hz

The number of pulses of a sinusoidal acoustic wave occurred in a time of one second. Is equivalent to the inverse of the period. The band of frequencies audible to humans spans from 20 Hz in the bass to 20,000 Hz (20 KHz) in the treble.

1.14 Harmonic

Any of a series of tones whose frequencies are integral multiples of the frequency of a fundamental tone. 2fo, $3f_0$...

1.15 Harmonic Distortion (THD)

Harmonic distortion is a technical standard used to define the audio signal coming out of a system.

Harmonic distortion occurs when the output signal of a system is not equivalent to the signal entered into him. This non-linearity affects the waveform, because the equipment has introduced harmonics that were not in the input signal.

The total harmonic distortion is measured by introducing a tone of 1 kHz and measuring the output signal. In the technical parameters of the equipment, often include total harmonic distortion and is given as a percentage.
The total harmonic distortion must never be over 1%.

1.16 Frequency Spectrum

Is a representation of the energy distribution of noise in terms of its component frequencies.

1.17 Frequency Bands

Frequency range characterized by an upper and lower frequency, and the center frequency which gives the name to the band.

1.17.1. Octave Bands

Usually used for soundproofing works. The upper frequency is the double of the lower frequency. And the central is the square root of the upper and lower frequency product.

1.17.2. Third Octave Bands

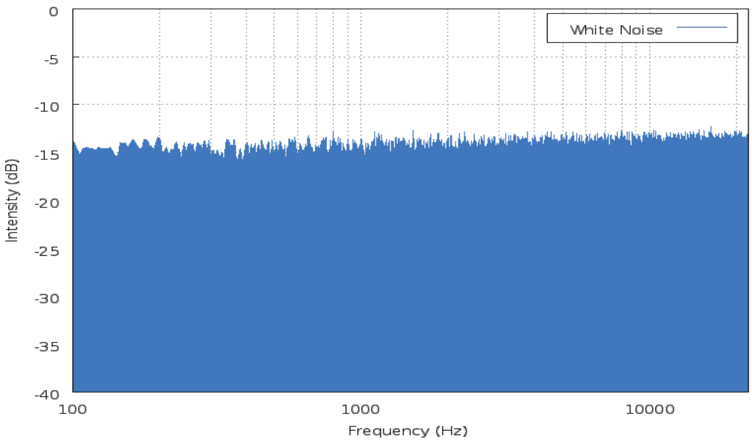
Usually used for insulation. The upper frequency is the cube root of two times the lower frequency. And the central is the square root of the upper and lower frequency product.

1.18 White and Pink Noises

Noises are random signals used to carry out standard measurements.

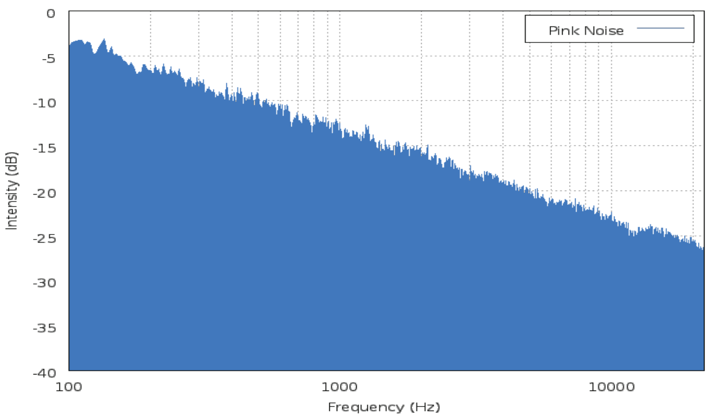
1.18.1. White Noise

Acoustical noise of which the intensity is the same at all frequencies within a given band. Its octave spectrum is a line of slope 3 dB / octave.



1.18.2. Pink Noise

Its frequency energy content decreases 3 dB per octave. This causes each frequency band of equal width (in octaves) contain the same total energy.



1.19 Sound Pressure Level

Symbol: L_p

Unit: Decibel dB

Is a logarithmic measure of the effective sound pressure of a sound relative to a reference value. Being P_0= 20 µPa.

$$L_p = 10\log\frac{P_{ef}^2}{P_0^2} = 20\log\frac{P_{ef}}{P_0}$$

$$P_{ef} = \sqrt{\frac{1}{t_2 - t_1}\int P^2(t)\,dt}$$

$$P_{ef} = \frac{P_{max}}{\sqrt{2}}$$

1.20 Sound intensity level

Symbol= L_I

Unit= Decibel dB

The intensity is the transported energy by a sound wave per area and time units. The sound intensity level is a logarithmic measure of the sound intensity in comparison to a reference value. Being I_0= 10^{-12} W/m^2.

$$L_I = 10\log\frac{I}{I_0}$$

1.21 Sound Power Level

Is a logarithmic measure of the sound power in comparison to a specified reference level. While sound pressure level is given in decibels SPL, or dB SPL, sound power is given in dB SWL. The dimensionless term "SWL" can be thought of as "sound watts level", the acoustic output power measured relative to a very low base level of watts given as 10^{-12} watts.

$$L_p = 10\log\frac{W}{W_0}$$

1.22 Free Field

Sound field in a homogeneous, isotropic medium, free of boundaries, where the sound decrease according to the inverse square law.

1.23 Inverse square law

Due to the free field, the are not reflections and any phenomenon which could increase the sound energy in any point. So the sound pressure level decrease with the distance in the way of the inverse square law.

$$SPL = L_w - 20\log r - 11$$

Where:

L_w is the power of the sound source

r the distance where the SPL want to be known

1.24 Measurement time interval

Time interval for which the time-averaged sound pressure level is determined

1.25 Weighting

The weight is used to compensate the differences in sensitivity which the human ear has to different frequencies within the hearing range. There are A, B, C, D and now Z weightings. But which is important in our project is the A weighting, which is commonly used for the measurement of environmental noise and industrial noise, and for other areas such as hearing damage and other noise health effects at all sound levels.

In the table can see the weighting value for the A-weight curve, and in the Fig. 3-1 the graphic of the curves of A, B and C weightings.

F (Hz)	20	25	31.5	40	50	63	80	100	125	160	200	250
A (dB)	-50,39	-44,82	-39,52	-34,54	-30,27	-26,22	-22,4	-19,14	-16,19	-13,24	-10,85	-8,67

F (Hz)	315	400	500	630	800	1000	1250	1600	2000	2500	3150	4000
A (dB)	-6,64	-4,47	-3,25	-1,91	-0,79	0	0,58	0,99	1,20	1,27	1,20	0,96

F (Hz)	5000	6300	8000	10000	12500	16000	20000
A (dB)	0,55	-0,12	-1,15	-2,49	-4,25	-6,71	-9,35

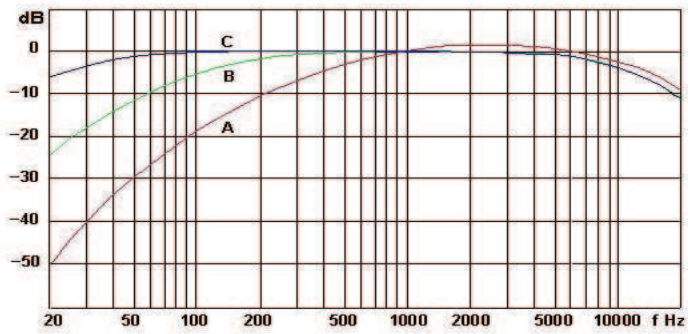
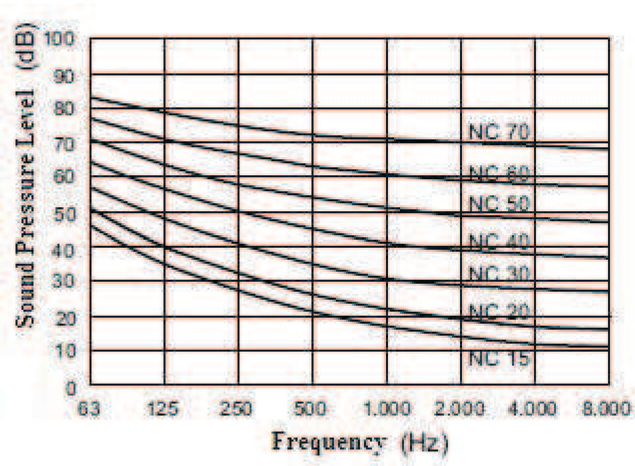


Fig. 3-1: A, B, C weighting curves

1.26 Noise Criteria Curves

To evaluate the degree of annoyance at a listener of some background noise, a comparison is made between the levels of existing background noise for each octave band between 63 Hz and 8 kHz, with the reference curves or Noise Criteria NC.

These curves are approximated to the hearing sensitivity as a function of frequency. This means that for every NC curve, so allowed pressure levels at low frequencies are higher which those for high frequencies, because the ear is less sensitive to low frequencies.



1.27 Absorption Coefficients

Symbol: α

Is the relationship between the acoustic energy absorbed by a material and the acoustic energy incident on the material, per unit area. Have to know that the absorption coefficient of a material varies considerably depending on the angle of incidence.

1.28 Absorption

Symbol: A

Unit: m^2

Is the magnitude that quantifies the energy extracted from the acoustic field when the sound wave passes through a given medium or the impact of the same surfaces with precinct boundaries.

Can be calculated by the following expressions:

$A_f = \alpha_f \cdot S$

$A = \alpha_m \cdot S$

where:
A_f is the absorption for a given frequency in m^2.
α_f is the absorption coefficient for a given frequency.
A is the average absorption in m^2.
α_m is the average absorption coefficient of a material.

1.29 Reverberation

Is the phenomenon of persistence of sound at a given point inside an enclosure, due to successive reflections in the walls of it.

1.30 Reverberation Time

Symbol: T.

Unit: seconds s.

Is the time it takes to reduce the pressure level in 60 dB after the cessation of emission of the sound source. The reverberation time varies depending on the frequency . Can be calculated by the T20 or T30 which are the measurement of 20 or 30 dB decay and after multiplied by 3 or 2. These have the advantage that the SNR have not to be so great as for the 60 dB decay measurement method of reverberation time.

Assuming diffuse field conditions (local reverberant enough such that the energy is distributed evenly throughout the enclosure and volumetric energy density constant) the reverberation time can be calculated by the expresion:

$$T = 0.161 \frac{V}{A}$$

Where:
V is the volume of the room in m^3.
A is the absorption of the local in m^2.

1.31 Signal Noise Ratio

Symbol: SNR or S/R

Unit: Decibel dB

The signal to noise ratio is defined as the margin between the power of the transmitted signal and noise power that corrupts.

A high value means that the signal is further from the existing noise and the smaller states that the noise is very close to the signal, being able to mix.

The SNR can be calculated by the power of two signals, or by the amplitude of them (voltage or pressure).

The linear expression is:

$$SNR = \frac{W_S}{W_N} = \frac{A_S^2}{A_N^2}$$

The logarithmic expresion are, for the power:

$$SNR_{dB} = 10\log\frac{W_S}{W_N} = W_{S,dB} - W_{N,dB}$$

And for the voltage or pressure:

$$SNR_{dB} = 20\log\frac{A_S}{A_N}$$

1.32 Porous Materials

Absorbent materials of granular structure, fibrous, etc.., acting through degradation of the mechanical energy into heat due to air friction with the surfaces of the material. Absorption coefficient increases with frequency.

Annex B. Loudspeakers

A loudspeaker is an electro-acoustic transducer that converts an electrical signal in sound. The speaker moves in accordance with the variations of an electrical signal causing pressure variations in the medium.

B.1 The parts of a common dynamic speaker

– Diaphragm or cone: The movement of it transforms the movement of the voice coil in pressure variations. The ideal material would be:

1) Rigid, to prevent uncontrolled cone motions.

2) Low mass, to minimize starting force requirements and energy storage issues.

3) Well damped, to reduce vibrations continuing after the signal has stopped with little or no audible ringing due to its resonance frequency as determined by its usage.

The most common are paper, plastic, and metal.

– Voice coil: Usually made of copper, though aluminum. The coil is oriented co-axially inside the gap; it moves back and forth within a small circular volume (a hole, slot, or groove) in the magnetic structure. The gap establishes a concentrated magnetic field between the two poles of a permanent magnet; the outside of the gap being one pole, and the center post (called the pole piece) being the other. The pole piece and back plate are often a single piece, called the pole plate.

– Magnet: Modern driver magnets are almost always permanent and made of ceramic, Ferrite, Alnico, or, more recently, rare earth such as Neodymium and Samarium cobalt. A trend in design is the use of the last instead of heavier ferrite types.

– Frame, Basket or Chassis: I designed to be rigid, avoiding deformation which would change critical alignments with the magnet gap, perhaps causing the voice coil to rub against the sides of the gap. Typically cast from aluminum alloy, or stamped from thin steel sheet, which can play an important role in conducting heat away from the voice coil, or plastic for inexpensive, low-mass drivers.

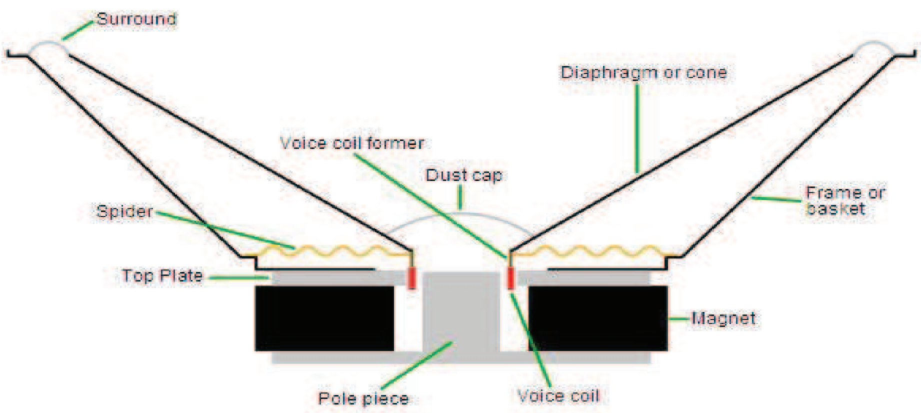
– Suspension system: In a loudspeaker can be two suspensions, the spider and the surround.

1) Spider: Made of rigid crumpled plastic, limits the displacement of the diaphragm, to prevent the coil leaves the air gap and at the same time, keep the coil perfectly centered in the air gap to avoid friction with the walls of it, which would end up breaking it.

2) Surround: Consists on rubber polyester foam, which performs a similar mission of the spider, allowing with this, only axial displacements of the cone.

– Air gap: In the air gap should concentrate the magnetic flux generated by the magnet, so it must be very narrow. Must also be long to support a great length of coil. Manufacturers often use coils longer than the gap, so that the magnetic flux intercepted by the coil remains constant, although the coil undergoes a large displacement.

– Dust Cap: The dust cap is a gently curved dome mounted in convex orientation over the central hole of most loudspeaker diaphragms and typically made of the same material of it. In some loudspeaker designs dust caps can also be part of the acoustic design of the driver by radiating high frequency energy (tweeter).



B.2 The parts of a compression driver

The compression driver speaker is intended only for indirect radiation systems, ie, to be attached to a horn. The compression driver is actually a moving coil electrodynamic loudspeaker, although it has some peculiarities, such as a compression chamber, a diaphragm, small and light and the structure to be anchored to the speaker.

Before explain all the parts of a compression driver, which work made the horn will be explained.

– Horn: Its purpose is to match the impedance of air at the speaker. In the case of speaker impedance matching is determined by the ratio between the surface of the loudspeaker membrane and the surrounding medium (air). Being lower the surface of the membrane with respect to the dimensions of the medium, the impedance matching is not good.

Known that, the parts of the compression driver are:

– The mobile set: comprises a dome-shaped diaphragm, a ring-shaped suspension, which usually is of the same material that the diaphragm, and a coil of wire. The operating principle is the same as in the case of direct radiator, it is a moving coil electrodynamic transducer.

– The phase corrector: Whose purpose is to ensure that the contributions of each part of the dome to radiation interfere positively, so keep the sound in phase. This is a solid piece which he underwent a series of concentric slots that guide the passage of sound waves from the membrane to the compression chamber. Its shape fits on top to the dome and in the bottom to compression chamber. This ensures on the one hand that the distance between the membrane and the corrective phase is constant and completely fills the compression chamber.
The operation is based on that the base of the corrector combines different sound waves coming from different areas of the dome, so the correction ensures that the way they have traveled is the same, so are in phase. If it were not the phase corrector the waves would travel on their own within the compression chamber and the result to the output of this would be poor, because it would give an arbitrary phase relationship.

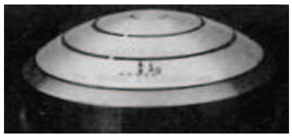


Fig. 3-2: Top view of a phase corrector

– Compression chamber: Is the volume of air between the dome and the phase corrector. If we consider that the dome is placed over the phase corrector with a small gap, we are forcing all the air in that small size cavity to pass through the slots in the phase corrector, which occupy an area much smaller than that occupied by the dome. This abrupt area change is what produces the beneficial effect of compression. This effect leads to an increase in the radiation impedance of the speaker, with the result that provides a better impedance matching.

B.3 The Plated diaphragm tweeter

Special consideration will take to that devices, being it really different from the others, in the way to transduced the AC electric current in pressure variations and with the efficiency which it makes that.

The generation of the sound takes place in a very different from the habitual one in ribbon tweeters. We can see in Fig. 3-3 that in this device, the diaphragm is formed by longitudinal folds, like in an accordion. In the straight face of each one of these folds, there is a printed by a conductive copper thread.

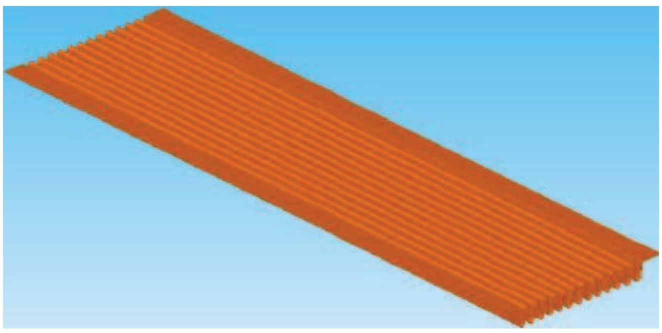


Fig. 3-3: Plated diaphragm of the TPL150H

This diaphragm is made in Beyma, on a material able to support extreme temperatures and of a great flexibility. It is used in military and aerospace applications to manufacture flexible printed circuits. It is of a great durability and reliability. The operation principle is the same one that in a regular tweeter, but geometry is very different:

Fig. 3-4: Magnet system of the TPL150H

As it is possible to be seen in Fig. 3-4, the magnetic field is closed happening by where the diaphragm is located. When an alternating electrical current circulates along the copper tracks of the membrane, a movement takes place in the folds from left to right. The sound wave is generated when the air between the folds is compressed. This causes that, being the displacement of each fold very small, the air moves at great speed, producing an important sound pressure.

So the advantages of this type of tweeter can be resumed in:

- Radiation surface four times the one of a tweeter or an equivalent compression driver, thanks to its folded geometry.
- Mobility transformer of air (Air Motion Transformer) because it causes in the air a speed four times greater than the one of the folds themselves (relation 4:1).
- This confers an enormous dynamics and an incredible transient response, vastly superior to that of any conventional tweeter, including ribbon tweeters and compression drivers.

B.4 Classification in depends on the manufacturing technology

Loudspeakers can be classified according to the manufacturing technology.

- Dynamic speaker or voice coil speaker: The electrical input signal acting on the coil creates a magnetic field that varies in meaning according to the signal. This flux interacts with a second continuous flux usually generated by a permanent magnet which is part of the body of the speaker, producing a magnetic attraction or repulsion moves the voice coil, and thus the diaphragm attached to it. To vibrate the diaphragm moves the air from in front of him, thereby generating pressure variations in it, or what is, sound waves.

– Electrostatic loudspeaker: These speakers have a capacitor structure with a fixed and moving plate (diaphragm), among which stores the energy supplied by a DC source. When increasing the energy stored between the plates, it produces a force of attraction and electrical repulsion between them, leading to the movable plate moves, creating a useful pressure. Advantages of electrostatic loudspeakers include levels of distortion one to two orders of magnitude lower than conventional cone drivers in a box, the extremely light weight of the diaphragm which is driven across its whole surface and exemplary frequency response because the principle of generating force and pressure is almost free from resonances unlike the more common electrodynamics driver. Disadvantages include a lack of bass response due to phase cancellation from a lack of enclosure

– Ribbon tweeter: A ribbon tweeter uses a very thin diaphragm (often of aluminum, or perhaps metalized plastic film) which supports a planar coil frequently made by deposition of aluminum vapor, suspended in a powerful magnetic field. The ribbon is of very lightweight material and so capable of very high acceleration and extended high frequency response.

– Indirect radiation: A horn adapt the high impedance of the loudspeaker to the low impedance of the air. This will improve the performance of the speaker. That is, more power is converted into sound, whether the horn was not used, the loudspeaker would use the same power getting less acoustic energy. They are more bulky than a usual loudspeakers and are used in professional sound reinforcement for large enclosures or large speakers mounted. The speaker of indirect radiation are composed of two parts, horn and compression driver.

B.5 Classification in depends on the frequency range

– Sub woofer: A speaker typically between 8 and 21 inches (20.3 to 53.4 cm) in diameter designed to reproduce only the lowest part of the audio spectrum, the bass frequencies. The frequency range for a sub-woofer can is about 20-200 Hz, changing depending of the speaker.

– Woofer: A driver designed to produce low frequency sounds, typically from around 40 Hz up to about a kilohertz or higher.

– Mid-range driver: A speaker designed specifically to reproduce the mid-range frequencies from approximately 300–5000 Hz, with a range typically between 6.5 and 15 inches (16.5 to 38.1 cm) in diameter.

– Tweeter: A speaker designed to reproduce the high frequencies region of the audio spectrum. Typically from around 2,000 Hz to 20,000 Hz (generally considered to be the upper limit of human hearing). Those hare categorized by their voice coil diameter, and range from 0.75 to 1.5 in (19 to 38 mm).

- Coaxial driver: A loudspeaker driver with two combined concentric drivers, a woofer with a tweeter. It frequencies responds cover the mid and high range frequencies. Have the advantage of the tweeter and the woofer being mounted very close together (coincident, hence the name coaxial (coincident axes)) which makes all of the sounds come from one point. The coaxial loudspeaker has another good characteristic which is the transition between the bass channel and the treble is less problematic than in the case of a classical system in two ways. This fact is related to the phase differences between channels: the distance between the two units causes a phase difference exists between the signal of serious and acute as you move away upwards or downwards from the central axis;
- There are 3 types of coaxial loudspeakers, which are:
    - Class A: The high frequency unit is assembled on the the back of the center of the woofer. The radiation from the high frequency unit is through a perforation made in the central core of bass. This air duct has an increasing transverse section. This air duct has a transverse section increasing: from the output of the treble unit to the top of the core of bass, in a way that provides a certain impedance matching (as a horn). At the end of this duct diffuser cap is put, whose function is to extend the impedance adapter that has been made within the nucleus of bass. Finally, elements of all bass mobile are also used for the benefit of radiation from the high frequency unit. As the cone works also like and impedance adapter, being a transverse section which increase.
    - Class B: The high frequency unit is assembled on the the back of the core of the bass unit and coupled to a small horn that is placed in the bass cone. This devices presents some inconvenient One is that because the proximity of the form to the bass cone causes some reflections which affect to the mid frequency response, being it very irregular. And the other is that despite the horn, there is still a big difference between the bass a high frequency loudspeakers.
    - Class C: The high frequency unit is assembled on the the back of the core of the woofer. Both, the high frequency unit and the bass are coupled to a horn, so the tweeters are placed within the sub-woofers The principal advantages of these devices are the high efficiency because these are indirect radiation devices, and the precision on the directive control for the whole system. But despite have these advantages also have some disadvantage such as the poor response in low frequencies, and distortion bigger for the low frequencies and introduced by the horn.

- Full-range: A driver with the speaker units mounted on substantially the same axis in the mechanical assembly; designed to have the widest frequency response possible. These drivers are typically 3 to 12 inches (7.6 to 30.5 cm) in diameter to permit reasonable high frequency response and carefully designed to give low-distortion output at low frequencies, though with reduced maximum output level.

B.6 Application Parameters

– Nominal Impedance: "Opposition" of a driver or a circuit to the flow of an alternating current. The market values of impedance speakers and woofers are 8, 4, 2 and 1 ohm. None of these values is better or worse, one must have direct correspondence with the output impedance of the amplifier to prevent damage to your circuit.

– Power Handling (P_e): Indicates the maximum amount of heat that support the copper wire of the coil before permanent thermal damage occurs. The larger value leads to a greater power will support and stronger will be the sound it produces. Power handling is typically specified in watts (W).
The beyma loudspeakers change that specification from RMS watts to the Standard of the Audio Engineering Society of North America (AES).
The fatigue of the speakers should be done with pink noise, filtered in the bandwidth of the speaker application. This is done by air, no box or panel of any kind, for two hours. The power will be taken as correct one for which no speaker settings have been altered more than 10% of its nominal value.
The difference is that in the RMS value the power was calculated with the nominal impedance. And for AES watts with the minimum impedance. So being both impedance really close the value of the watt will not change so much. With the expression of the power can be calculated both values. Changing the value of the impedance between the minimum or the nominal impedance.

$$P=\frac{V^2}{Z}$$

– Sensitivity: Sound pressure level produced by a speaker at 1 meter with 1 watt of power. Between two speakers of the same characteristics of frequency response, power and impedance, stronger will sound which has greater sensitivity. Specified in decibels (dB).

– Frequency Response: Frequency range covered by the device with a maximum deviation of + / - 3 dB.

B.7 Thiele/Small parameters

“Thiele/Small” commonly refers to a set of electromechanical parameters that define the specified low frequency performance of a driver. These parameters are published in specification sheets by driver manufacturers so that designers have a guide in selecting off-the-shelf drivers for loudspeaker designs. Many of the parameters are strictly defined only at the resonant frequency, but the approach is generally applicable in the frequency range where the diaphragm motion is largely pistonic, when the entire cone moves in and out as a unit without cone breakup.

Using these parameters, a loudspeaker designer may simulate the position, velocity and acceleration of the diaphragm, the input impedance and the sound output of a system comprising a loudspeaker and enclosure. Rather than purchase off-the-shelf components, loudspeaker design engineers often define desired performance and work backwards to a set of parameters and manufacture a driver with said characteristics or order it from a driver manufacturer. This process of generating parameters from a target response is known as synthesis.

B.7.1 Small Signal Parameters

- Resonance Frequency (F_s): The frequency at which the combination of the energy stored in the moving mass and suspension compliance is maximum, and results in maximum cone velocity. Also called F_0, measured in hertz (Hz).

- Mechanical Q factor (Q_{ms}): Characterizes the mechanical damping of the driver, that is, the losses in the suspension (surround and spider). It varies roughly between 0.5 and 10, with a typical value around 3. High Q_{ms} indicates lower mechanical losses, and low Q_{ms} indicates higher losses. The main effect of Q_{ms} is on the impedance of the driver, with high Q_{ms} drivers displaying a higher impedance peak. Is a unit-less measurement.

- Electrical Q factor (Q_{es}): Describes the electrical damping of the loudspeaker. As the coil of wire moves through the magnetic field, it generates a current which opposes the motion of the coil. This so-called "Back-EMF" (proportional to Bl * velocity) decreases the total current through the coil near the resonance frequency, reducing cone movement and increasing impedance. In most drivers, Q_{es} is the dominant factor in the voice coil damping. A unit-less measurement.

- Total Quality factor (Q_{ts}): Characterizes the combined electric and mechanical damping of the driver. In electronics, Q is the inverse of the damping ratio. The value of Q_{ts} is proportional to the energy stored, divided by the energy dissipated, and is defined at resonance (F_s). Most drivers have Q_{ts} values between 0.2 and 0.5, but there are valid (if unusual) reasons to have a value outside this range.

- Equivalent Volume (V_{as}): Is a measure of the 'stiffness' of the suspension with the driver mounted in free air. It represents the volume of air that has the same stiffness as the driver's suspension when acted on by a piston of the same area (S_d) as the cone. Larger values mean lower stiffness, and generally require larger enclosures. V_{as} varies with the square of the diameter. A typical factory tolerance for V_{as} spec is ±20–30%. Measured in liters (L) or cubic meters (m^3).

B.7.2 Fundamental Small Mechanical Parameters

- Moving Mass (M_{ms}): Is the mass of the cone, coil and other moving parts of a driver, including the acoustic load imposed by the air in contact with the driver cone. Measured in grams (g) or kilograms (Kg).

- DC Resistance (R_e): Is the DC resistance of the voice coil. American EIA standard RS-299A specifies that DCR should be at least 80% of the rated driver impedance. Measured in ohms (Ω).

– Mechanical Complement (C_{ms}): Describes the compliance (ie, the inverse of stiffness) of the suspension. The more compliant a suspension system is, the lower its stiffness, so the higher the V_{as} will be. C_{ms} is proportional to Vas and thus has the same tolerance ranges. Measured in meters per Newton (m/N).

– BL factor (BL): The product of magnet field strength in the voice coil gap and the length of wire in the magnetic field, also known as the 'force factor' because the force on the coil imposed by the magnet is $B\times l$ multiplied by the current through the coil. The higher the $B\times l$ value, the larger the force generated by a given current flowing through the voice coil. $B\times l$ has a very strong effect, indirectly lower Q_{es} and Q_{ts}. Measured in Tesla-metres (T·m).

– Effective Piston Area (S_d): The projected area of the driver diaphragm, depends largely on the shape and properties of the surround. Generally accepted as the cone body diameter plus one third to one half the width of the annulus (surround). Drivers with wide roll surrounds can have significantly less S_d than conventional types with the same frame diameter. Measured in square meters (m^2).

– Voice Coil Inductance (L_e): Is the inductance of the voice coil. The coil is a lossy inductor, in part due to losses in the pole piece, so the apparent inductance changes with frequency. Large L_e values limit the high frequency output of the driver and cause response changes near cutoff. Inductance varies with excursion because the voice coil moves relative to the pole piece, which acts as a sliding inductor core, increasing inductance on the inward stroke and decreasing it on the outward stroke in typical overhung magnet arrangements. This inductance modulation is an important source of non linearity (distortion) in loudspeakers. Measured in millihenries (mH).

B.7.3 Large Signal Parameters

– Max. Linear Excursion (X_{max}): Maximum linear peak (or sometimes peak-to-peak) excursion of the cone. Is an indicator of a loudspeaker motor's linear range. Note that, because of mechanical issues, the motion of a driver cone becomes non-linear with large excursions, especially those in excess of this parameter. Measured in mm or inch

– Max. Physical Excursion (X_{mech}): Maximum physical excursion of the driver before physical damage. With a sufficiently large input, the excursion will cause damage to the voice coil or other moving part of the driver.

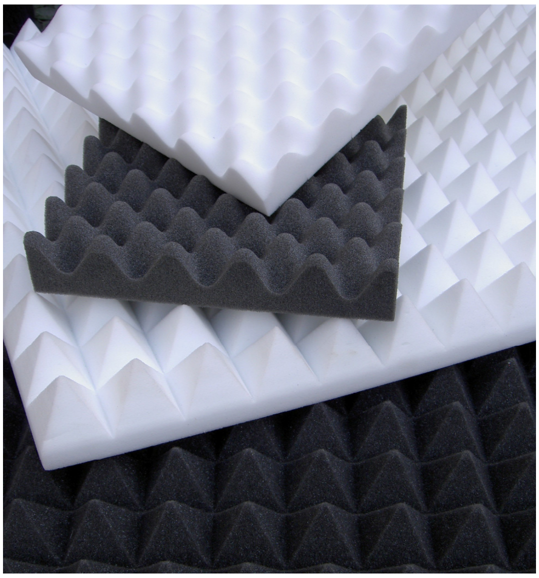
– Peak Displacement Volume (V_d): Refers to the volume displaced by the cone, equal to the cone area multiplied by Max.Linear Excursion ($V_d = S_d \cdot X_{max}$). A particular value may be achieved in any of several ways. For instance, by having a small cone with a large X_{max}, or a large cone with a small X_{max}. Comparing V_d values will give an indication of the maximum output of a driver at low frequencies. Specified in liters (L). High X_{max}, small cone diameter drivers are likely to be inefficient, since much of the voice coil winding will be outside the magnetic gap at any one time and will therefore contribute little or nothing to cone motion. Likewise, large cone diameter, small X_{max} drivers are likely to be more efficient as they will not need, and so may not have, long voice coils.

Annex C. Datasheets

Fiche Technique

PYRAMIDE

Le panneau **PYRAMIDE** est une mousse à haut pouvoir d'absorption, en mélamine ou polyuréthane (PU), à relief pyramidal. Les panneaux **PYRAMIDE** sont particulièrement adaptés à l'absorption acoustique dans les locaux et à la diminution de la réverbération.



COMPOSITION :

Mousse absorbante à relief pyramidal en mélamine ignifugée ou polyuréthane auto-extinguible (PU).

AVANTAGES :

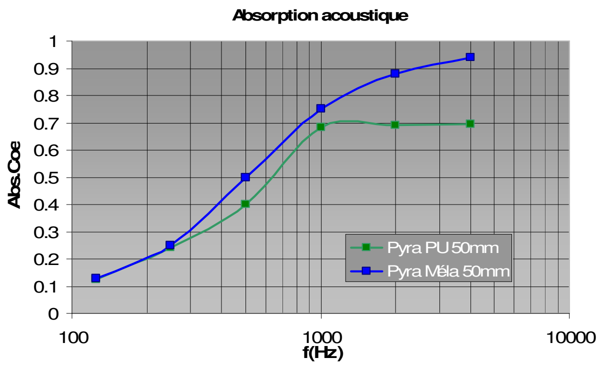
- Excellente absorption acoustique sur une large bande de fréquences.
- Amélioration du confort acoustique.
- Non-fibreux et non-volatile (contrairement aux laines minérales).
- Mise en œuvre collée économique et facilitée par son faible poids.
- Résistance au feu M1 pour les versions en mélamine.
- Possibilité de mise en œuvre en doublage de capotage pour la version plane…

CARACTERISTIQUES TECHNIQUES :

	Polyuréthane (PU)	**Mélamine**
Dimensions :	1000 x 1000 mm	1200 x 600 mm
Epaisseurs standard :	50 et 100 mm	50 mm
Aspect :	Gris anthracite	Gris clair ou blanc
Poids :	30 kg/m³	11 kg/m³
Combustibilité :	M4	M1-B1

PERFORMANCES ACOUSTIQUES :

f (Hz)	125	250	500	1000	2000	4000
Pyra PU 50mm	0.12	0.24	0.40	0.68	0.69	0.69
Pyra Méla 50mm	0.13	0.25	0.50	0.75	0.88	0.94

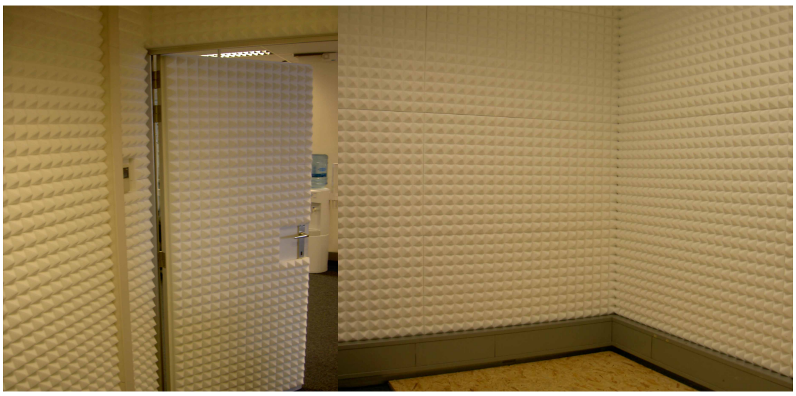


MISE EN ŒUVRE :

Panneaux à découper au couteau, et à coller directement sur le support propre et sec.

APPLICATIONS :

- Absorption acoustique dans les locaux (musique, voix,…)
- Diminution de la réverbération (acoustique des salles).
- Doublage absorbant de capotages et encoffrements de machines.



dB MUTE SPRL - INSONORISATION ET TRAITEMENT ACOUSTIQUE - ETUDES / REALISATIONS / MATERIAUX
57, rue de Theux B-1040 Bruxelles **Tél.** +32(0)2 538.63.10 **Fax.** +32(0)2 538.81.39 **e-mail:** info@dbmute.com
Website : www.dbmute.com - **TVA** 873.201.720 - **N° enr.** 03/16/01 - **ING** 310-1890717-79

beyma
PROFESSIONAL LOUDSPEAKERS

15XA38Nd

15XA38Nd

Coaxial

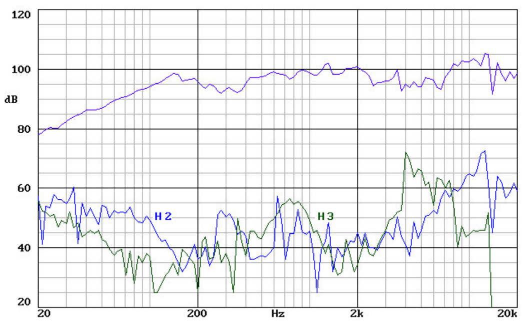
KEY FEATURES

- 15” bass loudspeaker and 2” exit compression driver combination
- Excellent power handling: 350 w AES (L.F. unit) and 90 w AES (H.F. unit)
- Extended and linear frequency response (25-20000 Hz)
- High sensitivity: 99 dB (L.F. unit) and 105 dB (H.F. unit)
- Low weight (common neodymium magnet system for both units)
- Bass loudspeaker designed for compact bass-reflex cabinets

GENERAL DESCRIPTION

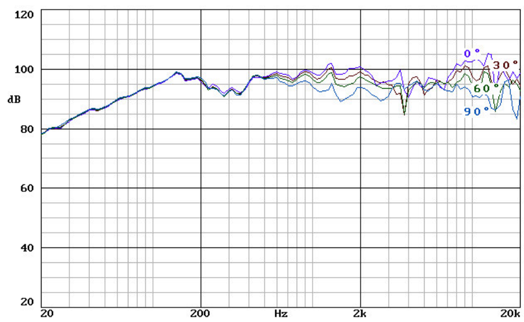
This 15” coaxial loudspeaker is intended for the most demanding professional applications. Its low frequency unit features a 4” (100 mm) edgewound aluminium ribbon voice coil capable of handle 350 w AES. This bass unit has been optimized in order to fit in with the most compact bass-reflex systems. The high frequency unit uses a 2.8” (72.2 mm) aluminium ribbon voice coil and a composite structure diaphragm, that is to say, a titanium dome and polyester surround combination. All these components give as a result a coherent and extended frequency response with low distortion that ranges from 25 Hz up to 20 kHz. Moreover, the use of a common neodymium magnet system for both units reduces the weight to 6.8 kg.

FREQUENCY RESPONSE AND DISTORTION CURVES

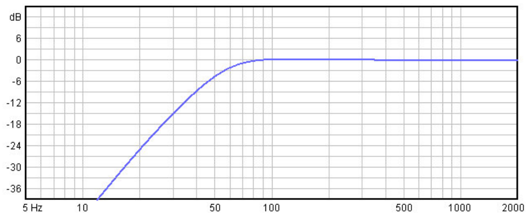


Note: on axis filtered frequency response measured with loudspeaker standing on infinite baffle in anechoic chamber, 1w @ 1m.

FREQUENCY RESPONSE OUT OF AXIS

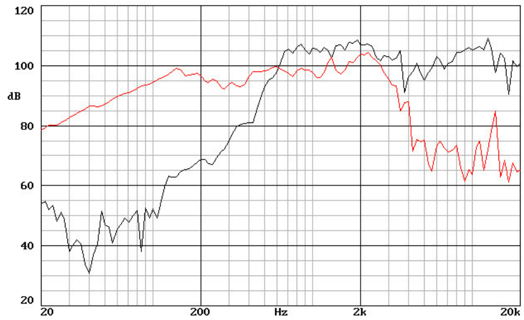


PREDICTED LOW FREQUENCY RESPONSE



Note: bass-reflex cabinet, Vb=100 l, fb=42 Hz

FREQUENCY RESPONSE OF LF & HF UNITS



Note: on axis frequency response of low and high frequency units, 1w @ 1m.

beyma
PROFESSIONAL LOUDSPEAKERS

15XA38Nd

Coaxial

TECHNICAL SPECIFICATIONS

L.F. UNIT

Nominal diameter	380 mm. 15 in.
Rated impedance	8 ohms.
Minimum impedance	7.6 ohms.
Power capacity*	350 w AES
Program Power	700 w
Sensitivity	99 dB 2.83v @ 1m @ 2π
Frequency range	25-3500 Hz
Recom. enclosure vol.	60 / 180 l 2.14 / 6.35 $ft.^3$
Voice coil diameter	100 mm. 4 in.
Magnetic assembly weight	4.2 kg. 9.24 lb.
BL factor	18.2 N/A
Moving mass	0.072 kg.
Voice coil length	16 mm.
Air gap height	9 mm.
X damage	28 mm.

H.F. UNIT

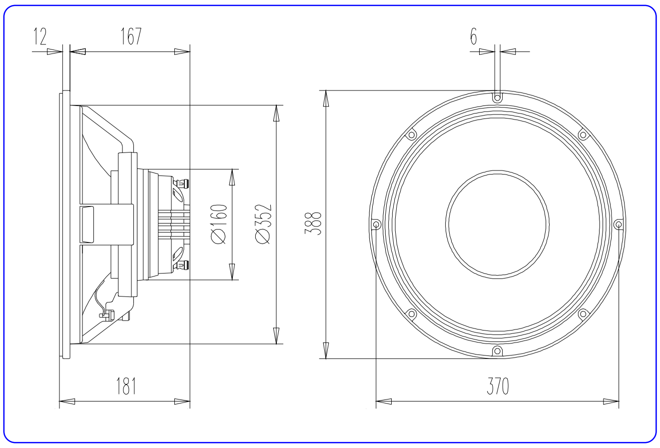
Rated impedance	16 ohms.
Minimum impedance	13.5 ohms.@ 3.5 kHz
Power capacity	90 w AES above 1 kHz
Frequency range	0.5 - 20 kHz
Sensitivity 1w @ 1m	105 dB
Voice coil diameter	72.2 mm. 2.87 in.
Flux density	1.6 T
BL factor	15.3 N/A
Dispersion	80° conical

THIELE-SMALL PARAMETERS

Resonant frequency, fs	33 Hz
D.C. Voice coil resistance, Re	6.8 ohms.
Mechanical Quality Factor, Qms	7.50
Electrical Quality Factor, Qes	0.30
Total Quality Factor, Qts	0.29
Equivalent Air Volume to Cms, Vas	359 l
Mechanical Compliance, Cms	326 μm / N
Mechanical Resistance, Rms	2 kg / s
Efficiency, ηo (%)	4
Effective Surface Area, Sd (m^2)	0.088 m^2
Maximum Displacement, Xmax	4 mm
Displacement Volume, Vd	350 cm^3
Voice Coil Inductance, Le @ 1 kHz	1.6 mH

Notes:
*The power capacity is determined according to AES2-1984 (r2003) standard.
Program power is defined as the transducer's ability to handle normal music program material.

DIMENSION DRAWINGS



MATERIALS

L.F. UNIT
- **Basket:** Die cast aluminium
- **Cone:** Paper
- **Surround:** Plasticised cloth
- **Voice coil:** Edgewound aluminium ribbon
- **Magnet:** Neodymium

H.F. UNIT
- **Dome:** Titanium
- **Surround:** Polyester
- **Voice coil:** Edgewound aluminium ribbon
- **Voice coil former:** Kapton

MOUNTING INFORMATION

Overall diameter	388 mm.	15.28 in.
Bolt circle diameter	370 mm.	14.56 in.
Baffle cutout diameter:		
- Front mount	352 mm.	13.85 in.
- Rear mount	352 mm.	13.85 in.
Depth	181 mm.	7.13 in.
Volume displaced by driver	7 l	0.25 $ft.^3$
Net weight	6.8 kg.	14.96 lb.
Shipping weight	7.4 kg.	16.28 lb.

****T-S parameters are measured after an exercise period using a preconditioning power test.**
The measurements are carried out with a velocity-current laser transducer and will reflect the long term parameters (once the loudspeaker has been working for a short period of time).

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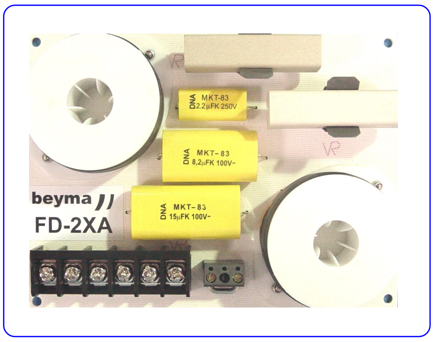
Polígono Industrial Moncada II · C/. Pont Sec, 1c · 46113 MONCADA - Valencia (Spain)
· Tel. (34) 96 130 13 75 · Fax (34) 96 130 15 07 · http://www.beyma.com · E-mail: beyma@beyma.com ·

beyma
PROFESSIONAL LOUDSPEAKERS

FD2XA / FD2XC1 / FD2XC2

FILTERS



KEY FEATURES

- 2 way filters
- 12 dB/oct. attenuation slope
- Additional high frequency attenuation
- High power components
- Specially designed for the use with Beyma coaxials

GENERAL DESCRIPTION

These 2 way passive filters have been specifically designed to be used with the XA and XC Beyma coaxials, in order to achieve an optimum performance of these drivers. Please, see attached filter selection guide to find the suitable filter model for each coaxial model.

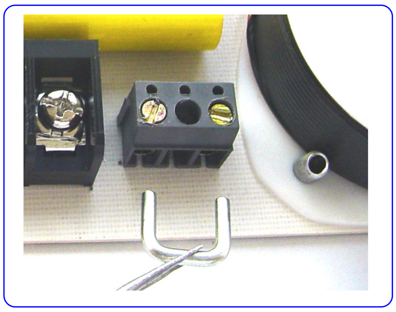
These filters offer the possibility of an additional attenuation of the high frequency unit. Anyway, this attenuation is only optional, optimum performance of the coaxial is achieved without using it. However, if you wish to use it you can activate it by simply removing the aluminium bridge labeled as «AT» on the back surface of the board (please see attached picture).

TECHNICAL SPECIFICATIONS

Type	2 way
Crossover frequency*	1.8 / 2 / 2.2 kHz
Power capacity	600 w
Low / High frequency impedance	8 / 16 ohms.
Attenuation slope	12 dB/oct.
H.F. attenuation	0, -1.5 dB

*Note: crossover frequency depends on model (FD2XA / FD2CX2 / FD2XC1).

ADDITIONAL HF ATTENUATION

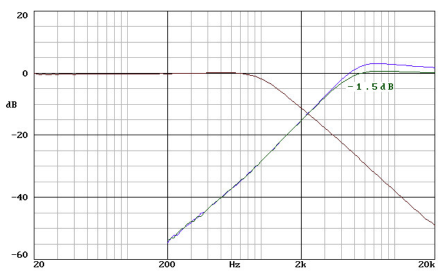


FILTER SELECTION GUIDE

FD2XA	Use with 15XA38Nd & 12XA30Nd
FD2XC1	Use with 12XC30 & 8XC20
FD2XC2	Use with 10XC25

TRANSFERENCE FUNCTIONS

Note: FD2XA transference function.



Note: FD2XC1 transference function.

Note: FD2XC2 transference function.

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beyma

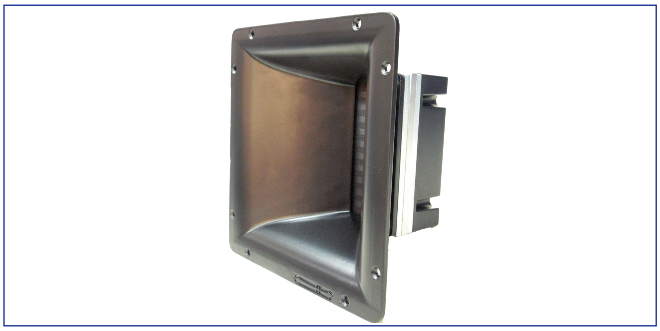
PROFESSIONAL LOUDSPEAKERS
www.beyma.com

TPL-150H

PLEATED DIAPHRAGM TWEETER WITH HORN

KEY FEATURES

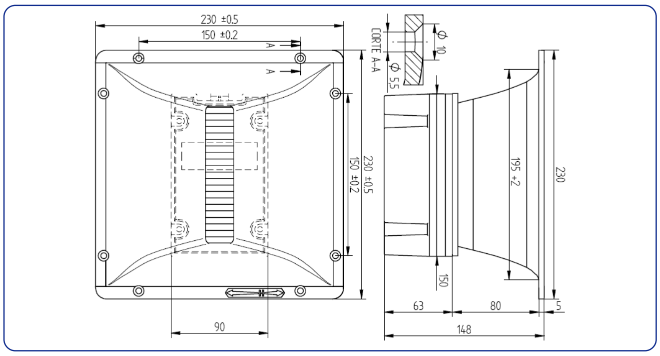
- Superior sound reproduction quality
- Extended frequency response (0.7 - 23 kHz)
- Coverage angles of 80º in the horizontal plane and 30º in the vertical plane
- High sensitivity (102dB)
- Extremely linear frequency response
- Low distortion
- Precise directivity control in the pass band



TECHNICAL SPECIFICATIONS

Rated impedance	8 ohms
D.C Resistance	4.9 ohms
Power capacity*	80 w AES above 1 kHz
Program power	160 w above 1 kHz
Sensitivity	102 dB 1w @ 1m
Frequency range	0.7-23 kHz
Recommended crossover	1kHz or higher (12dB/oct. min)
Horizontal beamwidth	80º(+9º,-20º) (--6dB, 1.2-16 kHz)
Vertical beamwidth	30º(+27º,-21º) (--6dB, 1.2-16 kHz)
Directivity factor (Q)	27 (average 1.2-16 kHz)
Directivity index (DI)	13 dB (+6 dB, -4.5)
Cutoff frequency	800 Hz
Overall dimensions (WxHxD)	230X230X148 mm. 9.05x9.05x5.8 in.
Cutout dimensions (WxH)	195x195 mm. 7.68x7.68 in.
Net Weight	3.68kg. 8.10 lb.
Shipping Weight	4.08 kg. 8.76 lb.

DIMENSION DRAWINGS

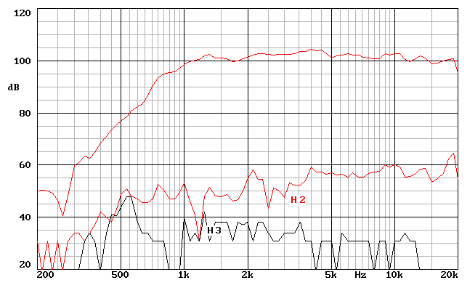


Notes:

*The power capacity is determined according to AES2-1984 (r2003) standard.
Program power is defined as the transducer's ability to handle normal music program material.

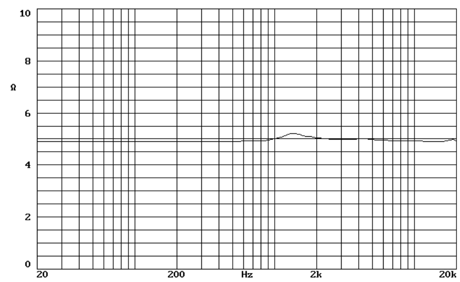
**Sensitivity was measured at 1 m distance, on axis, with 1 w input, averaged in the range 2.5 - 20 kHz.

FREQUENCY RESPONSE AND

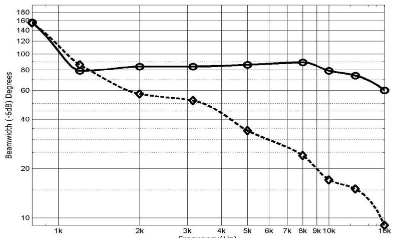


Note: on axis frequency response measured in anechoic chamber, 1w @ 1m.

FREE AIR IMPEDANCE CURVE

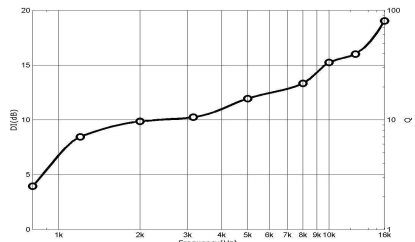


-6dB BEAMWIDTH



Note: Horizontal beamwidth is represented by heavy line.
Vertical beamwidth is represented by the discontinuos line.

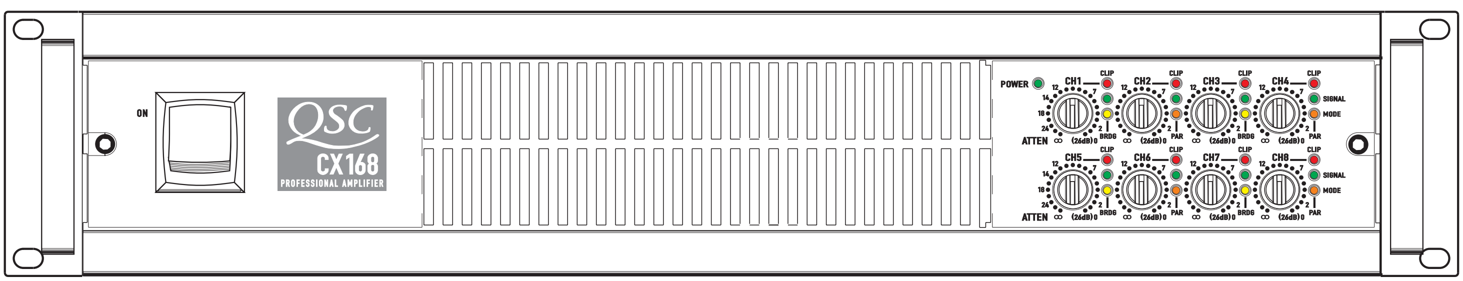
DIRECTIVITY INDEX



CX168

8-Channel Installed Sound Professional Audio Amplifier

User Manual



TD-000095-00 rev.C

IMPORTANT SAFETY PRECAUTIONS
& EXPLANATION OF SYMBOLS

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE THE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED PERSONNEL.

The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated "dangerous" voltage within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to humans.

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in this manual.

The lightning flashes printed next to the OUTPUT terminals of the CX168 amplifier are intended to alert the user to the risk of hazardous energy. Output connectors that could pose a risk are marked with the lightning flash. **Do not touch output terminals while amplifier power is on. Make all connections with amplifier turned off.**

WARNING: TO PREVENT FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE.

This amplifier has a serial number located on the rear panel.
Please write this and the model number down and keep them for your records.

Model: CX168
Serial Number:________________________________
Date of Purchase:________________________________
Purchased From:________________________________

FCC INTERFERENCE STATEMENT

NOTE: This equipment has been tested and found to comply with the limits for a class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance to the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by switching the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and the receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio or TV technician for help.

TABLE OF CONTENTS

INTRODUCTION

Congratulations on your purchase of the CX168 power amplifier. To help you obtain the best results from your purchase, we encourage you to carefully review this manual. With information including connections, configuration and operation, it offers many useful guidelines.

The CX168 provides a high level of channel density in a small, lightweight form factor ideal for multi-zone audio systems. Representing the culmination of QSC's extensive experience in power amplifier development, the CX168 is a highly versatile, reliable, and user-friendly tool that will likely remain a central component of your system for years to come.

Like the entire CX Series, the CX168 is equipped with QSC's exclusive PowerWave™ switching power supply technology to virtually eliminate noise and hum while reducing the unit's overall weight. Comprehensive protection circuitry includes DC, infrasonic, thermal overload and short circuit protection.

The CX168 includes 4 HD15 DataPorts (one per channel pair) for remote management or DSP. These ports allow each channel pair to be governed by QSControl, QSC's audio networking system—enabling the system operator to control amplifier gain levels, check the unit's clipping and thermal status, plus monitor numerous additional system parameters.

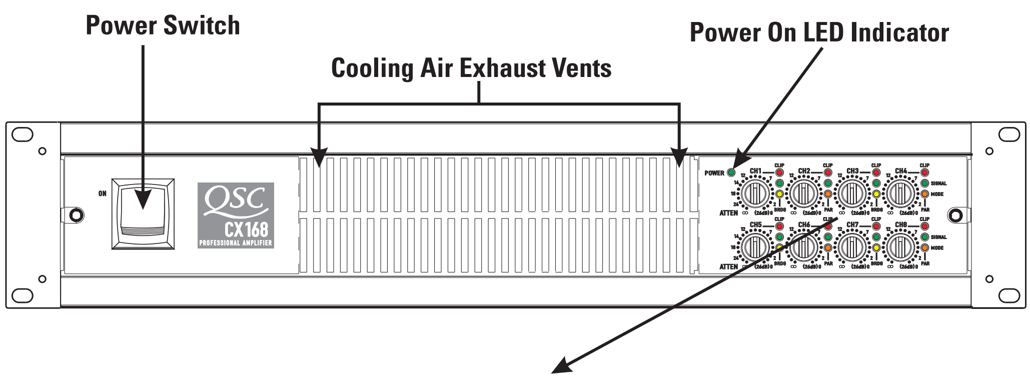
Alternatively, these DataPorts facilitate connection of signal processing equipment, such as QSC's DSP-3, which offers two channels of independent digital signal processing - including crossover filters, shelf filters, signal delay, compression, peak limiting, and parametric filters.

As individual channel pairs may be bridged, the CX168 is configurable as a 4-, 5-, 6-, 7-, or 8-channel unit. Further, the CX168 features detachable Phoenix-style input and output connectors—enabling the audio system to be wired with greater ease. Other features include 1-dB recessed and detented gain controls and a security cover for tamper-proof installations.

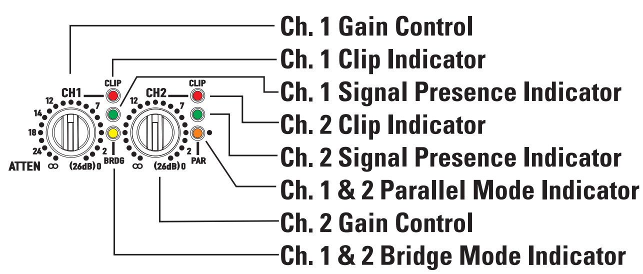
The CX168 is easy to use. All operating mode switches are grouped together on the rear panel, with one switchblock per channel pair. These switches enable clip limiter on/off, LF filter on/off, LF filter select and operation modes. The amplifier's front panel includes corresponding LEDs to indicate Bridged or Parallel modes in addition to Signal and Clip status.

Your CX168 power amplifier was designed to provide many years of trouble-free, great-sounding operation. We hope you enjoy your new CX168.

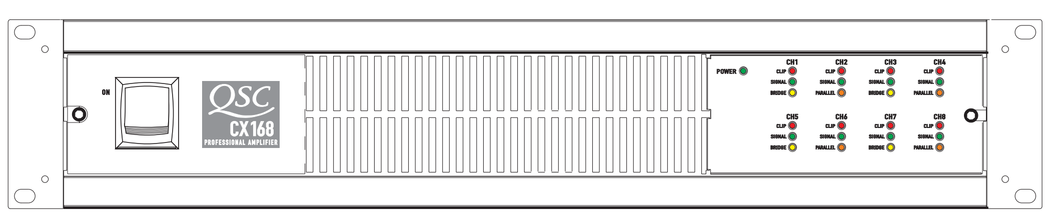
FRONT PANEL



GAIN CONTROL SECTION (Ch. 1 and 2 shown)

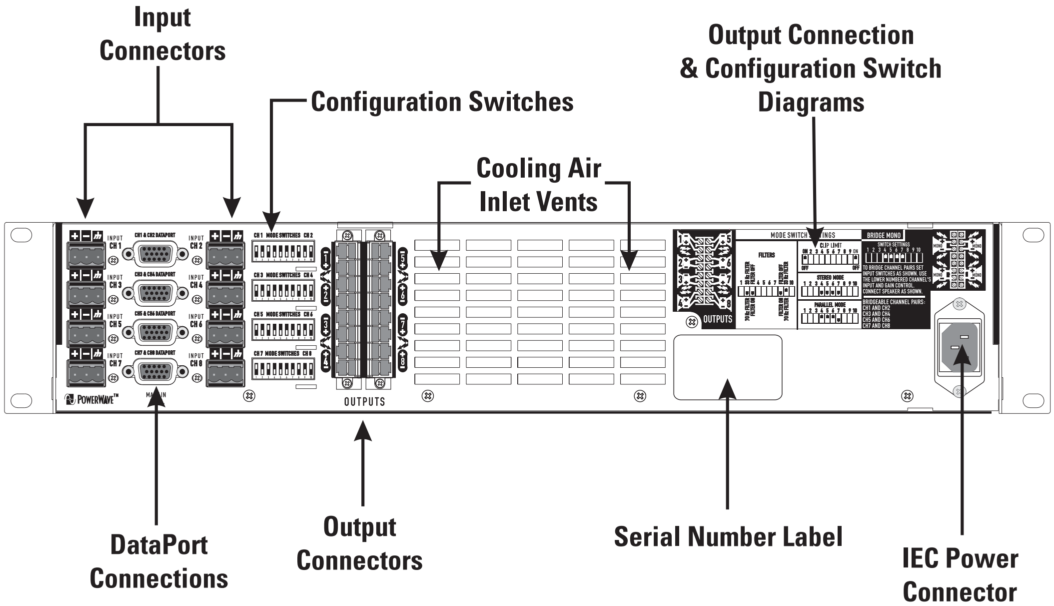


FRONT PANEL WITH SECURITY PLATE

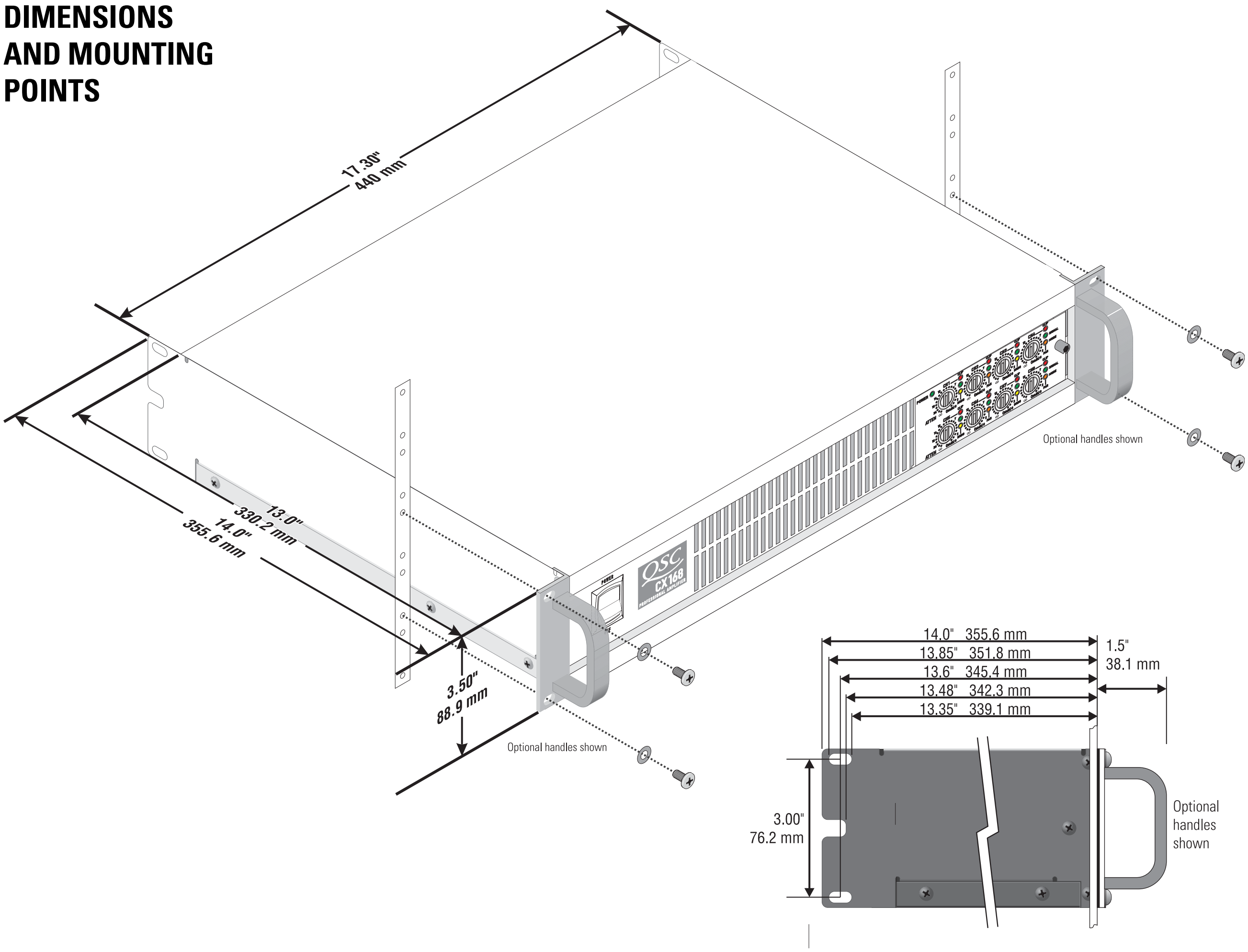


INTRODUCTION

REAR PANEL



DIMENSIONS AND MOUNTING POINTS



INSTALLATION: UNPACKING & RACK MOUNTING

What is Included

Your CX168 shipping container, as shipped from the factory, includes:

- CX168 eight channel audio power amplifier
- this user's manual
- security cover (plate) for gain controls
- self-adhesive rubber feet (use for non-rack mount applications)
- eight 3-pin terminal block input connectors
- two 8-pin terminal block output connectors
- #14AWG IEC-type detachable power cord

Save the container and packing material so the amplifier may be shipped without damage if service is ever required. If the original container is not available, be sure to use a strong shipping container with enough packing material to prevent the amplifier from being damaged in transit.

Rack Mounting

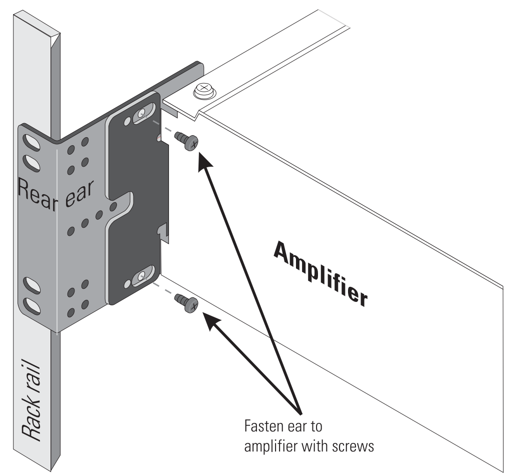
Use four screws and washers to secure the amplifier to the equipment rack rails. Support the weight of the amplifier while securing it to the rails to avoid bending or distorting the mounting ears. The amplifier may be used in an equipment rack or as a stand-alone unit. Rack mounting is optional. Self-adhesive rubber feet are provided for non-rack mount applications.

Supporting the Rear of the Amplifier

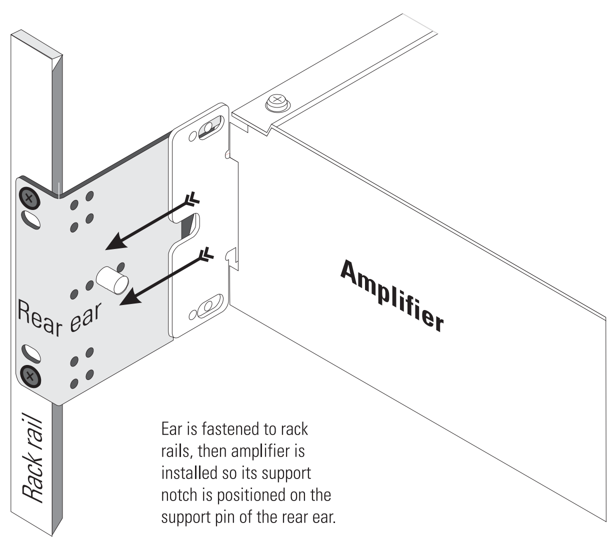
Unless the amplifier is being installed in its final, fixed location, we strongly recommend supporting the rear of the amplifier.

Supporting the rear of the amplifier is required for mobile and portable use. If the amplifier is to be transported in any way, install the optional rear rack mounting ear kit to support the rear of the amplifier. During transport, the shock loads encountered on the chassis and rack can easily damage an unsupported amplifier and the rack rails. With proper support, reliability is enhanced.

Rear rack mounting ear kits are an accessory item and are available from QSC's Technical Services Department or from your dealer or distributor. Refer to the literature included with the rack mounting ear kit for installation instructions. Two methods of rear rack mounting ear attachment are possible; use the method that best suits you application.



Optional rear rack mounting ear attachment methods- refer to the rear rack mounting ear kit's documentation for details

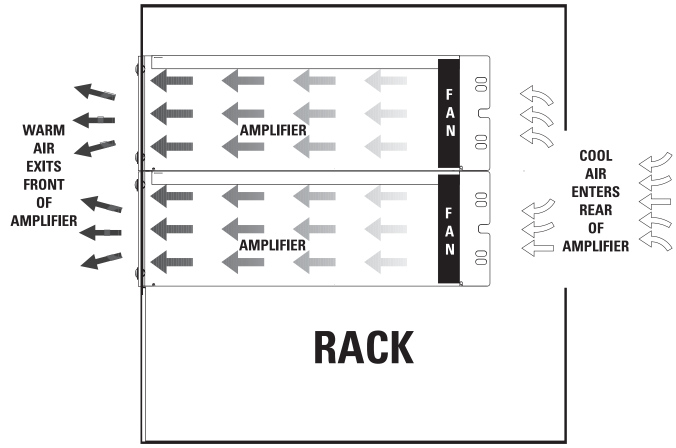


INSTALLATION: COOLING & AC MAINS REQUIREMENTS

Fan Cooling

The CX168 amplifier draws cool air into the rear of the amplifier and exhausts hot air from the front. This is done so the equipment in the rack stays as cool as possible. This method of cooling gives the operator "direct" air temperature feedback at the front of the rack, where it is the most convenient. The front panel's temperature indicates "how hard" the amplifier is working.

The fan varies speed automatically to maintain safe internal temperatures and minimize noise. Keep the front and rear vents clear to allow full air flow.

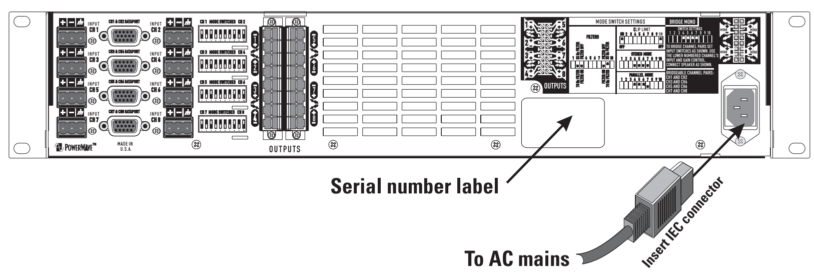


Air flow in QSC amplifiers

Do not obstruct the front or rear air vents! Make sure that plenty of cool air can enter the rack, especially if there are other units which exhaust hot air into it.

AC Mains Connection

The correct AC line voltage is shown on the serial number label. Make sure the AC mains is the correct voltage. Connecting to the wrong line voltage is dangerous and may damage the amplifier.



Connect the AC line cord by orienting the IEC plug correctly and pushing the plug firmly into the IEC receptacle. It should seat tightly. It can only be inserted when it is properly oriented.

Use the cord supplied with the amplifier, or an equivalent. Insure that the wire gauge of the cord is **#14AWG**. Use of #16 or #18 AWG can be dangerous and is not recommended.

The correct AC line voltage is shown on the serial number label.

Use the best possible connection to the AC power source. Avoid extension cords as they will cause some voltage drop between the AC source and your amplifier. If the use of an extension cord is required, ensure that it is the shortest length possible and is at least #14 AWG. Ensure that all grounding connections are maintained.

Connecting to the wrong line voltage is dangerous and may damage the amplifier or constitute the risk of electric shock. Verify the correct AC line voltage by checking the specification printed on the serial number label on the rear panel.

NOTE: Excessive length or inadequate gauge may result in short muting episodes if all channels are driven to full power.

SETUP: SETTING THE MODE CONFIGURATION DIP SWITCHES

Setup

By using the mode switches and connecting to the amplifier properly, the CX168 amplifier can be configured as a 1 to 8 channel amplifier. This flexibility enables the CX168 to be used for most any multichannel application.

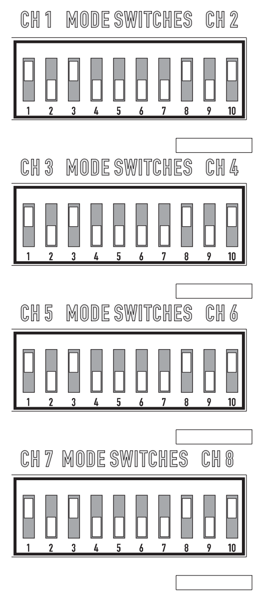
The CX168 can be thought of as four separate 2-channel amplifiers on one chassis. Each of the four amplifier sections may be operated in Stereo, Parallel, or Bridge mode independently of the others. For example, channels 1 & 2 could be set for stereo operation while channels 3 & 4 are set for parallel operation; all while channels 5 & 6 and channels 7 & 8 are set for bridge mode.

The channels are grouped as follows:

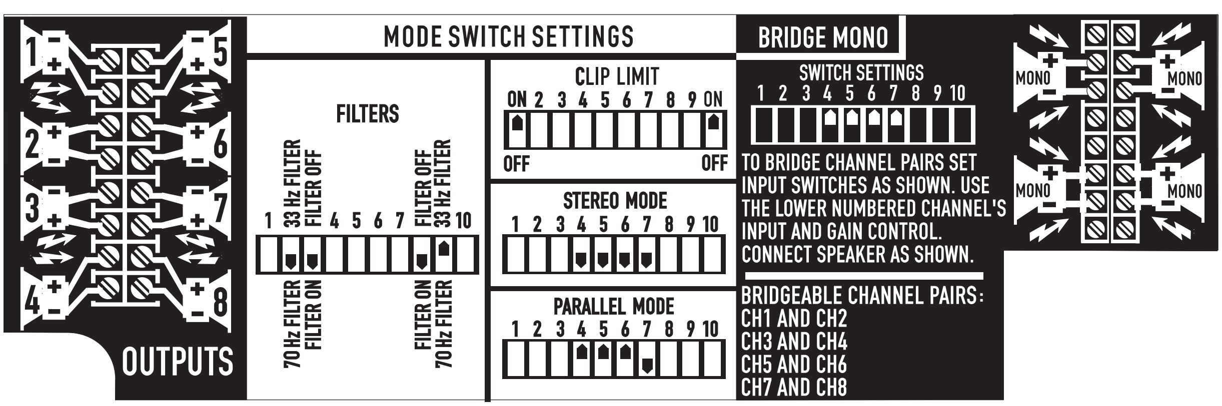
Channels 1 and 2
Channels 3 and 4
Channels 5 and 6
Channels 7 and 8

You can set the operating mode for each group by setting that group's configuration DIP switches in the desired positions. It is NOT POSSIBLE to group the channels differently. For instance, you can not bridge channels 3 & 5 because they are not in the same channel group.

Before setting the configuration DIP switches, you must first decide how the amplifier will connect to the speaker system.



"MODE" configuration DIP switches are located on the rear panel and look like the illustration above. Switch positions shown are for example only. Set switches as required for your application.



"MODE" switch settings and speaker connection diagrams are printed on a rear panel label, as shown, above.

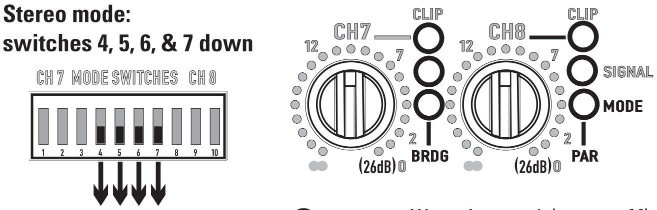
SETUP: SETTING THE MODE: STEREO, PARALLEL, OR BRIDGE

Modes

The following descriptions apply to a channel pair (such as Ch. 1 & 2 or Ch. 7 & 8). It is possible to set each channel pair's mode differently and customize the system configuration. All possible combinations will not be shown. The following describes the behavior of one channel pair in the three modes of operation and the *BRDG* and *PAR* LED's:

Stereo- Stereo mode supports two completely separate audio channels, usually referred to as 'left' and 'right'. Stereo configurations have two separate input signals and two separate output signals.

Stereo mode:
switches 4, 5, 6, & 7 down



CH 7 / CH 8 mode switch shown.
Other channel pairs similar.

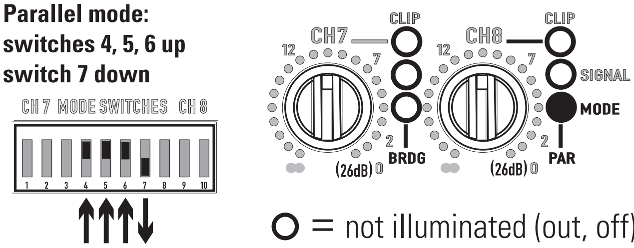
○ = not illuminated (out, off)

● = illuminated (lit, on)

Mode switch settings and LED indication for stereo mode operation. BRDG and PAR indicators should not be illuminated when in stereo mode.

Parallel- Parallel mode applies one input signal to both channels. Both inputs of a channel pair are connected in parallel when the mode switch is set for parallel, therefore, connect only one input per channel pair when in parallel mode. The outputs are connected the same as stereo mode. Each speaker will be supplied the same signal, which is still controlled by that channel's gain control.

Parallel mode:
switches 4, 5, 6 up
switch 7 down



CH 7 / CH 8 mode switch shown.
Other channel pairs similar.

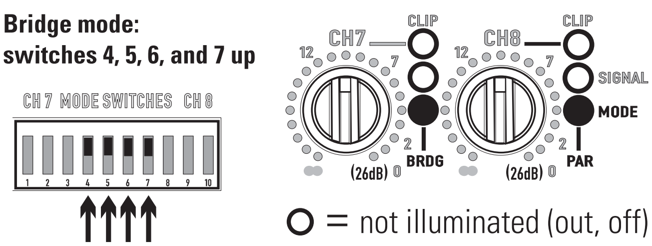
○ = not illuminated (out, off)

● = illuminated (lit, on)

Mode switch settings and LED indication for parallel mode operation. PAR indicator should be illuminated and BRDG indicator should not be illuminated when in parallel mode.

Bridge- Bridge mode combines the two channels of a pair (such as Ch. 3 & Ch. 4) into one higher-powered channel. Like parallel mode, both inputs of a channel pair are connected in parallel when the mode switch is set for parallel, therefore, connect only one input per channel pair when in bridge mode. There are 4 possible channel pairs that can be bridged on the CX168. They are Ch.1-2 Bridge, Ch. 3-4 Bridge, Ch. 5-6 Bridge, and Ch. 7-8 Bridge. Any pair or number of pairs can be independently configured in bridge mode. Use the first channel's input and gain control when in bridge mode. The second channel's gain control is non-operative and should be turned all the way down.

Bridge mode:
switches 4, 5, 6, and 7 up



CH 7 / CH 8 mode switch shown.
Other channel pairs similar.

○ = not illuminated (out, off)

● = illuminated (lit, on)

Mode switch settings and LED indication for bridge mode operation. BRDG indicator and PAR indicator should be illuminated when in bridge mode.

Note! Do not connect more than one input per channel pair when operating in parallel or bridge mode. However, in parallel or bridge mode, the unused input terminals may be used for daisy-chaining the input signal to other channels of the CX168 or to other amplifiers.

For pictorial description of operating modes, see Appendix.

SETUP: MODE SWITCHES: CLIP LIMITER AND LOW FREQUENCY FILTERS

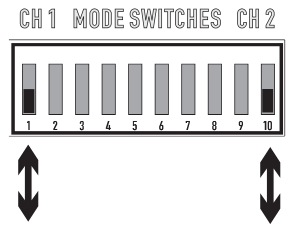
Clip Limiter

The CX168 amplifier has separate clip limiters for each of the 8 channels. These clip limiters respond only to actual amplifier clipping. Amplifier clipping generates internal error signals which cause the clip limiter to quickly reduce gain and minimize the overdrive. To preserve as much of the program dynamics as possible, limiting occurs only during actual clipping. Each channel's clip limiter can be switched on or off individually.

The clip limiter is internally set to respond as fast as possible after clipping is detected. For program material that is primarily "full-range", the effect on the overall audio quality should be imperceptible. We recommend using the clip limiters for almost all applications, especially full-range audio applications. Clipping can cause high-frequency artifacts to be output to the speakers, potentially damaging fragile high-frequency drivers.

For program material that is primarily low-frequency in nature (low-frequency or subwoofer drive) this may be perceived as a "rubbery" effect on the audio. If this is the case, it may be preferable to turn the clip limiters off and let the amplifier clip occasionally. With robust, low-frequency drivers, the occasional clipping should cause no problems.

Clip Limiter mode:
switches 1 & 10 up = ON
switches 1 & 10 down = OFF



CH 1 / CH 2 mode switch shown. Other channel pairs similar. Switch 1 operates the first channel's limiter, switch 10 operates the second channel's limiter.

CAUTION! Clip limiting reduces extreme overdrive peaks, allowing a higher average signal level without distortion. Increasing the gain with the clip limiter engaged until clipping is again audible, can double the average output power. Be careful not to exceed the power rating of the speakers!

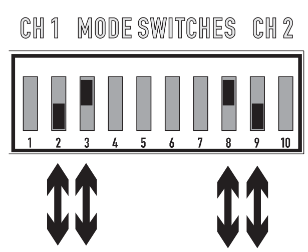
Low Frequency Filter

When driving speakers with limited low frequency response, it is important to limit the low frequency response of the amplifier. Doing so can result in more usable bass response since the speaker is not being overloaded by very low frequencies it can't handle.

Explanation- Low frequency sound waves require much more speaker cone motion to produce the same apparent loudness as higher frequencies. Properly designed speaker enclosures help the speaker to move more air with less motion using techniques like porting. Such enclosures only benefit from porting down to a certain frequency. Below this frequency, the speaker is "unloaded" and is basically free to move around uncontrollably without producing much bass. Limiting the frequency range of the low frequency content enables the speaker to behave the way it was designed to. If frequencies lower than designed are supplied to the speaker, performance will degrade.

All low frequency filters will change the character of low frequency transients. For best results, the cabinet design, speaker capabilities, and program material must be taken into account when configuring low frequency filtering.

Low Frequency Filter mode:
switch 3: up = first channel's filter OFF
switch 3: down = first channel's filter ON
switch 8: up = second channel's filter OFF
switch 8: down = second channel's filter ON
switch 2: up = first channel's filter frequency is 33 Hz.
switch 2: down = first channel's filter frequency is 70 Hz.
switch 9: up = second channel's filter frequency is 33 Hz.
switch 9: down = second channel's filter frequency is 70 Hz.

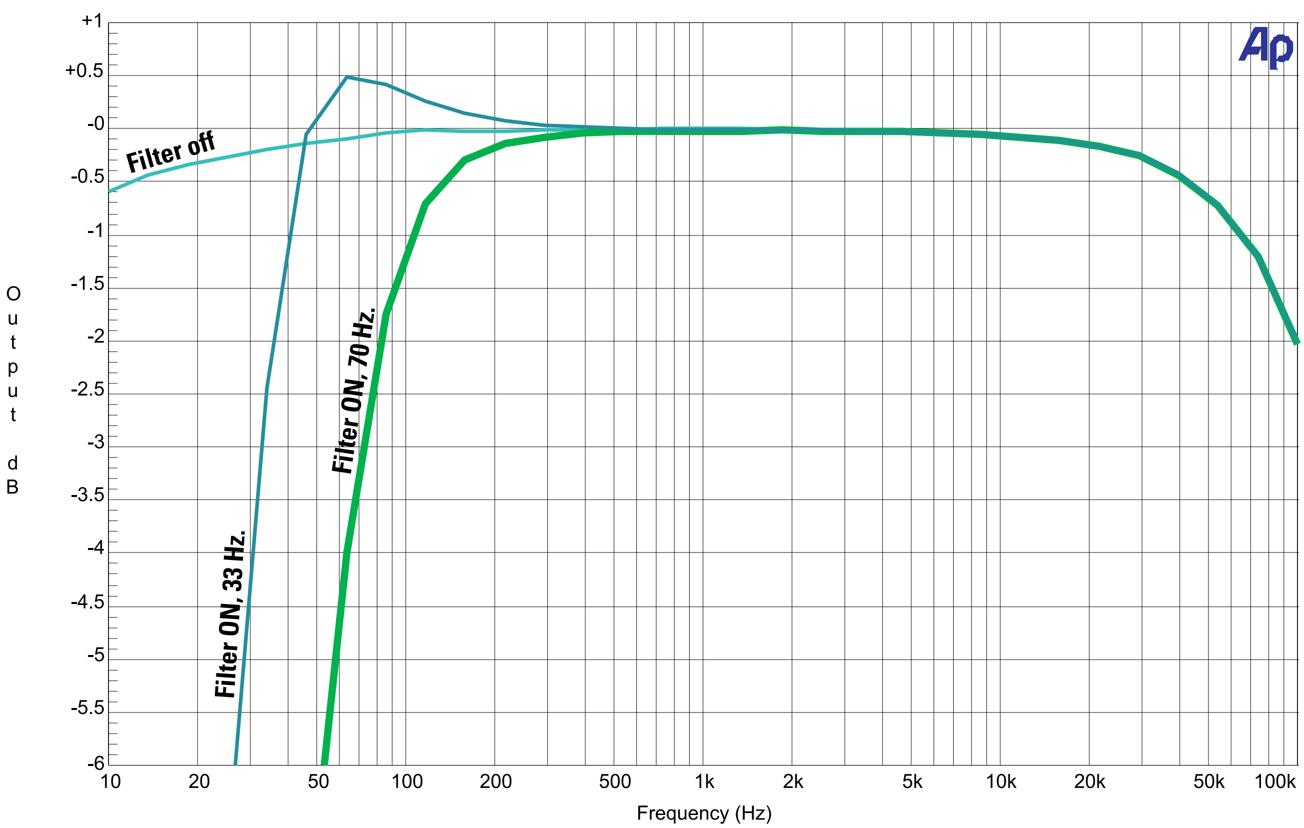


CH 1 / CH 2 mode switch shown. Other channel pairs similar. Switches 2 and 3 operate the first channel's filter, switches 8 and 9 operate the second channel's filter.

Low Frequency Filtering Tips:

- The OFF position should be used only for subwoofer systems with rated frequency response below 33 Hz. or if low frequency filtering is provided by other devices.
- Know the specification of the speaker cabinet you are driving. Match the low frequency roll off setting to the specified low frequency capability of the speaker cabinet. Do not drive the speaker with frequencies below its rating.
- Unless you have low frequency filtering before the amplifier, use the low frequency filter to protect your loudspeakers from cone over-excursion caused by frequencies below the speaker's limits.
- The 33 Hz. rolloff is a good "all purpose" setting. Turn the filter on by setting the appropriate DIP switch to ON position and select 33 Hz. using its 33/70 Hz. DIP switch. This setting is a good starting point for most large, full-range cabinets.
- The 70 Hz. rolloff is a good setting to use with smaller, compact speaker cabinets having limited bass capability. Turn the filter ON and select the 70 Hz. setting when using smaller cabinets.
- The frequency selection (33 or 70 Hertz) has no effect unless the filter is set to the ON position.

Frequency Response Curves

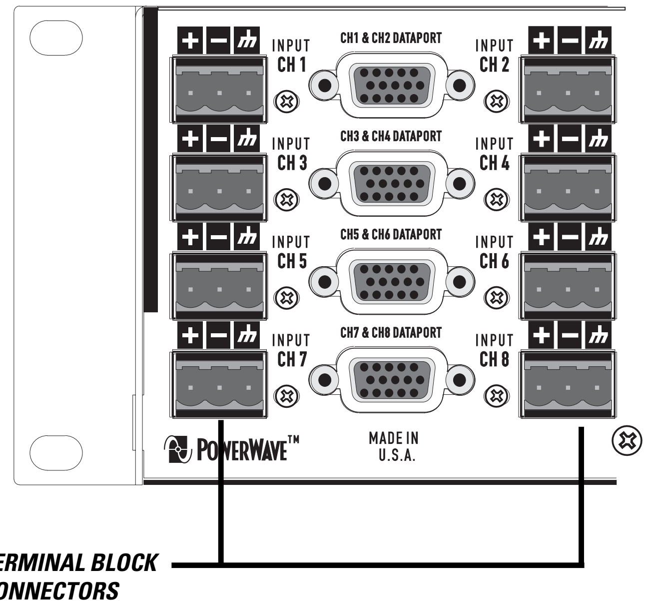


CONNECTIONS: INPUTS- Using the Terminal Block Inputs

Each channel has an active balanced "Euro-style" terminal block input jack. These terminal blocks allow the input wiring to be terminated using simple hand tools and allows for quick reconfiguration when needed. The input impedance is 20k ohm balanced or 10k ohm unbalanced.

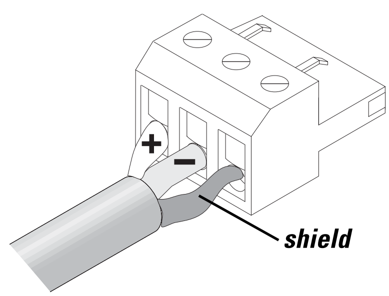
Balanced connection is recommended for all audio inputs. Balanced signals are less prone to AC hum and other electrical noise. Unbalanced signals can be suitable for short cable runs. The signal source's output impedance should be less than 600 ohms to avoid high frequency loss in long cables.

If the DataPort is being used for the input signal source, the terminal block connections should not be used for inputs. However, they may be used for daisy chaining the DataPort input signal to other channels or amplifiers. The signal available from the terminal block input connections will be about 10 dB lower than the signal presented to the DataPort.



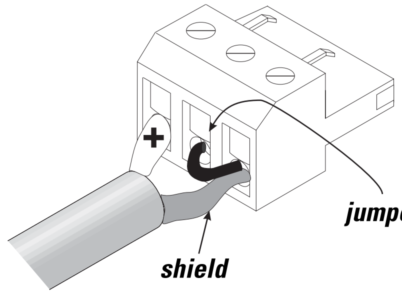
Terminal Block Connectors:

Balanced inputs: Connect the conductors to the connector as shown.



Terminal block: balanced connections

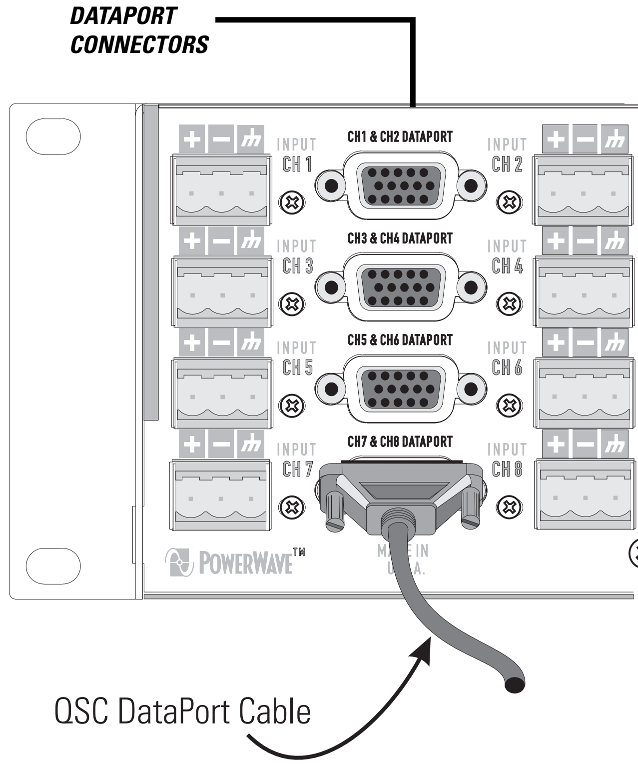
Unbalanced inputs: Connect the conductors to the connector as shown. Make sure that the unused side of the balanced input is connected to ground, as shown.



Terminal block: unbalanced connections

CONNECTIONS: INPUTS- Using the DataPorts

If used, the DataPort must be connected to a QSC DataPort product using a QSC DataPort cable. Do not use computer cables; they look similar but can cause operational problems with your amplifier and/or DataPort accessories.



The CX168 amplifier is equipped with four QSC DataPort connections that may be used for connecting to accessory QSC DataPort devices. Each DataPort services a specific channel pair; the channels are paired as:

Ch. 1 and Ch. 2
Ch. 3 and Ch. 4
Ch. 5 and Ch. 6
Ch. 7 and Ch. 8

Each DataPort and its associated terminal block input pair can be configured independently (there is no requirement to use a DataPort connection for all channels). Only the Ch.1-Ch.2 DataPort can control the power supply standby mode.

The DataPort connection will supply the input signals to the amplifier. If using the DataPort connection for audio input signals, do not apply inputs to the terminal block input connectors. If the DataPort connection is used only for monitoring amplifier operating conditions and NOT for providing audio inputs to the amplifier, then the terminal block inputs may be used to supply the audio inputs to the amplifier.

How to Connect to the DataPort:

Direct mounting of DataPort accessories is NOT supported by the CX168 amplifier. Devices that are normally mounted directly to the amplifier must be mounted remotely and connected to the DataPort using a DataPort cable. This is due to the high density of connectors on an eight-channel amplifier; there is not enough space to allow for accessories to be mounted directly to the back of the amplifier.

If the accessory attaches with a QSC DataPort cable, orient the HD-15 male plug correctly with the DataPort socket on the amplifier (it is "D" shaped and will fit only one way). Push the plug onto the socket firmly and ensure it is seated properly. Finger-tighten the 2 retaining screws. Do not over tighten.

Basic DataPort Notes:

1- The amplifier Gain controls will need to be set at their anticipated high-level setting. Use reduced level setting during setup & test.

2- If using the DataPort for audio input signals, control of the audio level will be accomplished with the DataPort accessory device to which the amp is connected.

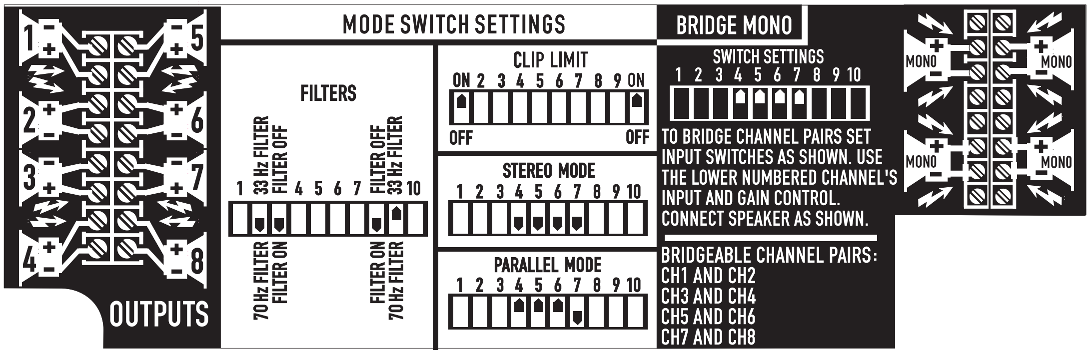
3- Control of the amplifier's AC power standby mode will only be possible via the Ch.1-2 DataPort. The amplifier's POWER switch must be physically set to the "ON" position to use the standby control feature.

4- The order in which the four DataPorts are connected to external devices makes no difference to the amplifier. However, the host controller to which the amplifier is connected will show the channels in the order that cables are connected.

CONNECTIONS: OUTPUTS

Outputs

Refer to the label on the rear panel of the amplifier for proper wiring connections. The output connection for STEREO and PARALLEL modes is on the left side of the label , while the output connection for BRIDGE mode is shown to the right side of the label. Note: Polarity changes from channel to channel and is different for bridge mode. Be certain of polarity of connections before applying power. Reversed polarity may degrade audio frequency response.

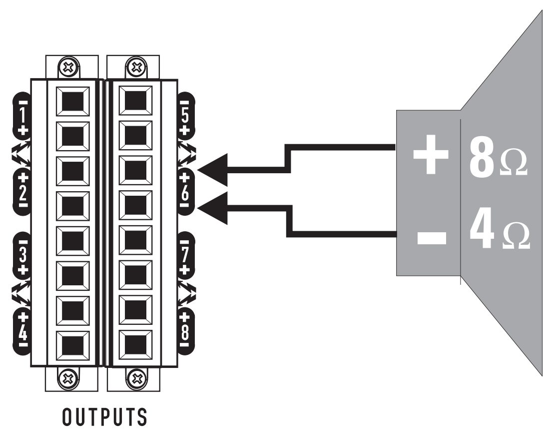


Speaker connection diagrams and "MODE" switch settings are printed on a rear panel label, as shown, above.

Stereo and Parallel Mode

In stereo or parallel mode, each speaker is connected to its own individual channel of the amplifier. This connection method is shown on the left side of the connection diagram (rear panel) or in the diagram, right.

Use 4 ohm minimum impedance in stereo or parallel mode. Ensure the mode configuration switches are set for stereo or parallel mode when connecting speakers to each channel's output.

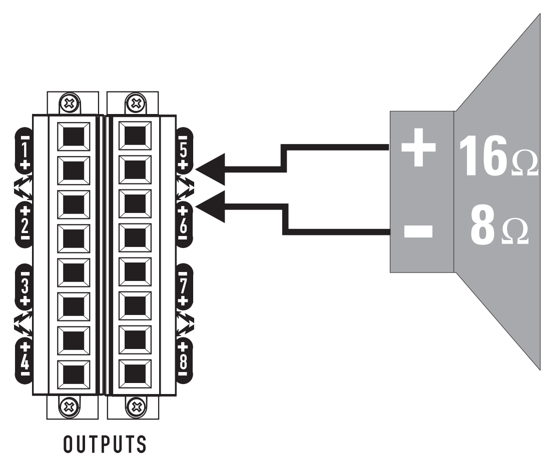


Stereo or parallel connection- single speaker shown connected to amplifier channel 6 output. Ensure that all speaker connections maintain proper polarity (+ to +, - to -).

Bridge Mode

In bridge mode, each speaker is connected to a bridged-pair of outputs. The channels must first be "bridged" by setting the mode configuration switches to the bridge settings. Then connect the speakers as shown on the right side of the connection diagram (rear panel) or in the diagram, right.

Use 8 ohm minimum impedance in bridged mode. Ensure the mode configuration switches are set for bridge mode when connecting speakers to bridged output pairs .



Bridge connection- single speaker shown connected to amplifier channels 5 and 6 (bridged with mode switch settings). Ensure that all speaker connections maintain proper polarity (+ to +, - to -).

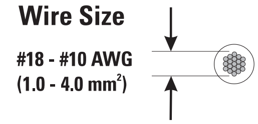
BRIDGE MODE PRECAUTIONS:

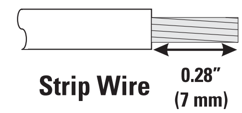
Do not use 2 ohm or 4 ohm loads in bridge mode! 8 ohms is the minimum impedance for bridge mode operation. This mode puts a high demand on the amplifier and speaker. Excessive clipping may cause protective muting or speaker damage. Ensure the speaker has a sufficient power rating.

CONNECTIONS: OUTPUTS- USING THE TERMINAL BLOCK CONNECTORS

OUTPUT TERMINAL SAFETY WARNING!
Do not touch output terminals while amplifier power is on. Make all connections with amplifier turned off. Risk of hazardous energy!

When selecting speaker cable (wire), always use the largest wire size and shortest length of wire practical for an installation. Larger wire sizes and shorter lengths minimize power loss and degradation of damping factor. Do not place speaker cables next to input wiring.





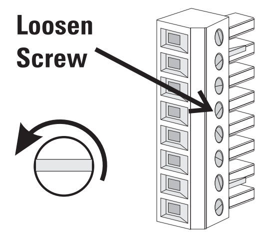
Terminal Block Connections

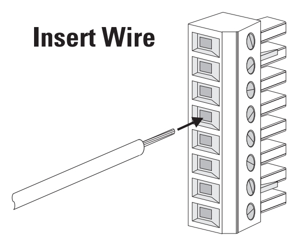
The output connections are made by attaching speaker wires to the 8-pin terminal block connectors. Once these connectors have been “populated” with the necessary speaker wire connections, plug the terminal blocks into their respective output jack.

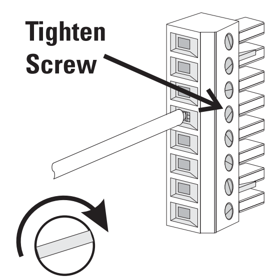
The output jacks are configured in two discrete connectors. One connector carries the connections for channels 1 through 4. The other, channels 5 through 8. Each connector has 8 pins, each of which connects to the speaker wire using a screw-clamp. Refer to the diagram, right, for the basic connection procedure. Be sure that speaker polarity is maintained. The locations of the “+” and “-” terminals on the amplifier is not the same for each channel.

Ensure that all wires are neatly terminated with no loose strands. Do not strip the insulation back excessively. Loose strands and exposed wire beyond the terminals may cause a short circuit and cause protective muting of the amplifier.

Once all the required connections have been made to the terminal block connectors, they may be inserted into the OUTPUTS jack on the rear panel. Orient the connector properly (it only fits one way) and push it in until it is fully seated in the receptacle. Note each connector is oriented so its wires face outwards.







How to make connections to the terminal block:

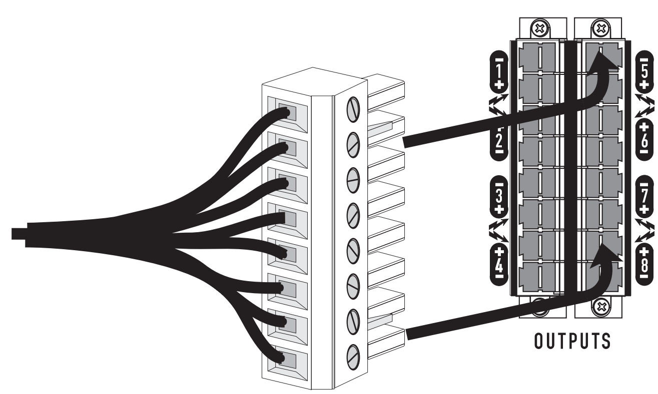
1- Strip the wires approximately 0.28” (7mm).

2- Twist any loose strands of wire together.

3- Loosen the appropriate retaining screw fully.

4- Insert the stripped wire into the receptacle.

5- Tighten the appropriate retaining screw to secure.



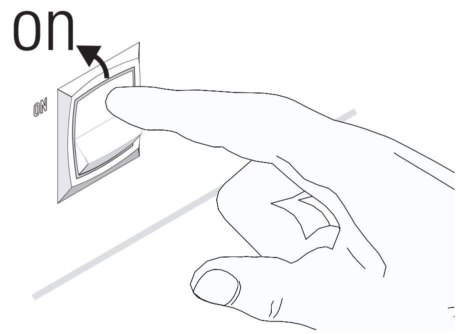
OPERATION: POWER SWITCH and GAIN CONTROLS

Power Switch

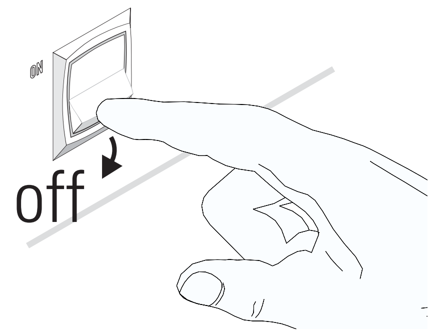
The power switch is a rocker-type switch. It is located on the left side of the front panel. To turn the amplifier on, push in on the top of the switch. To turn the amplifier off, push in on the bottom of the switch.

The green power LED should light up when the switch is in the on position.

When the power is switched off, the LED may takes several seconds to go out; this is normal.



To turn the amplifier 'on', push in on the top portion of the rocker switch firmly. It should move inward and snap into the 'on' position. After switching on, the power LED should illuminate.



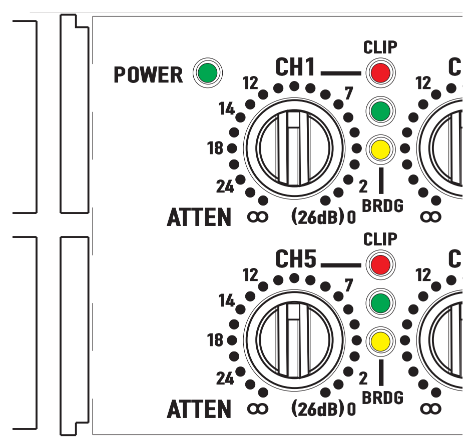
To turn the amplifier 'off', push in on the bottom portion of the rocker switch firmly. It should move inward and snap into the 'off' position. After switching off, the power LED should extinguish.

If the POWER LED fails to illuminate when the switch is in the ON position:

1- Check the AC cord and insure that both ends are fully inserted into their receptacles. If the power LED still fails to illuminate,

2- Check the AC outlet for voltage with a circuit tester or known good device (lamp, etc.).

3- Make sure that the amplifier is not in STANDBY mode. If the amplifier is being controlled by one of its DataPort connectors, the amplifier may be in standby mode. Verify the AC power status from the DataPort device controlling the amplifier.

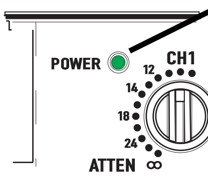


The 'POWER' LED is located at the upper-left of the gain controls section of the front panel. It is a green LED that illuminates when the power is turned on (and the amplifier is properly connected to the correct AC source). The power LED extinguishes when the power is turned off.

OPERATION: LED INDICATORS

LED Indicators

The LED indicators provide basic operation information to the operator. The 'normal' indications of the LED's are shown, below. NOTE! The BRDG and PAR mode indicator LED's are discussed on page 9.

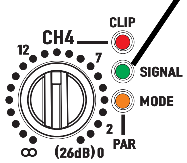


POWER: The power 'on' indicator LED is located at the upper-left of the gain control cluster. It is green and illuminates when the power switch is set to the on position and AC power is present at the IEC cord receptacle.

Normal indication: at power-on the LED will illuminate.

If no indication: Check AC power cord, AC mains, and that the DataPort is not being used to force the amplifier into 'Standby' mode.

When power is switched off: the LED may take several seconds to extinguish (go out); this is normal.



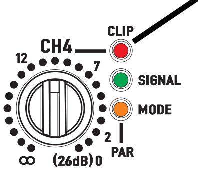
SIGNAL: Each channel has a green 'signal' LED located to the right of its gain control. The 'signal' LED lights up when the input signal is sufficiently strong.

Normal indication: illuminates when the input signal is present and sufficiently strong.

If no indication: check gain settings, input cables, connections and audio source. If audio source is extremely low signal strength, signal LED may not illuminate; this is normal but indicates that the input signal strength should be increased.

Occasional illumination: normal for weak input signal strength.

Fully illuminated (on): normal.



CLIP : Each channel has a red 'clip' LED that illuminates when the channel clips. It is located to the upper-right of each channel's gain control.

Normal indication: illuminates briefly at extreme output power peaks. Occasional clipping at high power levels may be normal. At power-on, the clip LEDs may briefly flash during turn-on muting.

Continuous operation at high power may trigger thermal protection circuitry, shutting down the channel pair concerned. The CLIP LEDs for the channel pair will fully illuminate to indicate thermal protection activation. The channels are paired as:

Ch. 1 and Ch. 2
Ch. 3 and Ch. 4
Ch. 5 and Ch. 6
Ch. 7 and Ch. 8

If no indication: normal if the amplifier is operated at nominal output levels. If the input signal is weak, it may not be capable of driving the amplifier into clipping, even at full gain.

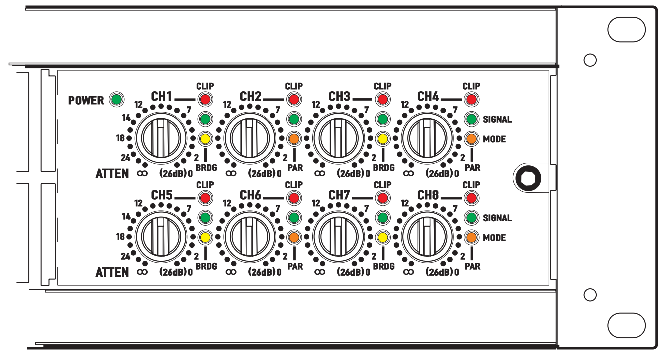
Occasional illumination: Occasional clipping (once briefly every few seconds) when operation at high power levels does not indicate trouble. Continuous operation at high power may trigger the thermal protection circuitry, shutting down the amplifier and fully illuminating the clip LED.

Illuminated most of the time: Not normal; for cleaner sound, reduce the output of the amplifier and/or input signal to avoid thermal shutdown of the amplifier and possible speaker damage. If clipping persists at normal levels, check for shorted or abnormal load impedances.

OPERATION: GAIN CONTROLS

Gain Controls

The gain controls are located on the right side of the front panel. They are operated by rotating the control clockwise to increase gain or counterclockwise to decrease gain. When turned fully counterclockwise, gain is fully reduced and no output signal will be present. When turned fully clockwise, the voltage gain of the amplifier is +26dB and the output signal will potentially be full strength (provided the input is fully driven).



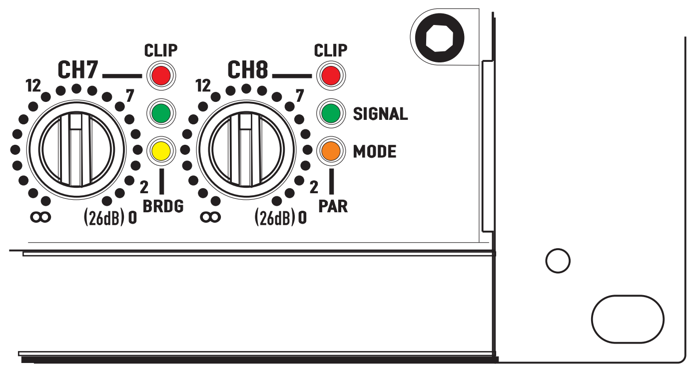
Location of gain controls on the front panel of the CX168

The gain controls are detented in 21 steps for repeatable adjustment. Surrounding the Gain control, the power attenuation level is shown in dB. Each detent is approximately 1dB change in amplifier gain.

To operate the gain control, rotate the control's knob so that the desired level is achieved. A small, flat-tipped screwdriver (#1 blade size) is typically used. Be sure not to use large screwdrivers as it is possible to damage the gain control by applying excessive torque. Gain controls are sensitive, electronic components and should be treated with care.

Each channel pair's gain controls and indicator LED's are grouped together as shown, right. The channels are paired in the following manner:

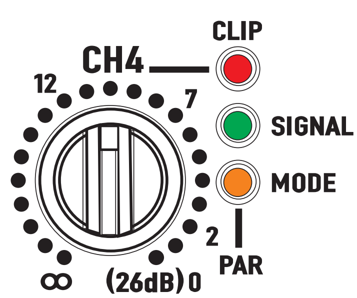
Channel Pair #1-	Ch. 1 and Ch. 2
Channel Pair #2-	Ch. 3 and Ch. 4
Channel Pair #3-	Ch. 5 and Ch. 6
Channel Pair #4-	Ch. 7 and Ch. 8



Typical channel-pair control and indicator group on the CX168

As detailed in the 'Setup' section of this manual, channels can only be paralleled and bridged within their channel pair. Example- Channel 3 can be bridged with channel 4 only; it can not be bridged with any other channel.

The dB markings around the gain controls are attenuation. They show attenuation from full gain. At 0db attenuation, the voltage gain of the amplifier is 26dB. If the gain control is set at the 7dB attenuation position, then the voltage gain of the amplifier is 19dB (26dB - 7 dB).



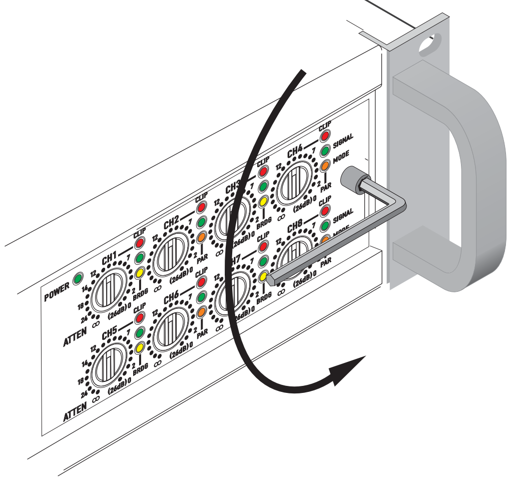
Typical gain control showing attenuation level markings and LED's for the channel. Note that there is only one BRDG (bridge) and one PAR (parallel) indicator per channel pair.

OPERATION: GAIN CONTROL SECURITY PLATE

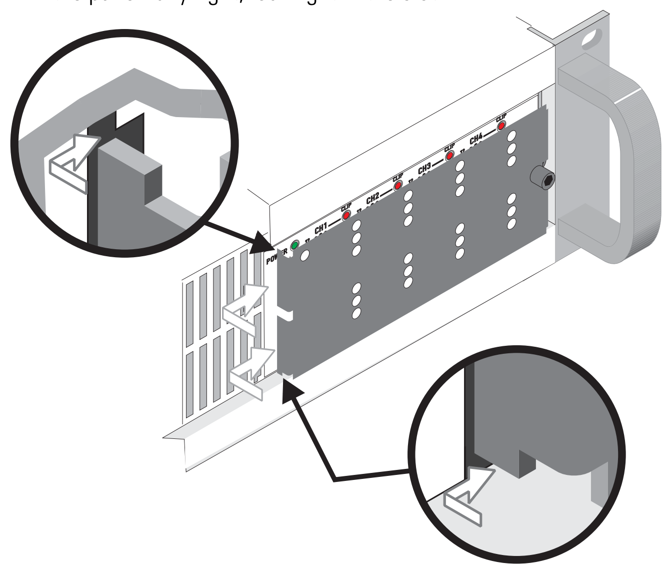
Gain Control Security Plate

After final gain adjustments have been made, the gain controls can be covered with a security plate. This easy-to-use security plate makes it difficult for others to adjust the gain. This may be desirable in instances where the gain controls should not be tampered with. To install the security plate:

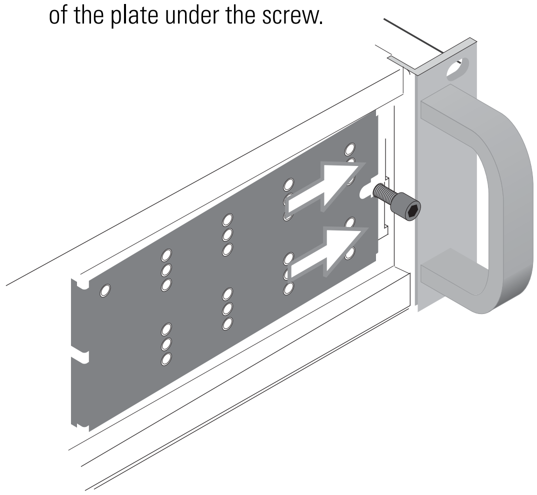
1. Use a 9/64" or 3.5mm hex key to loosen the screw. Do not remove; just loosen several turns.



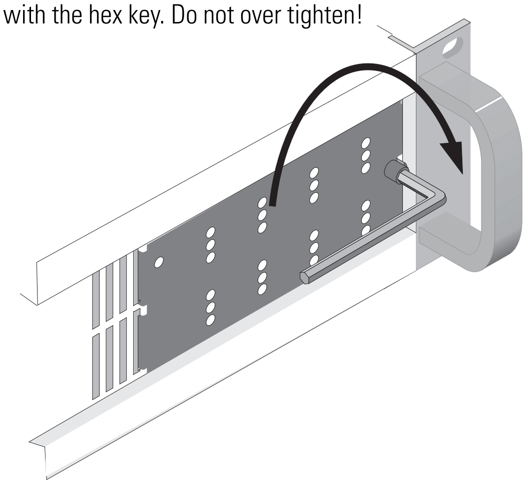
3. Line up the tabs with the keyed portion of the ventilation slots, insert into the slot and then slide the panel fully right, locking it in the slot.



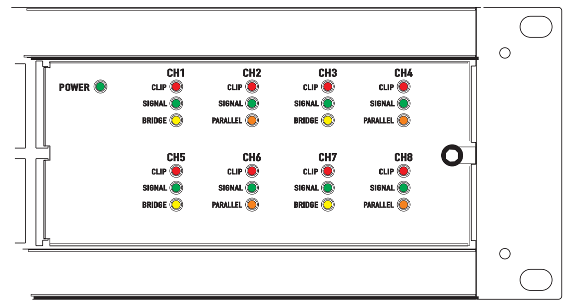
2. Orient the panel as shown and slide the right end of the plate under the screw.



4. Make sure the LED's are visible, then tighten the screw with the hex key. Do not over tighten!



When properly installed, the security panel will still allow you to monitor all the channel status LED's and power LED. The amplifier front panel will look like the illustration, right, after installing the security plate.

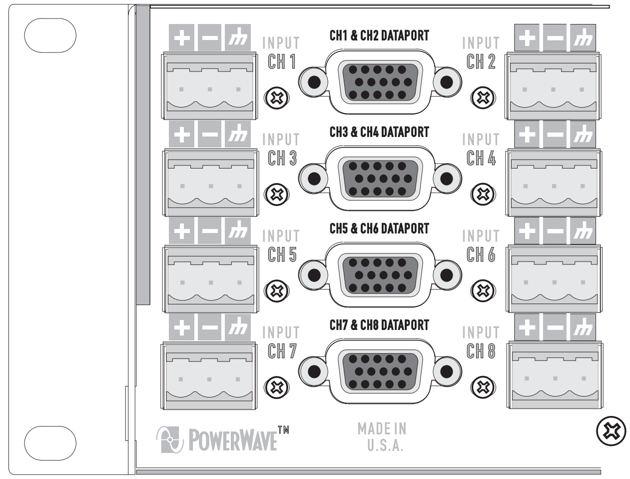


DataPort Connectors

QSC's DataPort equipped amplifiers offer operators of larger audio systems a high degree of system monitoring and operation from remote locations. A system operator in a sports complex might be located hundreds of yards (meters) from the actual amplifiers, making 'normal' control and monitoring of the system difficult, at best. By integrating QSC's DataPort-equipped amplifiers with QSC's DataPort accessories, a dependable and robust control and monitoring solution can easily be implemented.

The DataPort is used for connection to optional QSC DataPort accessories. DataPort accessory devices can provide remote cinema monitoring, DSP, filter and crossover functions.

The DataPort connections on the CX168 offer full-featured implementation of DataPort accessories and supply the required accessory voltage to DataPort devices that require it. CX168 amplifiers DO NOT support direct mounting of smaller accessories, like the DSP-3, due to the high density of connectors on the rear panel but they may be connected via cables.



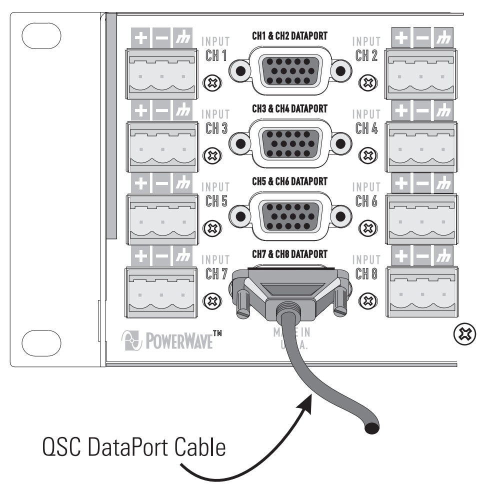
A single DataPort is assigned to each channel pair. Each DataPort is marked with the channel numbers served by that DataPort. Note, above, that each DataPort is clearly labeled and is located between its respective channel's terminal block connectors.

Using the DataPort

Use a QSC DataPort cable to connect the CX168 DataPorts to DataPort accessories. Consult the accessory's documentation for recommended mounting and interconnect information.

Do not use terminal block connections for inputs when the DataPort is used as the input signal source. If you do, the signals from the DataPort and terminal block inputs will be mixed, possibly with unexpected results.

See the following section for general operating notes regarding the DataPorts on the CX168.



Connect to the DataPort using a QSC DataPort cable. Align the connector with the socket housing and insert fully. Finger-tighten the retaining screws.

OPERATION: USING THE QSC DATAPORT CONNECTIONS

What Can I Connect to the DataPort?

Consult your QSC representative for up-to-date accessory compatibility. QSC has several DataPort-based accessories available like the:

CM16a Amplifier Controller & Monitor
DSP-3 Digital Signal Processing Module
DSP-4 Digital Signal Processor Module
XC-3 Crossover
SF-3 Subwoofer Filter
LF-3 Low Frequency Filter
DCM Crossovers & Monitors

General Guidelines to using the DataPorts on the CX168

1. Each channel pair (Ch.1-2, Ch.3-4, Ch.5-6, and Ch.7-8) has its own dedicated DataPort connector that applies control and monitoring functions to its specific channel pair.

2. The DataPort for channels 1 and 2 has a special function on the CX168. It is the 'master' DataPort for controlling AND reporting power supply status of the amplifier. It is only possible to control and report the status of the power supply using the Ch.1 & Ch.2 DataPort.

3. In order for the Ch.1 & Ch.2 DataPort to control the power status of the amplifier, the front-panel power switch MUST be in the ON position. If the power switch is in the 'off' position, the amplifier will not respond to any commands sent to its Ch.1 & Ch.2 DataPort.

4. The DataPort connection can be used for amplifier status monitoring only, using the terminal block inputs for supplying input signals to the amplifier. Ensure that the DataPort connection supplies NO INPUT SIGNAL when operating in this fashion. If signal is applied to both the DataPort and the terminal block input connector at the same time, the signals will be summed and amplified. The result may be undesirable.

5. If using the DataPort for supplying the audio input signals to the amplifier, we don't recommend using the terminal block input connectors, even if they are connected to non-operating audio sources. The DataPort and terminal block inputs are connected in parallel and isolated by about 20k ohms of resistance. Even with this isolation, it is possible that connection to other signal sources may significantly alter the input impedance at the DataPort. This could affect output levels as connections are altered.

6. If using the DataPort for supplying the audio input signals to the amplifier, do not use the unused terminal block inputs for daisy-chaining the DataPort signals to other channel pairs or other amplifiers. The DataPort device supplying the audio signal will not accurately report gain and levels will be effected.

7. The heat sink temperatures of the CX168 are reported on the first channel of each channel pair's DataPort. In other words, to monitor all four heat sink temperatures, all four DataPorts must be connected to the monitoring device (such as QSC's CM16a). Temperatures will be reported on channels 1, 3, 5 and 7.

APPLICATIONS: TYPICAL APPLICATION EXAMPLES

Four Room Stereo Feed

A simple application of the CX168 would be an installation with 4 rooms of coverage; each room to be provided with a stereo audio feed. All speakers are 4 ohm or greater impedance and have a frequency response rated down to 45 Hz., making them suitable for the 70 Hz. low frequency setting. Clip limiting will be used with these full-range speakers. Operating mode of all four channel pairs will be set to 'stereo' because the audio inputs are stereo and a stereo result is desired in each room.

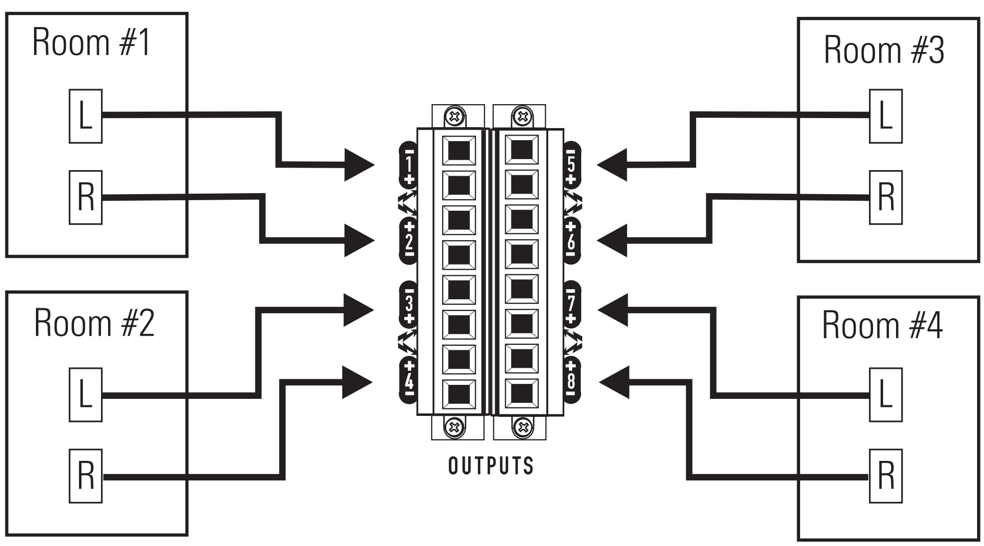
Input connections can vary too much to provide meaningful examples. For this example, assume that each input has its own discrete signal provided at the terminal block input connectors.

Routing would make sense if kept in room number order:

Room #1:	Fed by channel pair #1 (Ch.1&2)
Room #2:	Fed by channel pair #2 (Ch.3&4)
Room #3:	Fed by channel pair #3 (Ch.5&6)
Room #4:	Fed by channel pair #4 (Ch.7&8)

MODE switch configuration:

Ch.1/ Ch.2 mode switch:	Clip Limiters ON Low Frequency Filter ON Low Frequency Setting 70 Hz. Mode: stereo
Ch.3/Ch.4 mode switch:	same as first mode switch
Ch.5/Ch.6 mode switch:	same as first mode switch
Ch.7/Ch.8 mode switch:	same as first mode switch



Tri-Amp'd Stereo Cabinets with Subwoofer Driven by Bridged Channel Pair (next page)

Two discrete 3-way (tri-amp) cabinets are to be connected to the CX168 along with an 8 ohm subwoofer cabinet.

The subwoofer will require the most amount of power and therefore will be driven by two channels bridged into one. Its impedance meets the minimum requirement for bridged output. All other drivers are 4 ohms or greater and meet the single channel minimum impedance requirement. Each of the tri-amp cabinets will be driven with three discrete channels (totaling 6 channels). The total required channel count is 8; the CX168 is the perfect tool for the job.

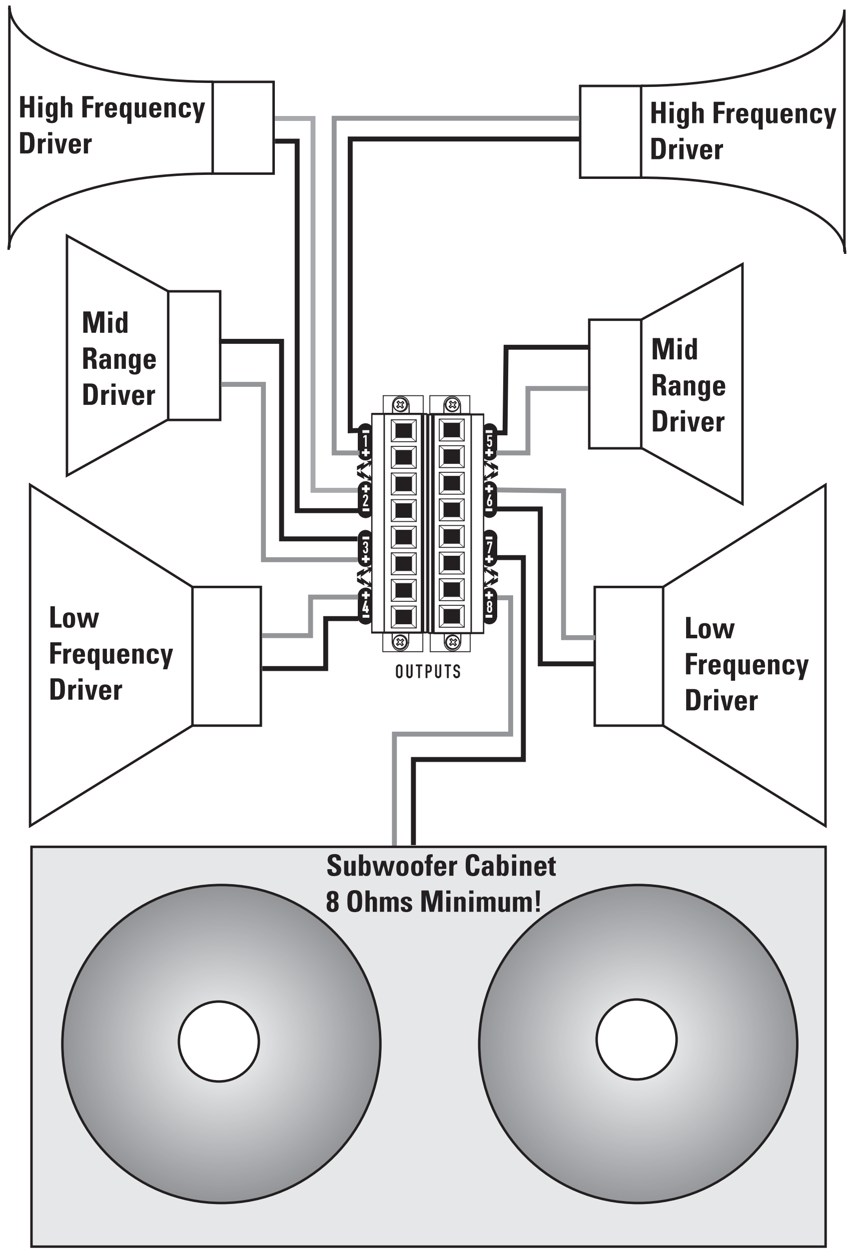
Tri-Amp'd Stereo Cabinets with Subwoofer Driven by Bridged Channel Pair (continued)

Routing is a function of 'easiest wiring' order and is:

LF Driver Left:	Ch. 4
Mid Driver Left	Ch. 3
HF Driver Left	Ch. 2
HF Driver Right	Ch. 1
Mid Driver Right	Ch. 5
LF Driver Right	Ch. 6
Subwoofer	Bridge Ch. 7 & 8

MODE switch configuration:

Ch.1/Ch.2 Mode:	Stereo
Ch.1 Clip Limiter:	ON
Ch.1 LF Filter:	OFF
Ch. 2 Clip Limiter	ON
Ch. 2 LF filter	OFF
Ch.3/Ch.4 Mode:	Stereo
Ch. 3 Clip Limiter	ON
Ch.3 LF Filter:	OFF
Ch. 4 Clip Limiter	OFF
Ch. 4 LF filter	ON
Ch. 4 LF frequency	70 Hertz
Ch. 5/Ch. 6 Mode:	Stereo
Ch. 5 Clip Limiter	ON
Ch.5 LF Filter:	OFF
Ch. 6 Clip Limiter	OFF
Ch. 6 LF filter	ON
Ch. 6 LF frequency	70 Hertz
Ch. 7/Ch. 8 Mode	Bridged
Ch. 7 Clip Limiter	OFF
Ch. 7 LF Filter	OFF
Ch. 8 Clip Limiter	no effect in bridge
Ch. 8 LF Filter	no effect in bridge



Channels 1 through 6 are configured for 'stereo' mode because all channels are completely separate from one another. Channels 7 and 8 are bridged to combine their power to drive the subwoofer (sub's are the most inefficient driver and therefore require the largest portion of the system power).

Clip limiters are used on the mid range drivers and the HF drivers because they are the most fragile and are damaged easily by clipping events. The LF drivers and the sub's are rugged enough to handle even prolonged clipping events without damage and thus no clip limiting is used.

Low frequency filtering is used only on the LF drivers. The HF and mid range drivers have filtering before the amplifier and need no filtering from the CX168. The subwoofer cabinet can handle the low frequencies without problem, so no filtering for them. The LF drivers response specification will dictate at what frequency (33 or 70 Hz.) the low frequency filter is set for.

APPENDIX: DESCRIPTION OF STEREO & PARALLEL OPERATING MODES

NOTE! In the following diagrams, channels 1 and 2 are shown as examples only. Any channel pair can be configured as shown here. The channels pairs on the CX168 are : Channels 1 and 2, Channels 3 and 4, Channels 5 and 6, and Channels 7 and 8. Therefore, in any example below, replace channel 1 and 2 with the channel pair numbers you are setting up.

Stereo Mode:

Each input signal is sent to its respective channel. Each channel has independent low-frequency (subaudio) filtering, clip limiting, gain control, and output connection.

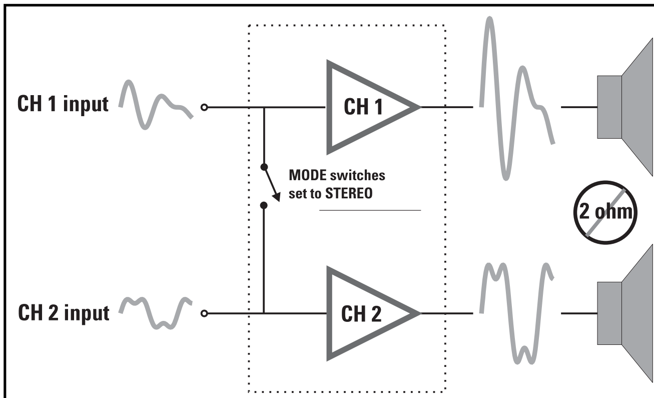
When to use STEREO input configuration: Use stereo mode for stereo sources (L-R inputs) and any other situation that requires each channel to be completely separate from the other.

How to use STEREO mode:

1- Set the MODE switches for stereo mode operation.

2- Connect the two input signals to CH1 and to CH2 (or signals may be provided from DataPort connection).

3- Connect the two speakers; one to channel 1's output terminals, one to channel 2's output terminals.



STEREO MODE OPERATION

NOTE: Ensure that the MODE SWITCHES are set to STEREO when feeding two separate signals to the two channels.

Parallel Mode:

The channel 1 and channel 2 inputs are connected together, applying a *single input signal* to both channels of the amplifier. A signal into any input jack will drive both channels. Each channel's low frequency filtering, clip limiting, and gain control still function independently. Each channel drives its own speaker load.

You can patch the input signal on to additional amplifiers (daisy chain) by using any of the remaining input jacks. This feature eliminates the need for "Y" cables.

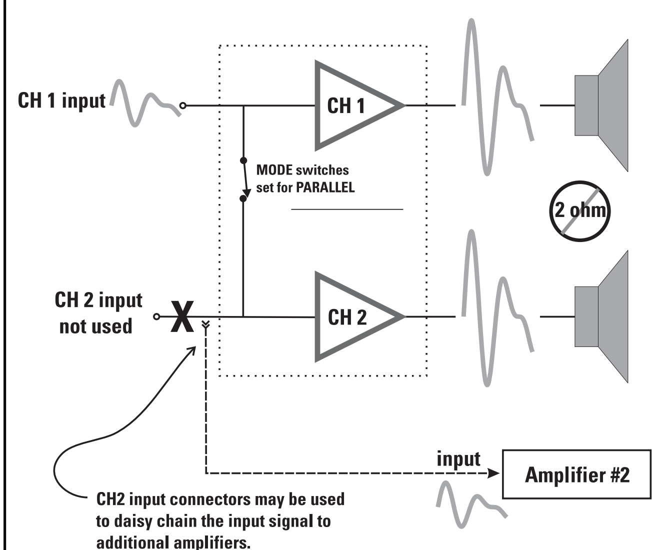
When to use PARALLEL input configuration: Use parallel mode when you need one signal to drive both channels; each channel having its own control (gain, clip limiter, low-frequency filter).

How to use PARALLEL mode:

1- Set the MODE switches for PARALLEL mode.

2- Connect the one input signal to either channel's input connector. The unused input may be used for daisy chaining the signal to another device or channel pair.

3- Connect the two speakers: one to channel 1's output terminals, one to channel 2's output terminals.



PARALLEL MODE OPERATION

APPENDIX: DESCRIPTION OF BRIDGE OPERATING MODE

Bridge Mode

Bridge mode combines both output channels into one output. This mode is for driving a single, high-power-rated load with twice the "normal" voltage swing. This results in about 4 times the peak power and about three times the sustained power of a single channel. It is also common to call this bridge *mono* mode.

When to use BRIDGE mode: Use BRIDGE mode when you need to deliver the power of two channels to a single 8 ohm load, such as a subwoofer. Do not use less than 8 ohm loads in bridge mode. Refer to the CONNECTIONS section of this manual for details.

How to use BRIDGE mode:

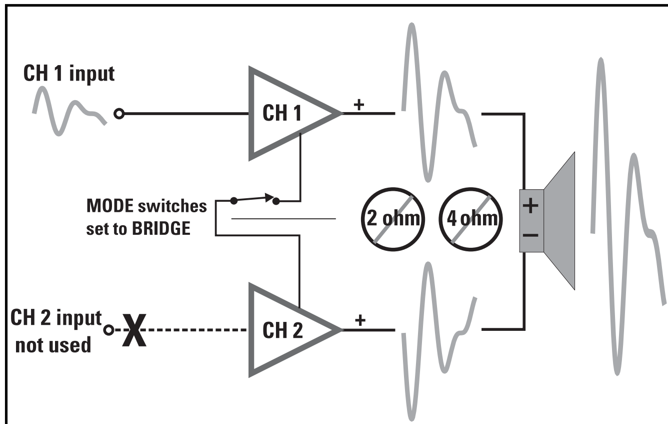
1- Set the MODE switches for bridge mode.

2- Connect the one input signal to channel 1 or channel 2's input.

3- Connect the one speaker load to the bridge output terminals

4- For a bridged pair, use the first channel's gain control, clip limiter and filter. The second channel's controls will have no effect.

5- Turn the second channel's gain to zero gain. Filters and limiter settings of second channel do not matter.



BRIDGE MODE OPERATION-

Note that speaker connection for bridge mono mode is different than other modes. See section on Connections: Outputs for proper bridge mode output connections.

Normal Operating Levels

The amplifier's protective muting system guards against excessive internal temperatures. With normal ventilation and loads, the amplifier will handle any signal level, including overdrive. Lower load impedances and higher signal levels produce more internal heating. Loads below 4 ohm loads may trigger thermal shutdown.

A Note on Sound Pressure Levels

Sound is heard as the ear converts vibration from sound waves into impulses in the nerves of the ear. Sounds above 90 decibels, particularly if the sound is prolonged, may cause such intense vibration that the inner ear is damaged. 90 decibels is about the loudness of a large truck about 5 yards away. A jackhammer emits sounds of about 120 dB from 3 feet away, and a jet engine emits sound of about 130 dB from 100 feet away. Motorcycles, snowmobiles, and similar engines range around 85 to 90 dB, and a rock concert may approach 100 dB. A general rule of thumb is that if you need to shout to be heard, the sound is in the range that can damage hearing.

Some jobs in the entertainment industry involving loud noise from music carry high risk for hearing loss. In the U.S., the maximum job noise exposure is regulated by law. Both the length of exposure and the extent (decibel level) of exposure are considered. ***If exposure is at or greater than the maximum exposure, protective measures must be taken.*** The table at right is referenced from OSHA's Permissible Noise Exposure table G16.

Duration of Exposure (per day) in Hours	Sound Level (dBA, slow response)
8	90
6	92
4	95
3	97
2	100
1.5	102
1	105
0.5	110
0.25	115

APPENDIX: MULTIPLE SPEAKER LOADS IN SERIES

Multiple Lo-Z Loads in Series:

Series connection is where the same signal current flows through each of the speaker loads. The signal goes into one terminal of the first speaker and out its other terminal; then the signal goes into one terminal of the next load, and so on. If any of the speakers in a series-connected branch of a system fails, the signal is prevented from flowing through any of the loads (no sound). Total impedance for series connections is the sum of each of the impedances in the chain (see illustration, below).

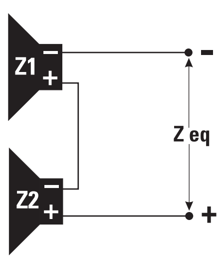
Series connection schemes are usually used with “raw” speakers (not often with speaker cabinets) for wiring convenience. The exposed speaker connections make this method easy for small “strings” of speakers in one area. Most PA-type speaker cabinets are NOT wired for series connection ability; consult your speaker’s documentation for connection information.

It is not recommended to connect speakers of different impedances in series because power will be divided unequally between them. If you are experienced with mixing and matching speaker loads of varying impedances, no damage to the amplifier will result as long as the total impedance is within 4 to 16 ohms per stereo/parallel channel or 8 to 16 ohms, bridged.

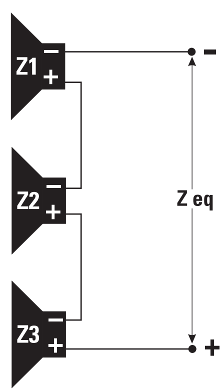
Below are examples of series connections for 2, 4, 8, and 16 ohm speaker loads. Series connections resulting in an impedance of 250 ohms or greater are not recommended.

SERIES LO-Z LOADS

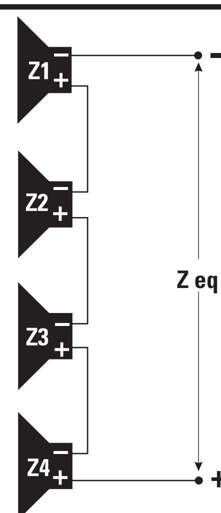
“Zeq” is the abbreviation for equivalent impedance and means “the impedance at the two connection points”



2 series loads



3 series loads



4 series loads

		2 series loads	3 series loads	4 series loads
Formula		$Z_{eq} = Z1 + Z2$	$Z_{eq} = Z1 + Z2 + Z3$	$Z_{eq} = Z1 + Z2 + Z3 + Z4$
Impedance of speakers	2 ohm	$Z_{eq} = 2 + 2 = 4$ ohms	$Z_{eq} = 2 + 2 + 2 = 6$ ohms	$Z_{eq} = 2 + 2 + 2 + 2 = 8$ ohms
	4 ohm	$Z_{eq} = 4 + 4 = 8$ ohms	$Z_{eq} = 4 + 4 + 4 = 12$ ohms	$Z_{eq} = 4 + 4 + 4 + 4 = 16$ ohms
	8 ohm	$Z_{eq} = 8 + 8 = 16$ ohms	$Z_{eq} = 8 + 8 + 8 = 24$ ohms	$Z_{eq} = 8 + 8 + 8 + 8 = 32$ ohms

Note: Impedance has a phase angle also. This is typically neglected in basic speaker calculations.

APPENDIX: MULTIPLE SPEAKER LOADS IN PARALLEL

Multiple Lo-Z Loads in Parallel:

If each speaker load is connected across the same signal source (the output of the amplifier), then the loads are in parallel with one another. If one of the speakers fails in a parallel-connected system branch, the remaining speakers will continue to operate. If all speaker loads are the same impedance, the total impedance of a parallel-connected system branch is the impedance of one speaker load divided by the number of loads in the branch (example: three 8-ohm loads in parallel: 8/3=2.667 or roughly 2.7 ohms, too low to use on the CX168).

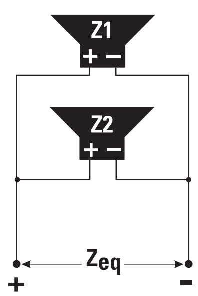
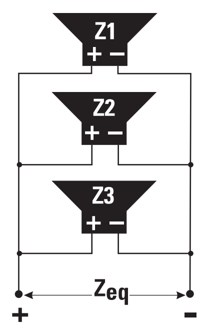
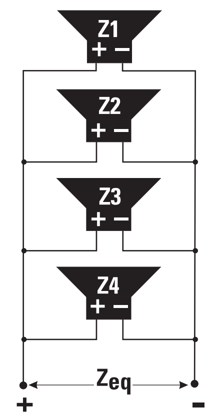
It is not recommended to connect speakers of different impedances in parallel because power will be divided unequally between them. If you are experienced with mixing and matching speaker loads of varying impedances, no damage to the amplifier will result as long as the total impedance is within 4 to 16 ohms per channel, dependant upon operating mode.

Most PA-type speaker cabinets are provided with two or more input jacks that are wired in parallel. If one cabinet is connected to the amplifier, the load impedance is equal to the one cabinet's specified impedance. If you attach another similar cabinet to the remaining jack of the first cabinet, the two cabinets are now in parallel with each other and the impedance will be half of the one-cabinet value. Consult your speaker's documentation for connection details as there are many connection methods in use.

Below are examples of parallel connections of 4, 8 and 16 ohm speakers. 2 and 4 ohm parallel loads are not usable because the parallel impedance of 4 ohm loads will always be 2 ohms or less; this is too low an impedance for the CX168 amplifier.

PARALLEL LO-Z LOADS

"Z_{eq}" is the abbreviation for equivalent impedance and means "the impedance at the two connection points"

		2 parallel loads	3 parallel loads	4 parallel loads
		 **2 parallel loads** all same impedance	 **3 parallel loads** all same impedance	 **4 parallel loads** all same impedance
	Formula:	$Z_{eq} = \dfrac{1}{\left(\frac{1}{Z1} + \frac{1}{Z2}\right)}$	$Z_{eq} = \dfrac{1}{\left(\frac{1}{Z1} + \frac{1}{Z2} + \frac{1}{Z3}\right)}$	$Z_{eq} = \dfrac{1}{\left(\frac{1}{Z1} + \frac{1}{Z2} + \frac{1}{Z3} + \frac{1}{Z4}\right)}$
Impedance of speakers	**2 ohm**	DO NOT PARALLEL 2 OHM LOADS FOR USE WITH CX168 AMPLIFIERS		
	4 ohm	DO NOT PARALLEL 4- OHM LOADS FOR USE WITH CX168 AMPLIFIERS		
	8 ohm	$Z_{eq} = \dfrac{1}{\left(\frac{1}{8} + \frac{1}{8}\right)} = 4$ ohms	DO NOT PARALLEL MORE THAN TWO 8 OHM LOADS FOR USE WITH CX168 AMPLIFIERS	DO NOT PARALLEL MORE THAN TWO 8 OHM LOADS FOR USE WITH CX168 AMPLIFIERS
	16 ohm	$Z_{eq} = \dfrac{1}{\left(\frac{1}{16} + \frac{1}{16}\right)} = 8$ ohms	$Z_{eq} = \dfrac{1}{\left(\frac{1}{16} + \frac{1}{16} + \frac{1}{16}\right)} = 5.3$ ohms	$Z_{eq} = \dfrac{1}{\left(\frac{1}{16} + \frac{1}{16} + \frac{1}{16} + \frac{1}{16}\right)} = 4$ ohms

Note: Impedance has a phase angle also. This is typically neglected in basic speaker calculations. Note: 4 ohms is the minimum impedance for stereo or parallel modes; 8 ohms is the minimum impedance for bridge mode.

TROUBLESHOOTING: NO SOUND

Problem: NO SOUND

• *INDICATION:* POWER LED INDICATOR NOT ILLUMINATED

- Check both ends of the AC line cord for proper connection. Both ends should be fully seated in their connectors.
- Check that the AC outlet works by plugging in a known-good device or by testing with a voltmeter. If too many amplifiers are used on one outlet, the building's circuit breaker may trip and shut off power. If this is the case, unplug some of the amplifiers from the outlet, reset the building's breaker and try again. For the other amps, use an outlet that is connected to another circuit breaker to distribute the load current.
- If using the DataPort connections, the channel 1 and 2 DataPort controls the power supply of the amplifier. Check whether or not the DataPort accessory connected to the CX168 is forcing the power supply into standby mode.
- An amplifier which keeps tripping the AC circuit breaker may have a serious internal fault. Turn it off, remove AC power, and have the amplifier serviced by a qualified technician.

• *INDICATION:* SIGNAL LED'S NOT RESPONDING TO INPUT SIGNAL LEVEL (NOT LIGHTING)

- If the green power indicator LED is lit, but the signal LEDs do not light up at all, check the input. Make sure the signal source is working and try another input cable. Connect the source to another channel or amplifier to check that it is working.
- If you are using the terminal block inputs, be sure the connectors are fully inserted at both ends of all interconnecting cables. If you are using the DataPort for your input signal, ensure the DataPort cable is properly connected at both ends and that the device supplying the audio signal to the DataPort is properly configured.
- Check gain controls.

• *INDICATION:* SIGNAL LEDS RESPONDING TO SIGNAL LEVEL

- If the green signal LED indicators are lighting normally, the fault is somewhere between the amplifier and the speakers. Check the speaker wiring for breaks. Check for proper connection at each end. Try another speaker and cable, if necessary. If your system has multiple speakers connected in series, any one speaker failure will cause all other speakers (in the same series circuit) to lose their signal.

• *INDICATION:* CLIP LED FLASHING

- If the red clip indicator flashes when signal is applied, the amplifier output may be shorted. Check that the speaker cable connections are tight and reliable at both ends. Check the speaker cables for stray strands or breaks in the insulation. Trim off all loose strands of wire and insulate any exposed wire with electrical tape. All connections should be made with great care and all wire terminations should be carefully checked to ensure quality workmanship.
- If the red clip indicator flashes when connecting the speaker cable between the + and - terminals, then that speaker cable (or speaker) may be shorted. Check the cable and the speaker.

• *INDICATION:* CLIP LEDS BRIGHT AND STEADY

The amplifier is in protective muting.

- Three seconds or so of muting is normal when the amp is turned on. After power-up muting, the Clip LED's should go out.
- Overheating will cause protective muting and the Clip LED to fully illuminate. If the amp is overheated, the fan will be running at full speed and the chassis will be very warm to the touch; sound should resume within a minute as the amplifier cools to a safe operating temperature. DO NOT TURN THE AMP OFF! This would stop the fan from cooling the inside of the amplifier. Instead, reduce the input signal (or amp gain setting) and wait for the amp to resume operation automatically.
- Check for proper ventilation. If the fan isn't running at all, the amplifier requires servicing.

TROUBLESHOOTING: DISTORTION, NO CHANNEL SEPARATION, HUM, HISS, FEEDBACK

PROBLEM: DISTORTED SOUND

- ***INDICATION:* CLIP LED FLASHING**
- If the red clip indicator flashes before the signal indicator does, the load impedance is abnormally low or shorted. Unplug each speaker one-by-one *at the amplifier.* If the clip LED goes out when you disconnect a cable, then that cable or speaker is shorted. Try another cable and speaker to locate and/or remove the fault.
- ***INDICATION:* CLIP INDICATOR NOT FLASHING**
- This could be caused by a faulty speaker or loose connection. Check the wiring and try another speaker.
- The signal source may be clipping. Keep the amplifier gain controls at least halfway up (-10 dB or less) so that the source does not have to be overdriven.

PROBLEM: NO CHANNEL SEPARATION

- Check the mode switch settings on the back of the amplifier. If the mode switches are set for BRIDGE or PARALLEL mode, there will be no channel separation. The mode switches must be set for STEREO for each channel to operate independently.
- Make sure other equipment in the signal path to the amplifier, such as mixers, preamps, etc., is set for stereo, not mono.

PROBLEM: HISS

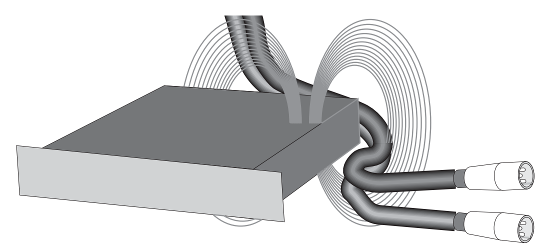
- Unplug the input cables to the amplifier. If the hiss goes away, then the problem is with the equipment or cables leading to the amplifier.
- If the hiss is present with no audio input cables connected, check that the AC line cord is properly grounded at its connection to the line. If the ground connection is OK and the hiss continues with no input cables connected, then the amplifier requires servicing.
- To keep the normal noise floor low, operate the primary signal source at full level, without clipping, and avoid boosting the signal further between the source and the amplifier.

PROBLEM: SQUEALS AND FEEDBACK

- Microphone feedback should be controlled with mixer controls. If noise continues to build up with zero mic gain, there is a serious fault in the signal processors or cables. Working in succession from the signal source towards the amplifier, check each device in the signal path by reducing its gain or unplugging it.

PROBLEM: HUM

- Use a common AC ground connection for all audio equipment.
- Check the input cables for broken ground (shield) connection.
- Use balanced connections for the entire signal chain.
- The design of the amplifier eliminates internal hum fields, but external transformers or other magnetic devices may cause hum. Move cabling and signal sources to identify "hot spots" in the system; then avoid those spots. Cables with faulty shielding are a common entry point for hum. Use top quality cabling. Another common source of magnetic fields are “wall warts” or pluggable transformers; keep input wiring away from them.



Magnetic field from power supplies in equipment can induce hum into cabling that is located in the field. If hum is a problem, try relocating cabling so that is away from power supplies, transformers and other magnetic field producing devices.

SPECIFICATIONS CX168

OUTPUT POWER *in watts*		
FTC:	8 ohms per channel (20 Hz.-20 kHz., 0.05% THD)	90
	4 ohms per channel (20 Hz.-20 kHz., 0.1% THD)	130
EIA:	8 ohms (1 kHz, 0.1% THD, 1 channel driven)	120 (100 watts, all channels driven)
	4 ohms (1 kHz, 0.1% THD, 1 channel driven)	180 (140 watts, all channels driven)
Bridged Mono:		
	16 ohms, 20 Hz.-20 kHz., 0.1% THD	180
	8 ohms, 20 Hz.-20 kHz., 0.1% THD	260
DYNAMIC HEADROOM		2 dB at 4 ohms
DISTORTION		
	SMPTE-IM	Less than 0.02%
	Typical 20 Hz.-20 kHz., 10 dB below rated power	Less than 0.05%
	Typical 1 kHz. and below, full rated power	Less than 0.02%
FREQUENCY RESPONSE		+0.0, -3.0 dB: 8 Hz. to 50 kHz. ±0.2 dB: 20 Hz.-20 kHz.
DAMPING FACTOR		>200 for 8 ohm load (5 kHz. and below)
SIGNAL to NOISE (unweighted, 20 Hz.-20 kHz.)		-107 dB
VOLTAGE GAIN (gain control set to 0 dB attenuation)		26 dB (20 X)
INPUT SENSITIVITY for rated power into 8 ohms		1.35 Vrms
INPUT CLIPPING, Vrms		10 Vrms (+22 dBu)
INPUT IMPEDANCE		10k ohms unbalanced , 20k ohms balanced

AMPLIFIER PROTECTION	Full short circuit, open circuit, thermal, ultrasonic and RF protection. Stable into reactive or mismatched loads.
COOLING	Continuously variable speed fan; back-to-front air flow through tunnel heat sink
INDICATORS	Front Panel: POWER LED (green), 8 each (1 per channel) SIGNAL (green) and CLIP (red) LEDs 4 each (1 per channel pair) BRDG LED (yellow), 4 each (1 per channel pair) PAR LED (orange)
CONNECTORS	Input: (8) 3-pin terminal block ("euro" or "Phoenix") Output: (2) 8-pin detachable terminal block connectors Control & Monitoring: (4) QSC DataPort connectors
LOAD PROTECTION	Turn-on/turnoff muting AC coupling (DC fault blocking) Clip limiting
OUTPUT CIRCUIT TYPE	AB + B
POWER REQUIREMENTS	SEE SERIAL NUMBER LABEL ON REAR PANEL FOR THE SPECIFIED OPERATING VOLTAGE Configured at factory for either 100, 120 or 230 VAC, 50- 60 Hz.
CURRENT CONSUMPTION *@ 120 VAC, typical (all channels driven) in Amperes*	
Idle	0.6
1/8 power, 8 ohms*	6.2
1/8 power, 4 ohms*	9.2
1/3 power, 8 ohms*	9.2
1/3 power, 4 ohms*	14.2

*NOTE: 1/8 power is representative of current draw with typical music program material with occasional clipping. *Pink noise*
1/3 power is representative of program material with severe clipping.

CONTROLS	**Front:** AC POWER switch, (8) gain controls **Back:** (4) DIP switch blocks with 10 poles (1 per channel pair): Clip Limiters, Stereo/Parallel/Bridge mode selection, low frequency filter on/off and frequency selection.
DIMENSIONS	19.0" (48.3 cm) wide, 3.5" (8.9 cm) tall (2 rack spaces) 14" (35.6 cm) deep (from front mounting rails) including rear support ears
WEIGHT	21 pounds (9.5 kg) net, 27 pounds (12.3 kg) shipping

SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE

WARRANTY INFORMATION & HOW TO CONTACT QSC

WARRANTY

(USA only; other countries, see your dealer or distributor)

Disclaimer

QSC Audio Products, Inc. is not liable for any damage to speakers, or any other equipment that is caused by negligence or improper installation and/or use of this amplifier product.

Product Warranty

QSC Audio Products, Inc. ("QSC") guarantees its products to be free from defective material and / or workmanship for a period of three (3) years from date of sale, and will replace defective parts and repair malfunctioning products under this warranty when the defect occurs under normal installation and use - provided the unit is returned to our factory or one of our authorized service stations via pre-paid transportation with a copy of proof of purchase (i.e., sales receipt). This warranty provides that the examination of the return product must indicate, in our judgment, a manufacturing defect. This warranty does not extend to any product which has been subjected to misuse, neglect, accident, improper installation, or where the date code has been removed or defaced. QSC shall not be liable for incidental and/or consequential damages. This warranty gives you specific legal rights, and you may also have other rights which vary from state to state. This limited warranty is freely transferable during the term of the warranty period.

HOW TO CONTACT QSC AUDIO PRODUCTS

Mailing address / Adresse postale / Postanschrift / Dirección postal:

QSC Audio Products, Inc.
1675 MacArthur Boulevard
Costa Mesa, CA 92626-1468 USA

Telephone Numbers / Numéros de téléphone / Telefonnummern / Números de teléfono:

Main Number / Numéro principal / Hauptnummer / Número principal	+(714) 754-6175	
Sales Direct Line / Ligne directe ventes / Verkauf-Direkt / Línea directo ventas	+(714) 957-7100	
Sales & Marketing / Ventes & marketing / Verkauf u. Marketing / Ventas y marketing	(800) 854-4079	*(toll-free in U.S.A. only)* *(sans frais aux É-U seulement)* *(zollfrei nur beim USA)* *(sin costo en EE. UU. solamente)*
Technical Service / Service à la clientèle / Kundendienst / Servicio a la clientela	+(714) 957-7150	
	(800) 772-2834	*(toll-free in U.S.A. only)* *(sans frais aux É-U seulement)* *(zollfrei nur beim USA)* *(sin costo en EE. UU. solamente)*

Facsimile Numbers / Numéros de télécopieur / Telefaxnummern / Número de FAX:

Sales & Marketing FAX / Télécopie ventes & marketing / Telefax der Verkauf u. Marketing / FAX ventas y marketing
+(714) 754-6174

Technical Service FAX / Télécopie service à la clientèle / Kundendienst-Telefax / FAX servicio a la clientela
+(714) 754-6173

World Wide Web: **www.qscaudio.com**

E-mail: info@qscaudio.com
tech_support@qscaudio.com

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QSC Audio Products, Inc. 1675 MacArthur Boulevard Costa Mesa, California 92626 USA