Acoustics'08 Paris
June 29-July 4, 2008
www.acoustics08-paris.org

Improving room acoustics at low frequencies with multiple loudspeakers and time based room correction

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Small and medium size rectangular rooms are often used for sound reproduction. These rooms have substantial acoustical problems at low frequencies primarily caused by the reflections from the room boundaries. The spatial variation in sound pressure level (SPL) can be up to 30 dB in a room at low frequencies, and appear not only at modal frequencies. The problem is an acoustical issue in time, and should therefore be analyzed in the time-domain, instead of the traditional steady state frequency domain. The construction of a finite-difference time-domain approximation program (FDTD) has lead to a simple and untraditional solution called CABS (Controlled Acoustical Bass System) that makes use of multiple loudspeakers. With the proper placement of low frequency loudspeakers, CABS can create a plane wave from the front wall which will be absorbed by additional low frequency loudspeakers at the back wall. With the back wall reflection removed a homogeneous sound field will be created in the whole room at low frequencies. Simulations and measurements of normal size listening rooms show that 4 loudspeakers are enough to even the sound field in a room. The CABS system is controlled by a developed DSP system.

1 Introduction

Loudspeakers placed in a room are used for reproduction of recorded sound, for generation or reinforcement of sound. The interaction between the loudspeakers and the listening room has always had a big attention by the critical listeners and the producers of audio systems. The storage and reproduction format for music is still normally stereo, so far only few music recordings are made for multi channel sound reproduction systems such as the 5.1 Surround Sound format. Especially the creation of low frequency sound in small rooms is in principal problematic, so a Ph.D. study [1] has been conducted on order to come nearer to an understanding and hopefully a solution of the major problems at low frequencies with the use of loudspeakers in small and medium size rectangular rooms, such as normal living rooms, control rooms and small concert halls.

2 Sound in rooms

It is very important to understand how sound interacts with a room if one wants to change or modify this, or to realize if it is possible to change the physical behaviour at all. Sound in a room in contrast to free field is a combination or sum of the direct sound from the source, in this case a loudspeaker, and the enormous number of reflections arriving later that the direct sound. Especially reflections in a room with parallel walls will give very big variation in the Sound Pressure Level (SPL) depending on the position of the sound source, the listening position and the frequency of the sound, not only at the so called modal frequencies, but basically at all frequencies. The modal or resonance frequencies in a room are those frequencies where the propagated and reflected wave will return to the source in phase with the source, and thereby increase the sound level. The lowest resonance frequency between parallel walls is the frequency which wavelength is twice the distance between the parallel walls. Eq. (1) is the well known formula used to calculate the resonance frequencies ($fn+$) in a rectangular room [2].

$$f_{n+} = \frac{c}{2}\sqrt{\left(\frac{n_x}{L_x}\right)^2 + \left(\frac{n_y}{L_y}\right)^2 + \left(\frac{n_z}{L_z}\right)^2} \qquad (1)$$

In Eq.(1) f_{n+} are the modal frequencies given by the dimensions of the room L_x, L_y, L_z and n_x, n_y, n_z are integers starting from 0, 1, 2 .. and c is the speed of sound in the air.

Anti-resonance frequencies ($fn-$) exist when the reflection returns to the source in opposite phase. These modal frequencies have most impact on the sound at low frequencies where the wavelength of the sound is close to the dimensions of the room and they will cause big distinct variations in the sound pressure level (SPL) in the room. At middle and high frequencies the relative number of modal frequencies will increase exponentially and there will be clusters of several individual modal frequencies. The difference in sound pressure level in a room between a resonance frequency and an anti-resonance frequency can exceed 30 dB, but big spatial differences exists more or less at all frequencies.

An IEC standard listening room at Aalborg University (room A) with dimensions: (L x W x H) = (7.08 x 4.12x 2.78m) has the first modal frequencies (n=1) shown in Table 1.

Room A	f_{n+} [Hz]	f_{n-} [Hz]	n_L	n_W	n_H
Length (7.08m)	22	11	1	0	0
Width (4.12m)	41	20	0	1	0
Height (2.78m)	63	32	0	0	1

Table 1. First Resonance frequencies f_{n+} and anti-resonance frequencies f_{n-} for an IEC room (room A)

2.1 Time and Frequency Analysis.

Measuring sound distribution in a room is very time consuming, especially considering more loudspeakers with different placement and the measurements do not really improve the understanding of what is going on. Normally the problem is looked at in the frequency domain, well knowing that sound propagates in time and reflects at obstacles in time and interacts in time. So to get a better understanding and prediction of sound in rooms a simulation program has been made in Matlab, which works in the time domain using Finite-Difference-Time-Domain approximations (FDTD) [3]. Multiple real pre-measured loudspeakers can be placed arbitrary in a virtual room, and a simulation in time and frequency of the sound pressure distribution in the room can be calculated and even animated. The Finite Difference Time Domain approximation has shown well performance and relatively short computation time at low frequencies.

The simulation of room (Room A) with a subwoofer placed at a front wall corner on the floor, and with 25 virtual microphone positions equally spaced by 48 cm at a height of 1.38 m. in a listening area of (1.92 x 1.92) m. in the middle of the room can be seen in Fig.1. The simulation of room A in Fig.1, shows that spatial variation within the 25 different positions at a single frequency can be of more than 30 dB, and that the variation in magnitude from 20 to 100 Hz at a single position can exceed 25 dB. This phenomenon is most audible at low frequencies, and is caused by the room as the sound pressure at any time and at any position in the room is the sum of the direct sound from the loudspeaker(s) and the enormous number of reflections from the room itself. A resonance frequency can be seen at 44 Hz and an anti-resonance frequency at 55 Hz in Fig. 2, but big spatial variations in SPL can be found at basically all frequencies.

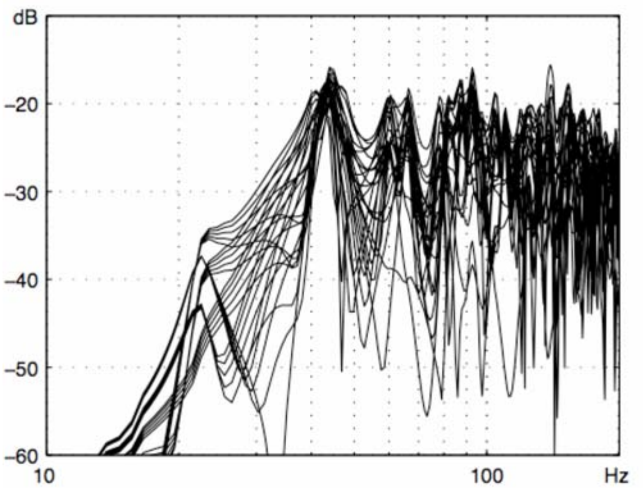

Fig. 1: Simulation of room A with 25 virtual microphone positions and a subwoofer on the floor in a frontal corner

We generally prefer listening to music or sound in a room instead of free field (anechoic) condition with no reflections, but the room also to some extend destroys the good experience by having very large impact on the quality of the sound. So basically it is the reflections in the room that gives the big unwanted variations in SPL, as well as the wanted feeling of being in a room.

3 Optimizing loudspeakers in room

A lot of research is going on in order to improve or optimize sound from loudspeakers in rooms, which is not at all an easy task. The classical solutions are:

- Placement of loudspeakers in the room
- Manual equalization of the signal to the loudspeakers
- Automatically equalization (room correction)
- Improve the quality of the loudspeakers
- Improve the acoustical properties of the room

Each of these solutions change the sound in the room, but there are an enormous number of parameters that can be adjusted, and the question is: What is perfect sound? Should the individual human sound perception be the judge, or objective measurements?

By looking at the simulation of a subwoofer placed in room A (Fig.1) one might wonder why loudspeaker manufactures spend so much effort in constructing the ‘perfect’ loudspeaker down to a fraction of a decibel, when we see what the room does to the sound. The often used solution for an improvement is equalizing using analog or digital signal processing at the signal send to the loudspeakers, either manually [4] or automatically - often named Room Correction. [5],[6],[7]. It is well known that the problem is very complicated. Equalizing the electrical signal sent to the loudspeaker can change the frequency spectrum of the radiated power from the loudspeaker, but it can not change the room’s manipulation of the sound. As an example take any single frequency at fig. 1 and change the gain at this frequency, say - 6 dB, that will change the amplitude at this frequency with - 6 dB at all positions in the room, but the spatial variations in the room at that frequency will not have changed at all, only the level. Another problem in equalizing is: where is the target point? One-point equalization will normally make the sound worse in any other listening position in the room. [5]

It is a good rule to solve an acoustical problem at the source, but in this case the sound from the loudspeaker is not the problem – the room is.

If the objective is to make a more even distribution of the sound pressure level (SPL) in a big listening area with no audible peaks and dips, the focus should be placed on the room’s interaction with the sound in the time domain, and not on equalizing the signal to the loudspeakers in the frequency domain.

The spatial distribution of the sound at low frequencies can be modified by the placement of the loudspeakers. Proper symmetrical placements of 2 low frequency loudspeakers in a simulation of room A can create a plane wave travelling towards the back wall, where it will be reflected towards the front wall for a new reflection and so on. To create a plane wave, the best placement of 2 loudspeakers at the front wall will be at the height $z = \frac{1}{2}Lz$, and at the width $x = \frac{1}{4}Lx$ and $\frac{3}{4}Lx$. see Fig 2. This placement of the loudspeakers will together with the mirror sources create a huge line array, with mainly longitude reflections at low frequencies. The simulations of room A can be seen at fig. 3 at a height z = 1,38m.

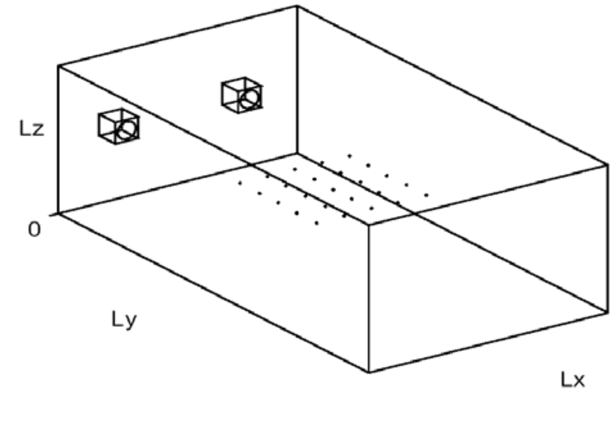

Fig.2: Setup of loudspeakers to create a plane wave.

The simulation is done at 44 Hz which is a resonance frequency with $ny+ = 2$, and at 60 Hz, which is not a resonance or anti-resonance frequency. The simulations show very big spatial variation in SPL along the length of the room, not only at the resonance frequency. Also to be observed is that at a resonance frequency (44 Hz) the SPL in general is much higher than at the arbitrary 60 Hz frequency.

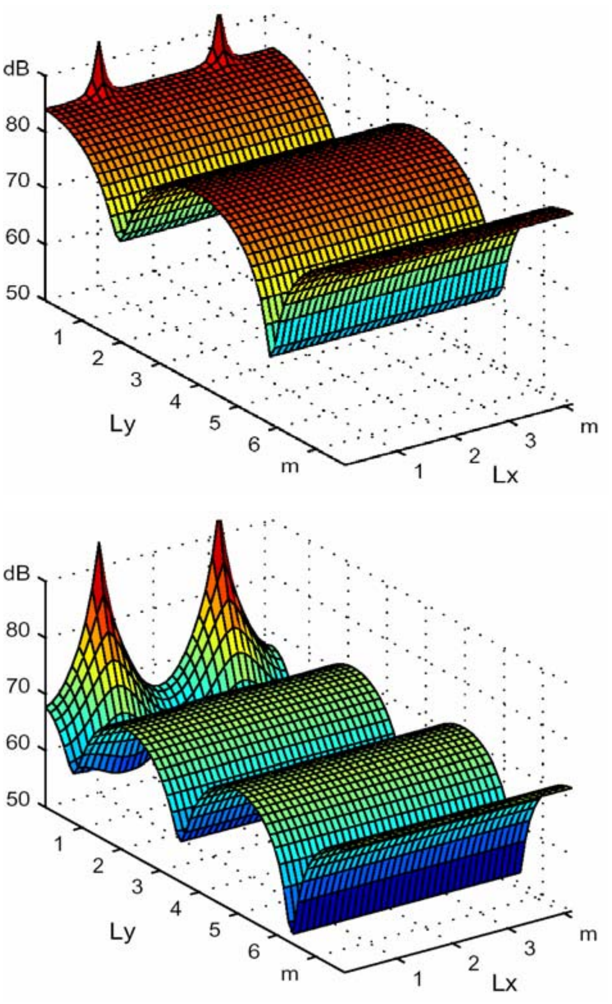

Fig.3: Sound pressure level distribution from a simulation of room A (see Fig. 2) at the height of z = 1.38m. Upper at 44 Hz a resonance frequency. Bottom at 60 Hz. The loudspeaker setup is 0.2.0. (see chapter 4.1)

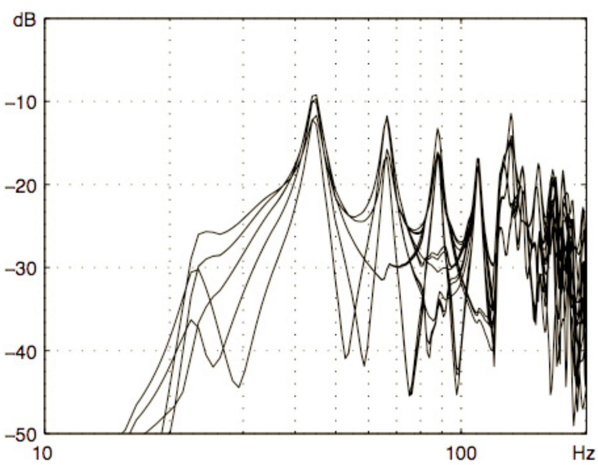

Fig.4: Simulation of room A setup 0.2.0 at 25 points.

A plane wave will create an even SPL distribution in the width of the room up to 100 Hz, which has been simulated in Fig. 4 where the SPL is simulated at the 25 microphone positions of the listening area. Only 5 curves are seen as the 5 microphone positions in a row in the width of the room have the same SPL. The calculated cumulative spectral decay (CSD) at one virtual microphone position (17) of the simulation can be seen in Fig. 5. It is obvious that the SPL at the resonance frequencies is higher, but also the duration in time is much longer. At resonance frequencies this booming bass will be very audible.

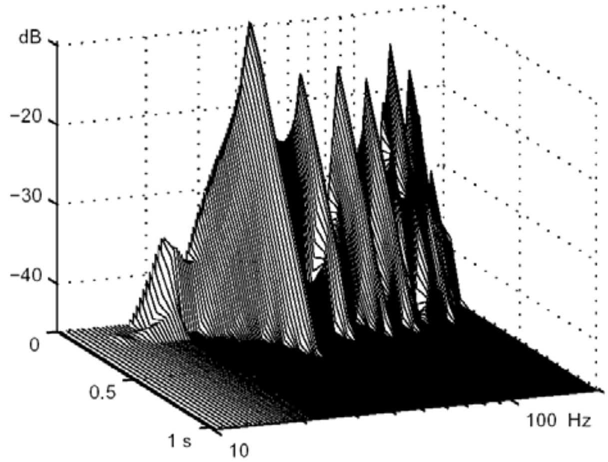

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Fig.5: Simulated Cumulative Spectra Decay, room A

4 CABS

By proper symmetrical placement of the loudspeakers a plane wave can be constructed. The construction of the plane way does not create a more homogeneous sound field in the room, but it gives a more simple field which might be homogeneous if it was not for the reflections at the back wall. Simulations of rooms A have shown, that if the back wall is removed the sound field will be homogeneous in the whole room up to 100 Hz. The obvious question is then: Is it possible acoustically to remove the back wall, or at least the back wall reflection?

Simulations with the FDTD simulation program shows, that by placing similar symmetrical loudspeakers at the back wall, and in principal feed these loudspeakers with delayed version of the signal to the front loudspeakers but in opposite phase and with the proper amplitude, then it is possible to remove the reflection from the back wall. This technique is known from active noise cancellation but here modified for another purpose. This System has been named CABS (Controlled Acoustically Bass System) and the principal block diagram can be seen in Fig. 6.

4.1 CABS notation.

In order to make it easier to identify the number of loudspeakers and their rough placement a notation is introduced:

CABS Fr. F. B.

Fr = Number of front wall loudspeakers (normally stereo).
F = Number of Front wall low frequency loudspeakers.
B = Number of Back wall low frequency loudspeaker.

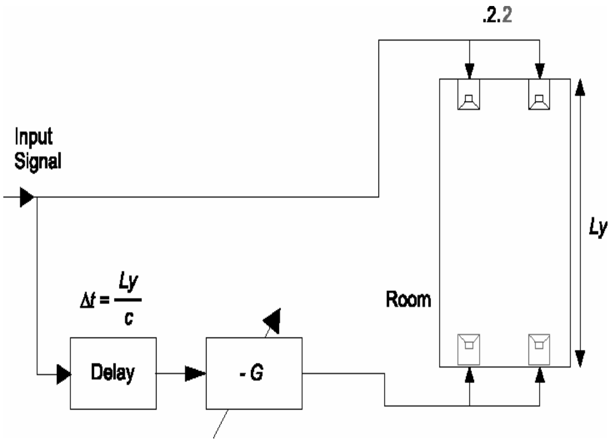


Fig.7: Block diagram of the implementation of CABS 0.2.2 to minimize the reflection from the back wall.

As CABS basically works at low frequencies it must be combined with the higher frequency area as well. The front loudspeakers can be full range loudspeakers as in CABS 2.0.2 or in a combination of front mid/high-range loudspeakers and front sub-woofers as CABS 2.2.2.

4.2 Simulation of CABS

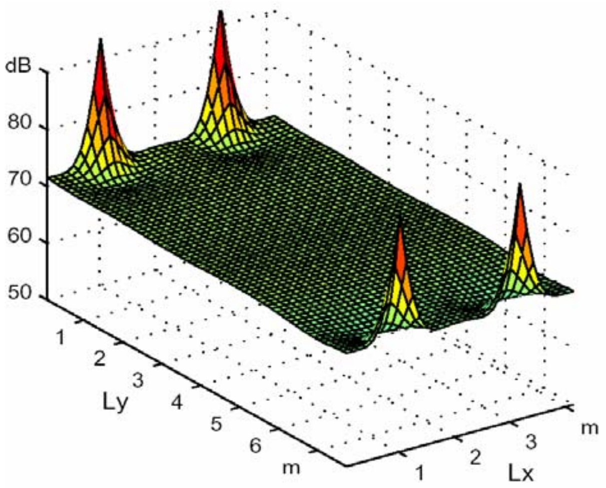
Simulations of room A with CABS 0.2.2 with 2 front sub woofers and 2 cancelling subwoofers at the back wall can be seen in Fig 8. The simulation of the SPL is made in a horizontal plane at the height z = 1.38 m at 2 frequencies: 44 Hz being the 2nd modal frequency in the length of the room, and at 60 Hz which is not a modal frequency. It can clearly be seen that the SPL is homogeneous in the whole room, not only in a restricted area.

By comparing Fig. 3 with CABS 0.2.0 and Fig 8 with CABS 0.2.2 both showing a simulation of room A at the same frequencies the improvement is clear, a homogeneous field can be created using CABS 0.2.2. The next step is to implement CABS and measure it in real rooms, to see and hear if the results in real life are as good as the simulations.

4.3 Implementation of CABS

CABS has been implemented in a DSP system made from scratch based on a 20 MIPS TMS320c50 16 bit fixed point signal processor, together with 4 channel of AD/DA converters running with 44,1 kHz sampling frequency [8]. The DSP system is built as a prototype with many options for various configurations of CABS. The system can take a normal stereo analog input signal e.g. from a CD-player, and deliver the analog outputs to the individual channels.

The DSP does all the signal processing, which includes cross over filters for splitting up the signals for the individual loudspeakers. CABS can be automatically as well as manually calibrated for the best performance, and calibration is only necessary at the setup of the system, or if the position of a loudspeaker or the room has been modified.



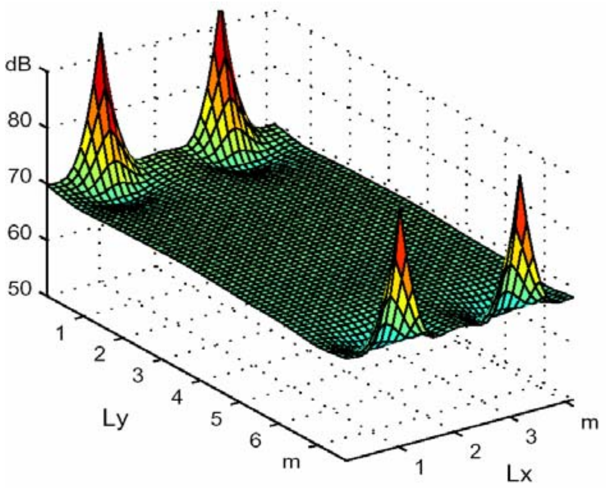


Fig.8: Sound level distribution from a simulation of room A with CABS 0.2.2 at a height z = 1,38m .
Upper at 44 Hz, a modal frequency for ny = 2.
Bottom at 60 Hz, not a modal frequency.

4.4 Measurement of CABS.

CABS 0.2.2 has been implemented in the real room A (the IEC standard listening room at Aalborg University) with the same loudspeaker placement as simulated with the FDTD program. The frequency spectrum has been measured at the same 25 microphone positions as in the simulations of room A. In Fig 9 the measurement of CABS 0.2.0 from the 25 microphone positions can be seen (with back loudspeakers turned off) and the cumulative spectral decay calculated from the measured data at one position (17) can be seen in Fig. 10. (the position is the same as the simulated position in Fig. 5 which is number 4 from the left in the 2nd row from the back wall.)

Measurements of CABS 0.2.2 (with the back loudspeakers turned on) at the 25 positions can be seen in Fig 11 showing a huge improvement compared with Fig 9. Up to 100 Hz the sound field in the listening area are almost homogeneous, and also as simulated in Fig 8 in the rest of the room. CSD has been calculated from the measurement in position 17 and can be seen in Fig. 12 showing a very short room impulse response at low frequencies.

CABS has also been simulated and measured with similar big improvements in a bigger room at Aalborg University a ITU multi channel listening room (L x W x H) = (8,12 x 7,39 x 2.88 m) = 173 m^3. [1]

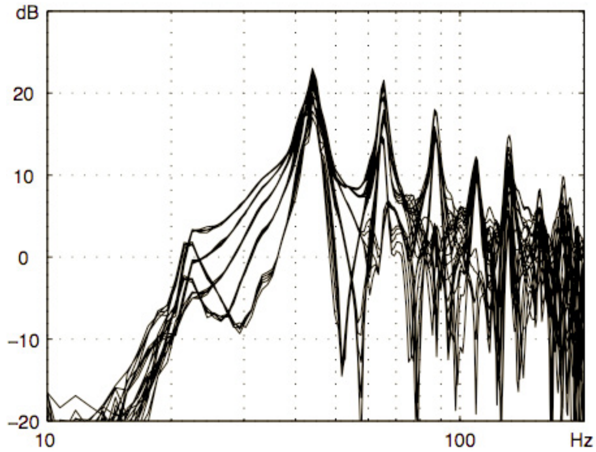


Fig.9: Measurements in the IEC room (room A) with CABS 0.2.0 with frequency response at 25 positions.

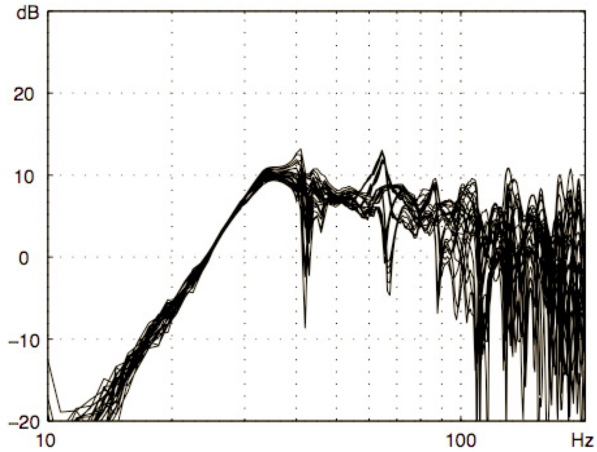


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Fig.11: Measurements in room A with CABS 0.2.2 with frequency response at 25 positions.

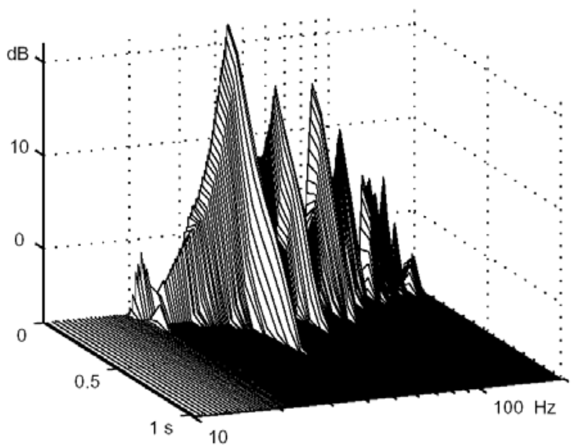

Fig.10: Cumulative spectral decay (CSD) measured at one position (17) with CABS 0.2.0 in the IEC room (room A).

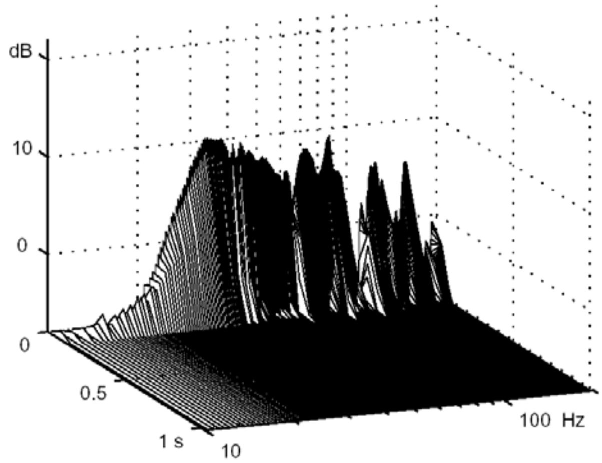

Fig.12: Cumulative spectra decay measured at position 17 with CABS 0.2.2 in room A.

5 Conclusion

CABS is a novel method to remove or reduce reflections in a rectangular listening room, and make a homogeneous sound field at low frequencies not only in a limited listening area, but in the whole room. By looking at the problem as an acoustical problem in the time-domain and solve it acoustically in the time-domain it will work for all frequencies as well as for transient signals. CABS will get rid of the resonance and anti-resonance frequencies at low frequencies, the sound gets clearer or sharper, less muddy and without booming bass and surprisingly the back loudspeakers are not heard at all.

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