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Applications of a plane wave based room correction system for low frequencies using multiple loudspeakers

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When low frequency sound is radiated inside small listening spaces by loudspeakers, large uniformities occur over the sound field. This is due to the multiple reflection, diffraction and scattering of sound on the walls and different objects in the room. A developed system named Controlled Acoustically Bass System (CABS) produces uniform sound field at low frequencies. This is performed by utilizing loudspeakers at the front wall and extra loudspeakers at the opposite wall, processed to remove the rear-wall reflection of a rectangular room. Effectiveness of CABS on a different room scenario has been evaluated by using a computer simulation program based on the Finite Difference Time Domain Method (FDTD). Non–ideal placement of loudspeakers in CABS have been evaluated. CABS has been simulated in an irregular room.

1 Introduction

Full range loudspeakers are typically used for high fidelity sound reproduction. These systems are typically placed in small or medium size listening spaces e.g. listening rooms, control rooms for studios, home theater rooms, cars etc. At the listener position the spectral response of the loudspeaker is extremely modified over the full frequency range. This is due to the combination of the direct sound and the multiple reflection, diffraction and scattering of sound at the walls and different objects in the sound path. Especially at low frequencies the sound level distribution over the room will experience differences of more than 20 dB. Mid and high frequencies can be controlled by acoustic treatment of the room, but when the loudspeaker radiates longer wavelengths e.g. from 10 m to 3 m (34 Hz – 114 Hz) the acoustic solutions become unpractical.

To tackle these problems several approaches have been investigated by a number of authors. Some efforts have been conducted towards the analysis and optimization of the placement of the loudspeakers in the room [1–4]. Other approaches have been directed to control the acoustic radiation power of the loudspeaker [5], by means of digital signal processing but most of the investigations have been conducted on the correction of the loudspeaker–room response by digital filters [6,7]. Another approach also making use of digital filters presented in [8] has been implemented in a low frequency sound reproduction chamber at Aalborg University. This approach is based on the simulation of a plane wave in a small room by the use of 2 x 20 loudspeakers build into two opposite walls. Every loudspeaker is controlled by an independent amplification channel and 72 Finite Impulse Response (FIR) filters. Differently from the solution proposed in Santillan's work the system presented in this paper utilizes less loudspeakers and can be implemented in larger rooms with a much simpler setup.

The idea of the Controlled Acoustically Bass System (CABS) is to built a plane wave traveling towards the opposite wall with the front loudspeakers by optimizing their placement. This will produce uniform sound field distribution only if the reflection is then cancelled out by similar loudspeakers at the rear wall with a delayed version of the signal but in anti–phase maintaining the plane wave along the room [9–11].

The main goals of this paper is to present how much precision is needed on the placement of the loudspeakers when utilizing CABS in rectangular rooms, also to present possible applications of CABS in irregular rooms.

2 Methods

2.1 Low frequency room simulation

Generally the problem of low frequency sound in rooms has been widely analyzed by the well–known formulations called *modal decomposition techniques* mainly based on the complex sound pressure in steady-state. Differently in this work the problem is inspected by a model based in the time domain. Other methods based on geometrical acoustics such as the Mirror Image model or Ray Tracing, are no longer sufficient when the wavelength is comparable with the dimensions of the room. A computer simulation program based on an element method was developed and described in [10] and [9]. The simulation program is based on the finite–difference time–domain method (FDTD) [12–14]. This model solves the linear lossless wave equation but in addition it applies the relation between the particle velocity and the acoustic pressure known as the force equation. The main difference with other methods is that both equations (lossless wave equation and force equation) calculate particle velocity and pressure as a function of time. In this fashion these two equations are utilized to compute the acoustic pressure produced by a number of sound sources in the entire enclosure. Since the particle velocity is always available the boundary conditions are defined by calculating the wall impedance from estimated absorption coefficients α and the normal component of the particle velocity to the wall. With this computational program written in MATLAB the sound field produced by multiple loudspeakers in a rectangular room can be calculated. Moreover irregular room shapes can also be modeled [9].

2.2 Mean sound field deviation

In order to quantify the deviations on the sound field distribution along the listening area the parameter *Mean Sound Field Deviation (MSFD)* is computed. This measure is calculated from the sound pressure levels on discrete frequencies at the 25 positions within the listening zone. The *MSFD* is expressed in dB as

$$MSFD = \left[SD \pm \ dB, MD \pm \ dB\right] \qquad (1)$$

and is conformed by two numbers, the *Spatial Deviation* (*SD*) which indicates the deviations within the space in ± dB and the *Magnitude Deviation* (*MD*) which reveals the magnitude spectral deviations also in ± dB.

To calculate this parameter the SPL at each discrete frequency (with a resolution of 1 Hz) is sampled at each microphone position. Then the whole listening area is represented in a table where the rows are the listening positions and the columns are the discrete frequencies from 20 Hz to 100 Hz. Next the standard deviation on each frequency column is calculated so that the Spatial Deviation SD is the mean of all standard deviations at the discrete frequencies along positions as expressed in

$$SD = \frac{1}{n_f} \sum_{i=f_{low}}^{f_{high}} \sqrt{\frac{1}{n_p - 1} \sum_{p=1}^{n_p} (X_{SPL_{p,i}} - \overline{X}_{SPL_i})^2} \quad [dB] \tag{2}$$

where p is the microphone position and i is a discrete frequency on the range between $f_{low} =$ 20 Hz to $f_{high} =$ 100 Hz, $X_{SPL_{p,i}}$ is the SPL in dB at microphone position p at the discrete frequency i and $\overline{X}_{SPL_i}$ is the mean of the SPL's of all microphone positions p at the discrete frequency i, n_p is the number of microphone positions and n_f is the number of discrete frequencies.

In the same fashion the standard deviation is calculated on each row position so that the Magnitude Deviation *MD* is the mean of all standard deviations on individual positions along frequencies as expressed in

$$MD = \frac{1}{n_p} \sum_{p=1}^{n_p} \sqrt{\frac{1}{n_f - 1} \sum_{i=f_{low}}^{f_{high}} (X_{SPL_{p,i}} - \overline{X}_{SPL_p})^2} \quad [dB] \tag{3}$$

where p is the microphone position and i is a discrete frequency on the range between $f_{low} = 20\,\text{Hz}$ to $f_{high} =$ 100 Hz, $X_{SPL_{p,i}}$ is the SPL in dB at microphone position p at the discrete frequency i and $\overline{X}SPL_i$ is the mean of the SPL's of all discrete frequencies i at microphone position p, n_p is the number of microphone positions and n_f is the number of discrete frequencies.

2.3 Controlled Acoustically Bass System

The Controlled Acoustically Bass System (CABS) is a plane wave based loudspeaker–room correction system that uses extra rear loudspeakers in order to give uniform sound field at low frequencies within the room. In order to identify the number of loudspeakers and their placement the following notation is introduced

$$CABS\ Fr.F.B. \tag{4}$$

where Fr is the number of front wall loudspeakers (normally stereo), F is the number of front wall low frequency loudspeakers and B is the number of back or rear wall loudspeakers. The principle of CABS is to build up a plane wave in one end of the room with the two front loudspeakers $(.F..)$ and at the opposite wall of the room cancel the reflection out with extra loudspeakers $(..B.)$. These extra loudspeakers are fed with the same signal as the front loudspeakers but in anti–phase including a delay $\Delta t \approx Ly/c$ (see Fig. 1). In addition to the delay the gain G of the extra loudspeakers has to be adjusted due to the attenuation of sound by the traveling distance and the damping characteristics of the room.

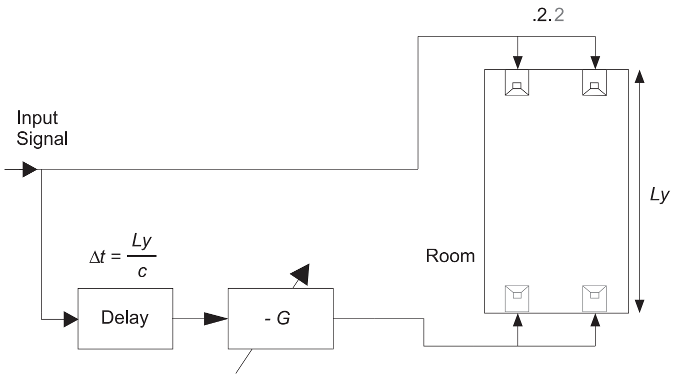


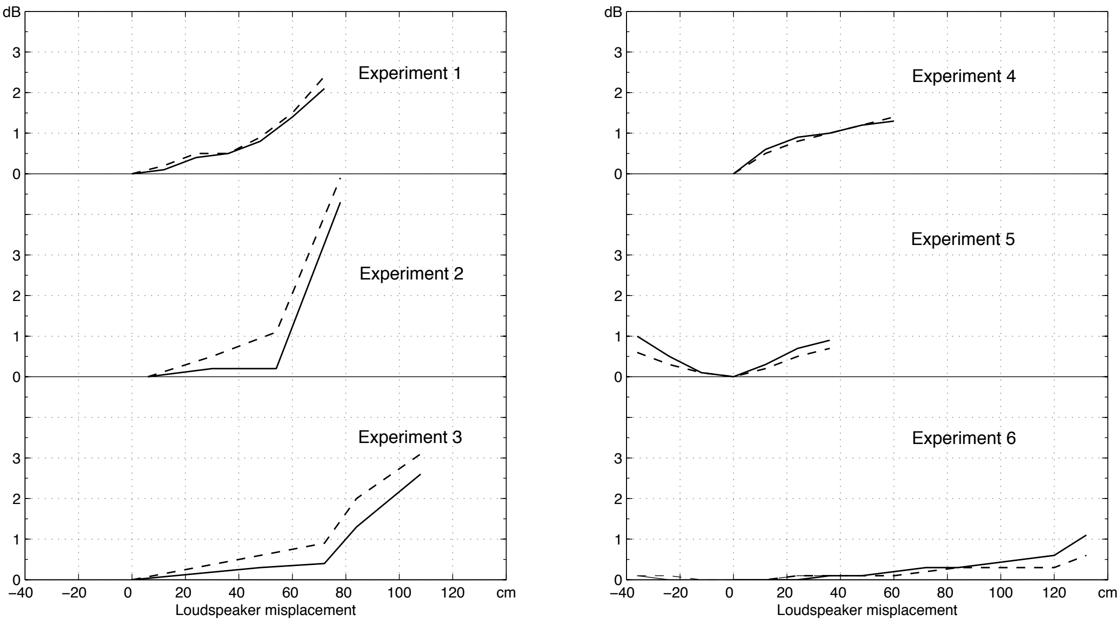
Figure 1: Block diagram of CABS .2.2, G is a factor according to the damping characteristics of the room and the attenuation of sound by the air.

In a rectangular room of width Lx, length Ly and height Lz CABS .2.2 will typically have the front loudspeakers (Fr or F) placed at $x = Lx\,1/4$, $x = Lx\,3/4$, $y = 0$ and at $z = Lz/2$. This manner the loudspeakers act as a line source in the horizontal plane up to approx. 100 Hz, therefore forming an uniform sound field within the room. The back or rear loudspeakers (B) are placed at $x = Lx\,1/4$, $x = Lx\,3/4$ respectively and at $y = Ly$ and $z = Lz/2$ to cancel only the reflection of the rear wall maintaining the plane wave along the room. Simulations and measurement of CABS .2.2 have shown good performance in small and middle size rectangular rooms. In contrary to the advanced room correction systems that typically optimize to a single or a few listening positions CABS .2.2 can achieve good responses not only in a single listening position but also over most of the room having values of *MSFD* in a large listening area of only $\left[SD \pm 1.3\ dB, MD \pm 2.1\ dB\right]$ in an ITU standard listening room and $\left[SD \pm 1.6\ dB, MD \pm 2.1\ dB\right]$ in an IEC standard listening room up to 100 Hz [9, 11].

3 Results

3.1 Loudspeaker Misplacement

To establish how much precision is needed on the placement of the loudspeakers simulations of CABS .2.2 have been performed in a rectangular room with similar dimensions as the IEC standard listening room at Aalborg University, being width $Lx = 4.20\,\text{m}$, length $Ly = 7.8\,\text{m}$ and height $Lz = 2.76\,\text{m}$. The loudspeakers are modeled as pulsating cubes of volume equal to 12 cm^3 inside the room. An absorption coefficient $\alpha = 0.12$ has been set on each wall. The sound field is sampled over the listening area of (1.92 × 1.92) m centered in the room and delimited by 25 virtual microphones equally spaced by 48 cm at a height of $z = 1.38$ m. The deterioration of the sound field over the listening area has been estimated as the difference between the actual *MSFD* computed and the best *MSFD* obtained at the optimal loudspeaker position. The *MSFD* has been calculated at a height $z = Lz/2$, the signals to the loudspeakers are not changed. Six different experiments have been simulated misplacing the loudspeakers on gradual steps:



(a) (b)

Figure 2: Deterioration of the Mean Sound Field Deviation (*MSFD*) over the listening area resulting from simulations of different loudspeaker misplacements. (a) Experiment 1 to 3. (b) Experiment 4 to 6.

1.- The rear loudspeakers (. .*B*.) have been misplaced towards the front wall 72 cm on steps of 12 cm from their optimal position at the rear wall.

2.- The front (.*F*. .) and the rear (. .*B*.) loudspeakers have been moved inside the room on steps of 24 cm up to 78 cm from their optimal placement at the front and rear wall.

3.- The front (.*F*. .) loudspeakers have been moved inside the room on steps of 48 cm up to 108 cm from their optimal position at the front wall.

4.- The rear loudspeakers (. .*B*.) have been separated from each other towards the left and right side wall on steps of 12 cm up to 60 cm away from from their optimal position.

5.- Only the right rear loudspeaker (. .*B*.) has been misplaced -36 cm from its optimal placement to the right wall. Then it has been moved to the left side wall on steps of 12 cm up to +36 cm of misplacement towards the right side wall.

6.- The height of the rear loudspeakers has been altered from 36 cm above the optimal position down to the floor on steps of 12 cm.

The result of these experiments is shown in Fig. 2 where on each experiment the misplacement in cm against the deterioration of the *MSFD* over the listening area from the optimal loudspeaker position is outlined.

3.2 CABS in Irregular Rooms

"L" shape room

By using the FDTD method basic irregular room shapes can be simulated. The simulation model of the rectangular room used in section 3.1 has been modified in order to built an "L" shape room with dimensions $Lx = 7.08\,\text{m}$, $Ly = 7.8\,\text{m}$, $Lz = 2.76\,\text{m}$ and $Lx_c = 4.20\,\text{m}$, $Ly_c = 4.5\,\text{m}$ (see Fig. 4(a)). The loudspeakers are modeled as pulsating cubes of volume equal to 12 cm^3 inside the room. The sound field is sampled over a listening area of (1.92×1.92) m centered in the room and delimited by 25 virtual microphones equally spaced by 48 cm at a height of $z = 1.38$ m. The simulation of a low frequency loudspeaker (subwoofer) placed at 1.62 m from the front wall F and 1.26 m from the lateral wall E on the floor in the irregular room is shown in Fig. 3. In Fig. 5 two loudspeakers have been simulated as in a standard stereo setup, each acoustic centre of the loudspeaker was located at approx. 1 m away from walls A and E respectively, at $x = 1.02\,\text{m}$ and $x = 3.18\,\text{m}$, and both at $y = 1.50\,\text{m}$ and height $z = 1.38\,\text{m}$. Both

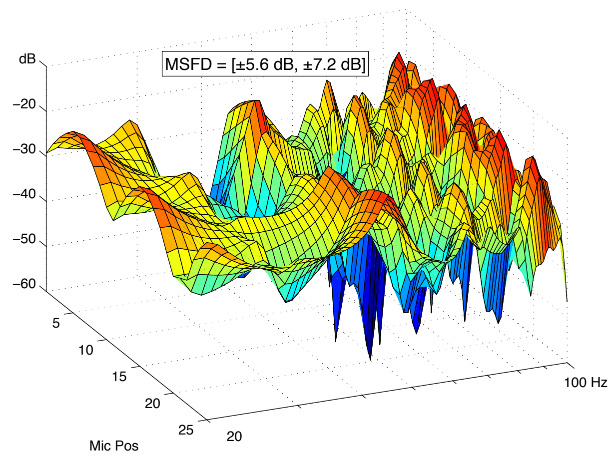


Figure 3: Mean sound field deviation at the 25 positions over the listening area in the irregular room using one loudspeaker at 1.62 m from the front wall F and 1.26 m from the lateral wall E on the floor.

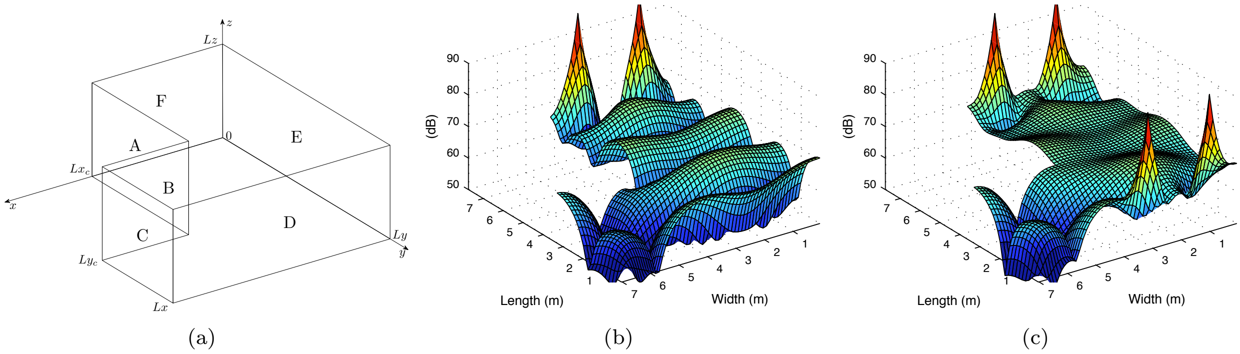


Figure 4: (a) Irregular room model. (b) Sound pressure level distribution in the room calculated on a plane at a height of $z = 1.38$ m, driven frequency 82 Hz using only the front loudspeakers at the wall F in the irregular room. (c) Sound pressure level distribution in the room calculated on a plane at a height of $z = 1.38$ m, driven frequency 82 Hz applying CABS .2.2 with two extra loudspeakers at the rear (D) wall.

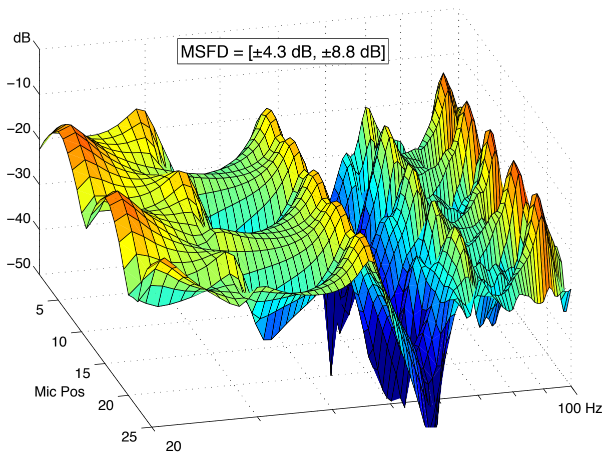


Figure 5: Mean sound field deviation at the 25 positions over the listening area in the irregular room using two loudspeakers as in an standard stereo setup, each loudspeaker was located at approx. 1 m away from walls A and E respectively, and both at $y = 1.50$ m.

loudspeakers producing the same signal. As it can be observed both situations present highly uneven sound field having differences of more than 30 dB between positions. These two loudspeakers have now been positioned at the front wall F in the room. Each acoustic centre of the loudspeaker was located at $y = 0.06$ m, and $z = 1.38$ m height maintaining 1/4 of the width from each lateral wall. It is assumed that both loudspeakers produce the same signal. The result of the simulation of this setup is presented in Fig. 6 (a). As it can be seen the sound field is extremely uneven. There are differences in magnitude of about 30 dB e.g. from 44 Hz to 55 Hz where deep notches appeared since the longitudinal room modes are strongly exited therefore the sound pressure level distribution will be different all over the room.

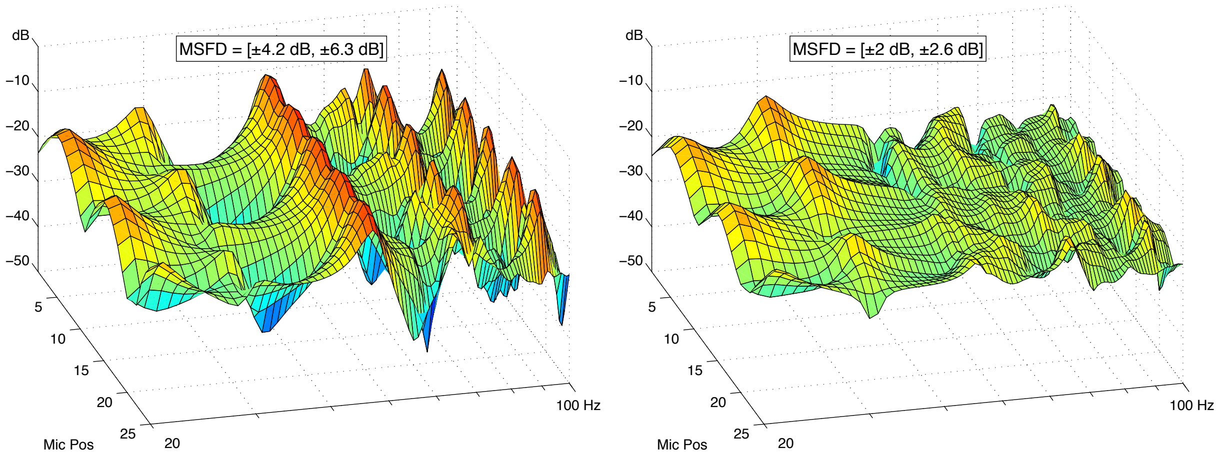
The benefit of placing the front loudspeakers at the front wall F at one quarter of the width from each lateral wall and at the same height, is that the loudspeakers and the reflections from the lateral, floor and ceiling will contribute to form an horizontal line array radiating traveling plane waves along the room (see Fig. 4 (b)). This will bring uniform level distribution along the width of the room at low frequencies, but to maintain the uniformity of the sound field the reflection of the back wall has to be canceled out. This is done by applying CABS .2.2 with the extra loudspeakers at the rear wall at $x = 3.90$ m and $x = 6.06$ m, and both at $y = 7.74$ m, and $z = 1.38$ m processed to cancel the back wall reflection (see Fig 4 (c)).

After applying CABS one can verify that the sound field is much more even over the listening area and in a large extension of the room having magnitude deviations of only ± 2.6 dB and spatial deviations of ± 2 dB (see Fig. 6 (b)). The system is less effective at frequencies from 20 Hz to 40 Hz due to part of the traveling waves along the room will diffract at the corner formed by wall A and B and will reflect to the wall C being unable to be canceled out.

4 Discussion and Conclusions

As seen from the simulation experiments the performance of CABS .2.2 in a rectangular room is very sensitive to misplacement of the loudspeakers away from the rear and front walls. On the contrary CABS .2.2 is less sensitive to lateral misplacement of the rear loudspeakers. The performance of CABS seems to be less sensitive to alteration of the height of the rear loudspeakers. The misplacement of loudspeakers will affect the mean sound field deviations mostly at higher frequencies near to 100 Hz. The implementation of CABS .2.2 in an irregular room presented fairly good performance comparing with standard loudspeaker sound reproduction at low frequencies. The sound field over the listening area and most of the room become more even up to 100 Hz. A possible optimization solution on these room shapes may be the addition of an extra loudspeaker at the wall C (see Fig. 4(a)) with the same signal as the rear loudspeakers but with a different delay and amplitude.

The system works acceptable in small and middle size irregular rooms. In contrary to the advanced room correction systems that typically optimize to a single or a few listening positions CABS .2.2 can achieve good responses not only on a single listening position but also over most of the room. Since the system works in the time domain it works at all frequencies up to around



(a) (b)

Figure 6: Mean sound field deviation at the 25 positions over the listening area. (a) Using two loudspeakers at the front wall in the irregular room. (b) Applying CABS .2.2 with two extra loudspeakers at the rear D wall.

100 Hz depending on the size and shape of the room and the number of loudspeakers utilized. The smaller the room the higher in frequency it works.

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