Info Acoustics

www.jocaviacousticpanels.com

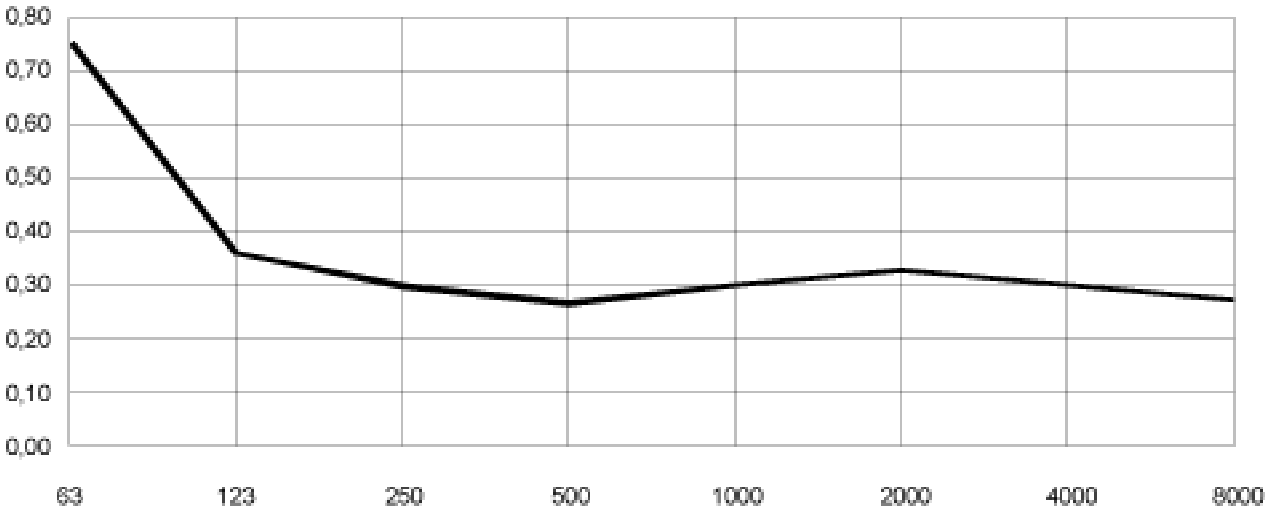
SABINE FORMULA.

Sabine is the father of modem acoustics. He found that reverberation time is described by a relationship between the room size and the amount of absorption in the room. Larger rooms – longer reverberation. More absorption – shorter reverberation.

T = 0,161 x V / A

Where
T: Reverberation time in seconds
V: Volume in m3
A: Absorption in m2 Sabine
0,161: Is a constant (to make the calculation right with the actual units)

Note:One square meter (1 m2) Sabine is comparable to an open window with an area of one square meter: The sound that hits the window will disappear and never return. One square meter Sabine is one square meter with full absorption. The basic formula sounds simple, but the problem is that the materials in the room will absorb differently at different frequencies. The absorption may range from nothing (fully reflective) to total absorption. A proper reverb time should be constant with frequency, but this is not always the case because of the behaviour of the materials in the room. The low frequencies are the most difficult to control.
This is why the reverberation time against frequency in practice may look like this:



Reverberation time measured in a control room. From 250 Hz and above the curve is nicely placed around 3 sec. But belov the reverb time rises to 75 sec. which is too much.

Info Acoustics

www.jocaviacousticpanels.com

ABSORBENT MATERIALS.

All materials in the room act acoustically even if they are not so-called acoustical materials. Basically we have three kinds of absorbers.

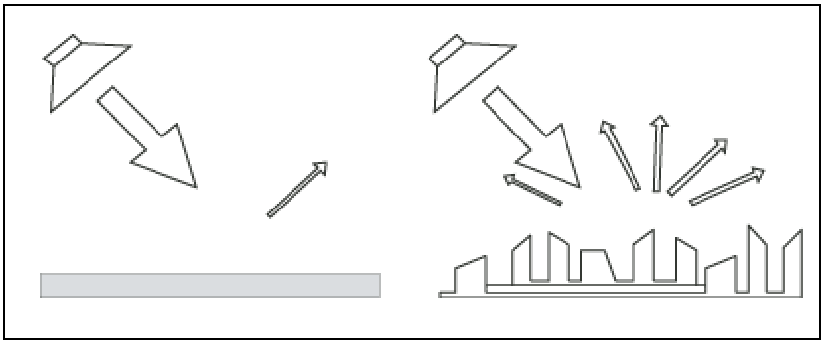
Membrane absorbers
This kind of absorber includes wooden floors, windows, doors, etc. This absorber provides absorption in the low end of the frequency range. The efficiency is normally not very high, but under normal conditions large areas are included in the basic room construction. Special designed membrane absorbers can be very effective.

Resonance Absorbers
Resonance absorbers include slit panels, perforated plate, Helmholtz resonators etc. These absorbers are normally used in the frequency midrange. The absorption is medium to high.

Porous absorbers
These absorbers include mineral wool, carpets, curtains, and so on. They can be very effective, but the thickness of the material has to be taken into account. Thin layers will only absorb the highest frequencies. (Think of a rehearsal room in the concrete basement where the only damping is a carpet on the door: Not good at low frequencies!).

To absorb a given frequency (and all frequencies above) the thickness of the absorber must be the quarter of the wavelength of that frequency. Or: the front of the material must be placed at a distance of one quarter of the wavelength.

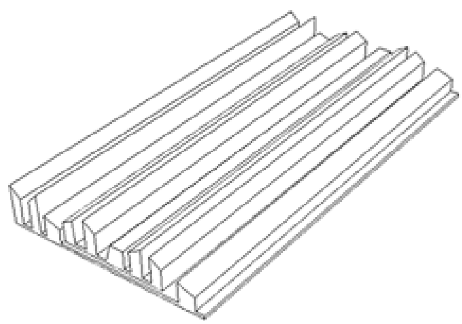
DIFFUSORS.



A diffuser provides diffuse reflection of the sound radiated against it. It can be a very useful solution in cases where reflections are disturbing the sound image and it is not advisory to ad further absorption. So in order to reduce flutter echoes, comb filtering etc., special elements can be placed on the “disturbing” surface. These elements must have dimensions comparable to the frequencies at which diffusion is wanted.
Absorbtion or diffusion?

A special technique developed by Manfred Schroeder is very capable in making a smooth and controlled diffusion. These diffusers normally referred as the "Woodiffusor" can be found as prefab modules.

Cross section of one example of the ."Woodifusor" Jocavi



Info Acoustics

www.jocaviacousticpanels.com

STANDING WAVES.

Standing waves exist in all kind of rooms. The shape of the room, the dimensions of the room, and the relationship between the dimensions of the room, are important parameters that will determine the frequencies around which the phenomenon exists as well as the distribution of these standing waves. But how do they occur?

Imagine a sound source. When the sound is emitted the sound is emitted the sound wave will propagate in all directions if no obstacles in sight. This will of course happen with the speed of sound. Now, if the sound source is placed inside a room the sound wave will hit the boundaries of the room.
If the boundaries consist of acoustically hard (reflective) surfaces, the sound is reflected. If the angle of incidence is 90º the sound will be reflected right back where it came from. Under certain circumstances the sound wave will meet itself again. For instance if the sound is reflected between to parallel walls.

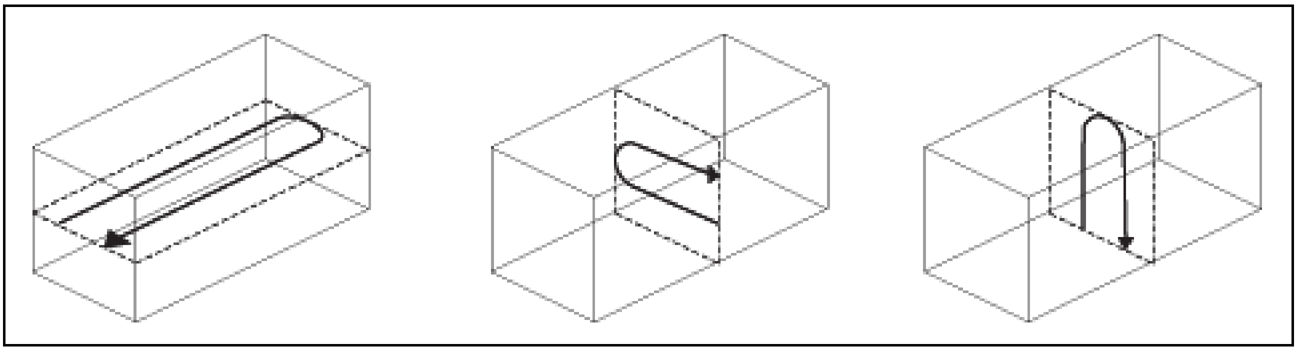
This Becomes a problem, when the sound wave not only meets itself, but when it meets itself in phase. And this will happen when the distance between the walls is half a wavelength of the radiated sound wave. Or one whole wavelength – or 1 ½, 2 ½ and so on.

This phenomenon is called standing waves. Actually the sound wave is not standing. But it is experienced like that because the sound pressure maxima and minima are positioned in fixed places in the room.

The sound field is initially radiated having a radical wave front but within a few reflections the sound field has obtained a plane wane front.

ROOM MODES.

The special frequencies are also called room modes. Standing waves between parallel walls are called axial modes. Other modes exit. For instance tangential and radical modes. (See the illustration). Normally the axial modes are the strongest.



Axial

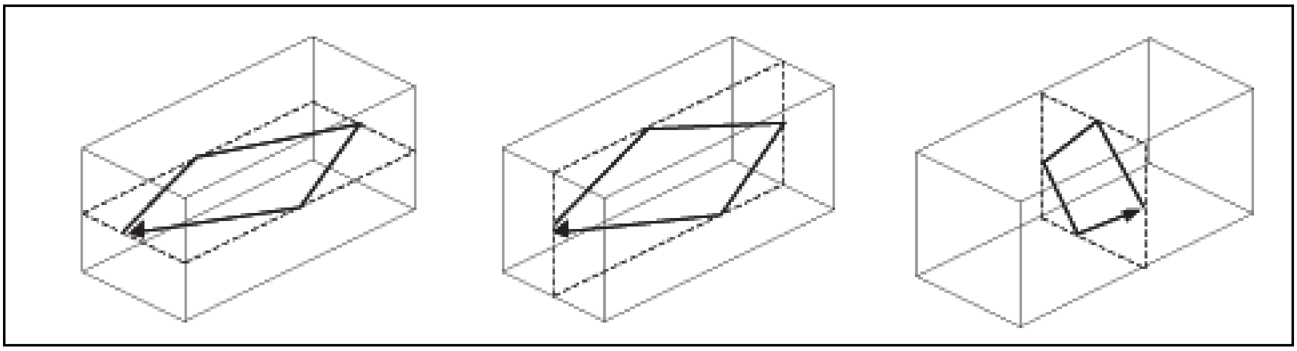
JOCAVI® - Acoustic Panels,Lda. - Avenida Pedro Álvares Cabral. C. Empresarial Sintra Estoril - V. - Armazém B18 2710-297 Sintra -Portugal.
info@jocavi.net - www.jocavi.net - Phone: +351 21 9243097 - Fax: +351 21 9243098

Info Acoustics

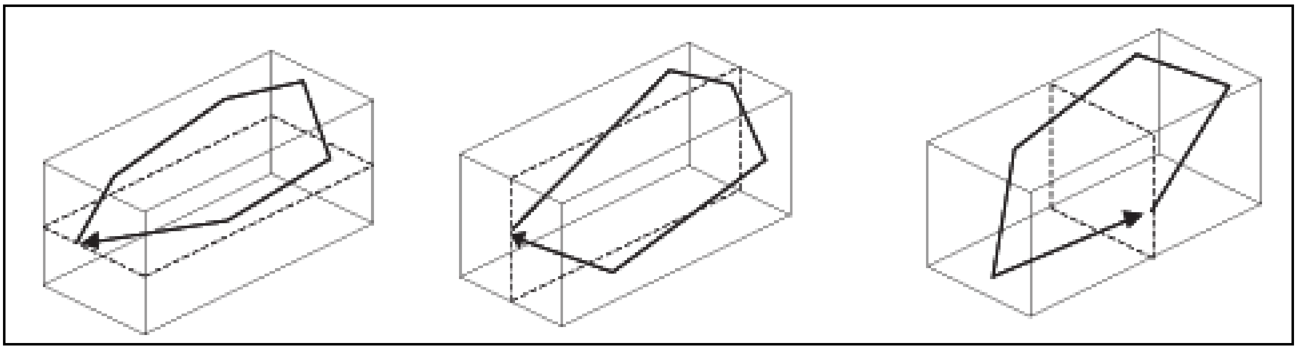
www.jocaviacousticpanels.com

ROOM MODES...*Part 2*

Tangencial



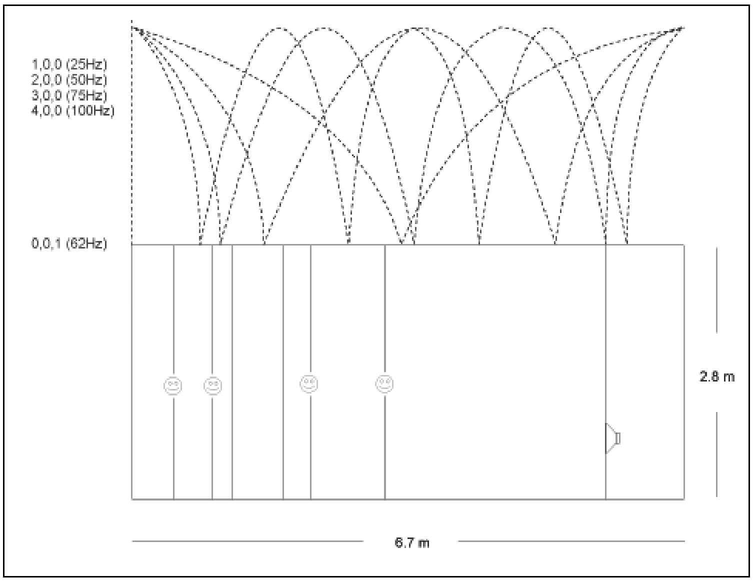
Radial



$$f = \frac{c}{2}\sqrt{\left(\frac{n_l}{l}\right) + \left(\frac{n_b}{w}\right) + \left(\frac{n_h}{h}\right)}$$

The standing waves are characterized by having a maximum sound pressure at the boundaries of the room. Depending on the frequency there are one or more dips across the room. In a box shaped room the frequencies can be calculated as follows:

where :
f = frequency in Hz
c =speed of sound (approx. 340 m/s or 1130 ft/s)
l = length of the room
w = width of the room
h = height of the room
n = integer from 0 and up



Info Acoustics

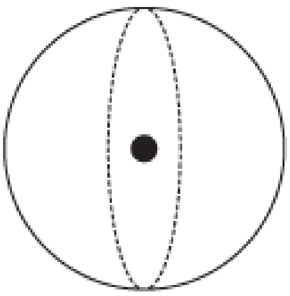
www.jocaviacousticpanels.com

HOW DOES THE STANDING WAVES UNFLUENCE THE SOUND FIELDS?

The maxima of the standing waves are shown in the figure. The curve expresses the area of the room where the actual frequency is audible. At the minima the frequency is represented at a much lower level (sometimes – 40 dB compared to the maximum).

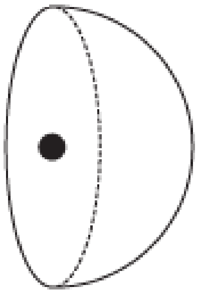
If the room has the same dimensions as length, width, and even height it is very problematic to obtain an even sound distribution.

How to prohibit standing waves?
Parallel walls in the room should be prevented. Then the strongest modes are suppressed. When placing the monitors it is important that as few modes as possible are excited. This is why the monitors should not be placed in a maximum of a standing wave. At low frequencies a monitor can be considered as to radiate the sound energy in all directions.



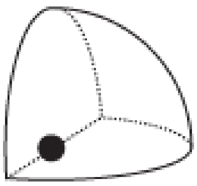
This is also called a 4π radiation

When placing the monitor close to a solid boundary – for instance a wall – the sound energy that should have been radiated in the direction of the wall instead is radiated into the free half space. Hence the sound pressure is doubled in the half space, which yields +6 dB.



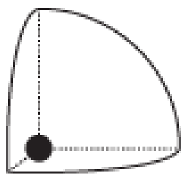
This is also called a 2π radiation

Placing the monitor against two boundaries – for instance in a corner limited by two walls – it is now radiating to the quarter space. Now the sound pressure is doubled twice, which yields +12 dB.



This is also called π radiation.

Colocando o monitor contra três barreiras – por exemplo um canto limitado pelo solo e duas paredes – o som é radiado num 1/8 de espaço. Comparado com o espaço livre, a pressão sonora aumenta agora em 18 dB.



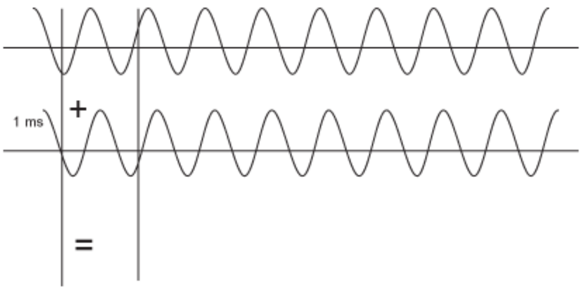
This is also called π/2 radiation

Na prática, a colocação perto de barreiras, paredes ou chão, influenciará a gama de frequências abaixo dos 125-150 Hz.

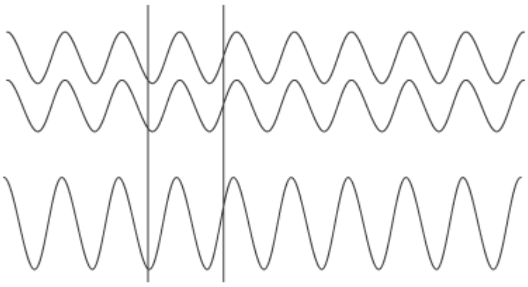
Info Acoustics

www.jocaviacousticpanels.com

COMB FILTERING.

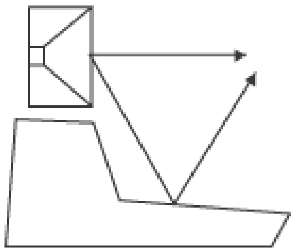


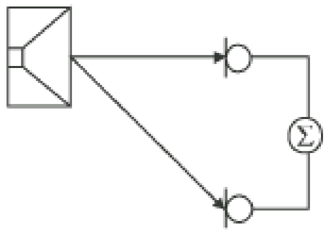
The filtering function that arises when a signal is added to itself after having been delayed in time is called a comb filter. The resulting frequency response resembles a comb, hence the name.



Two 500 Hz sinusoidal tones added. The second tone is delayed 1 ms hence the sum is zero. Two 1 kHz sinusoidal tones added. The second tone is delayed 1 ms hence the sum is the double (+6 dB).

The comb filter function is almost never intentional, but it is heard all the time in sound productions, where it can arise both acoustically and electrically. Acoustically, it typically occurs when the sound on its way from source to recipient takes in part a direct path and in part an indirect path via a single reflective surface. The reflection must be attenuated at least 10 dB and preferably 15 dB in order for it not to have an effect on the sound field at the recipient position. Electrically, the phenomenon arises when two microphones with a certain distance between them capture the same signal and the level from each microphone is of the same order of magnitude.

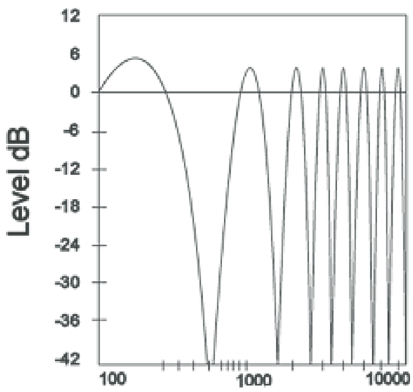




Two typical situations in which comb filters arise, either acoustically or electrically.
In general: All digital signal processing takes time. This means in practice that comb filter effects can arise if you loop a signal via, for example, a compressor and combine this signal with the original.

> dB Level Frequency– Hz
An example of a comb filter created by the combining of two signals with the same amplitude, but with a time delay between them of just 1 ms.
It can be seen that a dip occurs due to cancellation at 500 Hz, 1.5 kHz, 2.5 kHz, etc. It can also be seen that the two signals add to double their value (+6 dB) at low frequencies and with a full wavelength's delay at 1 kHz, 2 kHz, 3 kHz etc.

> Dip Frequency
Cancellation occurs for a comb filter at all the frequencies where the two signals are in opposite phase. This occurs when the time delay comprises duration of ½, 1 ½, 2 ½, etc. periods. At 1 kHz the period is 1 ms. Half of the period is 0.5 ms. If a time delay of precisely 0.5 ms occurs, it means that cancellation will arise, not just at 1 kHz, but also at 2 kHz, 3 kHz, 4 kHz etc



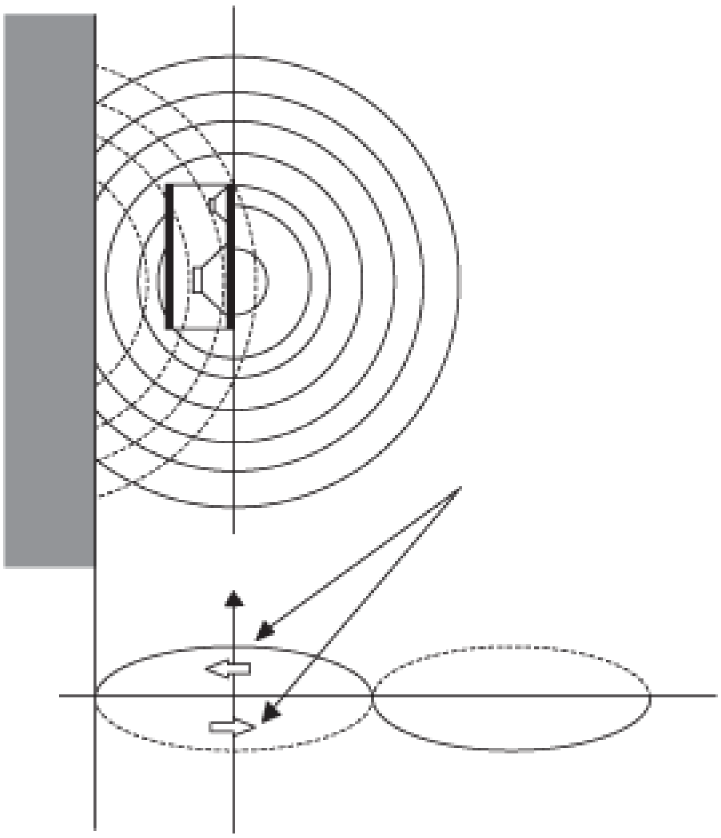
Info Acoustics

www.jocaviacousticpanels.com

REAR WALL CANCELLATION.

When the monitor is set up at some distance in front of a wall, reflections from the wall may occur and influence the perceived frequency response. This could result in comb filtering if all frequencies produced by the monitor were radiated in all directions. But the monitors are typically only omni- directional at low frequencies. The result of the reflection is a single or few dips in the frequency response perceived in front of the monitor. The frequency response may look like this: one cancellation at the frequency that has a wave length of four times the distance to the rear wall.

The dip – or cancellation frequency – is dependent on the distance to the wall. If the distance is 1 m, the first dip frequency with a wavelength of 4 m.



= c/f
Where :

l = wavelength (m) (ou ft)
c = speed of sound [m/sec] (or ft./sec.)
f = frequency (Hz)
Hence:

4 = 344 / f
F = 86 Hz

A closer position will result in a cancellation at a higher frequency. This is then limited by the frequency where the monitor becomes directional and does rot radiate sound to the rear.

A farther position will result in cancellation at a lower frequency. This is limited by the distance being so long, that the reflected sound is attenuated due to the long extra path travel.

JOCAVI® - Acoustic Panels,Lda. - Avenida Pedro Álvares Cabral. C. Empresarial Sintra Estoril - V. - Armazém B18 2710-297 Sintra -Portugal.
info@jocavi.net - www.jocavi.net - Phone: +351 21 9243097 - Fax: +351 21 9243098