



*DAAD*  
*and*  
*Audio Quality Test method*  
*user manual*



*Theory and practice*



*DaaD and Audio Quality Test method  
user manual*

**DAAD**  
*and*  
**Audio Quality Test method  
user manual**



***Theory and practice***



**4<sup>rd</sup> edition**

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## **INTRODUCTION**

Today the Acoustics, an important aspect of the sound reproduction, has the necessity to be dealt with, at last, in a comprehensive and concretely usable way. If as a result of trying to simplify this task, we show too many cuts and we open ourselves up to criticism, never mind! **The important thing is at least to make a start.**

The listening surrounding, origin of important distortions, is a factor that produces a big qualitative and quantitative deterioration of the reproduced sound.

**We venture to reach a provocative, but not too much, percentage: 30% of distortion is due to the chain of reproduction; 70% is due to the surrounding. The purpose of an acoustic treatment is to halve the percentage of the surrounding distortion.**

When we take into examination an acoustic test signal and we make two different measurements, the first immediately at the exit of the speakers and the second at the listening point, we will note that between the signal directly put out from the speakers and the same signal as it reaches the listening point, there are huge differences.

These differences will be noted in terms of tonal balance, with swerves of 6/10 dB between one frequency area and another, and in terms of dynamics, that for instance will decrease of 8/12 dB.

*We want to remind the reader that the scale of decibels( dB) is not linear, instead it is logarithmic: for instance the rise or decreasing of 6 dB means doubling or halving the sound pressure. Consequently if you read on your Sound Level Meter or on the graphic of the A.Q.T. (Audio Quality Test) a change of 6 dB, this means that the signal has doubled or halved; if you read a variation of 20 dB, the sound pressure has risen or decreased 10 times. At this point you understand how important it is, in your listening surroundings, to increase the dynamics (articulation) or to reduce the swerve between a frequency area and another (tonal balance), be it only 1 dB.*

This data, so self-evident, is witness to how critical the final interconnection ring is: the listening surroundings.



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Within current normal audio chains there is no other ring able to cause such a big deviation of the original audio message. In comparison the normally good quality of the electronics and cable components introduces to the audio chain much less significant variations and of a much smaller order.

The task of good acoustic treatment is to ensure that the audio signal coming from the speakers, reaches the listener's ears as unchanged as possible, without undergoing terrible surrounding influences. This is the only way the musical message, the recording and the audio chain characteristics can reveal themselves.

### **RESONANCES AND REFLECTIONS**

**The resonances generated in a room and the early perimetral reflections are the main causes of sonic surrounding distortion.**

#### ***Resonances***

The speakers' drivers make a piston motion that compresses and decompresses air. The human ear perceives a sound when the frequency of this motion is included between 20 and 20,000 times a seconds (Hertz!).

The speakers' drivers therefore produce waves of compression and waves of rarefaction of air. This is the way the acoustic energy is transferred from the speakers to the listening room. In a closed room, this energy does not fade quickly as it happens in open air, it hits the walls: it is partially absorbed, but a good portion of it bounces. It bounces and bounces again until it is completely absorbed. These rebounds are harmful because they interfere with the new following sounds. Their duration depends, for the longest ones, on the dimensions of the room, in this case these bouncing frequencies are named "fundamental resonances": They are strong, unpleasant and can be easily excited.



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Two examples explain how they create themselves.

a) In a half-full bath a baby causes squirts and small waves:

in moving his hands back and forth, it can be that sooner or later the baby finds a rhythm (a frequency) so that it creates a bigger wave that comes and goes; if the moving hand goes on with its (small) rhythmic movements, the wave can rise until it spills out of the bath. The appropriate frequency of the shaking hand has so excited the resonance of that bath: a big result with little force. The water level in the middle of the bath remains constant, since the big oscillations are at the two extremes of the bath. The same applies to the sound: an adequate acoustic sequence excites a resonance between two opposite walls, and a small woofer can move big air masses to give a big quantity of sound.

b) A stone hung by a thread, kept in hand.

With small and appropriate blows the stone oscillates like a pendulum: little force but with a sense of timing between blows (adequate frequency), with the result of big oscillations.

Here is the way the fundamental resonances between the three pairs of the opposite walls in a room, arise.

The compression and rarefaction waves generated by the surroundings add or subtract energy to the waves generated by the speakers, distorting them in a notable way.

### ***Reflections***

A speaker does not produce sound that reaches in its totality, only the listener's ear. At the lowest frequencies the speaker irradiates at 360 degrees. The highest frequencies travel instead in a very directional way. The frequencies comprised between the lowest and the highest ones have a radiation pattern that goes from wide to narrow depending on the rise of the frequency. Therefore, only a part of the sound coming from a speaker reaches the listener's ear. In a closed room,



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the remaining part bounces off the walls before reaching the listener with a time delay unlike the direct signal.

The point of the wall where the reflection arrives acts like a true speaker that emits a sound from a different position, with lower energy (because the wall absorbs a part of it), and with a time delay (because its path is longer) unlike the true speaker.

If this time delay is less than 25 milliseconds, we have the so-called "sound fusion".

The human brain cannot perceive the difference between two separate sounds that reach it within 25 milliseconds. Therefore the brain instead of locating the sound originating from the speaker and the sound originating from the wall, will perceive only one sound, duller and dirtier, located between the speaker and the wall: the brain has fused together the two sounds.

This effect, studied by Haas, is the fundamental point of the stereophonic theory, that is to say when two speakers emit two equal or different sounds.

A monophonic signal emitted from two speakers is the same for the two channels. The sound coming from the left speaker is the same as the sound from the right one. The sound fusion effect creates a sonic image placed only at the center of the soundstage.

Stereophonic technology implies also different sonic contents for each of the two channels. Since these sonic signals reach the ear at the same moment (the listener is sitting at the same distance from the two speakers), we do not perceive the differences existing between the two channels, instead, thanks to the sound fusion effect, we will be able to perceive a virtual image not exclusively placed in the center point, but which goes from the right to the left, more on the left or more on the right, occupying all the space behind the speakers to create a true sonic box.

Though the "sonic fusion" effect is of basic importance to the creation of the stereophony, it represents a negative factor whenever the early perimetral reflections work, that is to say everytime there is a sound in a closed room.

The "early reflections" are all those reflections gifted with high energy, which reach the human ear with a time delay lower than 25 milliseconds in respect to the signal coming directly from the speakers.



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When a reflected sound reaches the ear with a higher delay, the ear-brain system begins to consider it as a different sound, distinguished from the primary signal. It is at this point that it begins to be perceived as an echo, that is to say as a distinctive sound, temporarily successive to the initial sound.

A common living room has no such dimensions to allow the perception of the echo phenomenon.

In a normal living room, all of the high-energy sound reflections (that reach the ear after one or two reflections, conceding little energy on the walls because they bounced very little) reach the ear within the sound fusion period.

In a closed room with two speakers which produce each a signal, there are 8 primary reflections: 4 on each lateral wall, 2 on the ceiling, 2 on the floor.

So each surface generates many secondary sound reflections.

All of these sound reflections interact with the dynamics and the tonal balance of the original sound message, deteriorating it both in space and time terms.

### ***THE ACOUSTIC TREATMENT***

The acoustic treatment of a closed room serves to replenish the sound with its original features, avoiding bad room influences on it.

In a good listening room, resonances are controlled, they are not concentrated within one single frequency area. Early higher energy reflections are treated so to minimize their negative effects. The acoustic pressure, that has its maximum strength in the corners and close to the walls, is more uniformly distributed throughout the room.

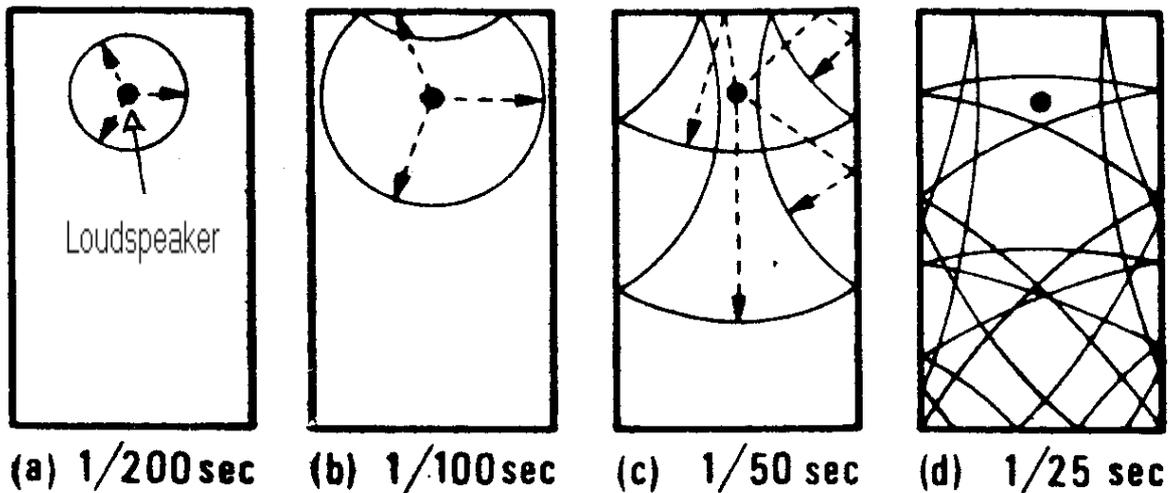
The sound energy should be maintained as diffused as possible, so that it is not concentrated in specific points of the room or in specific frequency areas.

The reverb times at all frequencies should be suitable to the musical listening.

The acoustics of a well-treated room are pleasant yet very much alive; are controlled yet natural.



**Fig. 1 - Propagation of a low frequency impulse**



**HOW TO REALIZE AN ACOUSTIC TREATMENT FOR STEREOPHONIC LISTENING**

An acoustic treatment plan must include 3 aspects:

- A) Positioning of the speakers and of the listening point
- B) Resonances
- C) Early Reflections

**A) The positioning of the speakers and of the listening point**

Depending on the position of the speakers, the air inside a room reacts in different ways. Therefore the types of resonance and the reflections change, in qualitative and quantitative terms, depending on the location of the speakers.



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Also the listening point feels the effect of its location inside the room, because each different location "sees" the rooms acoustic response in its own peculiar way, exalting some frequency ranges and penalizing others.

We could say that the listening point is a true "acoustic point of view".

It is therefore important to look for the best location for the speakers and for the listening point.

In any case it is possible to obtain more than acceptable results also in cases where we cannot freely search the best positions for the speakers and for the listening point in a room: the condition for good results, in this case of compromise, is to be allowed to make an acoustic treatment chunkier than the one sufficient with an optimum position of the hi-fi system.

Basically there are three ways to find the correct position of the speakers and the listening point.

### **1.The rule of the thirds, the rule of the fifths and more**

The so-called rules of the thirds and of the fifths are empiric methods. They apply to rectangular rooms and can be used separately or together. Divide the width and the length of the room by 3 and/or by 5. The resulting values show the best distances, where to place the speakers and the listening point.

Example: a room 7x5 meters

- Rule of the thirds: each speaker is 2.3 m. from the back wall and 1.6 m. from the lateral wall; the listening point is 2.3 m. from the rear wall and, naturally, in a central position between the lateral walls.
- Rule of the fifths: each speaker and the listening point are 1.4 m. from the rear wall and the speakers are 1 m. from the lateral wall.
- Rule of the thirds/fifths: we locate some points with the rule of the thirds and some points with the rule of the fifths. In our case we could place each speaker 2.3 m. from the rear wall and 1 m. from the lateral wall, with the listening point 1.4 m. from the rear wall, in a central position between the lateral walls.

In addition to these empiric rules there are more, very important ones, that should be considered in any circumstance:



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- Each speaker will not be equally distant from the back and the lateral wall: The difference should be at least 30 cm.;
- The speaker and the listening point should form an equilateral or a slightly isosceles triangle; the two longest sides of it will form the vertex for the listening point.
- In order to create an homogeneous soundstage, the speakers position should be as symmetrical as possible, in respect to the lateral walls and the back wall. This kind of symmetry should also be present regarding doors, windows and with bigger items of furniture.
- The listening point should have a good space behind its shoulders so to have the amplest reverberated field and to minimize the effects of the early back reflections.

### **2. Positioning "by ear"**

This means patient work of research by listening to the system in various locations, by evaluating the sonic results bearing in mind the effects on the sonic image and the response of the transients at low frequency (for instance the kettledrum beats). It is recommended to use a good and well known track from which we try to reconstruct an image which will have lateral and depth dimensions and the virtual images, especially at the center stage, will have optimum focus.

### **3. Searching for the best position by using specific software systems**

Some companies have developed software where is possible to draw a room plan where putting two loudspeakers and one listening positions, we can ask to the program to calculate the best setup simulation. The computer has the role of binding the many variants in play.



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This is probably the most valid way to a serious solution of the problem. Unfortunately at the moment the programs on the market are not sufficiently sophisticated to offer valid and constant results.

### **B) Resonances**

#### **The treatment of the resonances with DAAD**

(Please note: with "anterior" and "posterior" we intend "in front of" and "behind" the listener facing the speakers).

The resonances can only be properly treated when we know which ones they are. We should start from the dimensions of the room.

Let us assume we have a room 5.5x3.7x3 m.

We have already underlined that the principal resonances that we must combat are the axial ones.

To find them we need to divide half of the sound speed (172 m/s) by the dimensions of the room (fig. 2).

In this way we find the fundamental reflection. To find the other ways or multiples of the fundamental reflection, we multiply the values of this latter by 2, 3 or 4.

For practical purpose we will deal only with the resonances with a frequency lower than 150 Hz.

**Notes:** We remind you that the sound propagation speed, at 0° C and 760mmHg of atmospheric pressure, has been calculated as:  $S = 331,45 + 0,607T$  where T represents the surrounding temperature in centigrade. So it results an increase of speed of about 0,6 meters a second, for every degree of temperature; leaving out the parameters of air density too, we reach the definition of the sound speed at 21° C in 344 meters a second.



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The dimension 5.5m. has its fundamental resonance at 31,2 Hz (172:5.5); the resonance of second harmonic is at 62,4 Hz (31.2x2); the resonance of third harmonic is at 93,6 Hz (31,2x3); the resonance of fourth harmonic is at 124,8 Hz (31,2x4), and so on.

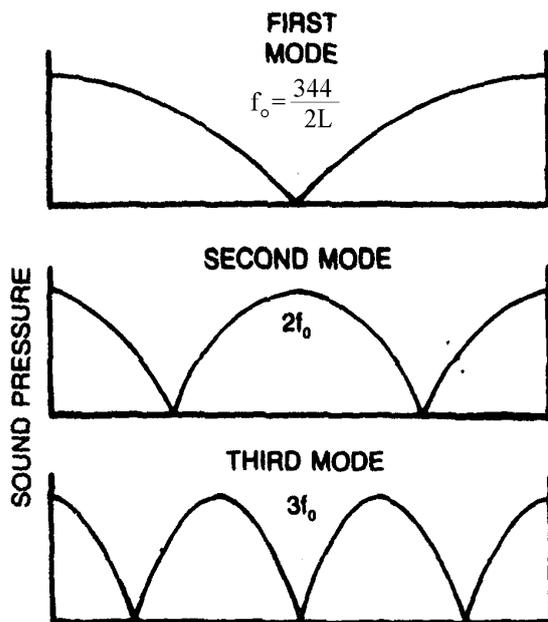
The dimension 3,7m. has: 46,4 Hz; 92,8 Hz; 139,2 Hz.

The dimension 3m has: 57,3 Hz; 114,6 Hz.

Now we know the resonant frequencies below 150 Hz.

Before understanding where we must apply treatment, let us observe the obtained values and let us make a reflection.

Let us line up the data in order of growth: 31,2; 46,4; 57,3; 62,4; 92,8; 93,6; 114,6; 124,8; 139,2.



**Fig. 2**

We will see intervals, which are quite similar between the implied frequencies: resonances are distributed quite well. This applies to all frequencies except at 90



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Hz, where we have the second harmonic resonance of the pair of walls 3,7m. very near to the third harmonic resonance of the pair of walls 5,5m. In this area of frequencies, the two different resonances add up to produce a stronger "boom".

### **Where are the points of higher sonic pressure in a room?**

**All of the resonances are particularly strong in the corners. Treating the corners, we treat all kinds of resonances.**

The fundamental resonance has its points of maximum acoustic pressure at the beginning and at the end of its dimension (and so close to the parallel walls). It has its minimum pressure point halfway up the length of the wall.

The second harmonic resonance has its point of maximum acoustic pressure at the beginning, at the end and exactly halfway. The points of minimum acoustic pressure are at the fourths of the dimension. The two extremes of the three obtained segments are those of maximum pressure. Instead their central points are those of minimum intensity.

Let us return to our example.

Let us consider the 5,5m. wall, which will have a beginning (0m.) and an end (5.5m.). For the fundamental resonance the points of maximum pressure will be 0m. and 5,5m. while the point of minimum pressure will be 2,75m.

For the second harmonic, the points of maximum intensity will be 0m. 2,75m.,5,5m. while the points of minimum intensity will be 1,375m. and 4,125m.

For the third harmonic we will have peaks at 0, 1.83, 3.6, 5.5m.

The points of minimum intensity will be 0.91, 2.75, 4.59m.

When treating our 5.5x3.7x3 room we will set up DAAD in all of the corners in order to reduce the intensity of all the resonances and we should treat the third harmonic of the dimensions 5.5 m. and the second harmonic of the dimensions 3.7m. by placing DAAD exactly in the points 1.83 m. and 3.6m. of the dimensions 5.5m. and/or in the points 1.85m. of the dimensions 3.7m.



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### **Which DAAD should be used?**

DAAD are broad band acoustic traps (this means that they do not work only within a limited range of frequencies) which have a cutting point at low frequencies which corresponds to the diameter of the trap. Since they have a lobe which diffuses the frequencies higher than 400Hz, the cutting point at the higher frequencies is determined by the degree of the trap rotation. Now, for reasons of simplicity, let us consider the DAAD only for their cutting points at the low frequencies, after we will treat the aspects concerning the diffusion.

Here are the cutting points of DAAD in different diameters.

- DAAD 4 : 55Hz
- DAAD 3 : 80Hz
- DAAD 2 : 110Hz

The height and the shape of DAAD modify the quantity and not the quality of the absorbed sound.

Knowing this and knowing the frequency characteristics of the specific resonances of every room, it is easy to choose the position and the type of the most suitable DAAD for every single circumstance.

Our 5.5x3.7x3 room will witness the use of the following DAAD.

Question: Will we need DAAD 4 in the corners?

Answer: The fundamental resonance of the long wall is at 31.2Hz. None of our DAAD absorb in an efficient way such a low frequency. Therefore it will be possible to treat the resonances of the 5.5m. wall only by starting from those of the second harmonic (62.4Hz).

To treat a resonance at 60Hz it is sufficient to use DAAD 4.

The traps will be put in the corners of the room so as to be efficient also for the other resonances. We prefer to use DAAD with wider diameters by positioning them in the corners behind the listener, making it ideal to treat the lowest frequencies.



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The front corners (the most important places for treatment) could be occupied by DAAD 4 (which deals with the resonances higher than 80Hz like those at 92.8, 93.6, 114.6, 124.8, 139.2Hz) and by a DAAD 3 (which acts for the resonances higher than 120Hz).

Since we have a boom around 90Hz, we will use one or two DAAD 4 for the treatment of the second harmonic of 3.7m. and/or the third harmonic of 5.5m.

At this point most of the resonant frequencies will have a better articulation and a better balance.

Let us suppose, always with a purpose of simplicity, to carry out an instrumental check in our room and to verify a residual resonance around 185Hz. We will observe that 185Hz corresponds to the fourth harmonic of the 3.7m. walls.

In order to tone it down we will place DAAD 3 or DAAD 2 (we are at 185Hz) in these points of the 3.7m. wall : 0.92 and/or 1.85 and/or 2.78.

**A correct treatment of the resonances is possible only by placing the right acoustic traps in the right places.**

**The corners are always the "right places", but when positioning DAAD along the walls of the room it is necessary to be careful and to act with accuracy so to install them in the points of larger acoustic pressure. By positioning the acoustic traps along the walls one makes a selective operation. If one misses the position an intervention may occur in the points with a minor acoustic pressure instead of in the points with major acoustic pressure, with the very bad result of accentuating the differences of the tonal balance.**



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### **C) The treatment of the early reflections**

Generally in a normally dimensioned room, the early reflections (=within 25 milliseconds) are those that reach the listener after bouncing one or two times off the walls.

The primary reflections (a single rebound off the wall) are the most troublesome because they consist of a lot of energy.

Many secondary reflections (those which rebound twice off the walls before reaching the listener) also stay within the sound fusion time, but often their reflection points coincide or are close to those of the corresponding primary reflections, but their energy is weaker.

For these reasons, we will give much more practical importance to the primary reflections and less importance to the secondary ones, even if they are early.

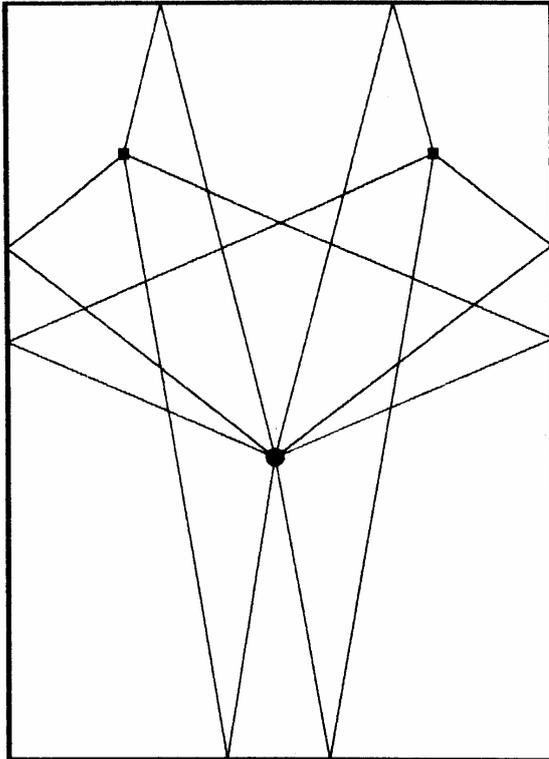
### **Search for the primary reflection points**

In a closed room with 4 walls, a floor and a ceiling, where a reproduction system with 2 speakers is playing, there are 12 primary sound reflection points. Two are on the front wall, (behind the speakers), 4 on the lateral walls, 2 on the back wall, 2 on the floor and 2 on the ceiling.

The reflections that we want to treat are mainly those on the walls. Why? For practical reasons: it is difficult to place DAAD on the floor, a thick carpet could be of some use; it is rarely possible to install DAAD on the ceiling....



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**Fig. 3 Primary perimetral reflections**

### *The front reflections*

They originate from two points of the wall behind the speakers. If the speakers are of the dynamic type, which do not emit trebles from the back, the reflections consist mainly of low frequencies (this is not the case with bipolar speakers). The flattening of the soundstage, a lack of intelligibility, a lack of sound articulation and a lack of focus are the principal damages they cause.

### *The lateral reflections*

There are four of them, 2 on the left and 2 on the right wall. Each speaker creates one reflection on the omolateral wall and one reflection on the controlateral wall.

soundstage and, if very strong, causes the so called "hole in the middle" of the sonic image that is to say, in the worst cases, the absence of sound from the center stage (the sound comes out of the speakers) or, in the less important cases, a dilated image at the center of the sonic stage.

The lateral controlateral reflection shrinks the sonic image and creates confusion : some think that this is the most dangerous reflection.

As a matter of fact we have for instance a sound that comes out off the right speaker and rebounds off the left wall, so it reaches the listener's ear with a little delay and with a certain amount of energy, but from the opposite part of where it should, therefore confusing itself with the signal coming from the left speaker.

This is the worst thing that can happen to a stereophonic sound, which in this way will lose most of its characteristics.

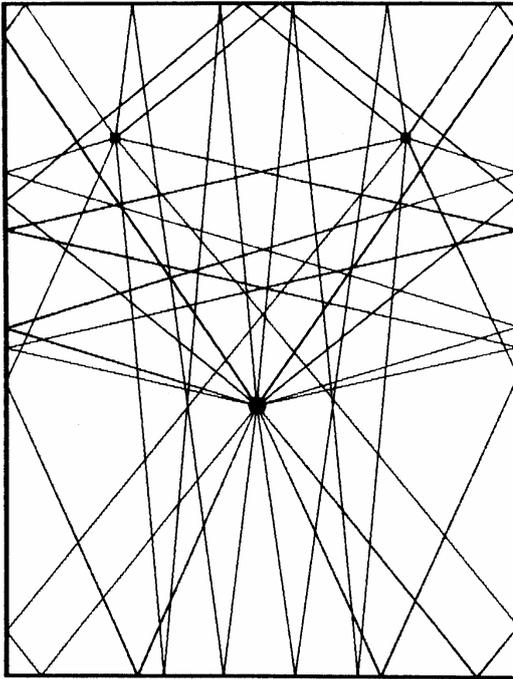
The lateral-controlateral reflection contributes to the widening of the sonic



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The spectral content of the reflections varies a great deal depending on the directivity and the position of the speakers. Generally the involved frequencies are located in the medium range and in the upper part of the warm zone.

### *The back reflections*



**Fig. 4 - Secondary perimetral reflections**

we will find one point of primary reflection. If we point the flashlight to the right wall and we illuminate the right speaker, we will locate the primary homolateral reflection point. We will carry out the same operation with the other lateral wall and we will find the other 2 primary reflection points. The same method will be used for the front reflections by pointing the flashlight on the strip of the wall behind the speakers.

For the back reflections, we will point the flashlight on the strip of the wall behind our shoulders. If we want to locate the primary reflections on the ceiling we will have to position a strip on the ceiling and to find with the flashlight the point on the strip that makes the speaker illuminated. A quite similar method consists of placing

They originate from 2 points of the wall behind the listener. They create tonal balance problems, especially at the higher frequencies. It is very easy to find the points of the primary reflections.

We will use a method that in the past centuries the designers of the best theatres were using. Light has the same behaviour as sound when rebounding off a reflecting surface: the entrance angle is the same as the exit one. For this reason, we will use a source of light in order to find the exact position of our reflection points.

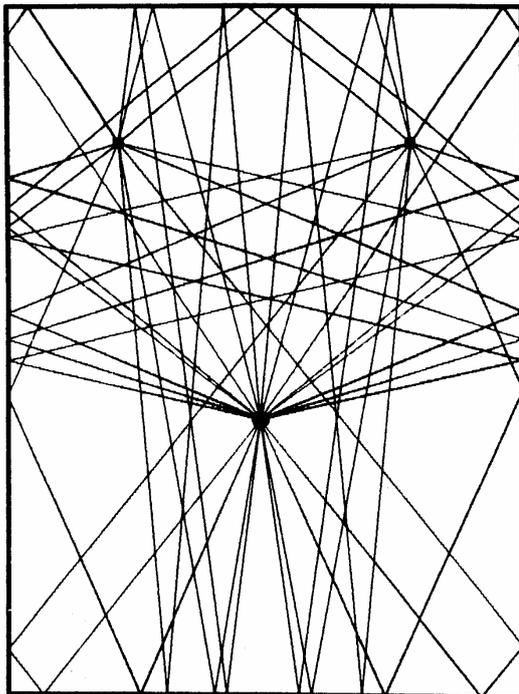
We will place a reflecting strip on the lateral walls, on the front and on the back wall.

Sitting in the listening point, we will point a flash light, for instance, to one lateral wall. When the light, reflected from the wall, illuminates the speaker,



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a light (for instance a lit candle), on top of the speaker, and the eye of the listener will see on the reflecting strip the points of primary reflections.



**Fig.5 - Secondary and primary perimetral reflections**

be made incorrectly if the acoustic traps are not located in the early reflection points, instead they stay in points of late reflections (many rebound off the walls) lack in energy.

The reflections that reach the listener after the sound fusion period, or that arrive weak are useful for a comfortable listening and they must not be destroyed. For this reason DAAD have one lobe of their lateral surface more sound diffusing than the other .

### **Reflections treatment**

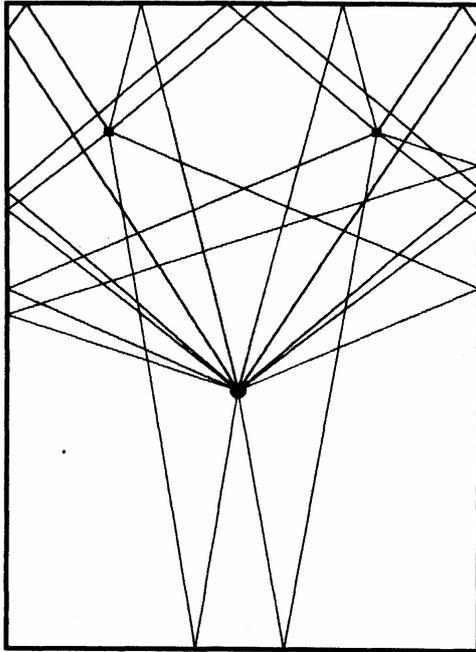
The purpose of the treatment is to subtract energy from the early reflections, so that they reach the listener's ear almost inaudible or very weak. To achieve this it is enough to use an acoustic trap as high as the speaker or at least as high as the listener's ears, which should be exactly in the reflection point.

Bearing in mind the reflections involved, acoustic traps that act from 120Hz and upwards (with the exception of the treatment of the back reflections) are sufficient. In most cases DAAD3, DAAD 2, are sufficient for a good treatment. In the presence of strong reflections, more powerful traps should be used (DAAD 4).

Like the treatment of the resonances, the treatment of the reflections can



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**Fig.6 - Perimetral reflections  
inferior to 9m.(7x5m. room)**

If rotated in a suitable way, a DAAD serves us by delaying the reflected sound, maintaining the surrounding energy not in a concentrated way, but in a diffused way (fig. 7).

Let us make an example:

in our treatment plans it is possible to see that a placed in the primary lateral homolateral reflection point has the seam ( the center of its absorbing half-equal surface) which faces the speaker. This way the sound that rebounds the wall is treated by a broad band trap since we do not want that any part of this sound reaches the listener's ear within (the sound)fusion period. The opposite part of the , the diffusing part, faces instead the back part of the room.

On this diffusing surface, due to its curving, the sounds coming from the lateral and the back walls, after numerous reflections, will strike and rebound. This diffusing surface will involve the acoustic energy once again, that is to say it will be reflected once again in a functional way as regards the musical listening.



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This possibility of rotation offers a powerful way to control, to set up and to vary the sound in a room, also because the primary reflections, with all of their additional and cancellation effects, are finally obscured. It is for this reason that it is possible to obtain a lively though well controlled listening room with DAAD.

By avoiding mistakes of bad positioning, one will not have to suffer a dry sound in a dull sounding room. Instead one will obtain a correct, articulate and lively sound.

Recent studies have shown that the major part of correct sound reproduction is determined in that brief time-lapse which follows the attack transient. That which comes in the first 25 milliseconds will determine some of the sound qualities (focusing, tonal balance etc.); the time delay included between 25 and 60 milliseconds is determinant for the ambience sensations, to project the listener in the recording sound dimension. The complicated acoustic events which happen within these two periods determine approx. 70% of the final sound quality ( the remaining 30% depends on the quality of the direct sound meaning the sound reproduction system)

Therefore, in projecting DAAD, it has been taken into consideration that a modern acoustic trap must be capable to attack the early reflections and the resonances, while conserving the acoustic energy with 25 to 60 milliseconds delay, broadcasting it within the listening room.

DAAD assure the most efficient treatment of the low frequencies and transform the real measures of a given listening room in dimensions which are virtually amplified. Rooms treated with DAAD possess a live acoustic, though controlled.

The “soundstage” opens and richens in depth. Single instruments become live, tangible and rich in plasticity.

These results are made possible by the use of new forms and new materials.

DAAD are made of drawn microperforated sheet and hard wood.

The internal part is made of three different filtering material, which insures high acoustic absorption performance, it is fireproof (first class) and anti-allergic.

The complete structure of the trap has been ideally designed for the conversion of the acoustic energy into heat and dull vibration.



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The external surface, with its particular structure, has the double function of broadcasting the sound without coloring it, and creating particular pressing conditions on the inside of the trap itself, rendering it efficient even at the lowest frequencies.

The design is, without doubt, new and attractive, conceived by a pool of architects and internal designers, who have aspired to emphasize forms instead of concealing them.

### **NECESSARY DATA TO PLAN AN ACOUSTIC TREATMENT.**

The above written considerations make it clear that the following data is necessary for a plan of treatment:

- 1) Shape and dimension of the room (including height).
- 2) Distance of the speakers and of the listening point from the walls.

Also useful are:

- 3) Position and dimensions of the furniture, windows and doors.
- 4) Listening impressions of the listener (harsh sound, confused basses, exceeding basses, narrow image, lack of focus, ecc.).

### **THE AUDIO QUALITY TEST ( A.Q.T. ): EVALUATION ELEMENTS**

It has been calculated that when listening to a stereophonic system in a normally dimensioned room, the listener perceives about 20% of sound directly coming from the speakers, and a 80% of sound originating "from the room", that is to say a sound that comes out the speakers, rebounds in various ways and finally reaches the listener.

This percentage changes depending on the dimensions of the room and on the distance speakers/listener : the variations are in the order of 3%. Anyway the reflected sound largely exceeds the direct one.

Do not get it wrong: these percentages do not show the ratio between the distortion brought by the reproduction system and the distortion brought by the surroundings. We do not say that a cable, a piece of electronics or a speaker have



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for instance 20% distortion while the room has 80%, we instead affirm that the sound of a Hi-fi system, apart from its components and its characteristics, reaches the listener's ears 80% in a reflected way and 20% in a direct way.

In order to obtain a correct stereophonic listening, because of the nature itself of the stereophony, it is necessary to modify this unfavorable ratio between direct sound and reflected sound in favor of direct sound.

Many people wrongly believe that direct sound means "boxed" sound. The contrary is true: when a higher percentage of direct sound reaches the listener's ears, the sound becomes free from the speakers cabinet, it becomes open and spread all over the space of the soundstage (that portion of the room that starting from the imaginary line which joins the speakers arrives at the wall behind the speakers) sometimes giving even the feeling to overcross the physical limits of the room itself.

A modern acoustic treatment must free the stereophonic sound from the limits created by the room.

**Stereophony consists of our brain ability to perceive two different sounds, given in a determined space of time (25 milliseconds), as a single sound, to which the brain attributes a different spatial location depending on the ratio between the intensity and the temporal distribution of the two sounds.**

In other words if we were in open air, if two speakers had the same distance from the listener, if both the left and the right speaker produced a 1000Hz sound with an intensity of 80dB, we would not perceive two distinct sounds, one from each speaker, but a single sound located exactly in the middle of the two speakers.

If the sound coming from the right speaker was at 70dB, we would perceive all the same a single sound, though moved towards the left side. And so on.

When the surrounding adds or subtracts something to or from the sound coming from the speakers and acts within the 25 milliseconds following the direct emission, this delicate and subtle balance between the speakers is broken. So our previous central sound, in presence of strong reflections behind the listener's head, will move towards the listener. In case of strong lateral reflections it will always set up in a central position, but its edges will become undefined. The entire sonic image will move to the side with the presence of stronger lateral reflections from one side, if the speakers are positioned in an asymmetrical way.



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For a real stereophonic listening it is necessary that the rooms acoustic treatment, the speakers and the listening point positioning are such as to reduce the percentage of the reflected sound in favor of the direct sound.

The A.Q.T. is a test able to give very interesting information about this matter.

It shows immediately the tonal balance behavior and the musical articulation degree.

When studying an A.Q.T. one can "see" the most important characteristics of a Hi-fi system sound in a specific room: one can understand if there are those acoustic basic requirements that allow a correct stereophonic reproduction.

The target is, as a matter of fact, to obtain from the examined Hi-fi system a "correct basic trim" where to work after to add all of the lawful, personal taste characteristics reserved to the listener's individual sphere, to his sensibility and to his listening needs.

Until now the sound reproduction world has entirely been dominated by personal evaluation elements, completely separated from whatever connection with scientific parameters. This is not right: it is our opinion that personal taste should be entitled to dictate its desires only within the limits of sound correctness, a lot like a writer that uses his preferred style within the limits of substantial grammatical accuracy.

An effective use of the A.Q.T. can lead to the threshold of the **CORRECT STEREOPHONIC REPRODUCTION**, through rigorous operative paths that are not bound to the insanities of an anarchic personal interpretation, though giving wide space to personal taste and exploiting it also.

The user will note from the beginning that there is a strict correlation between the data of the test and what he can perceive with his ears; it is a matter of fact that the A.Q.T. is a very rare example of an instrumental test that does not interfere with the direct listening experience. A better A.Q.T. always means a better listening experience. This way the A.Q.T. test is a very good learning tool to perfect the listening experience.

After the early practical experiences, the use and the interpretation of the A.Q.T. becomes simple and familiar: it is important not to be discouraged in the very first moments.

For this reason we would like to warn you about a number of initial misbehaviors, which are rather common and unpleasant, and also we would like to give some practical suggestions.



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On first examination, the A.Q.T. graph should be observed the same way as we look at a panorama picture, by trying to catch the general aspects without immediately stopping at the particulars.

The general aspect of the graph is always the most indicative parameter since it shows the behavior of the tonal balance and the musical articulation degree.

### **EVALUATION OF THE TONAL BALANCE**

Let us observe the of the graph peak line or let us select the proper window of our SOUND ANALYZER program: the continuous line shows the course of the tonal balance from 20 Hz to 2000 Hz.

Though the signal has been linear during all of the duration of the test, in a closed room it is practically impossible to obtain an equal linear curve.

The target is to obtain a course characterized by "soft" variations between the levels of close frequencies.

In not treated rooms, it is usual to observe a graph with an irregularly pointed course, similar to the ridge of a rocky chain of mountains. We want to obtain instead a resulting graph similar to the sweet profile of Siena's hills.

In the first case (rocky mountain chain), we observe differences of many dB (8-12) between close frequencies. For instance a value of +6dB at 200Hz, -4dB at 220Hz, +8dB at 250Hz, +2dB at 270Hz, etc. In the second case (sweet profile of grassy hills), we observe differences of many dB between far frequencies (for instance between 100Hz and 450Hz), but contained in 2-4dB between the close frequencies above-mentioned.

The first case is typical of an incoherent and inconsistent sound. The virtual images have no stability, they pulse and shift as long as the notes changes. It appears problematic to find a stable volume level because, with sudden changes, the Hi-fi system will seem to play now low and right after too loud and vice versa.

Voices can be from time to time nasal or dark or shrill. Low frequencies will be swollen or inconsistent. The high frequencies will be piercing or dull.

In the second case the sound is coherent and firm, well balanced and stable.



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In order to transform a "rocky chain of mountains" graph into a "sweet profile", you have to follow these two ways:

- Firstly find a better positioning for the listening point and the speakers and secondly carry out an acoustic treatment of the room.

In our opinion the tonal balance is particularly conditioned by the position of the speakers and by the listening point, while the musical articulation is very tightly bound to the degree of acoustic treatment.

Many cases of bad response in the tonal balance, have been well solved by an accurate re-positioning of the speakers and/or the listening point. Often the use of broadband acoustic traps, efficient also at the low frequencies, helps a lot in solving this kind of problem in an optimal way. Only in the most rebel cases it is necessary to intervene with an aimed- selective acoustic correction.

Anyway the first thing to do when searching for the "sweet profile of grassy hills" is to move the listening point and the speakers, checking the results of each step through the A.Q.T..

### **Tonal balance under 100Hz:**

On the graph it is necessary to check if there is a resonant boom (a sort of a high energy wave which rises at the center of two frequency areas which are more depressed regarding the average).

A resonant boom is an unpleasant accident (maybe the most harmful) which often occurs because of the pooling together of two close resonances around one frequency. The way to defeat it is to look for a new spatial relationship among the speaker and all the pair of parallel walls, especially avoiding to equal the distances between the walls behind the speakers and the lateral walls.

If the graph shows that low basses (under 50Hz) are stronger than the higher basses (50-100Hz) it is necessary to move the listening point forward, much more inside the room. Move instead the listening point toward the back wall in the case you look for a reinforcement of the low basses.

We will have the same result by moving the speakers forward and backward.

Since it is necessary for the speakers and the listening point to form an equilateral or a slightly isosceles triangle, often the two effects cannot be added, instead we



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have to find a balance between the room and the speakers-listening point triangle, considering this triangle as a whole.

### **Tonal balance within the warm zone:**

Up to 200Hz, the tonal balance of the warm zone is very sensitive to what occurs with the lower frequencies and it is also sensitive to the effects of the front reflections (those coming from the wall behind the speakers).

In this case the tonal balance has a bad response in this frequency area, it is necessary to distance the speakers out from the rear wall much more and/or to treat the early front reflections.

Never forget, when moving the speakers, what we have already written:

- it is necessary to consider the speakers-listening point triangle as a whole. According to necessity, the dimension of the triangle can be changed in order to keep a balance among different orders of needs.

Between 200 and 300Hz the lateral reflections start making the listener aware of their heavy presence.

In order to better this unpleasant presence it is useful to space the speakers out from the respective lateral walls, shrinking the triangle and so drawing the listening point nearer.

Sometimes sudden variations of the tonal balance in this area (and in the first part of the lower mediums frequencies) are due to cancellation problems depending on an asymmetrical position of the speakers (even of few centimeters) as regard their distance from the respective lateral walls.

For this reason we strongly insist that you check very carefully the proper symmetry of the distances between the lateral walls and the homolateral speakers, between the speakers and the rear wall and between the speakers and the listening point (the closer you are to the speakers, that is to say the more you are subjected to the direct emission field of the speakers, the more these measures must be extremely accurate).



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### **Tonal balance within the lower mediums:**

From 300 to 500Hz the lateral reflections continue to influence the sound in a significant way. In order to make the response more linear, a broadband traps acoustic treatment is strongly recommended together with the search for a different speakers orientation. Often the most immediate way to obtain a more linear response of the involved frequencies is to rotate the speakers toward the listener.

For a good response between 500 and 1000 Hz it is necessary to take into consideration the distance between the listener and the wall behind him. Often a listening too close to this wall creates, because of the primary back reflections, serious balance problems mainly at the highest frequencies, starting from the low mediums up and up with the mediums and the high ones.

A distant listening position from the wall behind the listener is always recommended also to enjoy the advantages coming from the presence of a fairly good reverberated field (surrounding sensation) behind the listener.

We have tried to give a general idea about the possible way for the listener to operate in the presence of tonal balance problems shown by the A.Q.T.

Again we would like to underline that the best results can be obtained by checking each step of modifications with the A.Q.T. Though the A.Q.T. test is not able to evaluate the sophisticated differences among different components (electronics, cables), it will clearly reveal the effects caused by moving the speakers and/or the listening point within few centimeters.

Do not be even surprised if bettering the response within a determined frequency area by modifying for instance the position of the listening point, it could result in a worse response within a different frequency area: this is logical and normal.

The important thing is to obtain the "most advanced" balance of the graph, as a whole.

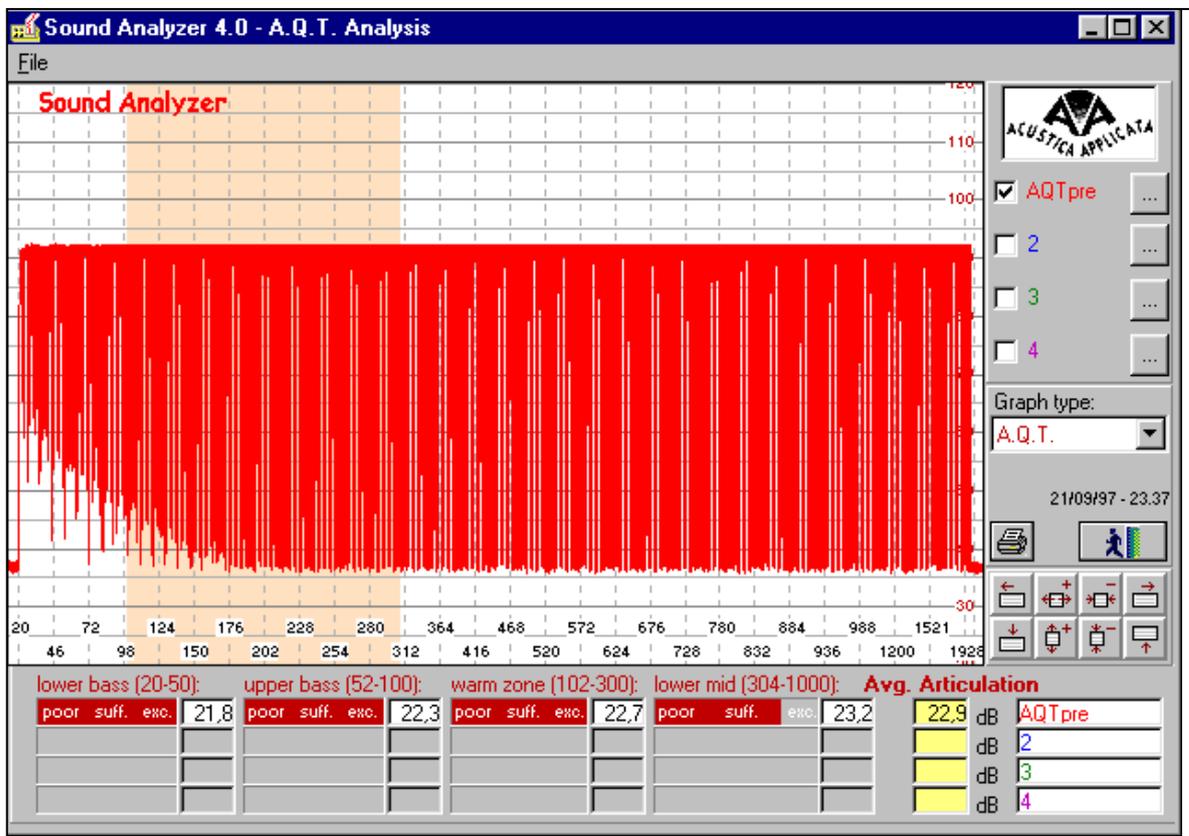
*The Audio Quality Test (A.Q.T.) is a test which "photographs" the behavior of the speakers-surrounding system: at low frequencies (from 20 to 50Hz and from 52 to 100 Hz), in the warm zone (102-300 Hz), at lower mid frequencies (from 304 to 1000 Hz).*



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To execute an A.Q.T. we need:

- a Sound Analyzer software as supplied by Acustica Applicata
- a Sound Level Meter positioned on the curve of ponderation "C", set up with the usual sound volume and in "fast" position.
- an A.Q.T. Sound Analyzer CD as supplied by Acustica Applicata.

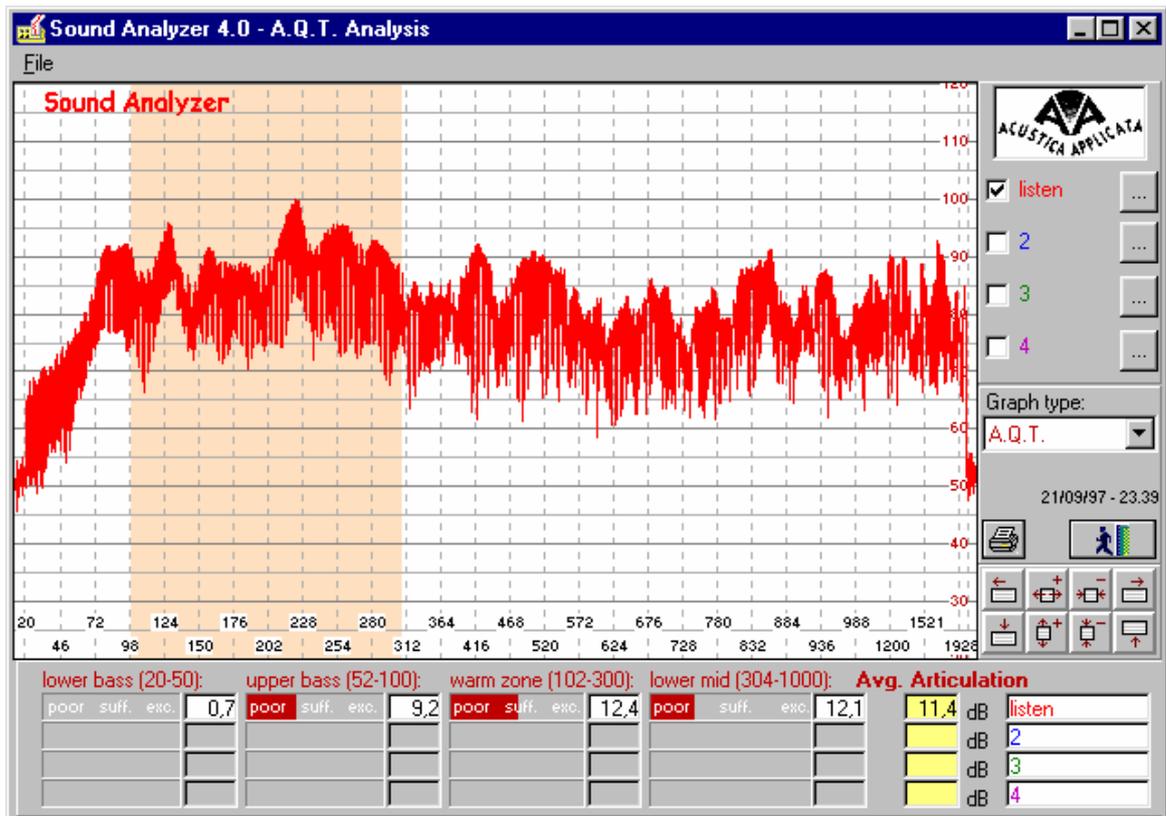


**Fig.8 - A.Q.T at the output of the preamplifier**

If you execute the A.Q.T. recording at the output of the preamplifier, you can observe a perfect response: linear, flat, absolutely free of peaks or depressions. (fig.8).



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**Fig.9 - A.Q.T. in the listening position**

If you execute the recording in the listening point of a room without acoustic treatment, you will instead obtain a graph with waves more or less steep and significant and you will see the different behaviour of the dynamics to the variation of the frequency (fig.9).



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### **EVALUATION OF THE MUSICAL ARTICULATION**

The Sound Analyzer Program automatically executes and shows the calculation of the average articulation. More it offers a qualitative evaluation of the articulation and the balance divided in the following four frequencies areas:

20/50 Hz = Ultra - Low Frequencies, 52/100 Hz = Low Frequencies, 102/300 Hz = Warm Zone, 304/2000 Hz = Lower Mid Frequencies.

The feature of the average articulation is definitely the most significant feature in order to evaluate the reproduction qualities of a speakers-listening room system, and in order to measure the improvements when an acoustic treatment is being set up.

Articulation means information, penetration of the sonic message, availability for dynamic variations. It is the contrary of the "dynamic monotony" (let us consider the typical tum-tum of the juke box).

We would like to add that the articulation carries two apparently contradictory components of the sonic message which are the music itself: the "punch", and the immersion into the musical abstract. The "punch" means dynamics, physical and visceral participation; the intellectual immersion into the musical abstract comes from the deep penetration of the sonic message which is allowed only if all of the music, not only a part, reaches our brain.

During the A.Q.T., a frequency coming out from the speakers, is followed by a brief period of silence. If it occurs that the room keeps the sound too long making it bounce in various directions, there will be a partial or total darkening of the frequency successively produced by the speakers. This way, part of the musical message is lost and the dynamic vibration is greatly softened.

A good listening room will allow the articulation, the relaxed and complex flowing of the music.

When reading the articulation graph, attention should given to the fact that you should observe high waves which rapidly rise and equally those in rapid descent.

Their descent means that, once the room has been excited by those frequencies, it offers a good degree of silence for the musical listening; if instead the graph waves have no wide excursion, this means that during the time of the persistence of an excessively high acoustic energy, one part of the new sound message has



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not been able to reach the listener, since it has been disguised by the long lasting acoustic memory of the surroundings.

The more we rise in frequency, the higher the level of musical articulation necessary. If 8dB of articulation is a good result for the frequencies below 100Hz, definitely they are not good for 300 or for 500 or for 800Hz. This explains the different evaluation scale for the four qualitative comparison windows of the Sound Analyzer Program.

Our experience indicates that, even in well treated rooms, it is practically impossible to obtain values superior to 10dB of dynamics under 50Hz. It is very difficult to obtain values superior to 12dB between 50 and 100Hz. Instead it is possible to obtain high values of dynamics above 100Hz, in an extremely important zone for the musical reproduction and that, rich in information, has the necessity of greater articulation.

The warm zone and the lower mid zone suffer a great deal from acoustic problems due to resonances and reflections.

Our casuistry shows, to who has used our investigation methods, that a huge part of the lamented sonic problems are attributed to the warm zone and to the first portion of the lower mid (from 100 to 500Hz). bettering the articulation and the tonal balance in this range of frequencies you can achieve a very notable sound improvement.

It is therefore in these ranges of frequencies we have to concentrate to obtain notable improvements and significant results.

In order to improve the articulation it is necessary to lower the reverb time (this operation is easier if we use broadband traps in the right number and positions).

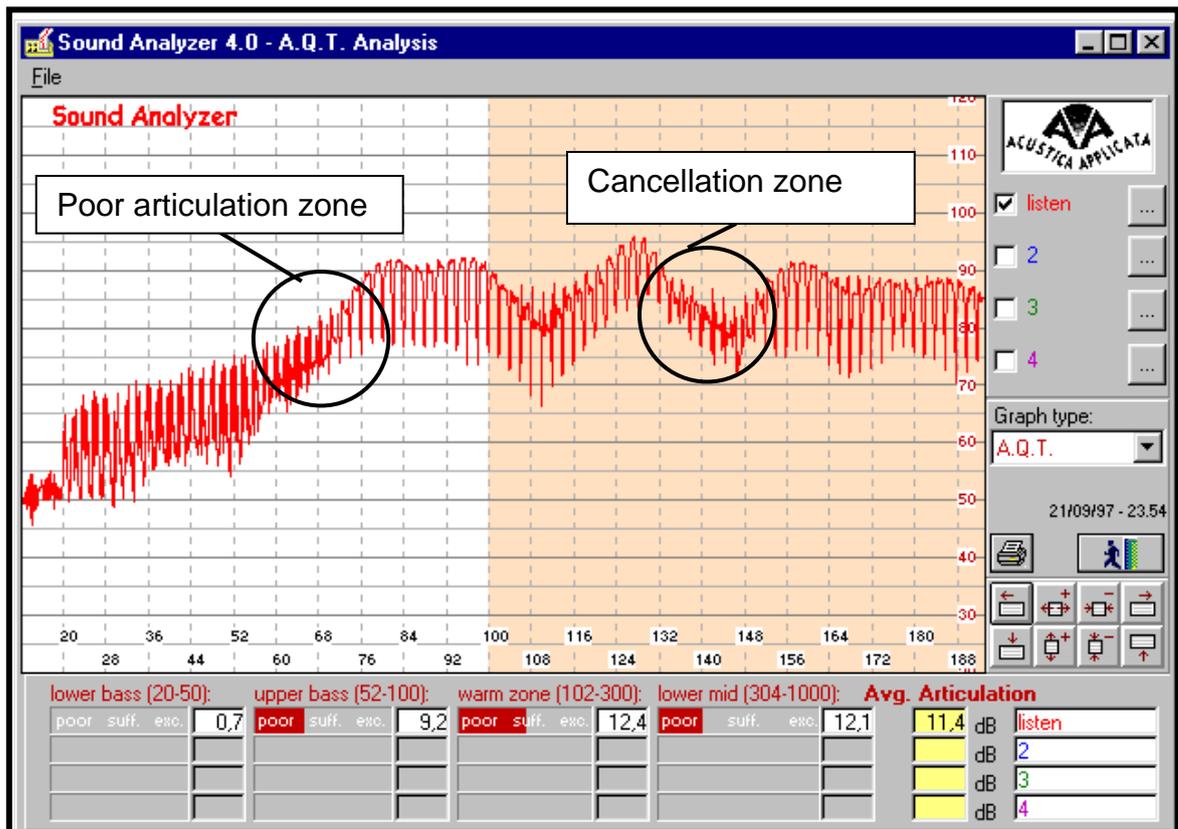
Within certain limits, the more numerous the acoustic traps are in a surrounding, the better the articulation value will be. But beyond these limits the articulation value keeps constant even raising the number of the acoustic traps. Their optimal number varies depending on the dimension of the room.

Using the A.Q.T. it is often possible to observe zones which particularly lack in articulation (the waves picture is indistinct, the articulation is measured in 2/4dB). These are called compression areas and they are originated by cancellation phenomena: too much reverberation and the algebraic sum of opposite frequencies. Acoustic treatment with broadband traps, brings back within acceptable limits the extension and the number of these areas.



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The important thing is to reduce the extension of defective areas and to make sure that the phenomenon involves few close frequencies.



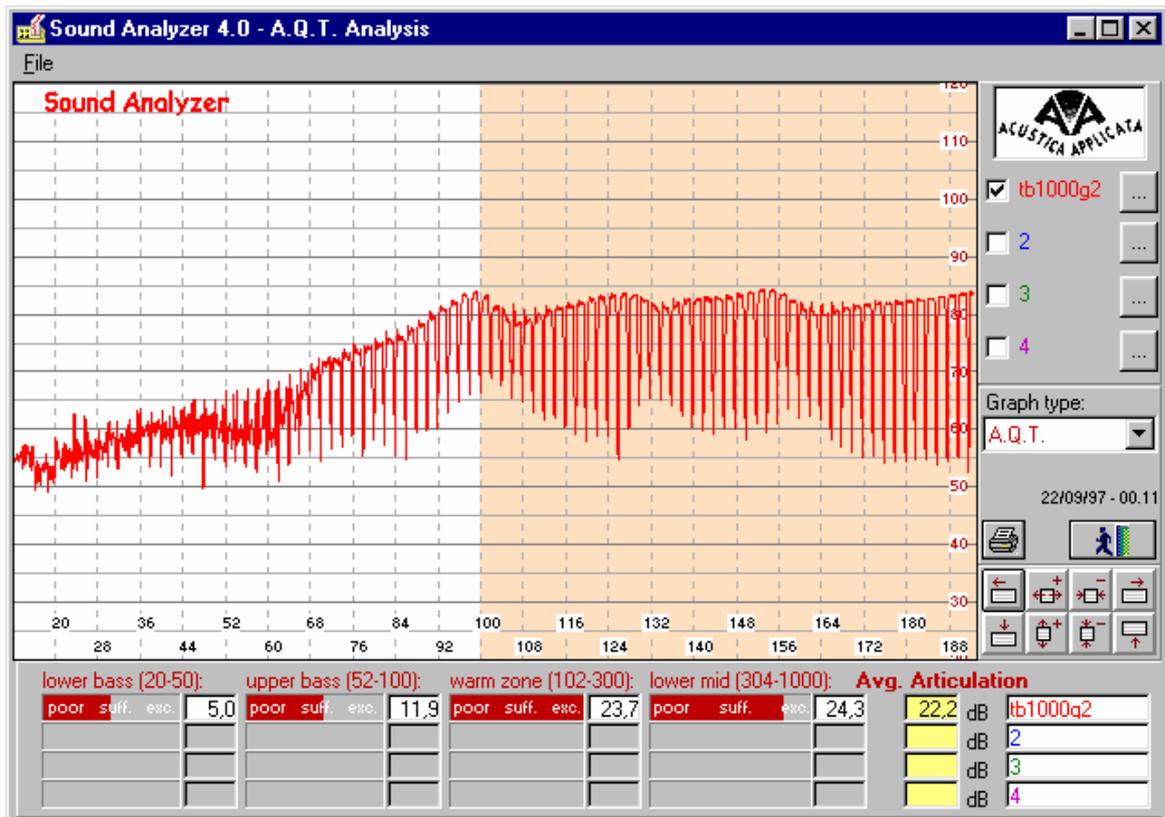
**Fig.10 - Initial part o.f an A.Q.T**

Many users of the A.Q.T. excessively linger over dynamic (articulation) compressions even when the graph offers a high degree of average articulation and the compressions points are limited a lot and they are not more than two or three. This is a mistake, because using the A.Q.T. we should never forget the general aspect: it is better to have a high level of average articulation with few



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compression points instead of a lower average articulation with no compression points.



**Fig.11- A.Q.T. after the acoustic treatment.**



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### **RELATIONSHIP BETWEEN TONAL BALANCE AND MUSICAL ARTICULATION**

The sharp distinction between tonal balance and musical articulation has been made mainly with teaching purposes.

In the daily practice you will see that if one improves, the other improves as well and vice versa.

We insist that:

- 1) the positioning of the speakers and the listening point especially affects the tonal balance.
- 2) the acoustic treatment significantly improves the musical articulation.

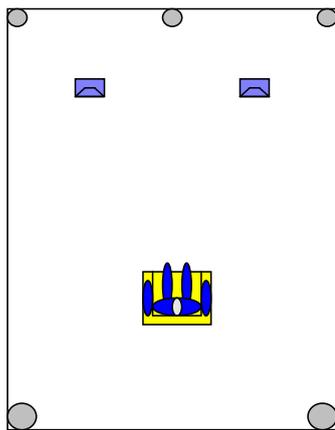
.....but if you put a certain number of broadband acoustic traps into a room you will see how the A.Q.T. graph, though maintaining constantly the value of the average musical articulation, will change depending on the position of the traps in the room itself; both the tonal balance and the position on the graph of the improvements of the musical articulation will change.

There are a number of rules for the positioning of the acoustic traps (see The treatment of the resonances with DAAD - pag.9) both for the treatment of the fundamental resonances and of the reflections.

In order to find the most efficient treatment typologies we send you to the relative part and we recommend that you execute A.Q.T. guidelines at every successive "step" of modification.



## PRACTICAL SOLUTIONS TO THE MOST FREQUENT PROBLEMS



**Fig. 12**

### **Confused and resonant basses, acoustic slime** (fig.12)

**Cause:**

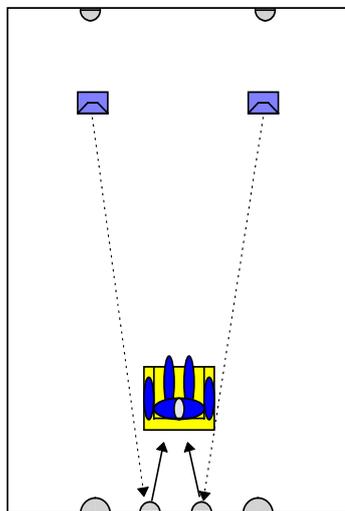
Addition of resonances in narrow frequency areas.  
Decaying times in respect to the frequencies of the warm zone are too long.

Speakers too close to the corners or equidistant from the lateral and back walls or too close to the back wall.

**Solution:**

Treat the resonances in the corners and where it is necessary.

Move the speakers away from the back wall or from the corners.



**Fig. 13**

### **Harsh sound** (fig.13)

**Cause:**

Excess of high frequencies caused by lack of low range or by very strong back reflections (especially when the listening point is a sofa against the wall) or by fluctuating echo.

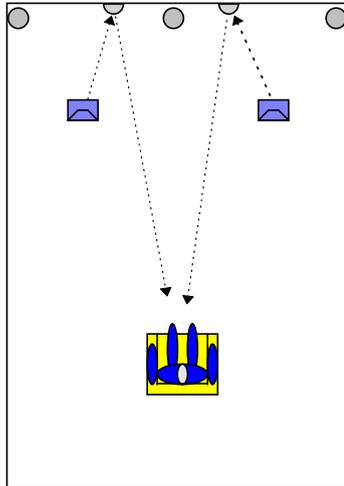
**Solution:**

Treat the back reflections and/or move the listening point away from the back wall; put an acoustic trap on the front and on the back wall exactly in their points of orthogonal projection.

Draw the speakers up to the back wall.



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**Fig. 14**

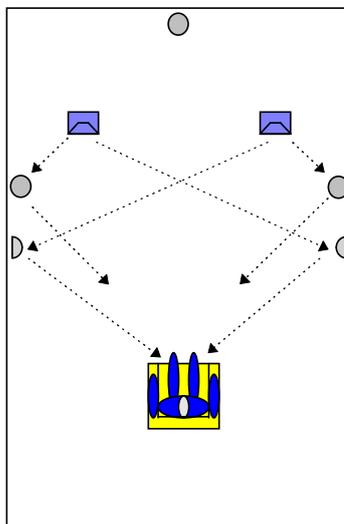
### **Squashed image** (fig.14)

**Cause:**

Strong front reflections.  
Listening point too far.

**Solution:**

Treatment of the front reflections.  
Draw the listening point in closer.



**Fig. 15**

### **Hole in the middle of the image, lack in focusing** (fig.15)

**Cause:**

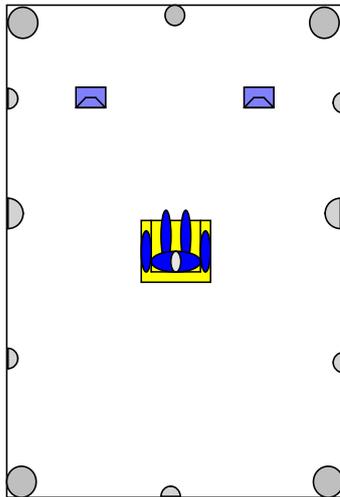
Too strong primary lateral reflections.  
Listening point too near.

**Solution:**

Treatment of the lateral homolateral reflection and/or  
the controlateral reflection.  
Move the listening point further away.



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**Fig. 16**

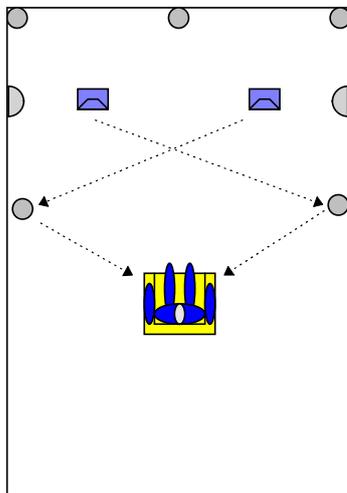
**Room which saturates at low volumes, lack in dynamics**  
(fig.16)

**Cause:**

Excess of reverberation: in the room too much resonant energy, which has not the time to fade, and so persists.

**Solution:**

Acoustic treatment to obtain shorter decaying times, that is to say faster, keeping the acoustics of the room alive.



**Fig. 17**

**Narrow image**  
(fig.17)

**Cause:**

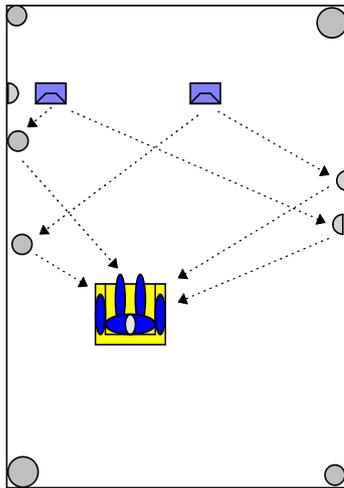
Lateral and back reflections being too strong.  
Lack in treatment in the front corners.

**Solution:**

Treatment of the early reflections and of the front corners.



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**Fig. 18**

### **Unbalanced image**

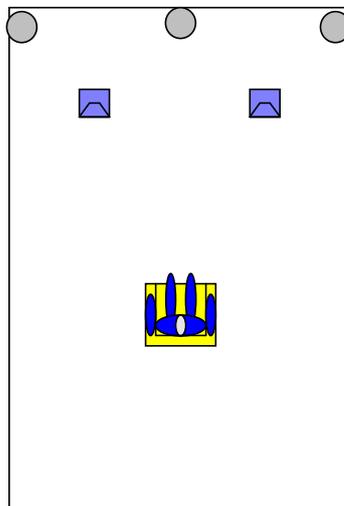
(fig.18)

#### **Cause:**

Bad positioning of the system.  
Bad acoustic treatment.

#### **Solution:**

Place the speakers in a way that the resonant front is homogeneous.  
Rebalance the differences with an appropriate acoustic treatment different for the lateral reflections, trying to re-establish an acoustic symmetry.



**Fig. 19**

### **Unbalanced image**

(fig.19)

#### **Cause:**

Bad positioning of the system.  
Bad acoustic treatment.

#### **Solution:**

Place the speakers in a way that the resonant front is homogeneous.  
Rebalance the differences with an appropriate acoustic treatment different for the lateral reflections, trying to re-establish an acoustic symmetry.



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### **Small losses of tonal balance (sound too brilliant or too dull) when DAAD acoustic treatment is being used**

**Cause:**

The setting of the acoustic treatment system is not accurate.

**Solution:**

Go on with the delicate setting through small rotations of DAAD.

Do not operate on all DAAD together, but consider everytime the symmetric columns, those with equal function (the two columns for the lateral homolateral reflections or those for the front reflections).

It is convenient to start the setting of the front corner columns, then the front wall columns, then the lateral walls columns and in the end the columns behind the listener.

### **Poor, dry, or oversharpe sound when a DAAD acoustic treatment is being used**

**Cause:**

Bad DAAD positioning.

**Solution:**

Place DAAD in the correct way and not at random.

### **Uselessness of DAAD**

**Cause:**

Problems with the electronics or with the speakers.

Bad recordings.

Wrong evaluation of DAAD in use.

Bad positioning of the speakers or of the acoustic treatment.

Scarce number of traps in use in respect to the surrounding acoustic problems.



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### **Solution:**

Check both the electronics and the speakers. Use good recordings (if for instance a recording has bad basses, it is impossible to think that DAAD can make them better; they can only give them back as they are).

Use DAAD with the right diameter and position them in a correct way.

Increase the number of DAAD.

We have so provided a schematic list of some of the most frequent problems one can find, with their most evident solutions.

It is obvious that we have considered only the acoustic defects, leaving the causes due to electronics (for instance a rough sound is frequently caused by a bad interfacing between the electronics, by speaker defects, or by the cables quality, etc.).

We cannot expect, from an acoustic treatment, to correct defects of other natures. The purpose of the treatment is to provide the listener with a sound identical to the original sound first emitted from the speakers, combating the complicated problems caused by the surroundings.

What happens before is obviously not related with the problems of the acoustic treatment which sometimes can compensate certain defects.

But when an acoustic treatment is able to correct the complicated problems we were dealing with in the introduction and it does it in such an efficient way as to show either the merits and/or the defects, and the true sonic nature of the electronics and the speakers, this means that we have well accomplished our task and we have made a big step ahead towards correct sound reproduction.

The operative rules you have read, are not many and they are surely incomplete. As a good justification for these limits, one should not forget that for the first time an attempt (successful according to us) to conciliate graphs and physical rules with concrete listening experiences, has been made; therefore we have proposed a practical guide concerning operative suggestions not quite bizarre or very questionable, but on the contrary, scientifically valid and also audibly repeatable and verifiable.



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### **THE QUESTIONS AND THE ANSWERS OF A METHOD - THE ACOUSTIC ANALYSIS WITH THE A.Q.T. SIGNAL**

It is always difficult to define “how” good a sound is. In the HI-FI world, superlatives are wasted and parameters like definition, airy sensation, naturalness, etc, are discussed; viceversa in the professional world, sound engineers, architects, sound technicians talk about reverb time, absorption coefficients, signal-noise ratio etc, to define the characteristics of a sound reproduced in a surrounding. It is pointless to deny that these two worlds have always been in opposition, sometimes vivid and polemic.

Our activity in the environmental acoustics field, has trained us to listen and try to solve the needs of both these worlds, paradoxically in contrast. Experience has taught us that most of the time the linguistic terms used to describe the problems, are responsible for adding to this conflict.

Example: A sound engineer who requests and looks for “punch” sitting at the desk of a recording studio is for us no different than an audiophile who wants ‘explosive’ dynamics from his system. An architect who wants better speech intelligibility in a congress room, rather than in a class room, is for us the HI-FI lover who talks about definition, selectivity or microcontrast of his system. An acoustic engineer who requests a reduction of the background uproar and noise of any public room, is for us the same as a private person who laments ‘sound difficulty’ in his HI-FI room, caused by uncontrolled reflections and reverberations.

In every one of our acoustic treatment interventions, we act parallelly with instrumental measurements and natural ear-listening. In our opinion these two factors are indispensable and complementary. With the A.Q.T. signal and the Sound Analyzer we have endeavoured to obtain a measuring method which gives us a rigorous and comparable mathematic result, but which shows precise acoustic behaviour in a room, in aspects usually considered important and exclusively for HI-FI.

The aim of the A.Q.T. analysis is to acoustically ‘photograph’ a room from the listening point and to obtain a synthesis of indications which are fundamental for us, not just for starting an acoustic treatment but above all to be able to follow its evolution. Even with minimum movement of the speakers and the listening point,



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or with the gradual emission in the room of sound absorbing material, graphic changes are easily noticeable and it is also easy to understand whether the situation is getting better or worse.

**An A.Q.T. graphic shows four parameters which for us are fundamental to improve the acoustics of a room:**

- 1. Dynamic response of the surrounding (articulation)**
- 2. Frequency response (tonal balance)**
- 3. Reverb time due to the frequency**
- 4. Phase and sound acoustic pressure coherence at the listening point (focusing).**

Let's remember that the A.Q.T. analysis investigates a physical phenomenon not electrical; it is up to us to measure and set up a graphic that effectively reaches our ears, after we have crossed the complete reproduction chain, including the last and most influential ring: the surrounding.

This consideration may seem banal but to realize how heavily the room influences the overall sound of your system, try to record the A.Q.T. signal only a few centimeters away from the speaker and compare the graphic with that taken from the listening point. If then you want to take advantage of the versatility of this measuring method, you can also compare the graphic between a recording taken at the CD output with that taken a few centimeters away from the speaker; if you notice a difference, this means that a portion of your sound has been lost along the way, but this problem is not up to us.



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### **HOW IS THE A.Q.T. SIGNAL**

The A.Q.T. (Audio Quality Test) is a particular kind of signal called “multiburst”. It is a sequence of pure, sinusoidal tones, which starts from 20 Hz up to 300 Hz with step of 2 Hz, from 300 Hz to 1000 Hz with step of 4 Hz, from 1000 Hz to 2000 Hz with a logarithmic raise.

Every tone takes 3/15 of a second (200 ms) and is separated from the successive by 1/15 (66ms) of silence for a total duration of 94,13 seconds.

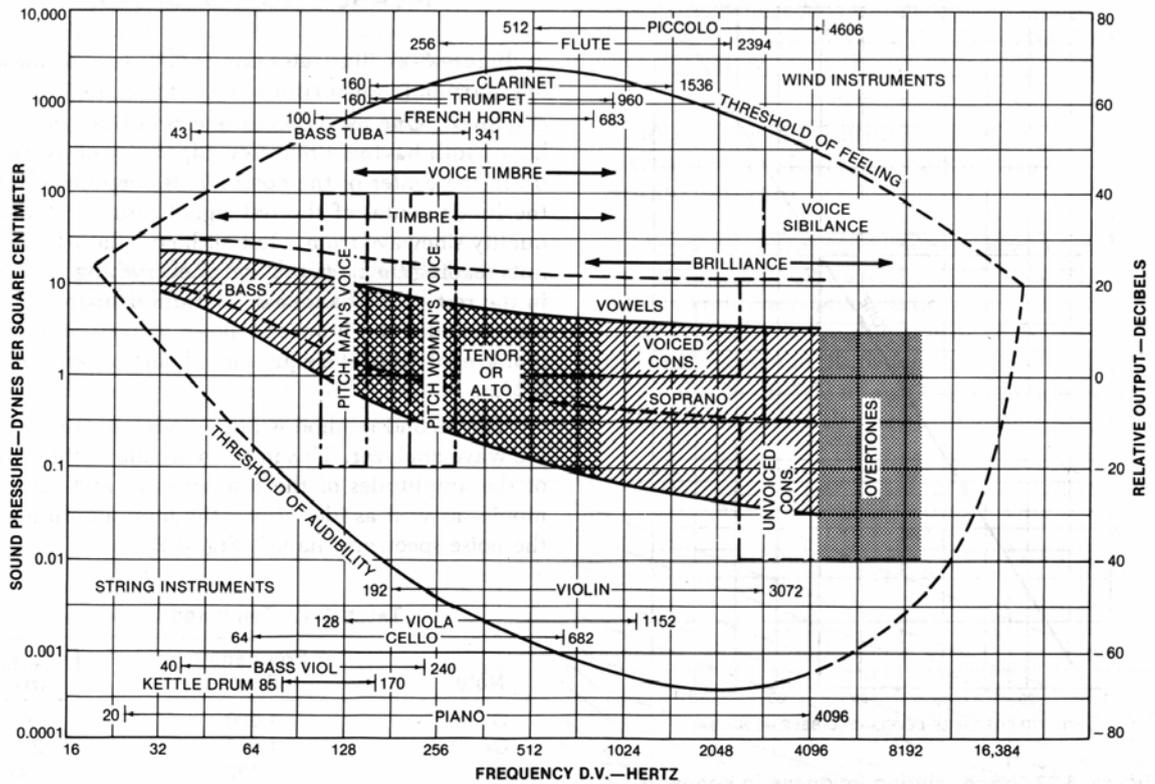
Naturally, the structure of the signal is not casual, the A.Q.T. must be the closest possible to a musical signal and like this, it has frequencies that change with time, intervalled with small spaces of silence.

In 3/15 of a second of tone, we have the possibility to investigate the sound of the surrounding, in 1/15 of a second of silence, we verify what we find in the “reverberated” field, such as to say all that portion of sound which comes to our ears not directly from the speakers, but from the walls of the room where we listen.

We can now describe, synthetically, the laws and phenomena of physics and psychoacoustics on which we base our A.Q.T. method.



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**Fig.20 - Range of the dynamics and of the frequencies for audio signals**

## ***Acoustic power and sound-pressure level***

In order to understand the essential difference between acoustic power and sound-pressure level, we will use as an example a phenomenon to which we are most familiar: heat.

If we want to warm a room up, we install a heater, but in order to quantify its power we will not say that it is able to produce a temperature of 20°, since this result will depend on the surrounding characteristics (dimensions, number of windows, thermic loss). It is correct to say that the heater has a power of 2 KW, that is to say



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it supplies a certain amount of heat in the unit of time. Similarly we will say that the fittest parameter to characterize a source is the sonic "power", while the sonic "pressure" evaluates the effect of the presence of a sound in the point we want to investigate.

Sonic power is defined by the expression :

$$G = 10 \text{Log}_{10} \frac{W_u}{W_i}$$

where  $W_u$  and  $W_i$  is the output power and the input power (taken as a reference) expressed in watt, of an amplifier.

If for example the gain of an amplifier is equal to 20dB, this means that the output power is equal to 100 times the input power, just like it is easily shown by the above expression.

We want to remind you that the dB is not a unit of measure but a quantity without physical dimensions, since it is always referred to as a relation between two values of the same quantity, where one of them is taken as a reference.

The measurement unity of the "acoustic pressure" is the Pascal (Pa)

$$1 \text{ Pa} = 1 \text{ Newton /m}^2 \quad 1 \text{ Newton} = 102\text{gr}$$

It has been measured that the human ear has a minimum audible level of 20µPa (20 microPascal = 0.00002 Pa) equal to 0dB, to arrive till 100Pa equal to 134dB which is the established pain threshold.

Therefore using a linear scale like the Pascal, in acoustic measures we should use numbers with 6 figures, entire or decimal, very difficult to use and to memorize.

If we also consider that the ear answer in a logarithmic way, we have decided to define the acoustic measures like the logarithm of the relation between the measured value and the reference value. The so expressed measures are called "levels".



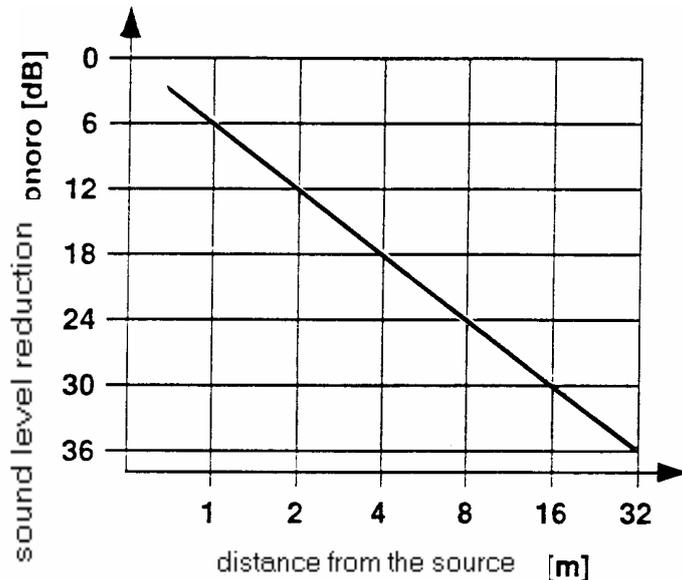
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Therefore the sonic "pressure" level, expressed through the dB logarithmic scale, is defined like:

$$\text{dBLps} = 20\text{Log}_{10} \frac{P}{P_0}$$

where P is the value measured in Pascal and P<sub>0</sub> is the reference level equal to 20µPa (inferior hearing threshold).

In the dB logarithmic scale it is interesting to observe that by doubling the sonic "pressure" we will have a rise of only 6dB. Let us pay attention therefore to evaluations about dB measures, since for each rise or reduction of 6dB the sonic "pressure" level will double or halve; instead in the case of the sonic "power", it doubles or halves with only 3dB (like it is easily calculated using the two expression above mentioned).



**Fig. 21**

### **Direct sonic field and reverberated field**

In a free field, moving away from a sonic omnidirectional source, the sonic pressure level halves everytime the distance from the source doubles (fig.21).

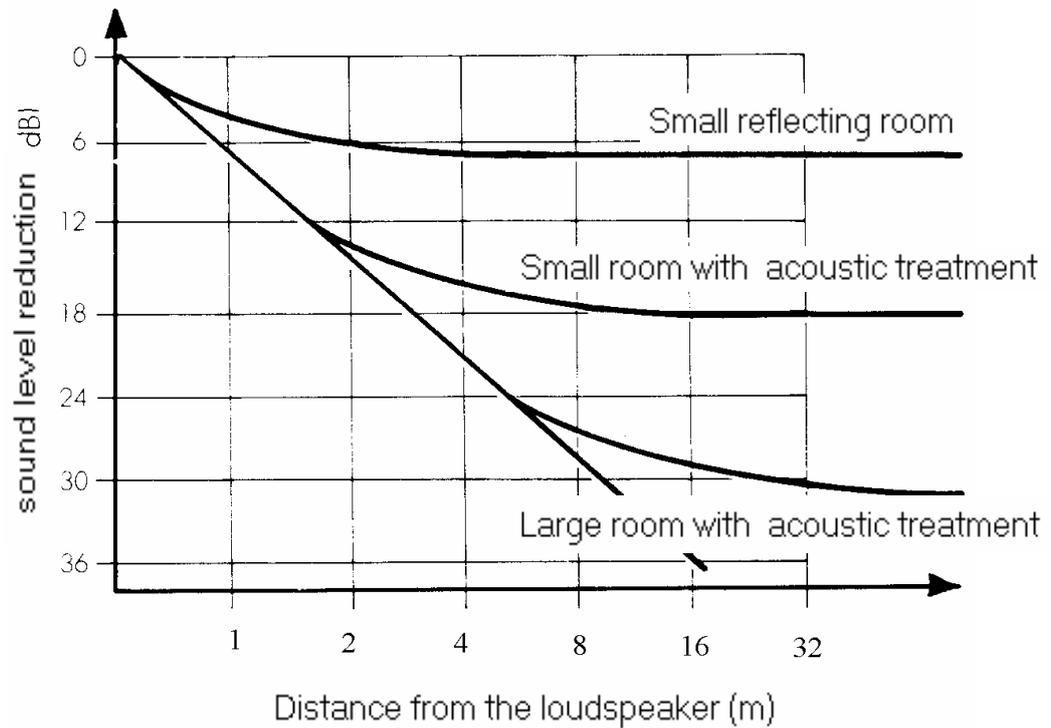
Instead in a closed room, the sound which reaches the listener is composed by direct waves coming from the source (speakers) and reflected waves which reach the listener after suffering single or multiple reflections on the walls of the

room where we are.

Our sonic pressure level will have a course as similar as the free field course (direct field), and another one which, because of reflections, will stabilize as an independent value from the distance between us and the source (reverberated field) (fig.22)



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**Fig.22 - Reverberated field in a room**



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### **The ear as an instrument of integration Precedence Effect - Haas Effect**

The ear works like a real and proper ballistic measuring instrument, which means it receives and integrates the sound intensity for brief intervals. This behaviour is similar to that of the eye which intakes a succession of single photograms, giving us the impression of continuous and fluid movement. At least 16 photograms a second (1 photogram every 62 seconds) are necessary so as not to have disturbed vision. The ear collects sounds which arrive within 25 milliseconds after direct sound, and integrates them giving us the impression that this is the maximum level and that these emanate from the same direction as the original source, even if in reality they are a group of reflections off the walls which reach us from different angles.

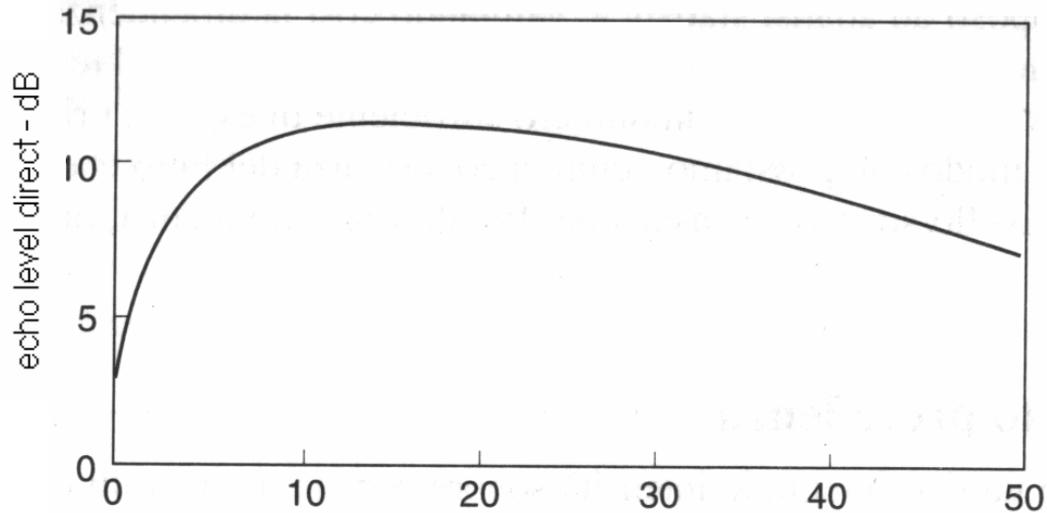
The sound fusion is guaranteed within the first 20/25 ms., after 50/80 ms., separate reflections begin to dominate, after more than 80/100 ms., real and distinct echoes become apparent.

The journey from a time zone to another is obviously very gradual and depends on the dB level and on the frequency, therefore it is not possible to stipulate a fixed border but a transition area which goes from the direct sound to the space sensation to the echo.

The Haas studies show how, within the delay between 5/25 ms., a sound must be more than 10 dB, superior to the main one in order to be perceived as a distinct echo (ref. fig. 23).



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Echo delay - ms

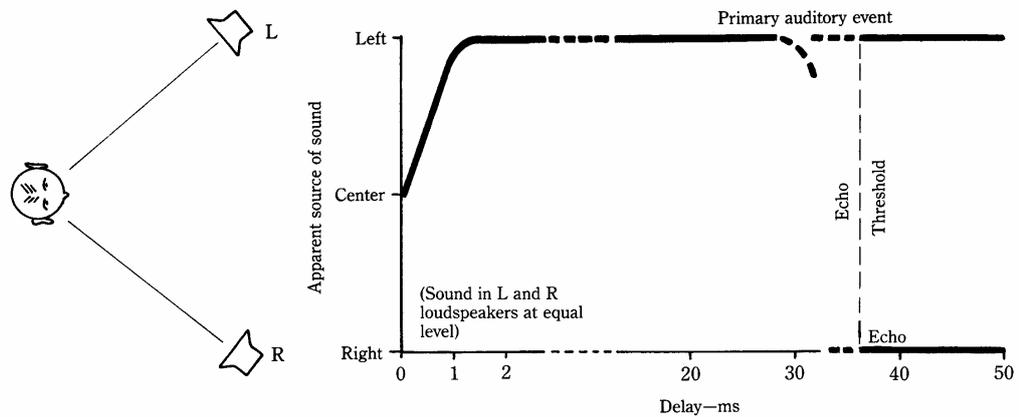
**Figure 23 - Precedence Effect (Haas Effect) in the human auditory apparatus. In the interval included between 5 and 35 ms., the echo level must be superior to the direct sound by at least 10 dB, to the point that the echo can be heard as it is. Infact, in this interval the reflected components which come from different directions, become fused by the ear in such a way that the resulting sound seems stronger and coming from the direct source. For delays of 50-100 ms. and more, reflections are instead perceived as a distinct echo.**

Another important law, for listening in close rooms, is that of the “First Wave Front”. Called this by **Cremer**, the very studious man who was the first in investigating this aspect of the surrounding acoustics, this law of physics shows how our ear-brain system is capable of establishing the exact sound direction also in listening conditions with a high level of reverberation.



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Within a delay of 20/25 ms., the ear, in the very first instants, (approx. 1,2 ms.), will determine immediately the whereabouts of the sound source, without being influenced by the multitude of reflections which will arrive immediately after from every direction. (ref. Fig. 24).



**Figure 24 - Law of the “First Wave Front”.** Up to 1,2 ms. there will be image movement, more than and up to approx. 25 ms., the image will be perceived coinciding with the direction of the signal which first arrives (LEFT in this case), while the level will be integrated. More than 35 ms., distinct echoes will be perceived.

The graphic shows the setup used to establish this law: a stereo system with a 45°- 60° angle in an anechoic room, a delayed signal, from 0 to 1,2 ms. compared with the second one, is sent to one of the two speakers (in this case the one on the right), to verify the image movement.

The two speakers emit sounds at the same level.



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Beyond the threshold of 20/25 ms., we will have a sensation of distinct echo, if the dB level of the reflections is sufficiently high; we will have ambience sensation, if the level of the reflections is lower.

Let's make a practical example:

Let's suppose we have our listening room with speakers at 50 cm. from the side wall. Especially if this is a discreetly reflecting surface (the common painted wall reflects approx. 96% of the acoustic energy), at this distance the listener will receive from the wall the reflected sound after 1,45 ms. with a dB level very slightly inferior to the direct sound.

Remembering the law of the First Wave Front, you can determine that below a distance of approx. 50 cm. from the side walls, these became real and proper supplementary acoustic speakers, being able to move the 'stereophonic image' decisively, or worse, in a different way depending on the frequency, since the absorption coefficient of the material which the wall is made of, is not constant at all frequencies.

Since sound speed is 344 m/sec. in 10 ms., the sound wave runs 3,44 mt. and in 25 ms. 8,6 mt., therefore in common dimension rooms, all primary reflections intervene in the interval where our ear-brain system is more capable of sound 'fusion'. (1ms. delay = 0,344 mt; 1cm.= 0,029 ms.; 50 cm. = 1,45 ms.).

Then you will understand that from a practical point of view, if we want to improve the listening of our room, it is necessary to move the speakers away from the walls (side and rear walls), as much as possible, and to treat the primary reflections with priority.

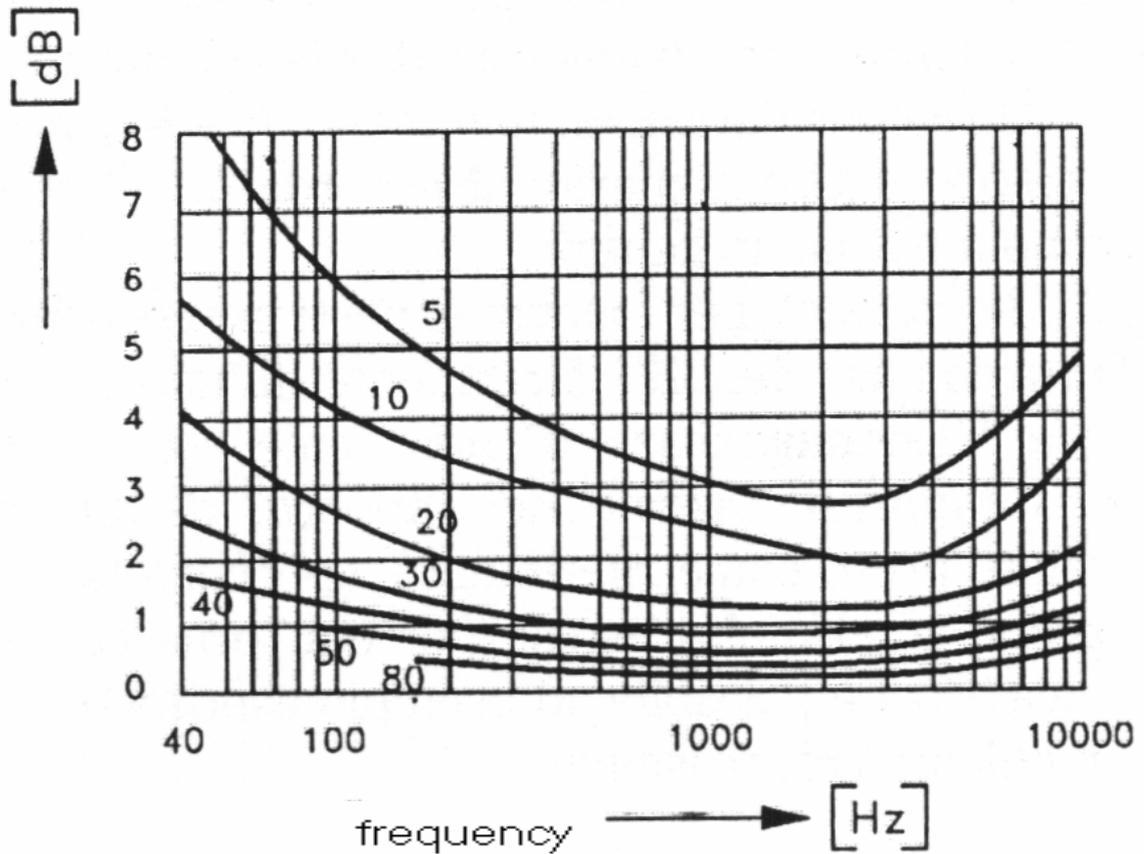
### **How the human ear perceives level differences. Critical bands**

The human ear is only able to distinguish acoustic level differences in dB, if these exceed a threshold which depends on the frequency and on the level itself. Sensibility is higher at mid - frequencies (600-3000 Hz), where the ear distinguishes 0,25 dB differences for high sound level, and 2-3 dB for lower level



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(ref. Fig. 25). Generally, sensibility is emphasized at high sound level and it decreases with lower sound level.



**Figure 25**



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The ear's behaviour is not the same with pure signals or with more complicated signals which have a certain bandwidth in Hz (for example: a single or prolonged organ note or symphonic music). The wider the occupied frequency band is, the higher the sound sensation given.

We have succeeded in defining the critical bands concept and according to the Moore and Glasberg's studies, these are:

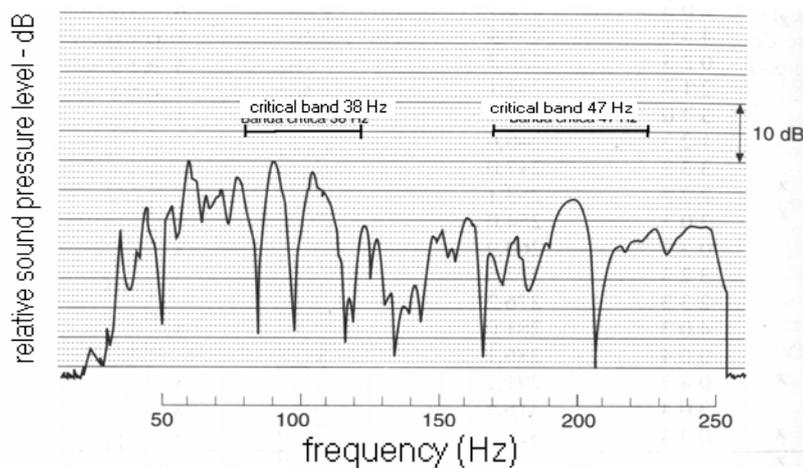
Central frequency (Hz)	Critical band width
100	38
200	47
500	77
1000	128
2000	240
5000	650

This ear's behaviour for 'critical bands', prevents us from perceiving imperfections in the frequency response for musical messages with a complex spectrum, unless these are sufficiently wide to 'exit' the 'critical band'.

With musical messages with complex and large spectrums, the ear is able to attenuate the imperfections of the surrounding frequency response. (The most complex audio compression algorithms, like ATRAC and M-PEG, are based on this concept). These imperfections are caused by the reflections and the standing waves and generate in the surrounding, what is defined as the so-called 'comb-filtering', a sort of 'rake' which heavily removes entire portions of sound from our listening.



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**Fig. 26 - Test with a swept sinusoidal wave**

### **Comb - Filtering**

Comb - Filtering is a phenomenon which manifests itself in the listening surrounding caused by interactions (constructive and destructive) between the main signal (direct sound) and a delayed copy of it (primary reflections from 0 to 25 ms.).

Let's make an example:

We have the main sound 'A' and a delayed sound 'B' of 1 ms.; the result between 'A' and 'B' is a sound 'C' with an outline which presents blanking zones (0 dB) and peak zones (+ 6 dB).

As a matter of fact, since B is a reflection on a wall which has absorption coefficient, the sound B reaches the listener not only late but also slightly reduced in level and therefore we don't have real and proper 'blankings' (holes at 0 dB) but consistent sound attenuations or exaltations.



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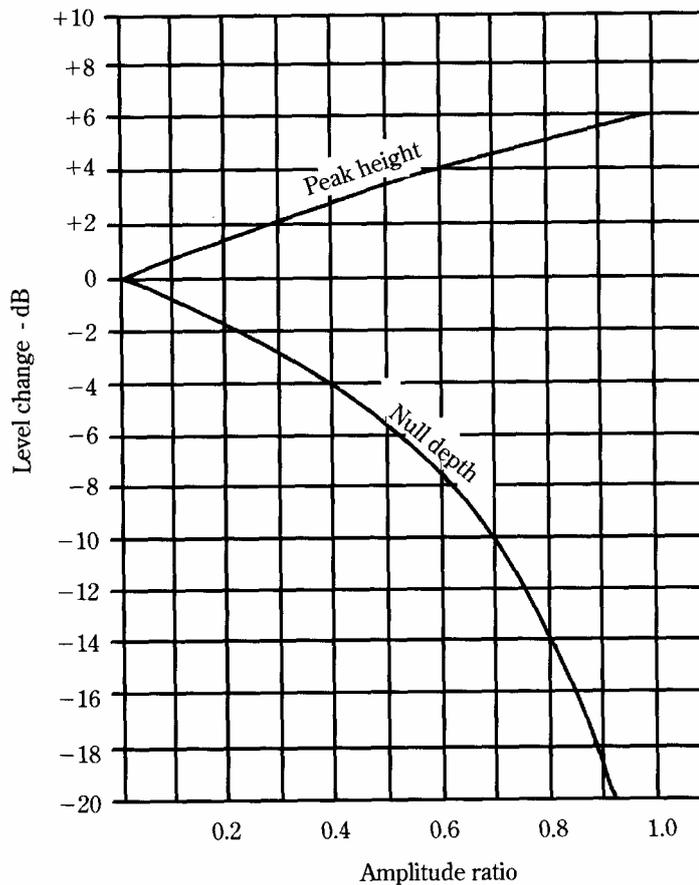
As we will see further ahead, common domestic walls (painted walls) have a very low absorption coefficient (0,95-0,9) which gives a dB level drop of the reflected sound on it approx. of - 0,9/0,45 dB. The frequency positioning of these 'peaks' and 'valleys' depends on the T delay of the reflected sound and follows this mathematic outline:

ms. Delay	Frequency of the First 0	Peaks distance
0,1	5000Hz	10000Hz
0,5	1000	2000
1	500	1000
10	50	100
100	5	10

The extent of the alterations depends on the attenuation of the signal B reflected to A and can be estimated according to the following graphic.



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**Fig. 27**

**The effect of amplitude ratios on comb-filter peak height and null depth**

With the signal B at - 3 dB (0,7 of amplitude) we have: approx. + 4 dB at the peak and -10 dB at the valley; with the signal B at - 20 dB alterations are minimum: + 0,3 dB at the peak and approx. -1,2 dB at the valley.

We conclude that in the time domain, short delays generate very distanced peaks and valleys, though when viceversa they generate very close alterations.



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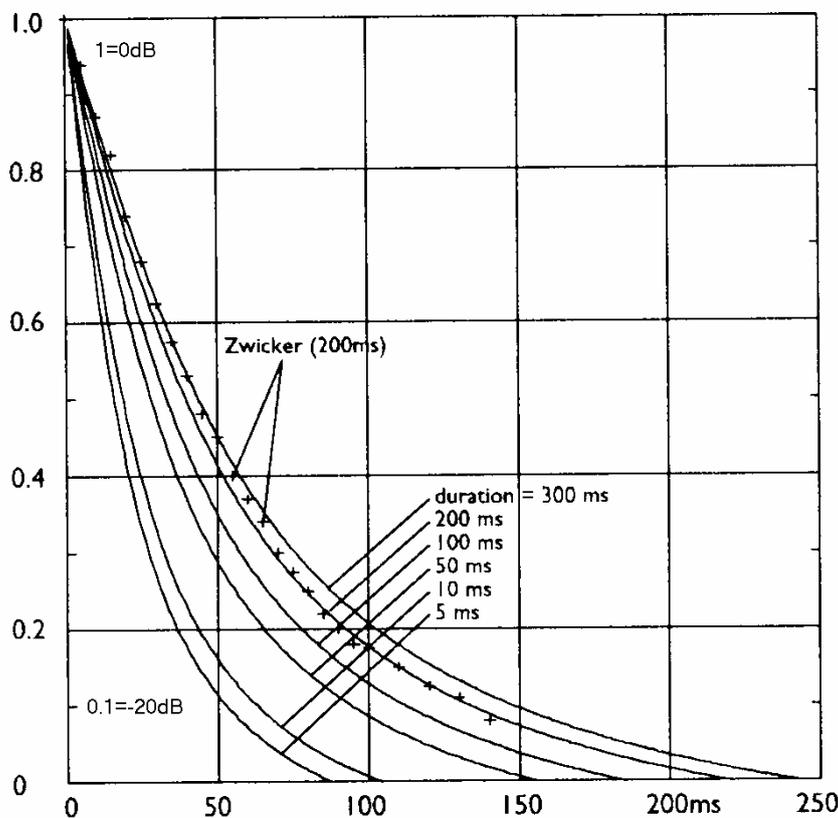
In the energy domain in dB, slightly attenuated signals generate deep alterations, while heavily attenuated signals, generate slight effects. We confirm again the necessity of primary reflection checking (walls, floor and ceiling).

Pay attention to the fact that, if the ear attenuates to a certain measure the surrounding defects, in the event of musical signals with a complex spectrum, it can singles out well the problems with pure notes (for example organ or piano notes), when these, by chance, slide to the bottom of a blanking 'valley'. The resulting sound is heavily falsified because the difference is of many dB.

### **Masking the defects of the surrounding in time**

The surrounding masking effect (also defined post-masking), is very interesting, above all for our type of acoustic research. The objective is to establish, to the point the ear is capable of perceiving the surrounding defects (primary reflections-standing waves).

**Fig. 28**



When a sound reaches the ear for the time-span 'd' of seconds, the famous integration ("fusion") of sound occurs. In order to re-establish full sensibility and to accept a successive sound message, a certain time, which depends on the time-span of sound event 'd', is necessary. Therefore, sound with a frequency component sounds very distant from



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the fundamentals of the preceding sound of time-span 'd' , there is no frequency masking and the sound is intelligible.

For sounds with frequencies close to those of the preceding event (sounds with fundamentals which slide inside the critical band signal of time-span 'd'), we have a primal masking phase, to the point that the original 'd' sound persists, and a consequent abandon of the masking with the progressive re-establishment of the ear full sensibility (Ref.Fig.28).

### **Consequences of the realization of the A.Q.T. signal - Why 20 dB.**

Our A.Q.T. signal, which is made up by frequency tones very close to each other (2-4 Hz), is capable of showing the masking phenomenon of surrounding defects. Studies carried out show that 60 ms. are necessary for a sound spanning = 5 ms. and approx. 135 ms. for a sound spanning = 200 ms.; this is the necessary time because the ear regains 20 dB. of sensibility (Ref.Fig.28). Therefore for sounds of a brief time-span (5 ms.), the ear masks the sounds, adjacent to the main one, of 20 dB. for approx. 60 ms., for more prolonged sounds the 'masking' lasts longer (135 ms. to have 20 bd. with " d" = 200 ms.).

If our surrounding allows, in 66 ms. = 1/15 sec. (silence space between one tone and another in the A.Q.T. signal), a sound level decline of 20 dB., this means that the ear is left free to perceive perfectly the successive tone. If in the same time the decline is superior (30-40 dB.), it is even better, but the ear would not be able anyway to re-adjust its sensibility with the same speed. This is why 20 dB. of decline, which we measure with the Sound Analyzer, are the result to aim for in the surrounding acoustic treatment.

Being the 'critical bands' of at least 25 Hz, with the A.Q.T. signal we are in the position to verify the masking of every tone on that which precedes; it is therefore a very severe test for our speaker-surrounding system and above all for our ear. If an articulation, which is the same or superior to 20 dB. (20 dB. in 1/15 sec. = 66 ms.) results from the analysis, we are sure that the surrounding is 'clean' and allows the passage of everything the ear can perceive. If the articulation is inferior to 20 dB., this means that tails and reflections which 'dirty' irrecoverably the sound, are persistent in the surrounding.



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### ***Evaluation of the surrounding reflections***

We know that the sound travels at 344 m/s, therefore:

1 ms = 0,344 mt.  
5 ms. = 1,72 mt.  
10 ms. = 3,44 mt.  
etc.

In common domestic rooms, the direct sound arrives after approx. 5-10 ms. from the emission (distance from the speakers = 1,72 mt. --> 3,44 mt.), the primary reflections of the walls after very few other milliseconds. If the speaker is at 34 cm. from the side wall, the reflection arrives after 1 ms., if it is at 100 cm. after 2,9 ms. and so on. Therefore we can affirm that all the primary reflections, (considering also the other room walls) reach our ear between 0,5 ms. and 25 ms. Due to “Comb-filtering”, the spacing of their effects on the spectrum of the signal which reaches our ear (see the table ‘Comb-Filtering’, page. 54) is very ample at 0,5 ms. 200 Hz. and very thick at 25 ms. 40 Hz.

### ***The level of the reflections***

The sound that reaches the ear after one or more reflections, is attenuated for both the distance run, and the characteristics of the wall reflection/absorption. While in a free space, the law of the inverse of the distance applies (these means doubling the distance we have an attenuation of 6 dB.), for the level of the reflected signal we can, in a first approximation, apply the following formula:

$$\text{reflection delay} = \frac{(\text{reflected path length}) - (\text{direct path length})}{\text{sound speed}}$$



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Presupposing a reflection of a surface of 100%:

reflection level at the listening point =  $20 \log \frac{\text{direct path length}}{\text{reflected path length}}$

Obviously, the walls will never have a reflection coefficient of 100% (coefficient 1 → 0 dB. of absorption - coefficient 0,5 → -6 dB. of absorption - coefficient 0,1 → -20 dB. of absorption). How we originally find them (plastered, painted etc.), they have the following characteristics:

Plaster wall: absorption coefficient 0,65 - 0,75 → reflection attenuation -2,5/-3,7 dB.

Painted wall: absorption coefficient 0,95 - 0,9 → reflection attenuation -0,9/-0,45 dB.

Therefore as you can see, the walls, as they are, slightly contribute to attenuate the level of the primary reflections; you can estimate that in rooms without acoustic treatment the level reduction goes from 10 dB. to -10 dB. So in light of all the considerations done, we can conclude that the primary reflections, together with the fundamental resonance are the principal responsible for the sound quality of a speaker- surrounding system.

In synthesis the primary reflection cause:

1. an energetic contribution of the perceived sound level; (due to the law of the First Front Wave). Their levels are integrated by our brain together with the direct sound and the sound level sensation is so determined.
2. Intertwisted with the surrounding fundamental resonances, they create comb-filtering effects both in frequency and in dB level, with heavy coloring of the spectrum and modification of the pattern (tonal balance - articulation).
3. Creation of changes on the stereo image (focusing).
4. Creation or cancellation of ambience effects, perception of the physical dimensions of the room where the recording has been carried out. For this reason we consider important to underline that the primary reflections (more than 2 ms. and not more than 35 ms.) do **not** give ambience effects to the listening but they cause pattern alterations and/ or image movement; therefore the ambience content of the room, where the recording has been carried out, must be present in the recording itself and it could never be perceived in rooms

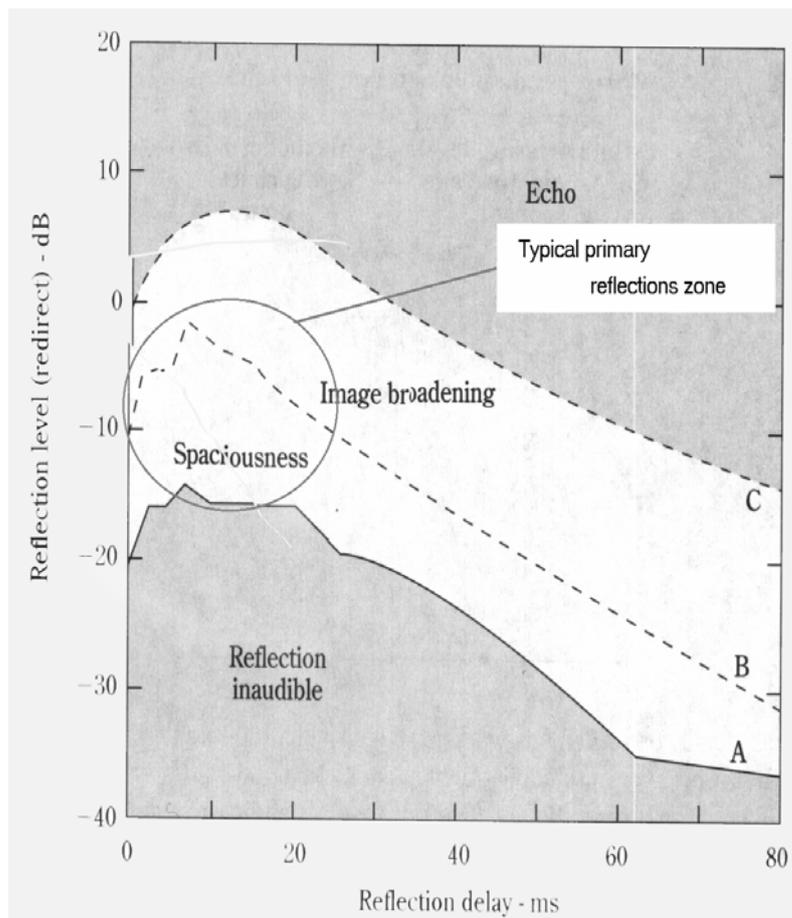


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with very controlled acoustics, if it is completely absent from the performance that we are listening to.

### ***The most advanced and recent studies. Olive and Tool's curves***

As you have understood, this subject is of an extraordinary interest for the listening in domestic rooms. Recently two very studios men, Olive and Tool, have tried to establish the relation between the dB level of the reflection and the ms. delay of it, in a more precise way. In the graphic the curves 'A', 'B' and 'C' are defined. These refer to the speech, taken as the reference sound source. The 0 dB level shows a reflection with the same level as the direct sound; -10dB shows a reflected signal which is inferior to the direct signal.



**Fig. 29 - The effects of lateral reflections on the perception of the direct sound in a simulated stereo arrangement. These measurements were made in anechoic conditions, lateral angles 45° - 90°, with speech as the signal. (A) Absolute threshold of the audibility of the reflection. (B) Image shift / broadening threshold. (A&B after Olive and Toole) (C) Lateral reflection perceived as a discrete echo.**



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The curve 'A' is defined as echo audibility threshold, below this, whatever the delay is, the reflection is not perceived by our hearing system. Notice how within 0-20 ms. of interval, the audibility level of the reflection slightly varies as regards the delay. If the reflections exceed the level of the threshold 'A', we will have space effects. If the threshold 'A' exceeds 10 dB, we will have a different curve (B) which is responsible for the effects that change both the dimension and the position of the primary image. If the level increase by 10 dB more, the previous sensations will be amplified to the point we perceive distinct echoes; curve 'C'.

The typical primary reflections of common listening rooms are underlined in the area 'D' (delays between 0,5 and 25 ms. with levels from -0,5 to -1,5 dB); these are included in the area between the ambience/space effects and the effects of change of the stereophonic image dimension/position. As you can see, the problem is not easy due to the high number of variable elements:

- Type of signal, music, speech, noise, etc
- dB level of the reflection.
- ms. delay of the reflection.
- Direction and source of the reflection.

However, the studies on this subject are becoming more frequent, because now the fundamental importance of the listening quality of these time areas of delayed sound, is not questioned any more: 0-25 ms. with an influence of 70% and more than 25 ms. for the remaining 30%.