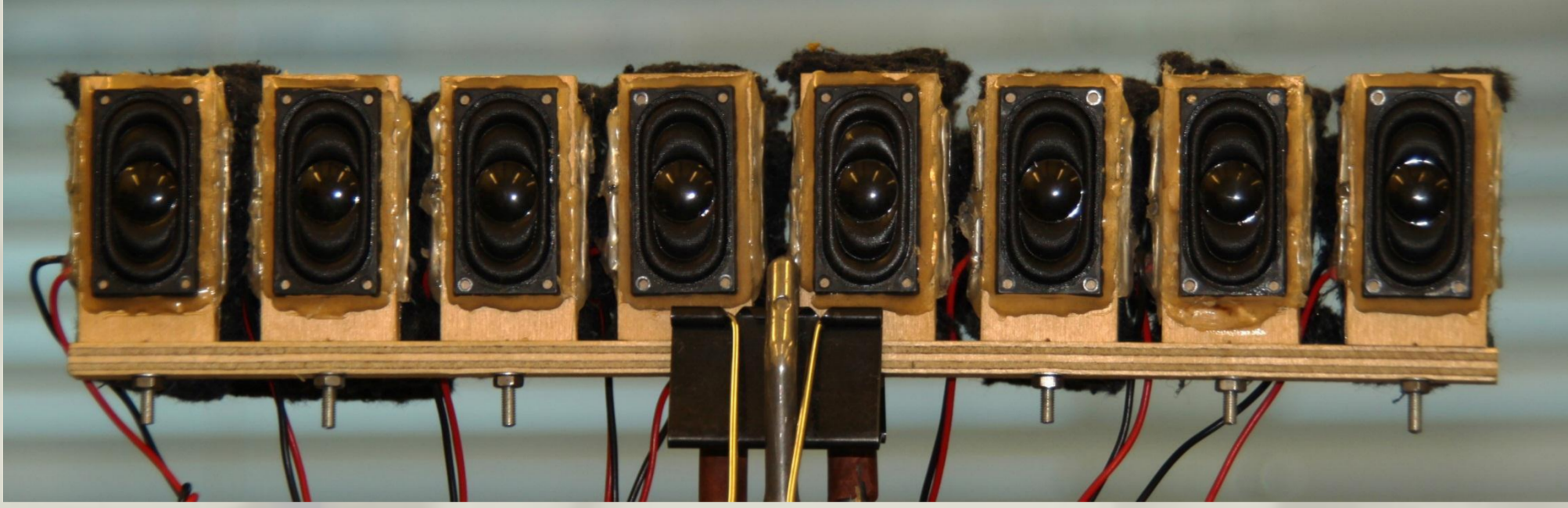# A superdirective array of phase shift sources

by Marcos F. Simón Gálvez & Stephen J.Elliott

isvr UNIVERSITY OF Southampton

The array as a Hearing Aid

A personal audio[1] application based on a miniature line array is being developed as an aid for the hard of hearing. The motivating application in this research is to boost the audio of a TV in a specific direction, and hence provide audio programs with levels adapted to the hearing of the TV viewers.

Based on the standard ISO 7029[2] and in a basic amplification procedure known as half gain rule 10 dB of amplification are needed at 3 kHz in order to overcome the hearing loss of a 70 years old person. Although this is a particular example of such a directive audio source, there are many other applications where it can be used.

By applying superdirective beamforming techniques[3], as the acoustic contrast maximisation[4] or the least squares inverse method[5], the performance of the array is greatly improved at low-medium frequencies, compared to the performance that is obtained with a conventional delay and sum array. Both superdirective techniques maximise the difference of mean acoustic pressure between two control zones, a *bright zone* with a maximum of acoustic energy where the hearing impaired person is placed, and a *dark zone* where the energy is wanted to be minimised.

These control techniques allow thus to provide a boost of the acoustic energy in one zone whilst that by minimising the energy radiated to the dark zone, the input to the reverberant field is also reduced, which is important to maintain a high acoustic contrast between bright and dark zones inside a real environment.

The phase shift source

In order to minimise the input to the reverberant field, it is needed that the line array is directive in a 3D sense. This means that the radiation of the array is also needed to be controlled along the axis *y* and *z*, which which here is implemented with a phase shift source.
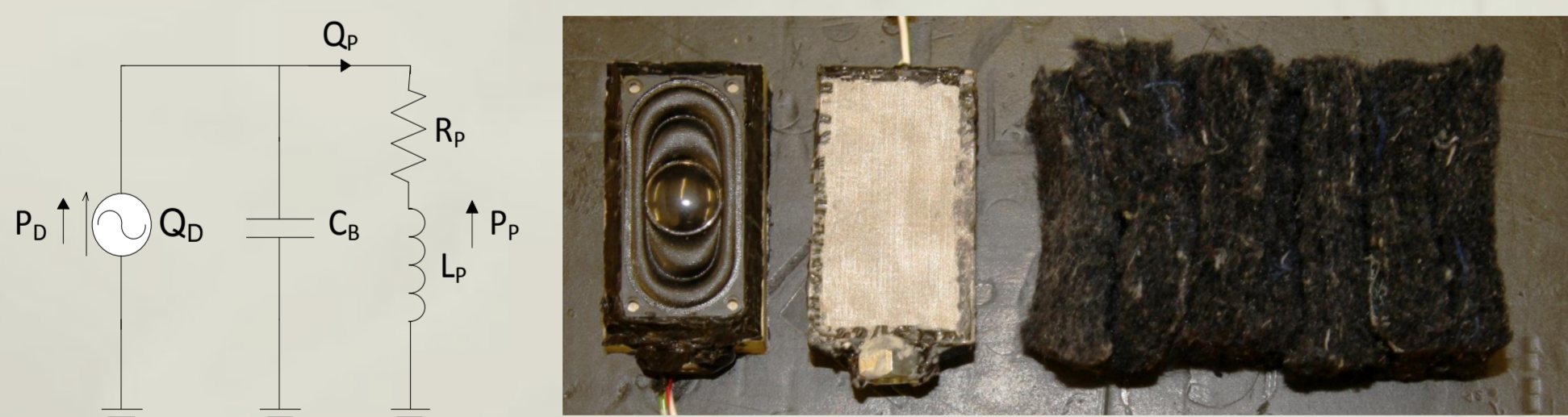
Fig 1. Lumped parameter model of phase shit source (left) and view of a phase shift loudspeaker with the inertance L_P (right).

The volume velocities of driver, Q_D, and port, Q_P, are related as

$$Q_P = \frac{-Q_D}{1 + j\omega R_P C_B - \omega^2 L_P C_B}.$$

By carefully selecting the values of R_P, C_B and L_P, a gradient loudspeaker with a cardioid type radiation pattern is obtained at the low frequencies, when the loudspeaker port is restive. At higher frequency, the effect of the inert000000ance L_P increases, and the source radiates as a closed box. By combining these two approaches, a directional radiator that works during a wide frequency range is obtained.

Results

A performance term is introduced here. This term is denoted as acoustic contrast and measures the ratio of mean square acoustic pressure between the desired control zones, it is defined as

$$C = \frac{N_D}{N_B}\frac{\boldsymbol{p}_B^H \boldsymbol{p}_B}{\boldsymbol{p}_D^H \boldsymbol{p}_D},$$

where N_B and N_D are the number of control points and $\boldsymbol{p}_B$ and $\boldsymbol{p}_D$ are the pressure vectors of bright and dark control zones respectively.
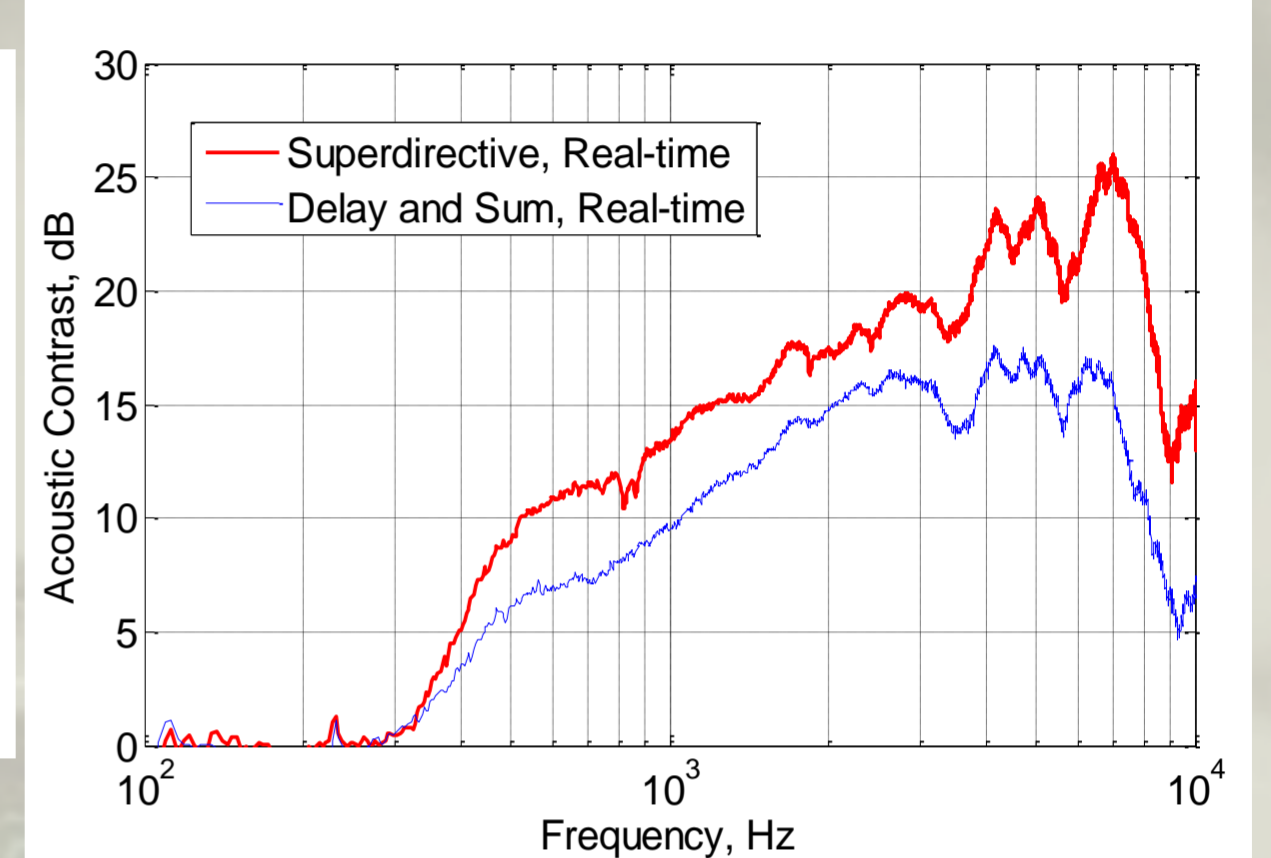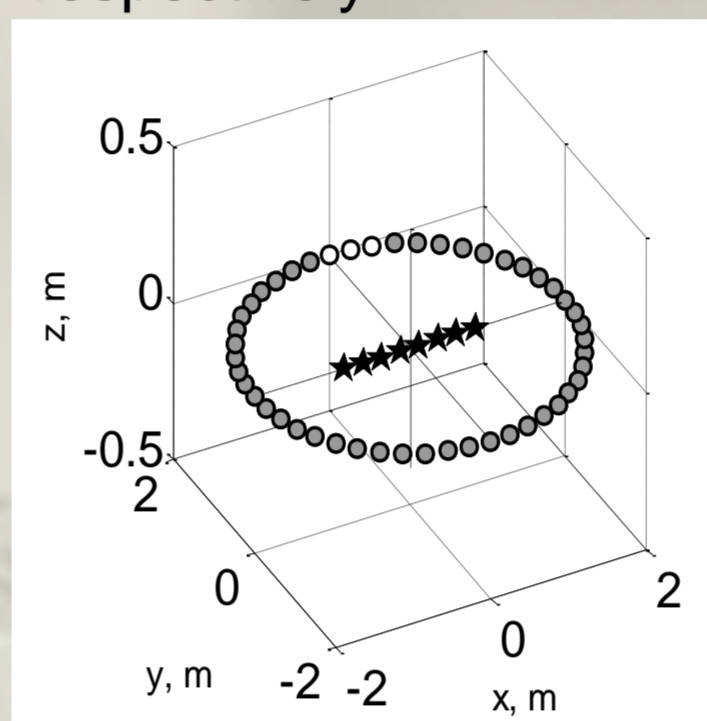
Fig. 1. Control geometry in which the performance of the array is measured. Three microphones constitute the bright zone (white points) and 45 microphones form the dark zone (grey points), all of them spaced 7.5°.

Fig. 2. Free-field acoustic contrast performance on the geometry of Fig. 1 when a superdirective approach, in this case the acoustic contrast maximisation, is used to drive the array's filters. The filters are limited to not to use more than 6 dB of the energy that a single source uses to create the same acoustic pressure on the bright zone. The performance of a delay and sum is also shown for comparison.

| 1000 Hz | 2000 Hz | 4000 Hz | 6000 Hz |
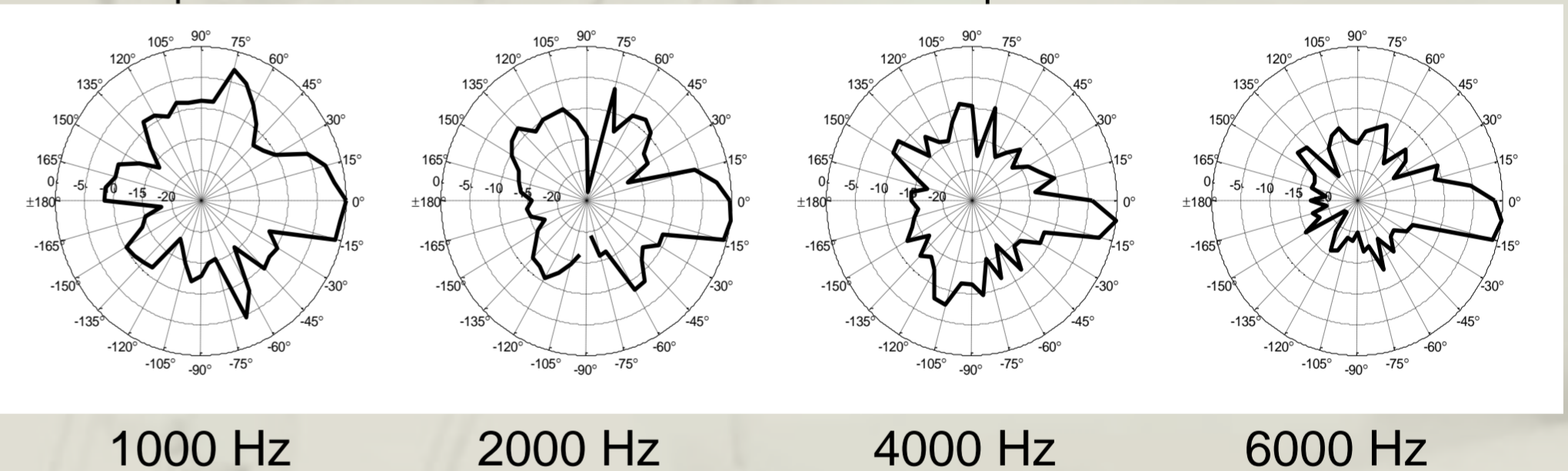
Conclusions

The phase shift sources have proved to form an efficient way of neglecting the array's back radiation.

An 8 sources loudspeaker array using phase shift sources that allow to obtain an acoustic contrast higher than 10 dB between 500 Hz and 8 kHz has been constructed .

Although the performance of the array in free-field is much greater than the original 10 dB target, its study in real environment conditions needs to be continued, in order to provide a 10 dB contrast in reverberant mediums.

References

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