Preliminary test repport

Acoustic System International RESONATOR

Preliminary test definition and test repport

Abstract :

Acoustic System International claims an invention of a set of accessories called RESONATORS that purify and amplify the sound, by a “modification” of the acoustic and magnetic parameters of a room.

PRO LINKS was asked to define and execute a simple test procedure to measure and verify the influence of this RESONATORS.

This document presents :

1) Our understanding of the RESONATOR role to modify the acoustic parameters
2) An introduction of the basic theoretical concepts of sound analysis to highlight the more important parameters that could be measured,
3) The first test results in the acoustic and in the electromagnetic spectrum,

For the acoustic applications PRO LINKS will propose to Acoustic System a formal procedure that could be used, not only to prove the actual influence of this accessories, but the basic tests to be done to simplify and improve the installation of this accessories.

More accurate tests must be defined to explain the effect of this accessories in the electromagnetic spectrum, specially for controlling the environment during the tests.

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PRO LINKS presentation

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Acoustic System RESONATOR

Acoustic System is offering a set of accessories that are placed in a room to purify and amplify the sound.

Their action was already proved to isolate or to enhance the acoustic characteristics of different rooms and auditoriums.

RESONATOR presentation

Physically, the RESONATOR is a small dimension “metal cup” placed over a support in different parts of the room.

There are 5 kinds of RESONATOR, each one defined by the main precious metal that is used. Their main “human perception“ characteristics are the following :

PLATINUM : Pure high frequency short range and long period of resonance.

GOLD : Complex medium to high frequency large range and long period of resonance.

GOLD SPECIAL : Pure high frequency large range and very long period of resonance.

SILVER : Complex low to medium frequency medium range and short period of resonance.

BASIC : Complex low frequency medium range and very short period of resonance

(the frequency and time of resonance must be confirmed in later report)

RESONATOR	Main resonance frequency range	Time
PLATINUM		
GOLD		
GOLD SPECIAL		
SILVER		
BASIC		

RESONATOR role

After a rough measurement action described in this document, we could see that this accessories have an actual role that can be measured mainly in the acoustic spectrum range, but also some interaction effects where seen in the electro magnetic spectrum.

From our understanding, in the **SOUND SPECTRUM** :

- The RESONATOR reacts as a complex diapason to the sound that he receives. (from the acoustic source and from the reflected waves).
- The resonance produced is an ECHO with an specific DISTORTION that enhance or attenuates part of the sound spectrum.
- The resonance interacts with the acoustic source and with all the reflected waves, modifying the human perception of the sound by modifying the acoustic parameters of the room.
- Each RESONATOR became a new sound source that interacts with the other RESONATORS installed in the room.

This effect is similar to the results that can be obtained by an specific signal processing most known as SOUND SPACIALISATION.

Even if the acoustic parameters modification theoretical model became to much complex, it seems possible to define a procedure to predict the action of a set of RESONATORS and to measure the effects.

The measurement protocol to reach this goal should be part of our mission when we will analyze in depth all the acoustic parameters as described further.

In the **ELECTRO MAGNETIC SPECTRUM** :

- The RESONATOR interaction in the electromagnetic field was observed even if the measurement conditions were not full under control.
- Specially, we could verify that an significant attenuation (with the GOLD model) at frequencies up to 300 kHz and some interactions up to 1, 8 Mhz

The physical action is not easy to explain, but it seems that a kind of shield or attenuation is produced by a resonant interaction (as an “passive” antenna) according to the positioning of the RESONATOR.

From the theoretical point of view, we know that

- Electromagnetic waves are reflected when they reach a surface, then an interference is created.
- A conductor in an electromagnetic field, acts as an antenna.
- An “spherical” or “parabolic” antenna concentrates the incident waves and can generates some amplification

Test conditions (and complementary testing) must be defined to formally prove that the measurement protocol can be reproduced with consistent results.

BASIC SOUND ANALYSIS CONCEPTS

Behavior of Sound in a Room

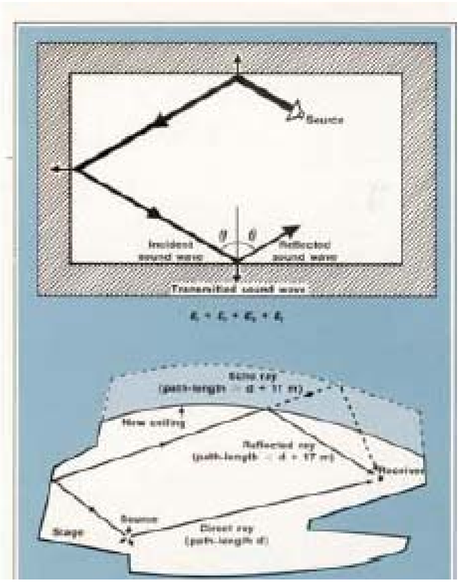
A knowledge of the behaviour of sound in a room is necessary if we wish to understand the involved parameters that should be measured to adapt the room for speech or music and if we want to attenuate external noise.

Consider the effect of placing a sound source in a room.

> *When sound energy (Ei) from the source strikes a room boundary, the reflected sound energy (Er) contributes to the sound-field in the room, the absorbed sound (Ea) dissipates as heat, and the transmitted sound energy (Et) propagates away through the boundary layer.*

Reflection of Sound

If the wavelength of an incident sound-wave **is much smaller** than the dimensions of the reflecting surface, then the angle of reflection of the sound-wave equals the angle of incidence.



We can use this geometrical behavior to predict the pattern of sound rays in a room, a limitation being that only the primary and possibly the secondary reflections can be studied before the reverberant field begins to mask the ray paths.

In larger rooms such as concert halls, 'ray tracing' can identify problematic echoes, an echo being defined as a reflection which arrives more than 50 ms after the direct sound.

An echo can also be thought of as a reflected ray with a path-length that is at least 17 m longer than that of the direct ray. Echo problems in large enclosures are solved by reducing the path length of the reflected ray.

This can be done either by lowering the ceiling or by suspending reflectors from the ceiling.

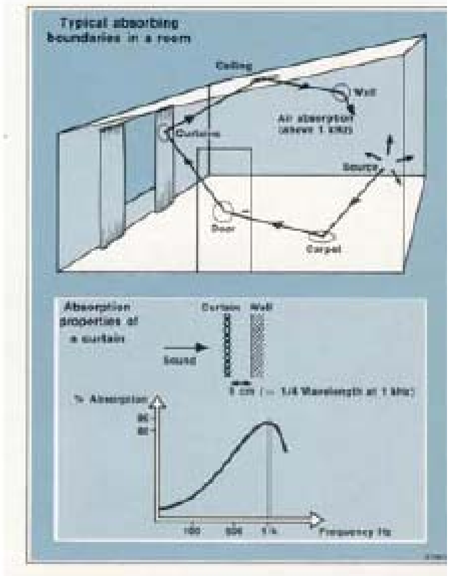
By observing the behaviour of the reflections in a room, we can control subjective properties such as intimacy, the quality of which depends on early arrival of reflections after the direct sound, and diffusion which is the evenness of the reverberant field.

Absorption of Sound

We can understand the effect of absorption by measuring, at a given position in a room, the sound pressure level caused by a steady sound power source. Instead of rising indefinitely as an increasing number of reflections arrive at the measuring position, the sound pressure level soon stabilizes.

This must mean that the rate of energy input is exactly compensated by the rate at which the energy is absorbed by the different surfaces of the room. If more absorption material is put in the room, the sound pressure level is less because the energy in the reflections is reduced.

Typical absorbing surfaces in a room include carpets and curtains.



These are simple porous absorbers which absorb sound energy by restricting the movement of air particles, the frictional forces causing the dissipation of energy as heat.

Porous absorbers are most effective when placed at a point on the sound wave which has maximum particle velocity. This position is a quarter wavelength away from a reflecting surface (when a wave is incident at right-angles) and is therefore frequency dependent.

A carpet is an example of a porous absorber close to a reflective boundary. It absorbs best at high frequencies because the dimensions of the quarter wavelengths are then comparable with the thickness of carpet.

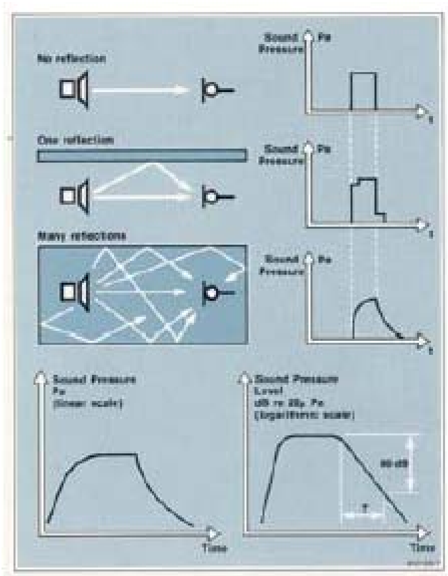
Other surfaces in the room absorb different frequencies to different extents, and by controlling the proportions of these absorbers it is possible to adjust the warmth of a room for music, or its clarity for speech.

Build-up and Decay of Sound in a Room

If we position a microphone in a room and then switch on a steady sound-source, we notice that the sound pressure level does not immediately reach a steady level. This is because the first reflection and subsequent reflections take a finite time to reach the microphone.

In the resulting equilibrium state, interference between the sound-waves causes a spatial distribution of pressure maximal and minima which can be detected by moving the microphone around the room.

These natural resonance or normal room modes are associated with the geometry of the room and the wavelengths emitted by the sound-source.



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Interesting consequences of these modes are that pressure doubling occurs at reflective boundaries, and that since all the room modes have antinodes at the corners of the room, they can all be "driven" by a sound-source placed there.

If the sound-source is now switched off, the collection of decaying room modes is called the **reverberant sound-field.**

The rate of decay depends on the amount and positioning of absorption in the room. Reverberation Time is defined as the time taken for the sound pressure level in a room to decay by 60 dB. This corresponds to a decrease in sound pressure by a factor of 1000.

Importance of Reverberation Time

In a room with highly reflecting surfaces, such as a bathroom, the reverberation time is relatively long, while in an anechoic chamber where all the walls, the ceiling and the floor are covered by a highly absorbent material, the reverberation time is nearly zero.

The absorption of different materials varies widely with the frequency of the incident sound and the angle of incidence. It follows that the reverberation time is liable to vary with frequency.

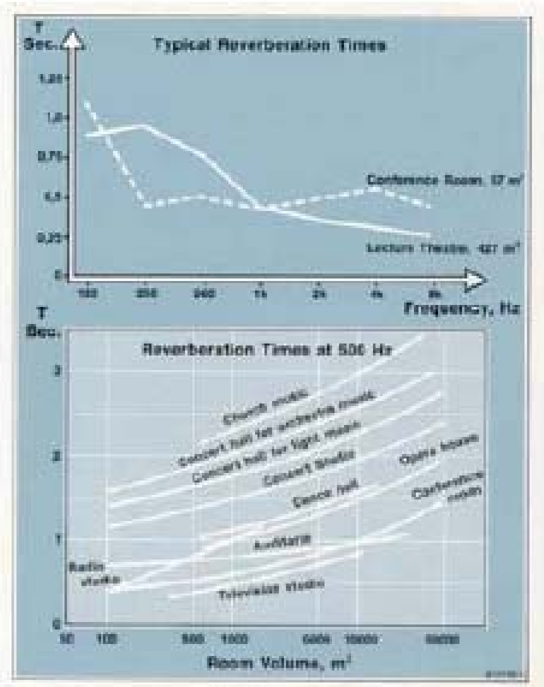
Generally, the reverberation time is longer at lower frequencies because these are usually less effectively absorbed than higher frequencies.

It is important that the reverberation time suits the intended use of the room :

> ***Too long a reverberation time renders speech less intelligible and music more cacophonous and produces higher background noise levels.***
>
> ***A short reverberation time deadens background noise, but muffles speech and makes music sound "thin" and staccato.***

Reverberation time measurement is defined by ISO 3382 - 1975(E) normalization and is part of the set of Building Acoustics Application instruments.

ISO 3382 - 1975(E) defines reverberation time as:



"The time that would be required for the sound pressure level to decrease by 60 dB after the source has stopped."

This is qualified by the note:

"This definition is based on the assumption that in an ideal case the dependence of sound pressure level upon time is a straight-line relationship and that the background level is sufficiently low."

As noted, it is not normally possible to achieve such decays in level during measurement, nor will the cessation of the source always be clearly defined, due to standing waves and early reflections.

Therefore, for practical reasons, the theoretical 60 dB decay time (T60) must be predicted from lesser decreases in level. Several values are in common use based on the time taken for the sound level to decay from 0 dB to -10 dB, from -5 dB to -25 dB, and from -5 dB to -35 dB:

EDT (Early Decay Time) 6 x time taken to decay 10 dB from the maximum level, T20 3 x time taken to decay 20 dB from the point 5 dB below the maximum level, T30 2 x time taken to decay 30 dB from the point 5 dB below the maximum level.

FIRST MEASURING RESULTS

SOUND SPECTRUM : Reverberation Time

The first test made in the office of Acoustic System was oriented to measure the effect of the resonator in the SOUND SPECTRUM using and amplifier with “white” noise and a sound analyzer to monitor the decay in sound pressure level after the sound source ceases.

The room was initially equipped with 9 RESONATORS as shown :

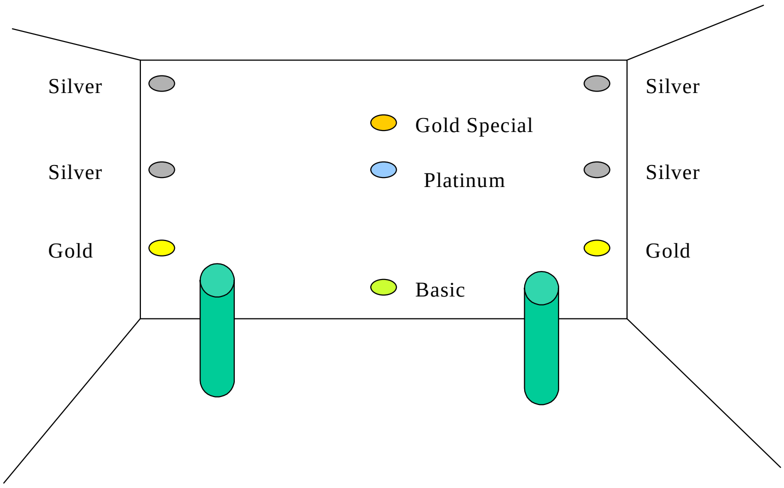
4 – SILVER

2 – GOLD

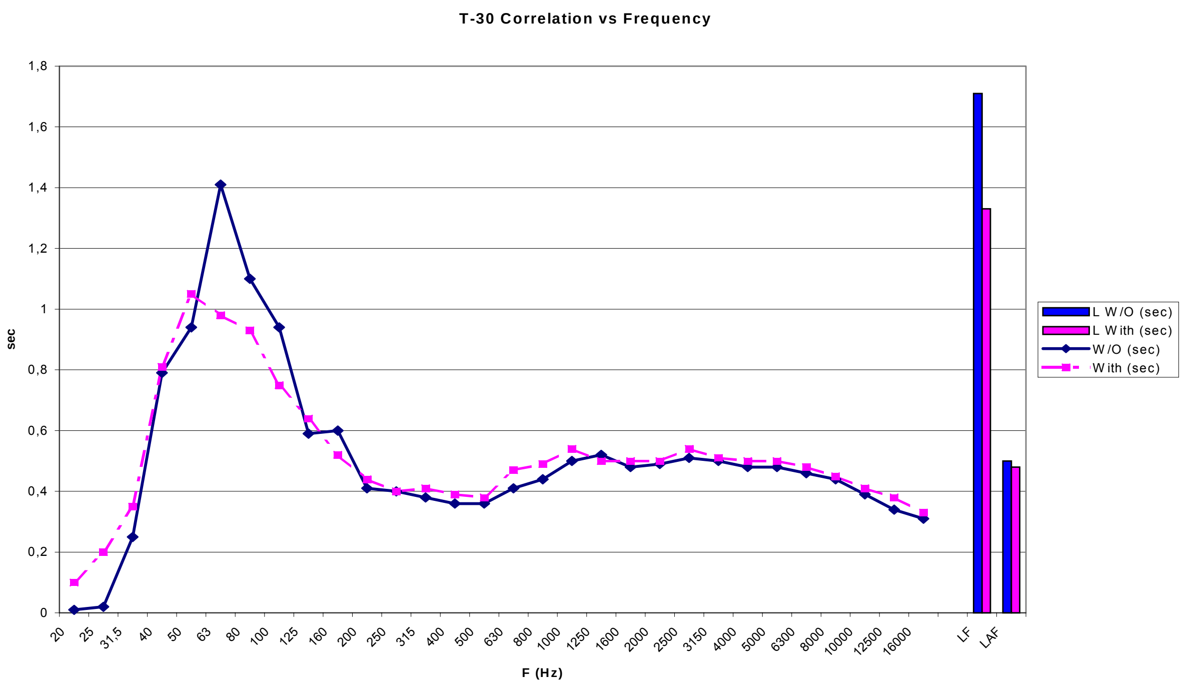
1 – GOLD SPECIAL

1 – PLATINUM

1 – BASIC



Following image shows the correlation curves at T30 level, according to the definition of the ISO 3382 - 1975(E) standard, for the same room with and without RESONATORS with no other modification.



From this curves we can see that :

- The response is strongly modified in the low frequencies (1/3 less at 70 Hz)
- The frequency response of the room is “equalized” reducing the time of the global response and equilibrating medium and high frequencies.

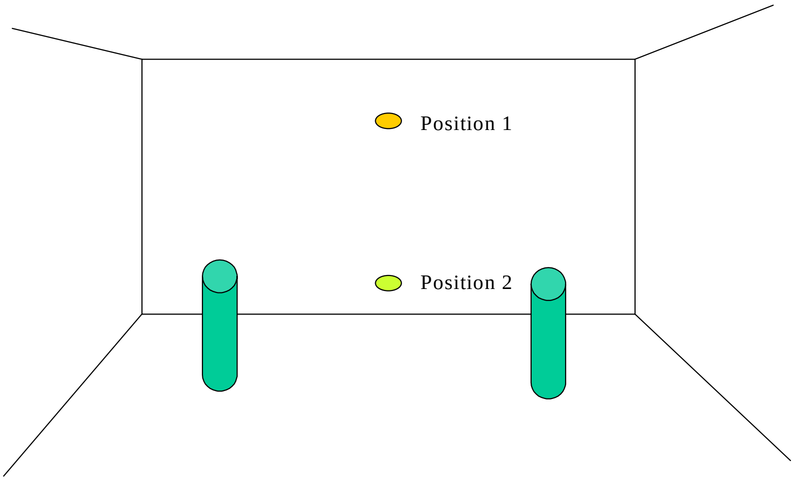
Full SPECTRUM : Electromagnetic Noise

The second test made in the same room was oriented to measure ”roughly” the effect of the resonator in the FULL SPECTRUM using a RF spectrum analyzer, to measure the amplitude at the noise level.

The room was initially equipped with each type of RESONATOR placed in position 1 and 2 as shown . (One at each time)

The noise was compared between 2 cases : with or without RESONATOR.

The ambient noise was not controlled, but we can see some interactions reduction far over the level of the human “ear” spectrum.



The following graphics and data values shows the most significant results.

The reference curve in each graphic is the average of the measurement without resonator before doing the measures with the RESONATOR installed.

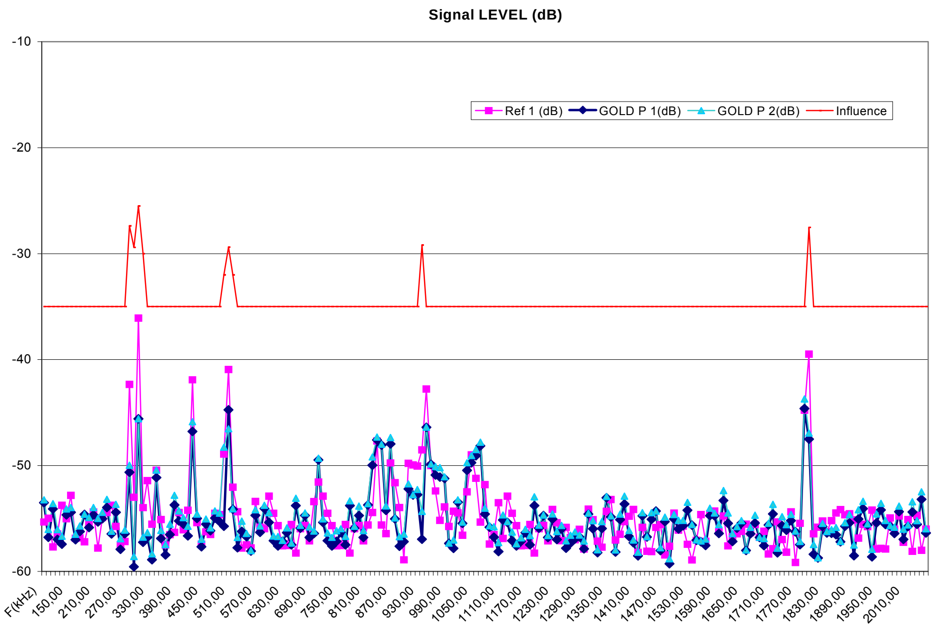
The shape of this reference shows some differences according to the time evolution, but the most significant signals where presents during all this measurements.

This was not a formal protocol to prove the influence of the RESONATORS but only the first rough tests that give us some orientations o define the right tests.

GOLD RESONATOR :

Frequencies with significant attenuation :

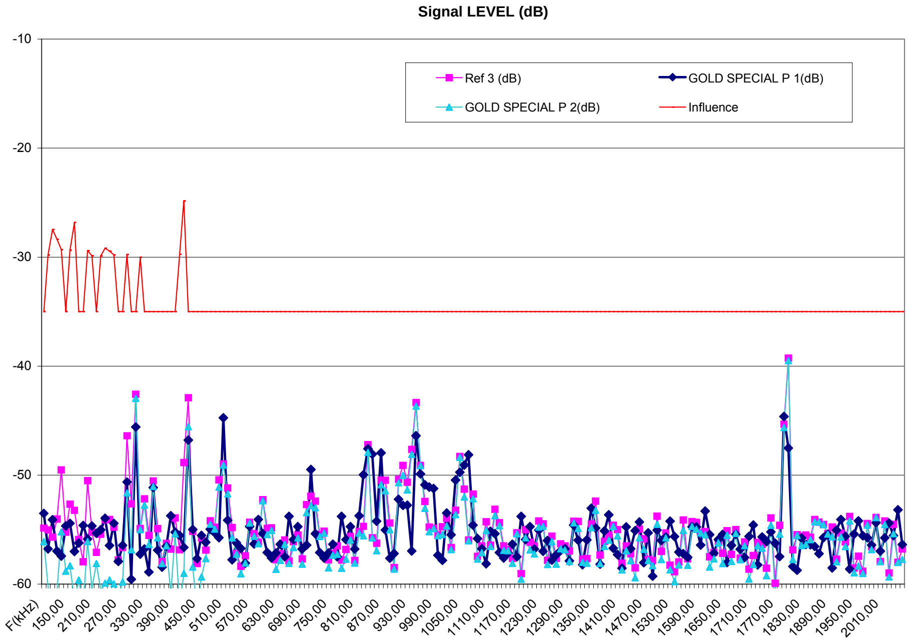
- 300 kHz,
- 500 kHz
- 900 kHz
- 1 800 kHz



GOLD SPECIAL RESONATOR :

Frequencies with significant attenuation :

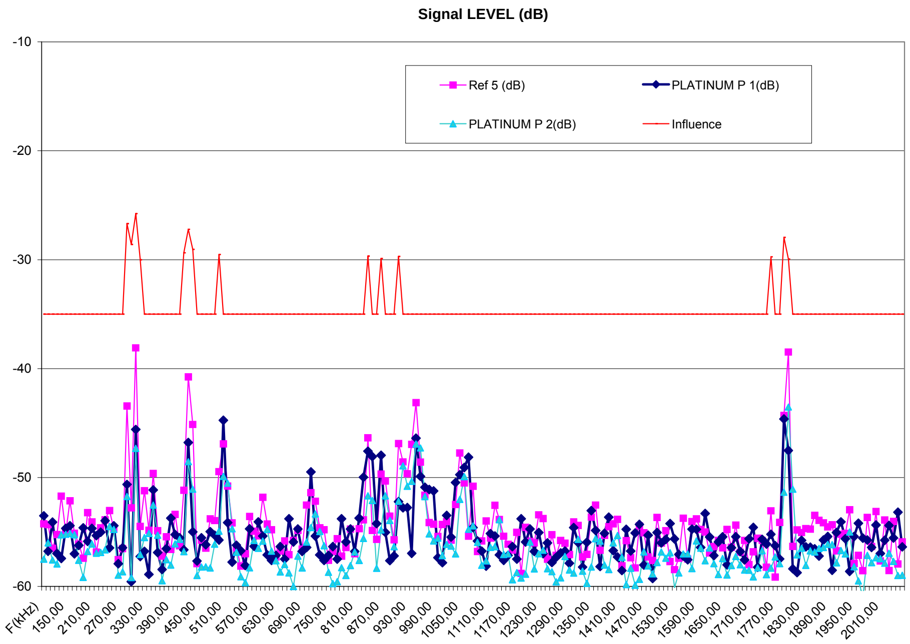
- up to 500 kHz



PLATINUM RESONATOR :

Frequencies with significant attenuation :

- 300 kHz
- 900 kHz
- 1 800 kHz

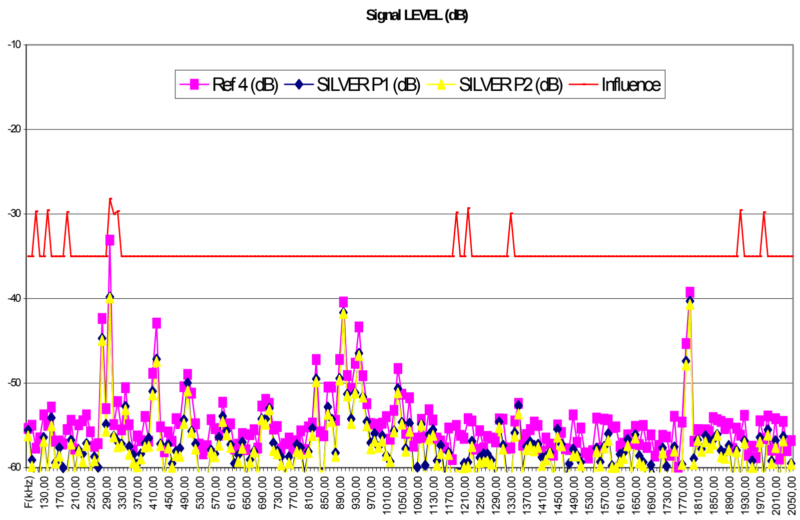


SILVER RESONATOR :

Frequencies with significant attenuation :

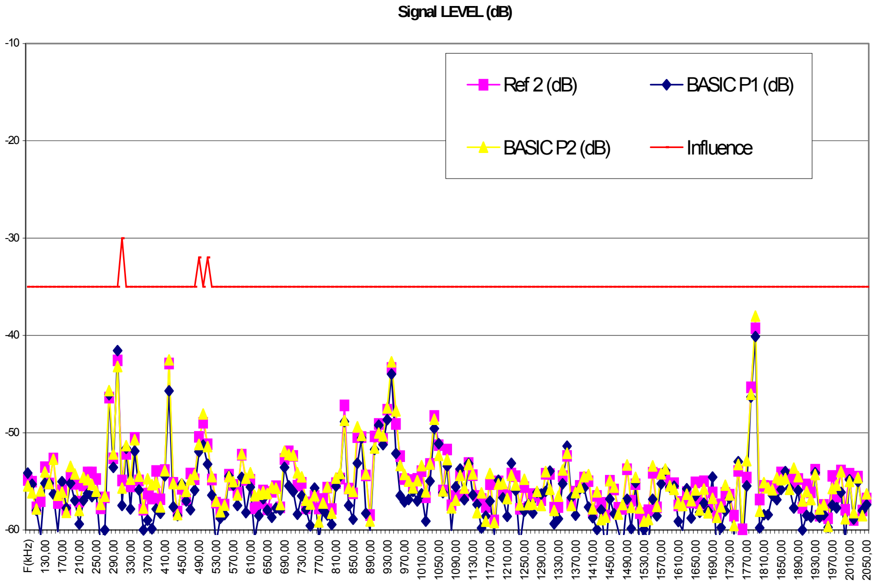
- up to 300 kHz

(Others : NOISE)



BASIC RESONATOR :

No Frequencies with significant attenuation



CONCLUSION

We can conclude that the RESONATORS provided by Acoustic System have an actual and measurable effect on the SOUND spectrum, and seems to have an effect at medium and high frequencies in the electro magnetic spectrum.

We see two actions that could be defined after this first trial :

For the acoustic applications :

Today, the installation of this equipment in a room is defined empirically by people that have the skills to “ear” the sounds reflection effects and that have enough experience within acoustic environment.

Since the acoustic parameters can be measured, and the effect of the RESONATORS could be predicted and measured, it seems possible to define a formal procedure that could help “non initiated” people to make better installations and get better results.

In order to develop this formal procedure, we suggest to start a new measurement campaign with the following goals :

- Define a procedure to measure the relevant acoustic characteristics of a room.
- Define a procedure to set the right RESONATOR or combination of RESONATORS that must be installed, and their positioning in the room.
- Define a procedure to verify the results and / or to “tune” the position or the combination of this RESONATORS.

To introduce this action we annex to this document a description of the measure of the acoustic parameters.

For the electromagnetic influence :

We know that the results of the tests described in this document are not enough to prove the action of the resonators at this frequencies, but at least they show that we can measure some effect.

The first limitation is that the environment was not controlled, and that the measures of the reference where not done simultaneously .

We propose to :

- Specify a formal procedure of the required test
- Make a campaign of testing with the defined procedure, in the site but also in collaboration with some Research Laboratories that have the possibility to make this tests under a controlled environment.

ANNEX
MAIN PARAMETERS MEASURING METHODS

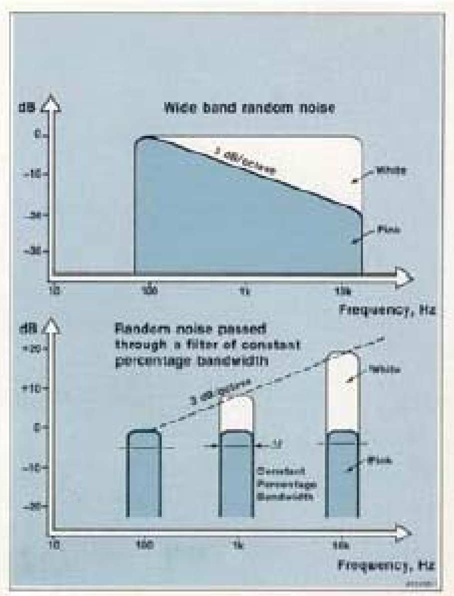
Measuring the Reverberation Time

To measure the reverberation time one needs a sound source to generate sound within the room and a receiving section to monitor the decay in sound pressure level after the soundsource ceases.

The Sound-Source

A starting pistol is a practical sound-source, but a pistol shot lacks both energy in the low frequency regions and reproducibility. A better way of excitation is to use a loudspeaker emitting noise in frequency bands. For a given power amplifier, this allows more energy to be transmitted into the room than with the starting pistol (which is important when high levels of background noise are present).

"White" noise, is a wide band of random noise (i.e. a signal containing all the frequencies of the spectrum with a random amplitude distribution) with a constant level per Hertz over the entire frequency spectrum. "Pink" noise is a wide band of random noise with a level decreasing by 3 dB per octave.



This attenuation is necessary to allow a constant energy to be transmitted through a filter with a bandwidth which becomes progressively wider (e.g. an oct. or 1/3 oct. filter), doubling the width for each octave.

Due to the presence of background noise, it is seldom possible to measure the full 60 dB reverberation decay and one has to be content with a 40 dB, 30 dB or even 20 dB decay extrapolated to 60 dB. It is usual to specify the decay over which the reverberation time was measured, e.g. Tr(30), Tr(20).

The noise can either be transmitted as a steady sound which is then cut off, or as a short pulse, the two methods having different receiving section requirements.

The Receiver

A typical receiving section may consist of a sound levelmeter fitted with an octave or a 1/3 octave filter set and a portable level recorder. A filter centred on the same frequency as the filter in the transmitting section reduces the influence of background noise. Since reverberation de creases in an exponential manner and is recorded on a logarithmic scale, the decay will be a straight line on the recording paper. The reverberation time result (for a given frequency band) is estimated directly from the recording.

The jagged appearance of the decays at low frequencies is due to the uneven distribution of the normal room modes at these frequencies.

When the pulse method of noise transmission is used, the graphical results represent the Impulse Response of the room and the reverberation time cannot be obtained directly from the decay. By using the appropriate software, it is possible to calculate reverberation time results from the impulse response. An advantage of the pulse (or Schroeder) method is that accurate and reproducible results are obtained faster than with the "cut-off" method.

Using a Building Acoustics Analyzer

A Building Acoustics Analyzer is an instrument containing both the transmitting and the receiving sections.

It supplies random noise in 1/3 octave bands to a power amplifier and a loudspeaker, analyzes the microphone signal through a second set of 1/3 octave band filters, and calculates the reverberation time for each frequency band.

Position of the Source and the Receiving Microphone

Due to room modes and echoes, the reverberation time of a room depends on the position of the source and the receiving microphone. In some cases the position of the source is obvious (e.g. the rostrum in a lecture theatre). To avoid exciting only some of the normal modes of the room, the sound-source is usually placed in a corner where every mode has a pressure maximum.

The receiving microphone should be placed at several positions in large rooms and auditoria because the reverberation time can vary from place to place. If required, the measured times should then be averaged for each frequency band by one of the following methods:

(a) a single microphone moved from place to place;
(b) several microphones scanned by a multiplexer;
(c) a single microphone on a rotating boom.

Measuring the Sound Absorption

The absorption coefficient of a material indicates the proportion of sound absorbed by the material relative to the total incident sound. The total absorption of a surface is given by the absorption coefficient multiplied by the area. The most usual measurement methods are:

Reverberation Chamber Method

The change in the reverberation time is measured when a 10m2 sample of absorption material is introduced into a reverberation chamber. From Sabine's Formula and the definition of absorption, α can then be found:

0,16 V (1 1) α = - S (Ts Te)

where

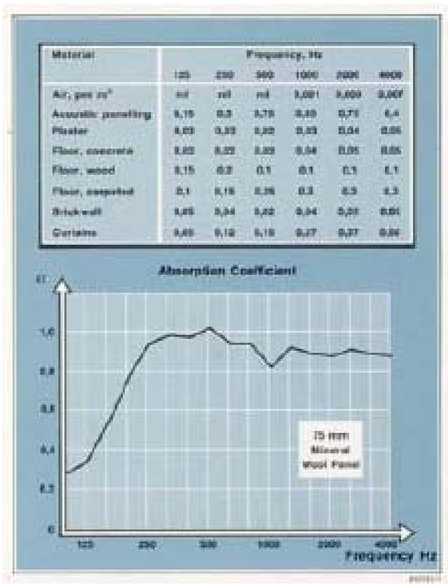
α is the absorption coefficient of the sample

S is the area of the sample of material

V is the volume of the chamber

Ts is the reverberation time, with the sample

Te is the reverberation time of the empty chamber



The measurements are performed by using an octave or 1/3 octave filter set to obtain α as function of the frequency.

Measuring the Change of Reverberation Time "in situ"

A similar method can be used in practical situations when determining the amount of absorbent material necessary to obtain a suitable reverberation time in a room. From the absorption coefficient, α, calculated from measurement in a reverberation chamber, one calculates the area of absorbent necessary to produce a required change in reverberation time in a particular room. The absorbent material is installed, the reverberation time is measured in the actual room and, if necessary, adjusted by adding or subtracting some of the absorbent material.

Standing Wave Method

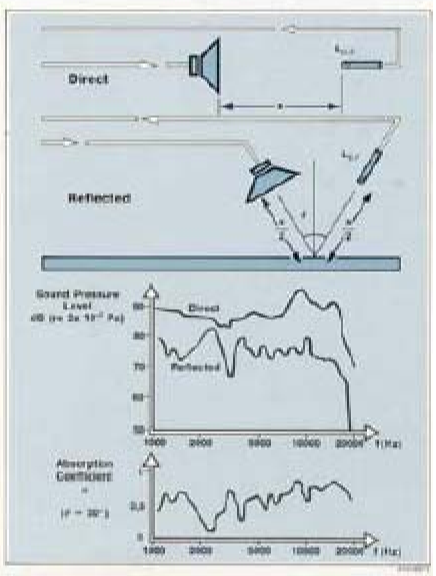
In this method a loudspeaker is used to produce standing waves in a tube terminated by the sample to be investigated. By measuring the ratio between the maximum and minimum sound pressures by means of a probe microphone moved along the axis of the tube, the absorption coefficient

can be calculated. The advantage of the method is that it only requires small samples of material, gives reproducible results and yields a direct scale reading for the value of a.

The disadvantages of the method are that α. is obtained for normal incidence only and that the method can only be used where the sample is representative of the material.

Tone Burst Method

This method enables the absorption coefficient of a material to be determined for various angles of incidence of sound energy.



No special reverberation room is required for this test.

A short tone burst is emitted from a loudspeaker into the room at a distance x from the receiving microphone.

The loudspeaker is then aimed at the test specimen at an angle of incidence, α, such that the total path length for the reflected sound is the same as in the first case.

By comparing the sound pressure level, Lp,r, of the reflected sound to the sound pressure level, Lp,d of the direct sound, the reflection coefficient can be calculated and the absorption coefficient determined from:

$$\alpha_{a,r} = 1 - r_{a,r}$$

where $\alpha_{a,r}$ = the absorption coefficient

$r_{a,r}$ = the reflection coefficient

$$= 10^{\frac{-(L_{p,d} - L_{p,r})}{10}}$$

Measuring the Sound Distribution

Sound distribution measurements are especially important in theatres and concert halls or other public halls where music and speech must be heard clearly throughout the volume of the auditoria.

Measurement in Existing Room

Measurements of sound distribution in a room can be made directly by placing a source in the most probable position of the actual source (theatre stage, church pulpit, etc.) and by using a sound level meter to measure the sound pressure levels at various positions in the room. The source should be a constant sound power source radiating a wide band signal (white or pink noise).

This method can be made more informative if measurements are made at the same positions but at different frequencies. Filters (octave or third octave) can be used in the emitting section to limit the necessary power of the source and/or in the receiving section to reduce the influence of background noise.