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Design and Measurement of First-Order, Horizontally Beam-Controlling Loudspeaker Cube

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ABSTRACT

This paper describes a loudspeaker cube with four transducers on its horizontal facets, designed to enable sound radiation with adjustable first-order beam control. Design and equipping of the cubical loudspeaker is presented along with two open data sets containing multipe-input-multiple-output impulse responses (MIMO-IRs) of our measurements. The first one contains 648×4 MIMO-IRs from input voltages to a grid of microphones at a fixed distance. The second set contains 4×4 MIMO-IRs from input voltages to loudspeaker cone velocities, and it characterizes the active and passive transducer coupling through the enclosure that we aim to equalize/decouple. Based on these measurements we present a simple FIR filter design required for beam control of which we discuss operation range and limitations.

1 Introduction

The loudspeaker cube discussed in this paper means to allow first-order beam control with a freely adjustable shape and direction of radiation, similar as, e.g., in Laitinen et al [1]. It was created with the idea in mind to provide a simple alternative to the icosahedral loudspeaker IKO [2]. A new target application especially suiting the loudspeaker cube is outlined in [3], where, aside from more elaborate control, the use of four loudspeaker cubes is suggested as a quadra-quadraphonic surround-with-depth system.

In section 2 we discuss the geometry of the device, followed by the description of the two measured impulse response sets. The first data set in section 3 is used for visualizing directivity patterns, which are enhanced by circular harmonic interpolation. The beam-control method is presented in section 4. The second data set, a

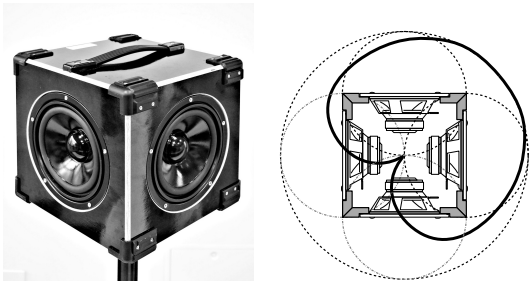


Fig. 1: First-order beam-controlling loudspeaker cube.

Laser-Doppler vibrometry (LDV) measurement of the loudspeaker cones' velocities, is described in section 5. We use its results to prove the independent control of all operation modes and for deriving a FIR filter required for beam control. The measurements show where the proposed beam control is effective and where its limitations are. All measurement data is made available

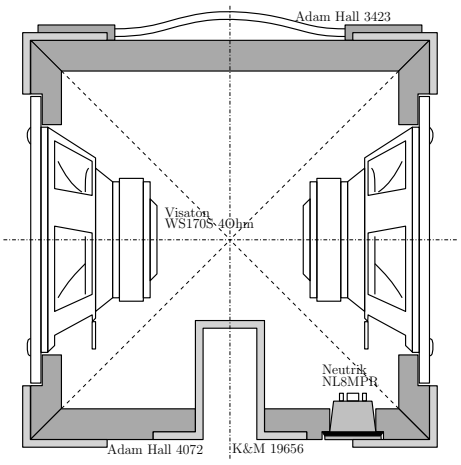


Fig. 2: Vertical cross section specifying the transducers, connector, handle, corners, flange adapter.

freely for further research [1].

2 Design

The cubical loudspeaker consists of a cube-shaped enclosure with a side length of 26 cm made of 18 mm thick plywood. To each of the four cardinal directions, there is a circular opening, 18.6 cm in diameter, each one accomodating a Visaton W 170 S driver (see Fig. 2). The top plane is equipped with a handle, and in the middle of the bottom plane there is a 8 cm deep, 36 mm wide hole for a standard loudspeaker stand. As indicated in Fig. 1, cabinet corners where mounted on the device, which makes it convenient to stack the loudspeakers on top of each each other.

3 Sound Pressure Measurement

Measurement Setup. For conducting a MIMO sound pressure measurement, the loudspeaker cube is mounted on a stand and placed on an remotely controlled turntable. Directional impulse response measurements were done with a concentric semicircular microphone array with a radius of 75 cm (cf. Fig. 3). It holds microphones at the zenith angles $\vartheta = 5^\circ, 15^\circ, \ldots, 175^\circ$ and the turntable is used to access all azimuth steps $\varphi = 0^\circ, 10^\circ, \ldots, 350^\circ$. A sine sweep was used as measurement signal for every loudspeaker at every azimuth stop, resulting in a $10^\circ \times 10^\circ$ directivity measurement set $p_{z,a,l}(t)$ of $18 \times 36 \times 4 = 2592$

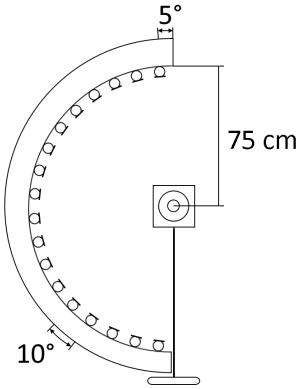


Fig. 3: Sound pressure measurement setting.

impulse responses, with the indices z, a, l for zenith $\vartheta_z = 10^\circ\,(z+\frac{1}{2})$, azimuth $\varphi_a = 10^\circ\,a$, and loudspeaker $l = 1, \ldots, 4$. To avoid the influence of room reflections, we cut the impulse responses to a a length of 512 Samples at a sampling rate of 44.1 kHz for the analysis. Measurement results are primarily important for evaluating the directivity patterns and for filter design.
The complete measurement data set was saved in the Spatially Oriented Format for Acoustics (SOFA), AES69-2015 and can be downloaded[2]. While we concentrate on 2D cuts through of the directivity patterns, interpolated 3D directivity-plots can be created using a newly introduced tool presented in [4].

Interpolation. We are primarily interested in the horizontal cross section of the directivity and therefore average the measurements acquired by the 9th and 10th microphone at the 85° and 95° zenith angles

$$p_{a,l}(t) = \frac{p_{9,a,l}(t) + p_{10,a,l}(t)}{2}. \tag{1}$$

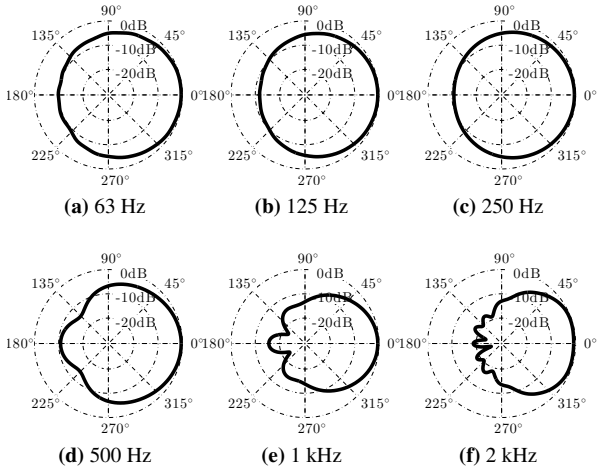
For being able to plot the directivity $p_{a,l}(t)$ over a continuous polar angle, we expand the measurement datapoints in terms of circular harmonics (Fourier series) of the order $\mathrm{N} = 17$,

$$p_l(\varphi,t) = \sum_{n=-\mathrm{N}}^{\mathrm{N}} w_n(\varphi)\; \psi_{n,l}(t), \tag{2}$$

$$\text{with } w_n(\varphi) = \begin{cases} \sqrt{2}\sin(n\varphi), & n<0, \\ 1, & n=0, \\ \sqrt{2}\cos(n\varphi), & n>0, \end{cases} \tag{3}$$

[1] http://phaidra.kug.ac.at/o:67631

[2] http://opendata.iem.at



(a) 63 Hz **(b)** 125 Hz **(c)** 250 Hz

(d) 500 Hz **(e)** 1 kHz **(f)** 2 kHz

Fig. 4: Measured and interpolated directivity pattern of the first loudspeaker.

where $w_n(\varphi)$ are the Fourier expansion functions over the $\mathrm{A} = 36$ azimuth stops. Tansformation to the circular harmonic domain yields the coefficients

$$\psi_{n,l}(t) = \frac{1}{\mathrm{A}} \sum_{a=0}^{\mathrm{A}-1} w_n\left(\tfrac{2\pi}{\mathrm{A}}\, a\right)\, p_{a,l}(t). \tag{4}$$

of the impulse responses. When transforming them into the frequency domain $\psi_{n,l}(\omega)$, we can plot directivity patterns, e.g. of the first loudspeaker $l = 1$, at different frequencies, cf. Fig. 4. Also, we can plot the frequency response of individual circular harmonic orders, whereby the zeroth order (monopole) and the first order (dipoles) are of particular interest for evaluating the first order control, cf. Fig. 5.

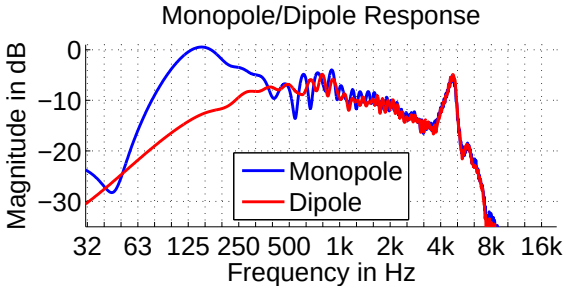


Fig. 5: Frequency responses of monopole $\psi_0^{\mathrm{mon}}(\omega)$ in blue and dipole $\sqrt{2}\psi_1^{\mathrm{dip-x}}(\omega)$ in red.

4 Beam Control

First-order beam-control is achieved by forming a weighted sum of the monopole and dipole patterns. Resulting first order directivity patterns are of the form

$$D(\varphi) = (1-\alpha) + \alpha\cos(\varphi - \varphi_0), \tag{5}$$

where α determines the shape between omni ($\alpha = 0$), cardoid ($\alpha = 0.5$) and figure-of-eight ($\alpha = 1$), while φ_0 modifies the beam direction. To control such a pattern, the signal $s(t)$ is distributed to the component $w(t)$ of the monopole and $x(t)$ and $y(t)$, of the dipoles pointing to the horizontal directions x and y.

$$\begin{bmatrix} w(t) \\ x(t) \\ y(t) \end{bmatrix} = \begin{bmatrix} (1-\alpha) \\ \alpha\cos(\varphi_0) \\ \alpha\sin(\varphi_0) \end{bmatrix} s(t). \tag{6}$$

Synthesis of monopole and dipole components. To obtain playback of a monopole radiation pattern w, all four loudspeakers play in-phase using the weights $[1, 1, 1, 1]$, i.e. $\psi_n^{\mathrm{mon}}(\omega) = \sum_{l=1}^{4} \psi_{n,l}(\omega)$.
To play to a dipole mode, a corresponding pair of opposing loudspeakers plays the signals out-of-phase. This means that the signal is distributed by the weights $[1, 0, -1, 0]$ for the x dipole aligned with the first and the opposing third loudspeaker, i.e. $\psi_n^{\mathrm{dip-x}}(\omega) = \psi_{n,1}(\omega) - \psi_{n,3}(\omega)$. The weights are $[0, 1, 0, -1]$ for the y dipole pattern $\psi_n^{\mathrm{dip-y}}(\omega) = \psi_{n,2}(\omega) - \psi_{n,4}(\omega)$. However, as we see from the frequency responses of the components $\psi_n^{\mathrm{dip-x}}(\omega)$ (dipole x) and $\psi_n^{\mathrm{mon}}(\omega)$ (monopole) in Fig. 5, the monopole and dipole patterns do not match without equalization.

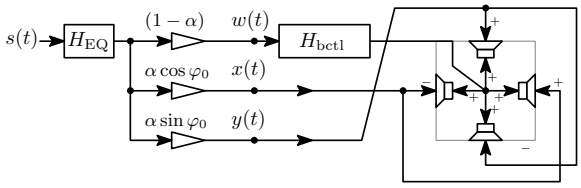
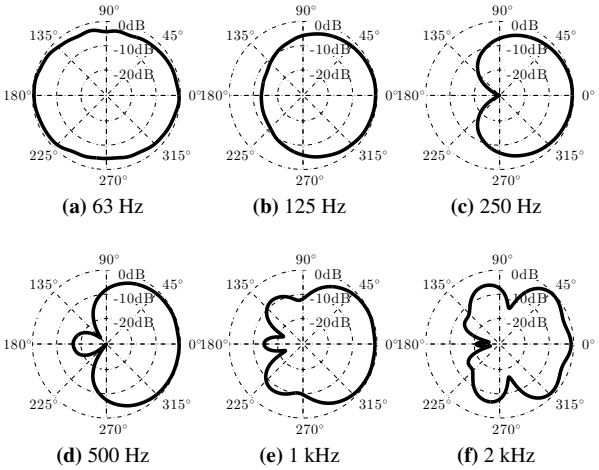


Fig. 6: Complete control scheme

Monopole to dipole equalization. Due to the better adjustment of the monopole component to radiation, low-frequency dominance develops between 200 Hz and 315 Hz. The dipole overcomes a certain part of its weaker adjustment to radiation since, other than the monopole, it is not impeded by the stiffness of the enclosed air below 200 Hz. To equalize the two responses

a filter needs to be designed. Although in theory the quotient of the meassured sound pressure responses could be used for this, oftentimes they exibit distortions due to reflections in the measurement chamber, cf. Fig. 3, which makes design based on sound pressure measurements complicated. As alternative strategy to develop less distorted filters, the reader is referred to [3] for IIR filter design or the LDV measurement based design presented in the next section.



(a) 63 Hz **(b)** 125 Hz **(c)** 250 Hz

(d) 500 Hz **(e)** 1 kHz **(f)** 2 kHz

Fig. 7: Created cardiod pattern $\alpha = 0.5$, $\varphi_0 = 0°$. Starting at 1kHz, the more concentrated radiation of the individual transducers distorts the shape.

5 Laser-Doppler-Vibromety Measurement

Filter Design. The design of beam control H_{bctl} would be greatly simplified if only we were able to control velocity vibration patterns of a loudspeaker in the monopole and dipole vibration modes. In such a case, it only remains to equalize the different farfield radiation of monopole and dipole, as in [3],

$$H_{\text{eq2}} = \frac{[s + (c/r_0)]\, s}{s^2 + 2(c/r_0)\, s + 2(c/r_0)^2}, \tag{7}$$

with $s = \mathrm{i}\omega$, $c = 343\ \frac{\text{m}}{\text{s}}$ and the effective radius $r_0 = 0.24$ m. This equalizer would only be one part of the required filter $H_{\text{bctl}} = H_{\text{eq1}}\, H_{\text{eq2}}$.

To approach control of the loudspeaker cone velocities H_{eq1}, they were measured and yield a 4×4 system of voltage-to-velocity transfer functions with all direct and all coupled responses

$$\boldsymbol{v}(\omega) = \boldsymbol{T}(\omega)\, \boldsymbol{u}(\omega). \tag{8}$$

The transfer functions $\boldsymbol{T}(\omega)$ can be used to control the velocities, which requires inversion. Fortunately, the system decouples when using the basis patterns we aim to control. According to the beam control described above and using the correct normalization, the monopole and the two dipole patterns require the voltage controls $\boldsymbol{u}_{mon} = \frac{1}{2}[1, 1, 1, 1]^{\text{T}}$ (monopole), $\boldsymbol{u}_{dx} = \frac{1}{\sqrt{2}}[1, 0, -1, 0]^{\text{T}}$ (x dipole), $\boldsymbol{u}_{dy} = \frac{1}{\sqrt{2}}[0, 1, 0, -1]^{\text{T}}$ (y dipole). Together with a quadrupole pattern $\boldsymbol{u}_q = \frac{1}{2}[1, -1, 1, -1]^{\text{T}}$, which isn't considered for control here, we get an orthogonal mode matrix $\boldsymbol{Q}$ that diagonalizes our system $\boldsymbol{T}$ sufficiently well,

$$\tilde{\boldsymbol{T}}(\omega) = \boldsymbol{Q}^{\text{T}} \boldsymbol{T}(\omega)\, \boldsymbol{Q}, \tag{9}$$

$$\boldsymbol{Q} = \frac{1}{2} \begin{bmatrix} 1 & \sqrt{2} & 0 & 1 \\ 1 & 0 & \sqrt{2} & -1 \\ 1 & -\sqrt{2} & 0 & 1 \\ 1 & 0 & -\sqrt{2} & -1 \end{bmatrix}. \tag{10}$$

Fig. 8 shows that $\boldsymbol{Q}$ decouples the output vibration modes sufficiently since the crosstalk between different voltage modes and velocity modes ($\tilde{t}_{ij}$ with $i \neq j$) is lower than -30 dB (black line).

A filter to equalize the monopole to the dipoles velocities now gets much simpler than system inversion of $\boldsymbol{T}$: in the decoupled domain of $\boldsymbol{Q}$, we can simply divide the active responses to define the missing filter

$$H_{\text{eq1}} = \frac{\tilde{t}_{22}(\omega)}{\tilde{t}_{11}(\omega)}. \tag{11}$$

Since control is only effective at frequencies below 1 kHz, and at high frequencies overcompensation is not useful, the H_{eq1} is cross-faded to unity from 1.5 kHz to 2 kHz. From this equalizer, an impulse response is obtained via inverse DFT, which can be cut to 512 samples. Amplitude and phase are shown in Fig. 9.

Vibration Measurement. The measurements were conducted by the help of polytec PDV-100 Laser-Doppler-Vibrometer. The velocity of every of the four loudspeaker cones $v_1, \ldots, v_4$ was measured at its center point, while all the loudspeakers were exited one by one by an exponential sine sweep.

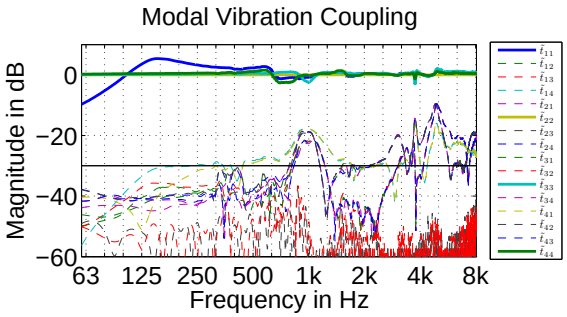


Fig. 8: Vibration coupling $\tilde{t}_{ij}$ between in and output indices: 1 monopole, 2 dipole x, 3 dipole y, and 4 quadrupole (unused); all equalized to $\tilde{t}_{22}$.

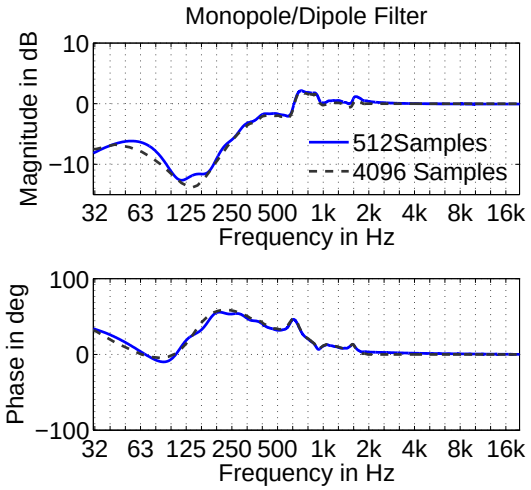


Fig. 9: FIR Filter $H_{\text{bctl}} = H_{\text{eq1}} H_{\text{eq2}}$ required to equalize radiation of the monopole control mode to radiation of the dipole control mode.

6 Summary

We presented the construction and control of a simple first-order beam-controlling loudspeaker. We showed a possible, robust design that can serve as a reference for new similar beam-controlling loudspeakers. And we outlined the control principle, which is easy to implement in an arbitrary programming environment, for instance as a Pd patch or as part of a VST plugin.

A related paper [3] shows applications further research into these directional loudspeakers leads to. For readers interested in pursuing their own research, for example in developing other control strategies, we provid measurement data as explained above.

Furthermore, the filter design concept and the interpolation by transformation to circular harmonics we presented can be of interest to anyone dealing with similar new sets of measurement data. By Laser-Doppler-Vibrometer measurement we have shown that the basis modes of a sufficiently symmetric cube loudspeaker array can be easily controlled independently of each other. By the directivity plots we have shown that the control is highly effective below 1 kHz. Open Source plugins for controlling the loudspeaker cube in different setups are offered online[3].

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