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LOUDSPEAKER ARRAYS FOR FAMILY TV

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As the hearing capabilities of humans are degraded with age, it becomes more difficult for people of different ages to enjoy television together, as they need different audio volumes. A 60 years old person needs a 10 dB boost at high frequencies, on average, in order to be able to listen properly. This paper describes an approach to obtain this boost. A line array using 8 phase-shift sources has been designed and constructed. This array produces a boost of 10 dB in a narrow spatial zone, where the person with the hearing loss is placed. The response in other directions is minimized to reduce the excitation of the reverberant field in the room, with the phase-shift sources controlling the radiation to the rear of the array. Free field simulations were used to design the array and also investigate various algorithms for the broad band reproduction of the audio signal. The real-time performance of the experimental array was then tested in free field and realistic environmental conditions.

1. Introduction

Our hearing system degrades with aging, especially at high frequencies, due to the loss of active amplification processes that occur in the inner ear. This degrading is known as *Presbyacusis* and leads to a severe deficit in auditory perception[1]. Little previous attention has been paid to the TV hearing environment for hearing impaired. The normal situation in a living room where members of a family with different ages watch TV, is that these members require different program volumes. Whilst the young members of the family will be happy with a lower volume, the old members would like to turn up the volume in order to listen to the TV clearly and without difficulties.

Applications and hearing aids which improve the intelligibility for hearing impaired TV listeners are present in the market, however, apart from the use of headphones, these do not provide a proper control of the overall sound field. According to the standard ISO 7029[2] a 60 years old female, on average, has a hearing loss of about 20 dB around 6 kHz.

In this paper we present a personal audio[3] application in order to address this problem. This personal audio system generates a bright acoustic zone[4, 5] in a spatial area where the people with hearing loss are placed, *bright zone*, and is designed not to excite the rest of the room, where people with normal hearing are watching the TV, *dark zone*. In order to amplify correctly the program that is sent to the hearing impaired, a linear procedure in which the amplification provided is the half of the hearing loss is used, which is known as *half gain rule*[6]. Using this procedure 10 dB of amplification are needed, which means that our target is an acoustic contrast of 10 dB from 500 Hz to 6 kHz between bright and dark zones.

The personal audio device presented here consists in a line array that uses small phase shift sources. The line array is going to be used in broadside, or close to broadside configuration, with the loudspeakers pointing towards the TV viewers. The directivity of a line array is proportional to the ratio between the array's aperture and the radiation wavelength[7], which for a small array is very low at low frequencies. In order to overcome this lack of directivity the superdirective array processing is introduced. Superdirective techniques have been widely researched for sensor arrays[8, 9], and have been applied previously to source arrays[10]. Although superdirective techniques greatly increase the directivity of an array at low frequency, they require a huge power amount to perform this increase. However, by the use of regularisation[11] the amount of superdirectivity used can be controlled, with what an effective directivity using a reasonable amount of power can be obtained[12]. A superdirective approach is used here, known as acoustic contrast maximisation [4]. This technique is based in the maximisation of the ratio of squared pressures between bright and dark zones. It has been recently used for applications as an active headrest[13] and a sound system for a mobile phone [5].

An important facet of the personal audio system presented here is the use of phase shift loudspeakers as the sources for the array. As a broadside array of omni directional sources radiates symetrically around its axis, it is needed to neglect the array's back radiation in order to decrease the power input to the reverberant field. The only way to decrease this radiation is to use an extra back source that will attenuate the arrays back radiation, however this means that the required filters for the array are doubled [14, 15]. The phase shift sources are based on a specific cabinet construction which creates an acoustic phase shift network[16] which leads to a cardioid type directivity pattern[17], resulting in a highly directional source.

2. Control strategy: The acoustic contrast maximisation

We assume that the sound field we need to control is sampled in a set of control points a certain distance from the line array. In order to determine the pressure at each control point, the individual transfer impedances between each source and each control point are used. The sound field is divided into two spatial areas, the bright zone, and the dark zone, where for every single frequency two correspondent control matrices are created.

The transfer impedances matrix correspondent to the bright zone is defined as $\mathbf{Z}_B$, which is a $(N \times M)$ matrix, where N denotes the number of control points in the bright zone and M for number of control sources:

$$\mathbf{Z}_B = \begin{bmatrix} Z_{11} & Z_{12} & \cdots & Z_{1M} \\ Z_{21} & Z_{22} & \cdots & Z_{2M} \\ \vdots & \vdots & \ddots & \vdots \\ Z_{N1} & Z_{N2} & \cdots & Z_{NM} \end{bmatrix}. \tag{1}$$

Another transfer impedances matrix is created for the dark zone, $\mathbf{Z}_D$. This an $L \times M$, where L represents the number of control points in the dark zone:

$$\mathbf{Z}_D = \begin{bmatrix} Z_{11} & Z_{12} & \cdots & Z_{1M} \\ Z_{21} & Z_{22} & \cdots & Z_{2M} \\ \vdots & \vdots & \ddots & \vdots \\ Z_{L1} & Z_{L2} & \cdots & Z_{LM} \end{bmatrix}. \tag{2}$$

The pressure vectors for bright and dark zone, $\mathbf{p}_B$ and $\mathbf{p}_D$, are obtained by multiplying the respective transfer impedances matrix by the optimal vector of complex source strengths $\mathbf{q}$

$$\mathbf{p}_B = \mathbf{Z}_B\mathbf{q} \text{ and } \mathbf{p}_D = \mathbf{Z}_D\mathbf{q}. \tag{3}$$

The performance of the array is measured in terms of acoustic contrast, C, which is defined by the ratio of squared pressures between bright and dark zones, i.e.,

$$C = \frac{\mathbf{p}_B^H \mathbf{p}_B}{\mathbf{p}_D^H \mathbf{p}_D} = \frac{\mathbf{q}^H \mathbf{Z}_B^H \mathbf{Z}_B \mathbf{q}}{\mathbf{q}^H \mathbf{Z}_D^H \mathbf{Z}_D \mathbf{q}}, \tag{4}$$

where ^{H} denotes the hermitian complex conjugate transpose. By using this equation, the performance of the line array can be evaluated.

The maximum value for the cost function C of Eq. 4 is obtained by solving a constrained optimisation where $\mathbf{p}_D^H \mathbf{p}_D$ is minimized with the condition that $\mathbf{p}_B^H \mathbf{p}_B$ is held constant to a value c. Using Lagrange multipliers the function to be minimized with respect to $\mathbf{q}$ and the Lagrange multiplier λ is

$$J = \mathbf{q}^H \mathbf{Z}_D^H \mathbf{Z}_D \mathbf{q} - \lambda(\mathbf{q}^H \mathbf{Z}_B^H \mathbf{Z}_B \mathbf{q} - c). \tag{5}$$

By setting the differential of this function with respect to the complex and real parts of $\mathbf{q}$ to zero the following equation is obtained

$$\mathbf{Z}_D^H \mathbf{Z}_D \mathbf{q} - \lambda \mathbf{Z}_B^H \mathbf{Z}_B \mathbf{q} = 0. \tag{6}$$

That rearranging gives

$$\mathbf{q} = \lambda [\mathbf{Z}_D^H \mathbf{Z}_D]^{-1} \mathbf{Z}_B^H \mathbf{Z}_B \mathbf{q}. \tag{7}$$

The optimal vector of source strengths is obtained when $\mathbf{q}$ is proportional to an eigenvector correspondent to the largest eigenvalue of the matrix $[\mathbf{Z}_D^H \mathbf{Z}_D]^{-1} \mathbf{Z}_B^H \mathbf{Z}_B$[4]. If frequency dependent regularisation is used the above equation can be rewritten as

$$\mathbf{q} = \lambda [\mathbf{Z}_D^H \mathbf{Z}_D + \beta I]^{-1} \mathbf{Z}_B^H \mathbf{Z}_B \mathbf{q}. \tag{8}$$

where the optimal vector of volume velocities is given by the eigenvector corresponding to the maximum eigenvalue of the matrix $[\mathbf{Z}_D^H \mathbf{Z}_D + \beta I]^{-1} \mathbf{Z}_B^H \mathbf{Z}_B$[18].

3. Free field simulations

In order to study the behavior and predict the performance of a line array, free field simulations using a M=8 sources line array with sources spaced 4 cm have been conducted. This simulations have been calculated using the formulations of Section 2 and are analysed in terms of C, the acoustic contrast, and array effort, E. The array effort is defined as the norm of the optimal set of source strengths divided by the source strength that a monopole needs to obtain the same acoustic pressure as that produced by the array in the center of the bright zone, q_M, i.e.,

$$E = \frac{\mathbf{q}^H \mathbf{q}}{|q_M|^2}. \tag{9}$$

The array effort determines the electrical power required to drive the array, assuming there is no electro-acoustical interaction between the sources. The acoustic contrast and array effort are dimensionless magnitudes whose levels are plotted here in dB.

The simulations introduced here have been carried out in a 3D control sphere with a quasi equal area distribution for each microphone. The geometry, together with the single and double rows of sources being considered, are shown in Fig. 1.

3.1 Point monopoles with hypercardioid directivity

These simulations have been conducted using point hypercardioid sources, in order to be able of estimating the performance that was able to be obtained using phase shift sources. The free field transfer function of a monopole is defined as

$$Z = j\omega\rho_0 \frac{e^{-jkr}}{4\pi r}, \tag{10}$$

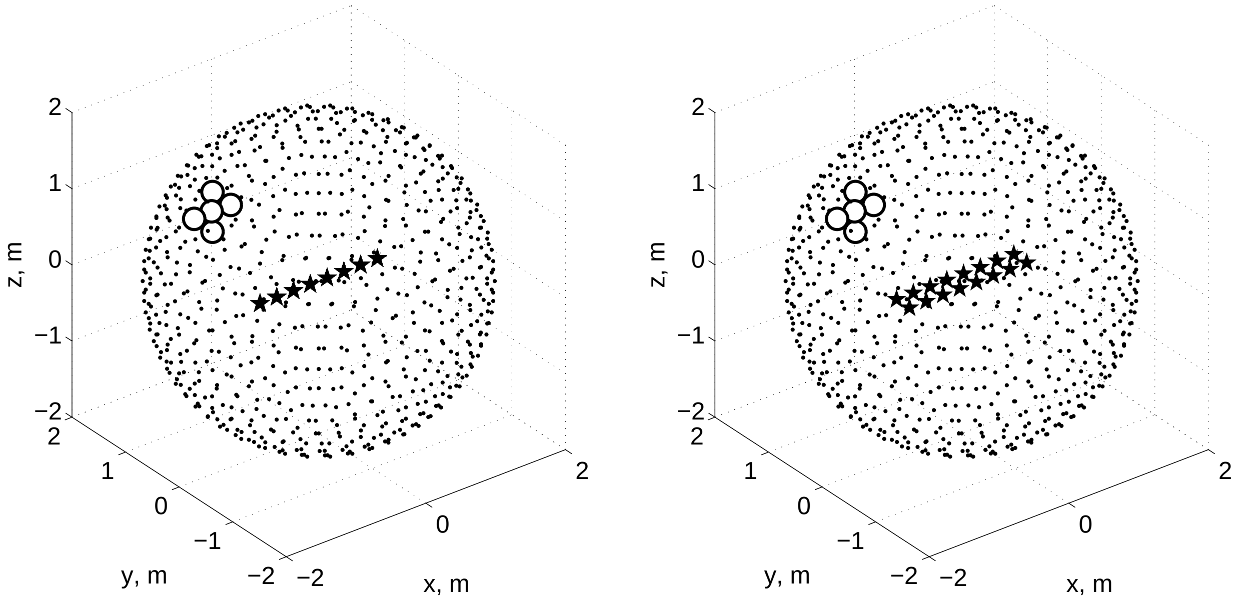

Figure 1: Geometric arrangement for double monopole and hypercardioid simulations. Black points represent the dark microphones. The white large points represents the bright microphones and the black stars represent the sources of the array.

where $j = \sqrt{-1}$, ω represents the angular frequency, k represents the wavenumber, $\rho_0 = 1.20$ kgm^{-3} is the air density at 20° C and r stands for the distance between the monopole and the spatial point where the transfer function is being measured. The far field directivity pattern of a gradient source is given by

$$D(\psi, \theta, \phi) = 1 - \psi + \psi \cos\theta \cos\phi, \tag{11}$$

where ψ is the directivity parameter. By the product theorem[19] Eq. 10 can be multiplied with Eq. 11 to give a transfer impedance that depends on θ and ϕ, i.e.,

$$Z_D = ZD(\psi, \theta, \phi) = j\omega\rho \frac{e^{-jkr}}{4\pi r} \left(1 - \psi + \psi \cos\theta \cos\phi\right). \tag{12}$$

If ψ=0.75 the source behaves as a hypercardioid source, whose radiation pattern maximizes its directivity index and hence has a minimum sound power input to the reverberant field [20].

3.2 Double monopoles array

In this case a double array, $2 \cdot M$ =16 monopole sources are used, as shown on the right hand side of Fig. 1. In this case the strengths of the back row sources are a delayed copy of the strengths of the front row sources, i.e.,

$$q_{m_2} = -q_{m_1} e^{-jkd_y/3}, \tag{13}$$

where the sub subscript 1 and 2 represents the front and rear row respectively and d_y represents the separation between front and rear sources. If the delay is selected to be $d/3$ a hypercardioid pattern is obtained at low frequencies[21]. Another design strategy is to use acoustic contrast maximisation to optimize independently its sources strengths according to the respective control geometry.

3.3 Comparison of Results

Fig. 2 show the comparison of results of the various design approaches described above. On the plots corresponding to the unlimited array effort, it is possible to see how, below 1kHz, the acoustic contrast for all the approaches described above provides an increase of around 6 dB with respect to and optimised array of 8 monopole sources, at the expense of a large increase in array effort. Going up in frequency, the approach based in two rows of independent monopoles achieves a contrast higher than 10 dB until around 8 kHz. The double row approach, based in delayed monopoles, presents a big null of acoustic contrast centered at 6.5 kHz, the frequency at which d_y is approximately 0.75 times the radiated wavelength. The point hypercardioid sources give the best result above 1 kHz, providing the highest value of acoustic contrast.

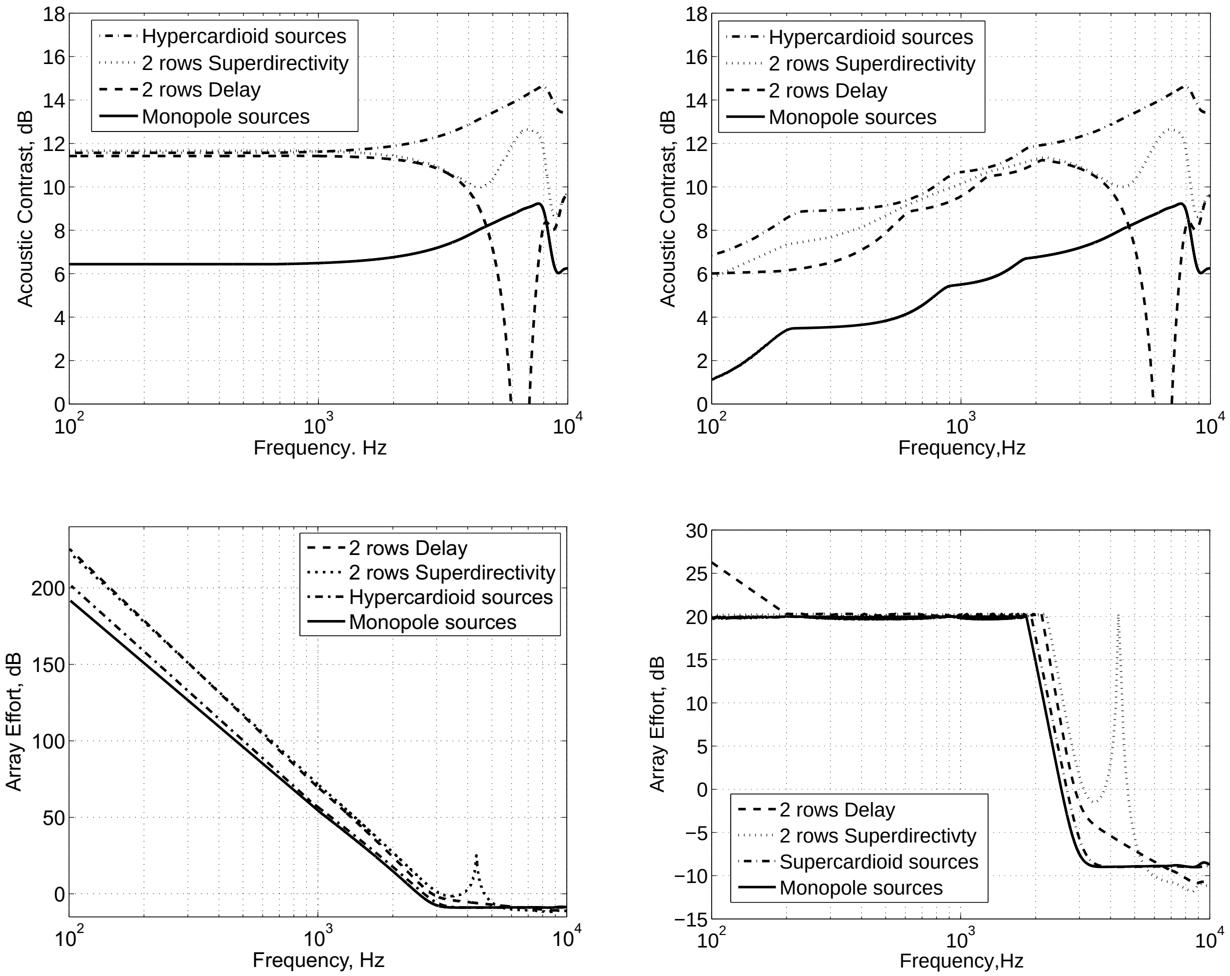

Figure 2: Comparison of acoustic contrast and array effort obtained in the geometries of Fig. 1. The plots of the left show the case when the array effort is unlimited and the plots of the right show the case when the array effort is limited to 20 dB.

When the array effort is limited to 20 dB, by adjusting the parameter β in Eq. 8 at each frequency, which represents a more realistic approach, the results based on point hypercardioid sources give a higher figure of acoustic contrast compared with the approaches based on double rows. The array based on hypercardioid sources then represents the best approach, needing only M filters and M sources, what achieves a simpler and more robust device.

4. The prototype array

Based on the previous simulations, an array of M=8 phase shift sources has been constructed and its performance measured in free-field conditions. The array's sources are placed 46 mm apart, so that the array presents a total aperture of 375 mm. The control geometry where the array's performance has been measured can be observed on the right hand side of Fig. 3.

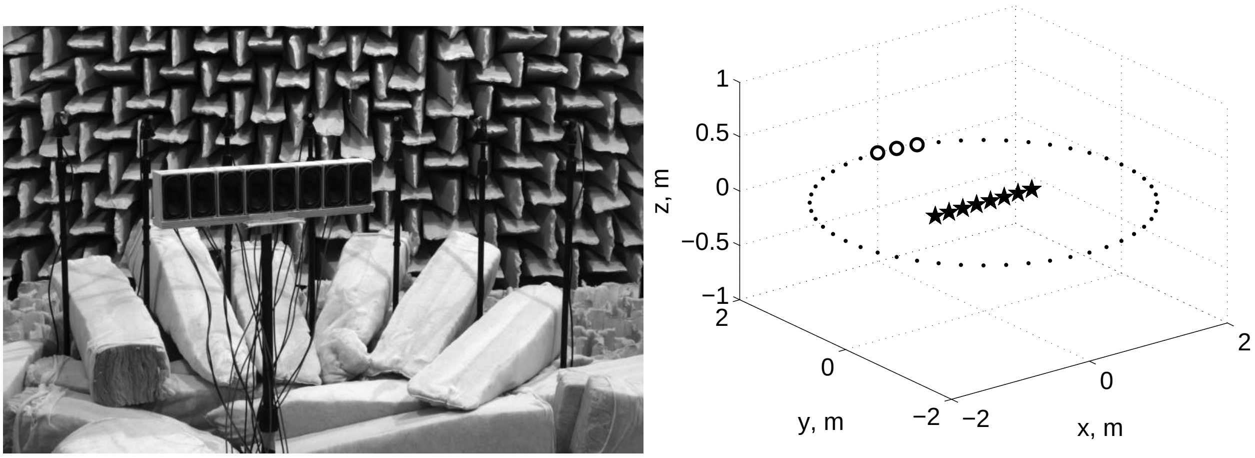

Figure 3: The prototype array in the anechoic chamber (left), and control zone used for the measurements in the anechoic chamber (right).

4.1 Phase shift drivers

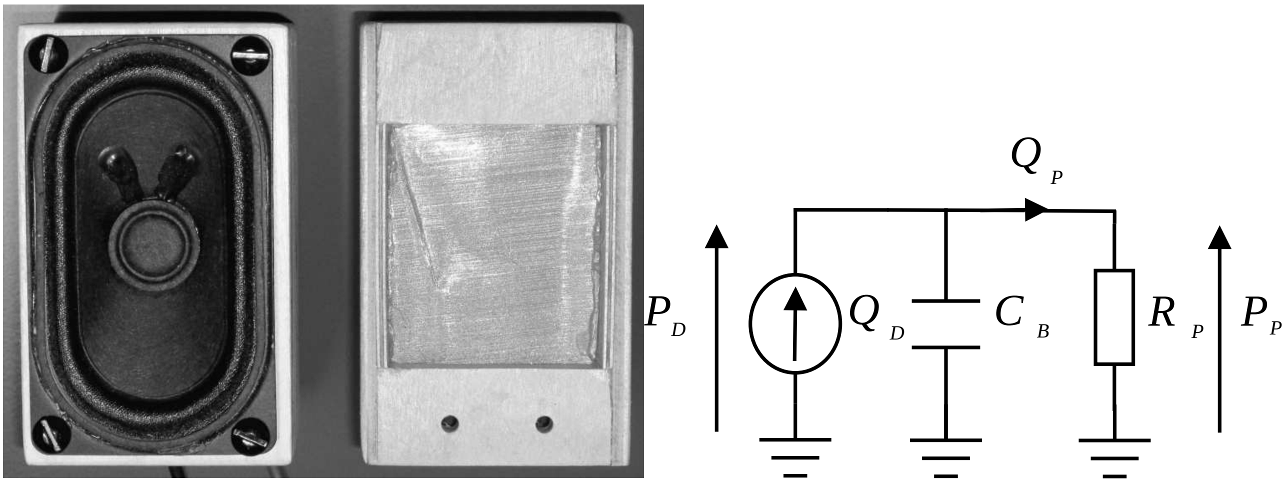

Figure 4: Close up of one of the phase shift sources used in the array (left), and low frequency electroacustical model of a phase shift source (right).

The phase shift drivers used to build the array are shown on the left hand side of Fig. 4. These sources are constructed using a small cabinet with an aperture at its back covered with an acoustic resistive material, so that the network formed by the cabinet's volume and the aperture resistance create a phase shift network. The delay of the phase shift network is then controlled by the size of the aperture at the back of the cabinet.

An electroacustic model of the phase shift loudspeaker can be used to approximate the behaviour at low frequencies. The model is shown in the right hand side of Fig. 4, in which Q_D represents the volume velocity created by the source's diaphragm, C_B is the acoustic compliance of the cabinet's volume, R_P the port's resistance and Q_P the resultant volume velocity on the cabinet's aperture, which can be written as

$$Q_P = -Q_D \frac{\left(\frac{1}{R_P} + j\omega C_B\right)^{-1}}{R_P} = -Q_D \frac{1}{1 + j\omega R_P C_B}, \tag{14}$$

where if $\omega < \frac{1}{R_P C_B}$ port and diaphragm velocities can be related by a delay, being both volume velocities related as

$$Q_P \approx e^{-jkc_0 R_P C_B}(-Q_D). \tag{15}$$

4.2 Real time measurements

Fig. 5 shows the result of a simulation using point hypercardioid sources together with an off-line simulation with the measured transfer impedances of the prototype array with phase shift sources in anechoic conditions, for the control geometry of Fig. 3. The real time measurement of acoustic contrast on the same control geometry is also included, obtained by driving the array with white noise filtered by 1000 coefficients filters designed to match the frequency responses via the acoustic contrast maximisation algorithm. It can be observed how the real time measured acoustic contrast is

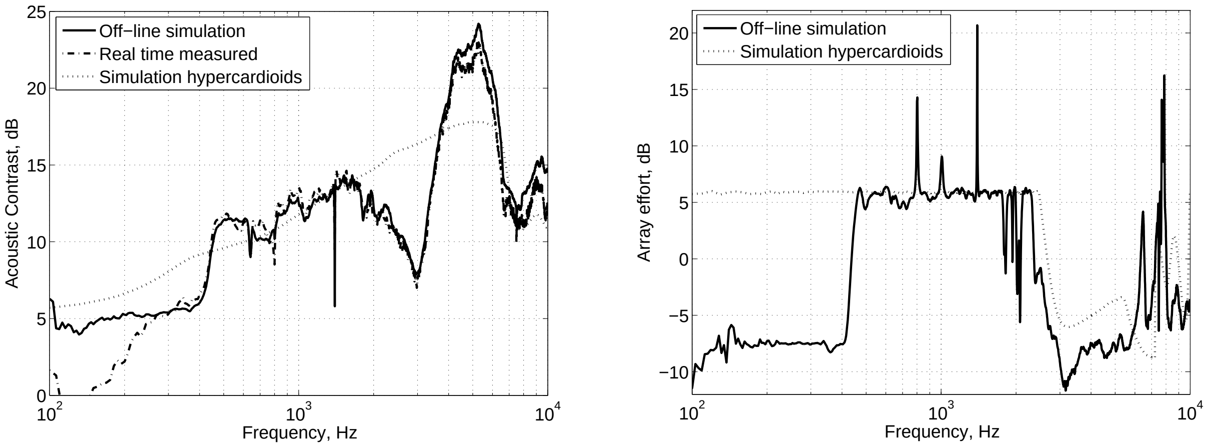

Figure 5: Comparison between the measured and simulated performance of the array in free field conditions using acoustic contrast maximisation.

greater than 10 dB for almost all the frequency band, for an array effort lower than 6 dB. It can also be observed that the real time respones closely match the off-line simulations. Below 1.5 kHz the array behaves similarly to the results of the simulation using hypercardioid sources. The phase shift sources present a null of its directional characteristics centered at about 2.8 kHz, which is similar to that null seen for the delayed monopole model in Fig. 2, but at a lower frequency since diffraction around the enclosures increases the effect separation. However above 4 kHz the sources start to be more directional due to beaming of the individual loudspeakers, and the contrast offered by the prototype array is greater than the simulation with hypercardioid sources. This contrast decreases up to 6.5 kHz, which represents the frequency at which spatial aliasing starts to occur.

5. Conclusion

The performance of different approaches of reducing the back radiation of the array has been studied. The use of sources with hypercardioid directivity offers the best acoustic contrast performance for a lower array effort on a 3D geometry. This approach is predicted to provide an acoustic contrast greater than 10 dB above 800 Hz, in a practical arrangement, with a 3D geometry.

A prototype line array using 8 phase shift sources has been constructed and measured in anechoic conditions. This array is able to achieve an acoustic contrast of more than 10 dB for almost the whole of the desired frequency band, which makes it a good candidate for the target application of boosting TV sound for the hearing impaired.

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