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Directional loudspeaker arrays for acoustic warning systems with minimised noise pollution

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ABSTRACT

This paper describes numerical and experimental results of beamforming algorithms for generation of directional sound. The intended application is a sound source for cars with the objective to warn vulnerable road users while minimising noise pollution. Nowadays, sensors exist which are able to reveal the position of the vulnerable road users, which information can be used by the warning signal generator. Based on this information, the signal generator is designed to generate the specified warning signal at the location of the vulnerable road user while the acoustic response at other locations is minimised. The directional sound beam was realised with an array of controlled acoustic moving coil sources. The paper compares different methods to generate the sound beam and investigates their effectiveness for this application by simulation and experimental results. The robustness under realistic conditions in simulations and in initial real-time experiments on a car are discussed.

1. Introduction

Studies [1,2] have shown that (hybrid) electrical vehicles pose an increased risk to pedestrians and bicyclists when compared to traditional vehicles with combustion engines. This increased risk is due to the low noise production at slower speeds, where tire noise is not yet dominant. These speeds are typically below 30 km h^{-1}. To improve safety, a sound source should be added so that the localisation of electrical vehicles becomes comparable to that of combustion engines.

It is suggested to make use of a directional sound beam in order to minimise the noise pollution. This directional beam is designed to produce a specified acoustic response in a given target direction whilst minimising the response in the other directions. The algorithms that are able to create such a beam through producing filters for use with transducer arrays are commonly referred to as beamformers. A comparison between end-fire arrays and broadside arrays by Cheer et al. [3] shows that end-fire arrays are able to provide higher directivity than broadside arrays. However, end-fire arrays radiate symmetrically with respect to their axis, and therefore, if the axis is aligned with the driving direction, cannot be used for steering at arbitrary angles in the forward direction as required for the present application. In principle the broadside array radiates equally in the forward and backward directions, but radiation in the backward direction can be shielded by the vehicle if the broadside array is mounted on the front of the vehicle.

Much work on beamforming has been done in the electromagnetic domain using antenna arrays. In [4], many of these earlier approaches are summarised and referenced. Beamforming for microphone arrays is discussed in [5]. An overview on beamforming in general is provided by van Trees [6]. Automatic steering of arrays is discussed in [7].

As a consequence of reciprocity, the algorithms that are designed for these receiving arrays can be applied to transmitting arrays of point-sources as well. The most simple method of beamforming is commonly known as delay and sum (DS) [6,8,9]. This method is designed to compensate for the different paths between the sources and the bright spot in order to ensure that they are constructive.

Another approach which is specifically designed for transmitting type arrays is the contrast control (CC) approach [10,11]. A variant known as acoustic energy difference (AED) was later proposed in [12]. These methods optimise the contrast between the bright and dark zones.

The least-squares (LS) algorithm described in [6,8] aims to find a sound pressure field that matches the desired field. In [13], a

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frequency-invariant approach is introduced for an omnidirectional receiving array.

Another approach is presented in [9], which could be considered as a variation to the minimum variance distortionless response (MVDR) method developed for microphone arrays, as discussed in [6, p. 441]. This beamformer type minimises the total pressure whilst constraining the pressure in the bright zone. This paper introduces a modified version for use with loudspeaker arrays that can be used with arbitrary transfer functions and multiple constraints. This algorithm will be called sound-power minimisation (SPM).

Yet another approach to creating a directional source was introduced in [14] under the name of the time-reversal approach. This algorithm reverses the impulse response from the loudspeaker to the focus point, effectively using the reflections that are present to focus the sound. As also noted in [15], the performance greatly depends on the surrounding which makes this algorithm impractical for dynamic environments. Compact sources with high directivity can be realised with parametric arrays [16]. Because of the sound pressure level requirements at low frequencies, moving coil loudspeakers may still be preferred for some applications.

Making use of a synthetic aperture [17] was considered, as this allows increasing the aperture of the array. However, this algorithm is not suitable given the real-time constraint for this application. This unsuitability can easily be seen by calculating the travelled distance of a car with a speed of $v = 30\ \mathrm{km\,h^{-1}}$ before the first generated wave-front arrives at the maximum distance of $D = 5\ \mathrm{m} : \Delta x = vDc^{-1} \approx 0.12\ \mathrm{m}$, with c the speed of sound. Since this distance is small compared to the wavelength c/f with f the frequency in Hz, this approach is not considered.

This paper contributes in investigating the applicability of algorithms for use with a steerable, directional warning system for automotive exterior using moving coil loudspeakers. The focus in this paper is on the comparison of different algorithms, algorithmic extensions for use in an acoustic environment representative for automotive exterior and experimental verification with a real-time system. The robustness in realistic simulation scenarios and the performance of the system on a car will be discussed.

2. Methods

We define N loudspeakers at positions $\mathbf{r} = [\mathbf{r}_1 \ \cdots \ \mathbf{r}_N]$ and a vector $\mathbf{G}(\mathbf{x})$ containing the transfer functions from the loudspeakers to the focus point $\mathbf{x}$:

$$\mathbf{G}(\mathbf{x}) = [\,G(\mathbf{r}_1|\mathbf{x}) \ \ \cdots \ \ G(\mathbf{r}_N|\mathbf{x})\,], \tag{1}$$

in which $G(\mathbf{r}_n|\mathbf{x})$ is the transfer function from a loudspeaker at position $\mathbf{r}_n$ to the pressure at point $\mathbf{x}$. The source strength vector $\mathbf{q} = [q_1 \ \cdots \ q_N]^\top$ is defined to describe the output of each loudspeaker, in which $^\top$ denotes the transpose operator. The pressure in the focus point is then given by

$$p(\mathbf{x}) = \sum_{n=1}^{N} G(\mathbf{r}_n|\mathbf{x}) q_n = \mathbf{G}(\mathbf{x})\mathbf{q}. \tag{2}$$

The mean square energy E is defined as the mean squared magnitude of the acoustic pressure in a region V:

$$E = \frac{1}{V} \iiint_V p(\mathbf{x})^* p(\mathbf{x})\ \mathrm{d}\mathbf{x}, \tag{3}$$

in which * denotes complex conjugate. The mean square energy E is proportional to the acoustic potential energy in V, and is proportional to the radiated sound power if the pressure is obtained by homogeneous sampling on a sector in the far field. In the discrete space domain, V can be approximated by sampling at sufficiently small intervals using M positions:

$$E \approx \frac{1}{M} \sum_{m=1}^{M} p(\mathbf{x}_m)^* p(\mathbf{x}_m) = \mathbf{q}^H \left(\frac{1}{M} \sum_{m=1}^{M} \mathbf{G}(\mathbf{x}_m)^H \mathbf{G}(\mathbf{x}_m) \right) \mathbf{q} = \mathbf{q}^H \mathbf{R}_V \mathbf{q}, \tag{4}$$

in which ^{H} denotes Hermitian transpose and in which a spatially averaged correlation matrix $\mathbf{R}_V$ of transfer functions $\mathbf{G}$ is defined by

$$\mathbf{R}_V = \frac{1}{M} \sum_{m=1}^{M} \mathbf{G}(\mathbf{x}_m)^H \mathbf{G}(\mathbf{x}_m). \tag{5}$$

As illustrated in Fig. 1, two distinct types of regions are defined: a bright and a dark region. Furthermore, the total region is defined as the combination of both bright and dark zones. These regions will be abbreviated by subscripting b (bright), d (dark) or t (total). Then let E_b, E_d and E_t denote the energy of these regions and let $\mathbf{R}_b$, $\mathbf{R}_d$ and $\mathbf{R}_t$ denote the corresponding correlation matrices as defined in Eq. (4). Additionally, we define the transfer functions $\mathbf{G}_b(\mathbf{x})$, $\mathbf{G}_d(\mathbf{x})$ and $\mathbf{G}_t(\mathbf{x})$, which are the transfer functions from the N loudspeakers to the bright region, dark region, and total region, respectively.

2.1. Experiment methodology

In order to find the most suitable algorithm for use with this application, experiments were performed to evaluate the beam shape, the stability and suitability for real-time implementation. The experiments have been performed using a uniform, i.e., equally spaced, array with eight elements. The frequency range of interest is 100–3000 Hz. The elements were spaced 57.2 mm apart and therefore comply to the spatial aliasing constraint [6]. They were placed at a height of 0.6 m above a reflecting ground surface. The area of interest was a half-circle in front of the array at a distance of 5 m and a height of 1.8 m.

Three sets of experiments were used with increasing order of realism, (a) simulations using ideal point sources in free-field conditions; (b) adding a fully reflective ground surface and (c) using measured transfer functions and a real-time implementation in the field.

For the first set of experiments, the following transfer functions were used

$$G(\mathbf{r}_n|\mathbf{x}) = \frac{1}{4\pi|\mathbf{r}_n - \mathbf{x}|} e^{-jk|\mathbf{r}_n - \mathbf{x}|}, \tag{6}$$

with $j = \sqrt{-1}$ and the wavenumber $k = 2\pi f/c$. For the assessment of the influence of a reflective ground surface in the second set of experiments the corresponding mirror sources were added.

The third set of experiments make use of measured transfer functions. These transfer functions are measured using B&K 4957 and 4958 array microphones connected to a PCB481 conditioning amplifier with $10\times$ gain. The microphones are positioned at 15 equally-spaced positions on the half-circle in front of the array (Fig. 2).

A real-time Linux-based system is used for both steering and measurement, where the algorithms have been implemented using

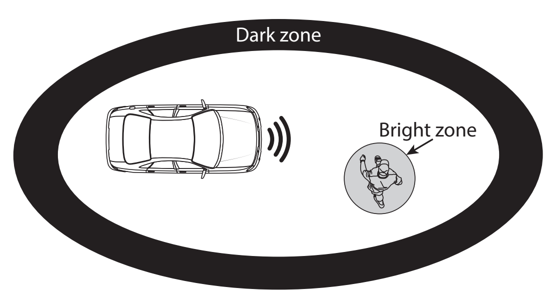

Fig. 1. Illustration of a bright and dark acoustic region.