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Sound field control for reduction of noise from outdoor concerts

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ABSTRACT

We investigate sound field control based on the concept of sound zones for the mitigation of low frequency noise from outdoor concerts to the surrounding area by adding secondary loudspeakers to the existing primary sound system. The filters for the secondary loudspeakers are the result of an optimization problem that minimizes the total sound pressure level of both primary and secondary loudspeakers in a sensitive area and the impact of the secondary loudspeakers on the audience area of the concert. We report results from three different experiments with increasing complexity and scale. The sound field control system was reducing the sound pressure level in the dark zone on average by 10 dB below 1 kHz in a small scale experiment in anechoic conditions, by up to 14 dB in a controlled large scale open-air experiment and by up to 6 dB at a pilot test at a music festival.

1 Introduction

Organizers of open air concerts are facing the challenge to give their guests an exciting audio experience while complying with regulations on noise exposure in the surrounding area. In a previous paper [1] we proposed the application of sound zoning to this problem as this collection of methods is trying to achieve just that: a bright zone with large sound pressure levels and high audio quality and a dark zone with low sound pressure levels.

In this paper we use one of these sound zoning methods, pressure-matching acoustic contrast control (PM-ACC) [2], and apply it to the outdoor concert scenario by adding a set of loudspeakers to the sound reinforcement system. We reformulate the PM-ACC optimization problem for the case when only the additional loudspeakers (secondary sources) are part of the control system while the main loudspeaker system (primary source) is left untouched. The optimal control filters minimize the total sound pressure level due to the primary and the secondary sources in the dark zone and minimize the sound pressure radiated from the secondary sources into the bright zone.

This work focuses at the control of low frequencies as these are the most critical frequencies in the noise problem of outdoor concerts and the control of sound over large spaces with a feasible number of loudspeakers is only possible in that range.

The paper is structured as follows. Section 2 presents the methodology. We reformulate PM-ACC for the case where only the signals of the secondary sources' can be controlled and define two metrics that help to quantify the performance of the control system. In section 3 the method is applied in three different experiments and results are presented and discussed. Section 3.1 describes a down scaled model setup of a concert scenario in an anechoic chamber, section 3.2 reports from a large scale outdoor experiment with professional audio equipment and section 3.3 presents the results from a pilot test at a music festival. Section 4 discusses the applied methodology in the light of the experimental results. Section 5 summarizes the present work.

2 Methods

2.1 The optimization problem

The objective function of the PM-ACC method minimizes the SPL in a dark zone and the reproduction error relative to a target field in a bright zone. In the context of outdoor concerts, the bright zone is representing the audience area and the dark zone corresponds to a noise sensitive area in the surroundings. The cost function for the control weights $\mathbf{w} \in \mathbb{C}^N$ at a single frequency is [2]

$$\min_{\mathbf{w}} \kappa \left\| \mathbf{h}^t - \mathbf{H}_B \mathbf{w} \right\|^2 + (1-\kappa) \left\| \mathbf{H}_D \mathbf{w} \right\|^2, \qquad (1)$$

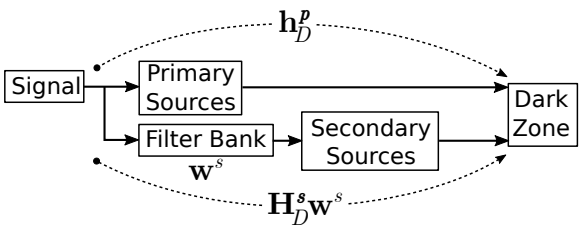

Fig. 1: Signal path diagram: the total sound pressure in the dark zone is the sum of sound pressure from the primary and secondary sources. The total system transfer-function to the dark zone is $\mathbf{h}_D^p + \mathbf{H}_D^s\mathbf{w}^s$. The equivalent holds for the bright zone.

where $\mathbf{h}^t \in \mathbb{C}^{M_B}$ is the target transfer function in the bright zone, $\mathbf{H}_{B/D} \in \mathbb{C}^{M_{B/D}\times N}$ are the transfer matrices between sources and the spatially sampled bright and dark zones and κ is a parameter that weights the reproduction error in the bright zone relative to the energy in the dark zone.

The sound field control (SFC) approach of this paper only controls the additional secondary sources, as illustrated in Fig. 1. Splitting control weights and transfer matrices into primary and secondary contributions, $\mathbf{w} =: [\mathbf{w}^p, \mathbf{w}^s]^T$, $\mathbf{H} =: [\mathbf{H}^p, \mathbf{H}^s]$, setting the primary weights $\mathbf{w}^p$ as constant and the target transfer-function as the transfer-function of the primary system, $\mathbf{h}^t := \mathbf{H}_B^p\mathbf{w}^p$, transforms the cost function to

$$\min_{\mathbf{w}^s} \kappa \left\|\mathbf{H}_B^s\mathbf{w}^s\right\|^2 + (1-\kappa)\left\|\mathbf{H}_D^s\mathbf{w}^s + \mathbf{h}_D^p\right\|^2 + \lambda\left\|\mathbf{w}\right\|^2, \tag{2}$$

where we added a Tikhonov regularization term with parameter λ and $\mathbf{h}_D^p := \mathbf{H}_D^p\mathbf{w}^p$. This cost function minimizes the radiation of the secondary sources into the bright zone and the total sound energy in the dark zone. The optimization problem can be efficiently solved by rewriting it as a linear least squares problem

$$\min_{\mathbf{w}^s} \left\|\mathbf{A}\mathbf{w}^s - \mathbf{b}\right\|^2, \tag{3}$$

where

$$\mathbf{A} = \begin{bmatrix} \sqrt{\kappa}\mathbf{H}_B^s \\ \sqrt{1-\kappa}\mathbf{H}_D^s \\ \sqrt{\lambda}\mathbf{I} \end{bmatrix} \quad \text{and} \quad \mathbf{b} = \begin{bmatrix} 0 \\ -\sqrt{1-k}\mathbf{h}_D^p \\ 0 \end{bmatrix}. \tag{4}$$

The purpose of the regularization term is two-fold: firstly, it enables us to solve the possibly ill-posed inverse problem by making the solution robust against noise in the measured transfer-functions [3, 4]. Secondly, it smoothly distributes the array effort over the control loudspeakers and limits the magnitude of the resulting control gains. This way realizable solutions, i.e. secondary sources play in their linear range, can be found by tuning λ. In this work, we have taken a λ that is constant with frequency for the sake of simplicity. In the experiments, λ was chosen manually such that the maximum gain of the filters was approximately 0 dB.

By solving the optimization problem of Eq. 2 for all relevant frequencies we get the complex gain of the secondary sources as a function of frequency. The discrete Fourier transform of the frequency domain filters is a set of FIR filters. The control loudspeaker driving signals are obtained by real time convolution of the audio signal with the corresponding FIR filters.

2.2 Performance metrics

We define two easily interpretable performance metrics which quantify how well the two objectives of the cost function are reached. The insertion loss

$$\mathrm{IL} = 10\log\left(\frac{1}{N_B}\left\|\mathbf{h}_D^p\right\|^2\right) - 10\log\left(\frac{1}{N_D}\left\|\mathbf{H}_D^s\mathbf{w}^s + \mathbf{h}_D^p\right\|^2\right) \tag{5}$$

represents the decrease in sound energy in the dark zone due to the control sources with a a large IL indicating a strong reduction.

In the bright zone, the primary to secondary ratio

$$\mathrm{PSR} = 10\log\left\|\mathbf{H}_B^p\mathbf{w}^p\right\|^2 - 10\log\left\|\mathbf{H}_B^s\mathbf{w}^s\right\|^2 \tag{6}$$

quantifies the ratio of sound energies coming from the primary and the secondary sources. A large PSR value means that the sound from the primary sources dominates the sound field in the bright zone.

3 Results

We investigated the sound field control method in three experiments with increasing dimensions and complexity of the surroundings.

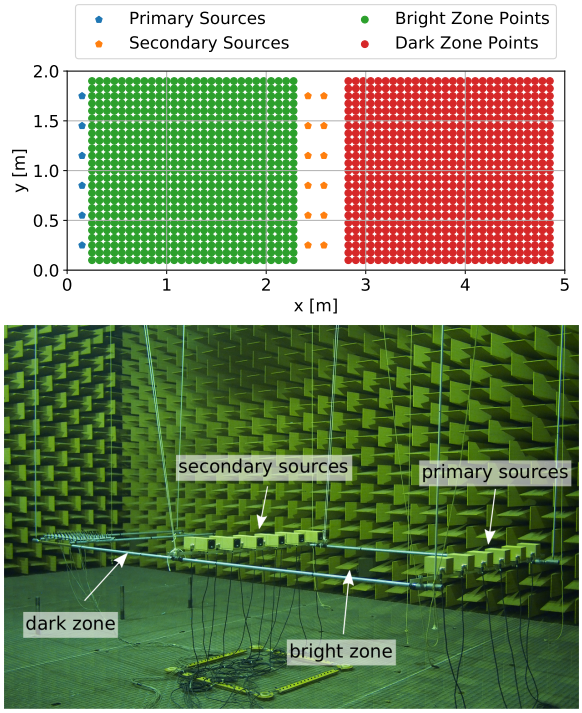

Fig. 2: Photo and geometry of anechoic chamber experiment

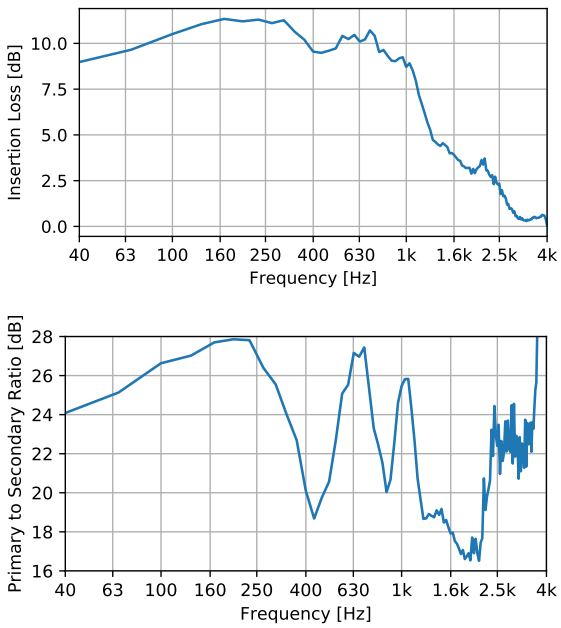

Fig. 3: Insertion loss and primary to secondary ratio in anechoic chamber experiment with $\kappa = 0.1$ and $\lambda = 0$

3.1 Anechoic chamber experiment

Fig. 2 shows the small scale experimental setup in the anechoic chamber of the Technical University of Denmark with a volume of $1000\,\mathrm{m}^3$. Six primary sources are simulating the subwoofer array of a typical sound reinforcement system. Behind the bright zone (the audience area), a secondary loudspeaker array consisting of 12 sources is placed in a double layer array with 6 loudspeakers facing the bright and 6 loudspeakers facing the dark zone. All loudspeakers are of the same type with a 4 inch driver. Directly behind the secondary sources, a dark zone is simulating a sensitive neighboring area. We measured the transfer functions from all loudspeakers to a densely sampled grid of 700 microphone positions per zone to construct the transfer function matrices. Half of the data is used for the filter calculation and the other half for performance estimation as suggested in [3].

The resulting IL and PSR is shown in Fig. 3 for parameters $\kappa = 0.1$, $\lambda = 0$. No regularization had to be used due to the large amount of measurement positions and low noise levels. We observe an IL of about 10 dB up to around 1 kHz, after which it drops quickly. PSR is larger than 18 dB in that frequency range, showing that the double layer array is directing most of the radiated sound energy to the dark zone. In fact, noticing the secondary sources in the bright zone during music playback was, if at all, only possible very close to them. The two dips at around 400 Hz and 800 Hz are due to the approx. 40 cm spacing between the membranes of the double layer array, as also observed in [5]. The distance corresponds to half of a wave-length at these frequencies and the sound from the secondary sources radiated into the bright zone can not be efficiently cancelled without also cancelling the sound radiated into the dark zone which is needed to cancel the sound from the primary sources. The optimization allows for a low PSR in favour of a large IL, because κ is small.

The difference in SPL over space due to activation of the secondary sources is shown in Fig. 4 for three frequencies. The SPL reduction is especially strong in the left half of the dark zone. This patch of strong reduction is getting smaller with frequency. However, it is also present above the Nyquist frequency of around 570 Hz of the line arrays as was also observed in [2, 5]. In general, the size of this patch will become large if the

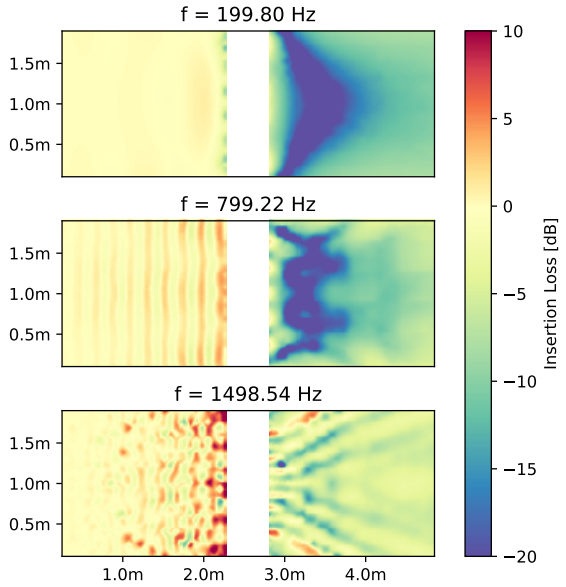

Fig. 4: Insertion loss over space in the bright zone (left) and dark zone(right) at different frequencies in the anechoic chamber experiment. $\kappa = 0.1$, $\lambda = 0$.

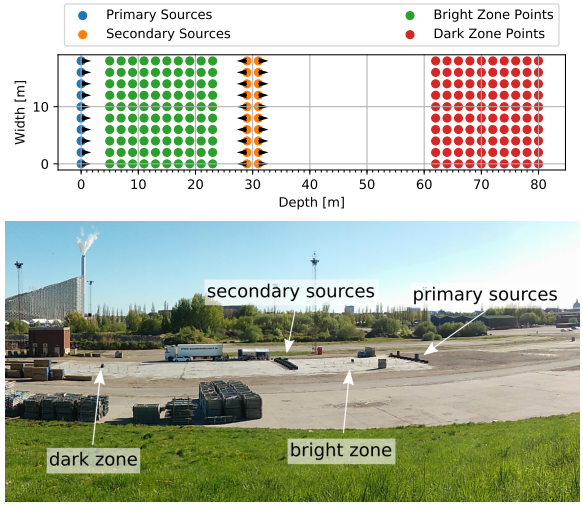

Fig. 5: Geometry and photo of large scale experiment

dark zone lies far away relative to the distance between primary and secondary sources. At around 800 Hz, the PSR value has a dip and one can see a standing wave forming in the bright zone due to the low directivity of the secondary array at this frequency. Above 1 kHz the secondary array can not resolve the complex sound field of the primary sources, which leads to a low IL.

3.2 Large scale experiment

We conducted a large scale experiment with professional audio equipment to test the sound field control system under conditions that are more similar to a typical outdoor concert. In comparison to the anechoic chamber measurement there were reflections from the ground and objects scattered around the experiment, changing temperature and a lower frequency range. Fig. 5 shows the geometry and a photo of the experimental setup, which spanned a region of $80\,\mathrm{m} \times 20\,\mathrm{m}$. The primary sources comprised 10 subwoofers arranged in a line array with 2 m spacing. The secondary sources consisted of 20 subwoofers of the same type arranged in a double layer line array. The subwoofer model had an intrinsic cardioid radiation pattern and a nominal frequency range of $37 - 115\,\mathrm{Hz}$ (-5dB). If a single layer of cardioid speakers creates enough PSR there is no need for a double layer arrangement and the amount of loudspeakers can be reduced by a factor 2. The transfer-functions to the bright and dark zones were sampled at 100 microphone positions in each zone, half of which were used for the computation of the control filters and the other half for the performance estimation.

Fig. 6 shows the performance metrics of the sound field control solution with $\kappa = 0.1$ and $\lambda = 50000$. We estimated a maximal IL of around 12-14 dB between 45-85 Hz when using either two (blue continuous) or just one layer (yellow dashed) of secondary sources. Using a double layer does not significantly increase the IL but increases the PSR by 2-10dB. The PSR is larger than 15 dB at all frequencies for both cases, suggesting that a single layer of cardioid secondary sources could be focusing its sound energy sufficiently well away from the bright zone.

We measured the transfer-functions of all single sources for computation of the filters and prediction of the results at noon in peaking temperatures. We also directly measured the sound field of the total system in the beginning of the night, when temperatures had fallen by 6-8 °C (green dash-dotted). The change in weather conditions resulted in a change of the speed of sound and thus changed the transfer-functions. This mismatch in conditions during transfer-function measurement and playback lead to a reduction of the IL by 2-10 dB (see discussion).

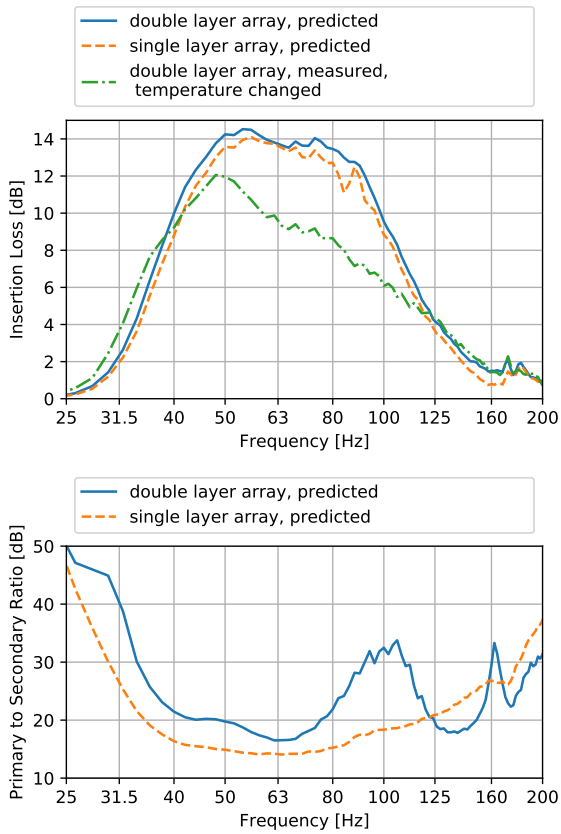

Fig. 6: Insertion loss and primary to secondary ratio for large scale experiment with $\kappa = 0.1$ and $\lambda = 50000$

3.3 Pilot experiment

In the largest of the three experiments, the sound field control system was put to test during a two day festival in Turin, Italy. We present here some first, preliminary results.

Compared to the previous experiments this scenario posed several new challenges: complex, uneven terrain with many reflecting structures and surfaces around the venue and dark zone, i.e. long reverberation tails; microphone positions neither in line of sight with the primary source nor the secondary sources, i.e. indirect sound must be cancelled by the indirect sound from the secondary sources; restrictions in placement of secondary sources and microphones, i.e. fewer samples of the transfer-functions to the dark zone; and time constrained measurements in noisy environment at large distances, i.e. low signal to noise ratio in transfer-function measurements.

Fig. 7 gives an overview of the venue, the setup of loudspeakers and the microphone positions in and around a part of the festival area that spans around 300 m. The primary source (main stage subwoofer system) comprised 20 cardioid subwoofers in a digitally curved line array configuration. The secondary source array consisted out of 16 subwoofers of the same type arranged in a single line with 2.55 m spacing (center-center) and facing the negative x-direction.

The stage sound system also featured two vertical line arrays for the higher frequencies, but these were not included in the transfer-function measurements and thus also not accounted for by the sound field control system.

We assumed that the secondary source configuration and the large distance of the secondary sources to the audience area would lead to a low impact of the control system onto the audience area, which was approximately the area between $x = 100$ and $x = 200$. In fact, a difference from switching the control speakers on and off could only be noticed up to around $x = 90$. Closer to the audience area the sound from the secondary sources was completely masked by the sound from the primary source.

In the optimization problem we set $\kappa = 0$ to focus all effort on the minimization of sound in the dark zone. The dark zone was defined as the area between $x = -100$ and $x = 0$. It was sampled at 20 microphone positions in an elevated courtyard and 30 positions on a rooftop.

After measurement of the transfer-functions from all sources to all microphone positions, a set of control filters was computed and the regularization parameter was chosen by hand such that the gain of the control filters was not overly extreme.

Figure 8 shows the measured magnitude response of the sound system with active and disabled secondary sources at the line of microphone positions around $x = -10$ and compares them to the prediction. They align fairly well, showing an IL up to approx. 6 dB. Assuming that we can use the prediction also for all other positions, we estimated the average IL at all microphone positions by offline prediction in Fig. 9.

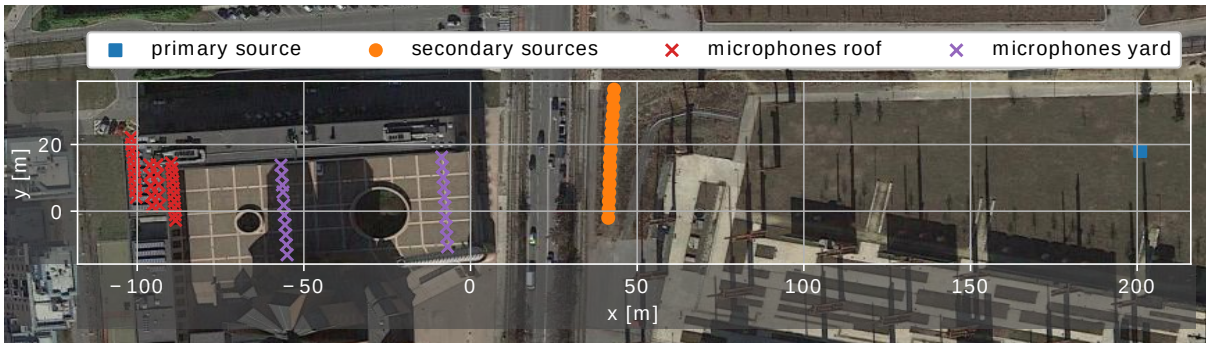

Fig. 7: Pilot experiment: positions of sources and microphones

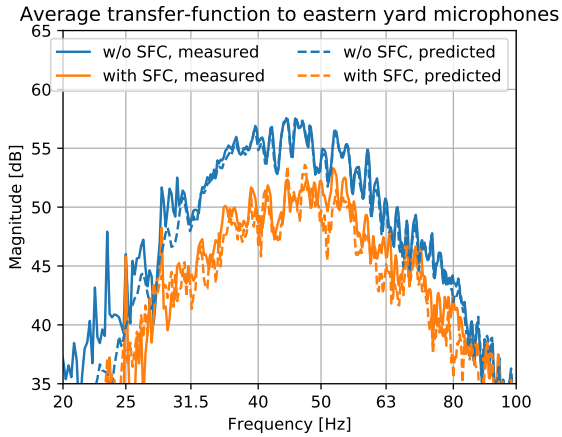

Fig. 8: Pilot experiment: measurement and prediction of the effect of the sound field control system on the sound system transfer function to the first row of microphones in the dark zone. $\kappa = 0$, $\lambda = 10$

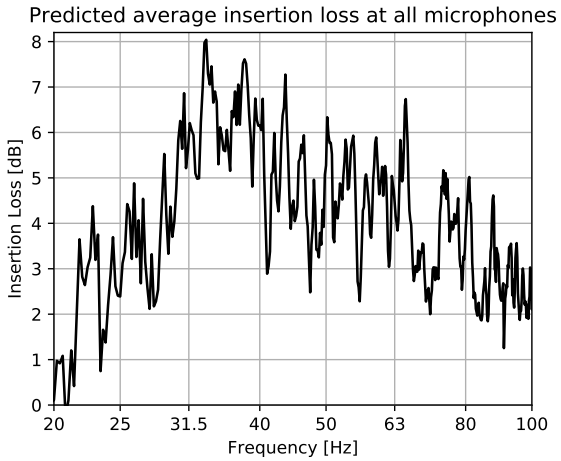

Fig. 9: Pilot experiment: predicted insertion loss at all microphone positions $\kappa = 0$, $\lambda = 10$

4 Discussion

There are several issues in this work that would benefit from an estimation of the transfer-functions using a computational model. First of all, the measurement of the transfer-functions in a sufficiently dense and large grid of points takes several hours, especially with many control sources and long excitation signals due to high background noise. Secondly, the transfer-functions should be updated according to changes that affect the propagation of sound, because any mismatch can degrade the system performance as it was shown in Fig. 6. A model that accounts for such changes would make the update simple. Finally, the dark zone is not necessarily an area that is accessible for direct measurements. Ideally, if the propagation between sources and dark zone is modelled properly, it is possible to estimate the transfer-functions on a dense grid of virtual microphones at arbitrary positions. Caviedes Nozal et al. showed the possibility of using a Bayesian framework to model the response of acoustic sources for use in sound field control scenarios from a small set of measurements [6]. However, they use a rather simple propagation model which is not able to take into account all the elements in the acoustic path between sources and dark zone (scattering, diffraction, reflections...).

The performance of the sound field control system in terms of IL is completely dependent on how accurate the secondary sources are able to reproduce the inverted sound field from the primary source in the dark

zone. The synthesis of very complex sound fields is possible by using many, distributed loudspeakers or by exploitation of small differences between loudspeaker transfer-functions, leading to large control gains and sound fields that rely heavily on destructive interference between speakers. The use of regularization favours solutions of the optimization problem with smaller gains and therefore less dependence on cancellation between speakers. It improves robustness at the cost of larger reproduction errors and a good choice of the regularization parameter carefully balances this compromise. In this work the regularization parameter λ was constant over frequency and chosen by hand, such that the control filter gains where lower than 0 dB at most frequencies, thus driving the secondary sources at lower or similar levels compared to the primary sources. The magnitude of the transfer-functions, however, is small at low and high frequencies, generally leading to smaller values of the cost function terms $\kappa \|\mathbf{H}_B^s \mathbf{w}^s\|^2 + (1-\kappa) \left\|\mathbf{H}_D^s \mathbf{w}^s + \mathbf{h}_D^p\right\|^2$, but not smaller $\lambda \|\mathbf{w}\|^2$. This leads to a stronger regularization at these frequencies. We expect that a well chosen, frequency dependent regularization parameter will widen the frequency band in which a large IL is achieved. Relevant work has been done already in this direction, including a loading to the matrix transfer-function matrix that is proportional to the uncertainties, ending in a regularization factor that comes directly from the measured data [7, 8]. Still, even with perfect knowledge of the transfer-functions on an infinitely dense grid or an optimal regularization method, the performance is going to be bounded by the geometry of the venue, its surroundings and the number of secondary sources and their placement in this space.

Additional secondary sources that are further away than half of a wave length to the primary source will increase the total sound power output in free-field conditions [9]. Decreasing the sound energy in the dark zone must therefore come in hand with a sound energy increase somewhere else. In the three experiments we have so far not experienced any regions where the sound field control system noticeable increases the sound pressure level (apart from very close to the secondary sources), but this will be topic of future investigations.

5 Summary

We have shown in three experiments that the noise from outdoor concerts can be reduced by actively controlling the sound from the main sound system in a dedicated area.

The secondary sources can be arranged in a double layer array to mitigate any negative impact from the secondary sources to the audience area. Alternatively, a single layer of cardoid loudspeakers seems to have a sufficiently strong directivity with a similar effect.

The observed reduction in sound pressure levels was among other things dependent on the geometric complexity of the venue and terrain, which can become a limiting factor for the performance of the sound field control system.

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References

[1] Heuchel, F. M., Caviedes Nozal, D., Brunskog, J., Fernandez Grande, E., and Agerkvist, F. T., "An Adaptive, Data Driven Sound Field Control Strategy for Outdoor Concerts," in *Audio Engineering Society Conference: 2017 AES International Conference on Sound Reinforcement – Open Air Venues*, 2017.

[2] Chang, J.-H. and Jacobsen, F., "Sound field control with a circular double-layer array of loudspeakers," *The Journal of the Acoustical Society of America*, 131(6), pp. 4518–4525, 2012, doi: 10.1121/1.4714349.

[3] Møller, M. B. and Olsen, M., "Sound Zones: On Performance Prediction of Contrast Control Methods," in *Audio Engineering Society Conference: 2016 AES International Conference on Sound Field Control*, 2016.

[4] Coleman, P., Jackson, P. J. B., Olik, M., Møller, M., Olsen, M., and Pedersen, J. A., "Acoustic contrast, planarity and robustness of sound zone methods using a circular loudspeaker array," *The Journal of the Acoustical Society of America*, 135, p. 1929, 2014, doi:10.1121/1.4866442.

[5] Chang, J.-H. and Jacobsen, F., "Experimental Validation of Sound Field Control with a Circular Double-Layer Array of Loudspeakers," *The Journal of the Acoustical Society of America*, 133(4), pp. 2046–2046, 2013, doi:10.1121/1.4792486.

[6] Caviedes Nozal, D. and Brunskog, J., "Parameter optimization of forward sound propagation models using Bayesian inference for sound field control purposes," in *Euronoise 2018 - Proceedings*, pp. 2301–2308, 2018.

[7] Pereira, A., Antoni, J., and Leclère, Q., "Empirical Bayesian regularization of the inverse acoustic problem," *Applied Acoustics*, 97, pp. 11–29, 2015, doi:10.1016/j.apacoust.2015.03.008.

[8] Zhu, Q., Coleman, P., Wu, M., and Yang, J., "Robust Acoustic Contrast Control with Reduced In-situ Measurement by Acoustic Modeling," *Journal of the Audio Engineering Society*, 65(6), pp. 460–473, 2017, doi:10.17743/jaes.2017.0016.

[9] Jacobsen, F. and Juhl, P. M., *Fundamentals of General Linear Acoustics*, Wiley, 2013.