ENHANCEMENT OF THE DYNAMIC RANGE IN ACOUSTIC PHOTOS BY MODIFIED TIME DOMAIN BEAMFORMING

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ABSTRACT

The fast identification of acoustic (noise-)sources is an important discipline in the process of automotive engineering. Based on time domain beamforming in the far-field, the *Acoustic Camera* [2] is capable of localizing sound sources by detecting the direction of incidence. The interpretation of measurement results is sometimes complicated because of inherent drawbacks of the system and the method itself, and moreover by bad acoustic conditions.
In this context self noise of the microphones can affect the acoustic image. By means of the well-known diagonal deletion in the cross-spectral matrix this effect can be reduced in the frequency domain. In this study the diagonal deletion technique is applied for time domain beamforming, suggested by Dougherty in [1]. The results are compared to the pure delay-and-sum algorithm and are discussed in terms of an extended applicability of the *Acoustic Camera.*

1 INTRODUCTION

Based on the classical delay-and-sum algorithm, sound localization systems produce acoustic photos that give information about the pressure level distribution in the far-field caused by sources in certain directions. To steer in these directions the corresponding delays have to be compensated for. This can be done both in the frequency domain and in the time domain. There are a lot of aspects that argue for one of these methods. Besides the computational efficiency the most important advantage using the time domain technique is considering the broad-band character of signals. Nevertheless, calculating the corresponding phase shifts in the frequency domain provides miscellaneous techniques of manipulating the cross spectral matrix (CSM). One of these data-dependent modifications is to remove the main diagonal of the CSM, that is, excluding the auto-spectra from the microphone signals.

The CSM reveals that the beamforming output is a “mix” of auto-terms and cross-terms formed by the microphone channels. The auto spectrum does not contain any important information in terms of high contrast in the acoustic picture. Moreover, it can obliterate the relevant information in measurement scenarios where incoherent signals are produced at the microphones (e.g. wind tunnel). Here, the amount of signal energy in each microphone channel, respectively in the auto-spectra, is high compared with the cross-spectra, because uncorrelated signals are averaged out in the beamforming process. Thus, eliminating the auto-spectra can improve the dynamic range of the acoustic photo.

Based on the algorithm proposed by Dougherty in [1], in this study the method of excluding the auto-spectra is implemented for time domain beamforming. Microphone data of practical measurements with the *Acoustic Camera* [2] is extracted and fed into a modified beamforming algorithm realized in MATLAB. Results of two measurement scenarios (wind-tunnel and a transient door-closing sound) are shown in comparison to the conventional time domain beamformer. In addition, the algorithm is applied on synchronized multiple measurements [3] in an engine test cabin.

2 BEAMFORMING FORMULAS

2.1 Time domain

The beamforming output of a conventional delay-and-sum beamformer in the time domain is the time averaged sum over each microphone signal p_m with a delay τ_{xm} corresponding to a pixel x in the focus plane:

$$b_T(x) = \frac{1}{M} \left\langle \frac{1}{T} \int \left[\sum_{m=1}^{M} w_m \; p_m(t - \tau_{xm}) \right]^2 dt \right\rangle \tag{1}$$

To normalise the output the averaged sum is divided by the number M of microphone channels. A weighting function w_m can be applied to form a wave number spectrum that satisfies certain conditions in terms of sidelobe levels and spatial resolution.

2.2 Frequency domain

Modifying the time signals with delays corresponds to applying phase shifts in the frequency domain for each frequency. The beamforming output can be split into a data-independent part that contains the phase shifts in the so called steering vector $\vec{S}$ and in a data-dependent part that is formed by the cross spectral matrix C. Based on a short time DFT the averaged microphone signals are cross correlated with each other and listed in C. Thus, the beamforming output becomes

$$b_F(x) = \vec{S}^*(x)\, C\, \vec{S}(x)\,, \tag{2}$$

with

$$\vec{S} = \exp(j\omega\tau_{xm})\,.$$

The beamforming output has to be calculated for each frequency and it yields much more computational effort than in the time domain.

2.3 Excluding the auto-spectra

In general, the sound field is a mixture of correlated and uncorrelated parts arriving at the microphones. In a beamforming process it is assumed that the correlated parts of a source are added coherently whereas stochastic, uncorrelated parts are averaged out, providing a sufficient average time. Suppressing of uncorrelated components is reflected in the off-diagonal elements in the CSM. On the other hand the mean energy value $\psi_{pp}(0)$ in the auto-correlation function of each microphone signal is not affected by the averaging process, it contains the complete signal energy of the soundfield (at the microphone) and, even though of secondary importance, the electronic noise of the microphone channel. Thus, excluding the auto-terms is an effective method. Eliminating the squared fourier coefficients $|P(f)|^2$ in the main diagonal of the CSM is, according to *Parsevals Theorem*

$$\psi_{uu}(0) = \int_{-\infty}^{\infty} |P(f)|^2\, df = \int_{-\infty}^{\infty} p^2(t)\, dt\ , \tag{3}$$

equivalent to removing the (mean) power of the time signals $p(t)$, which are found at $\psi_{uu}(0)$ in the auto-correlation function ψ_{uu}. This leads to the modified delay-and-sum formula including the diagonal deletion in the time domain with uniform shading:

$$b_{TDD}(x) = \frac{1}{M}\left\langle \frac{1}{T}\int \left[\sum_{m=1}^{M} p_m(t-\tau_{xm})\right]^2 - \sum_{m=1}^{M} p_m^2(t-\tau_{xm})\, dt \right\rangle \tag{4}$$

This “cleaned” version of eq. (1) can be interpreted as a centered beamforming output *without* the auto power “offset”.

3 RESULTS IN PRACTICAL MEASUREMENTS

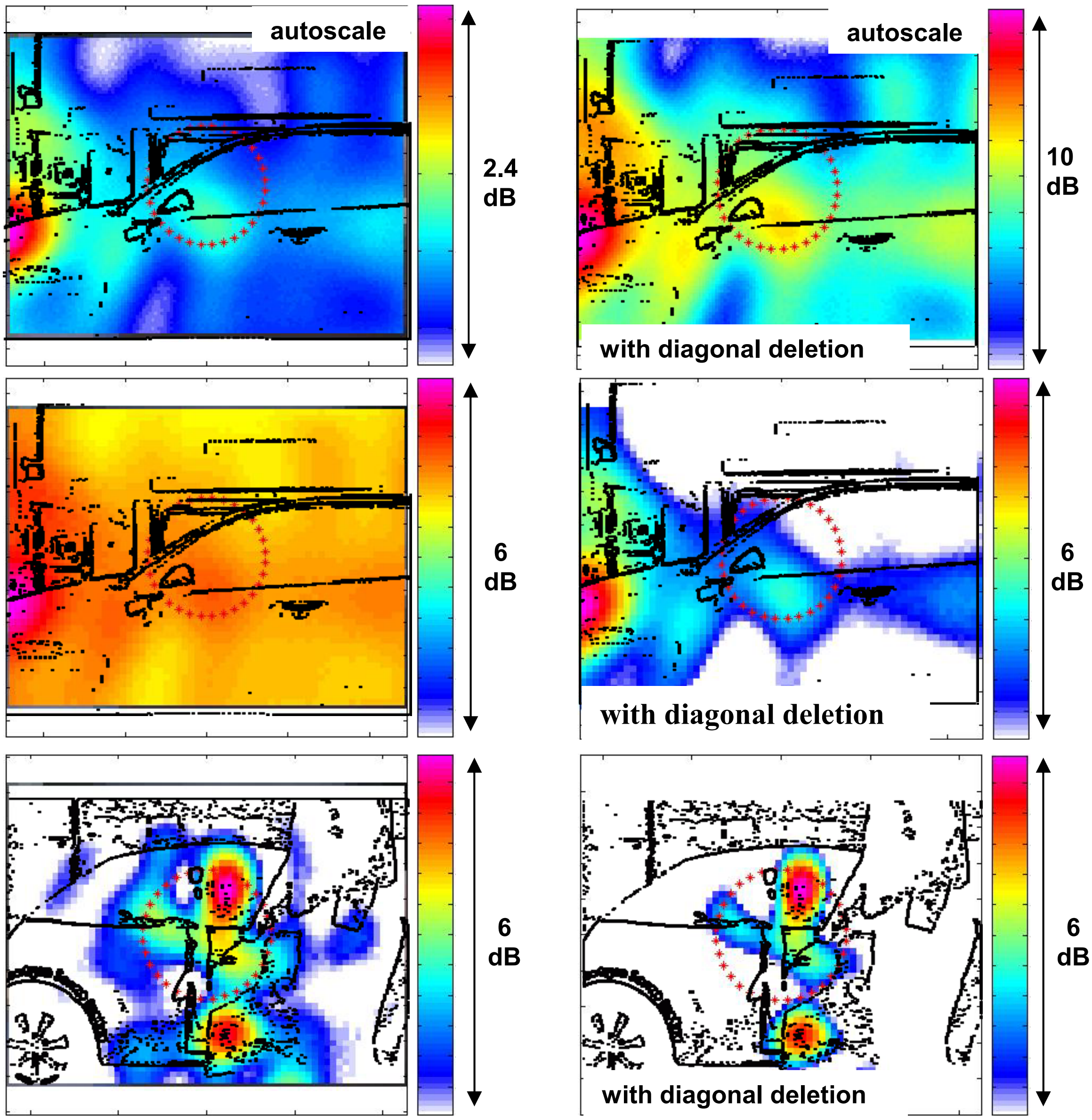

Fig. 1. Time domain beamforming in wind tunnel and investigating closing sound (bottom): conventional beamforming (left); with removing the auto power (right)

Fig. 1 shows the results of beamforming processing without and with diagonal deletion, respectively excluding the auto power of each microphone channel. The dynamic range is increased by approximately 7dB in the wind tunnel measurement (upper pictures) and up to 10dB in the closing sound measurement (bottom pictures).

In Fig. 2 the results of a synchronized multiple measurement in an engine test cabin are shown for the 5 kHz third octave band.

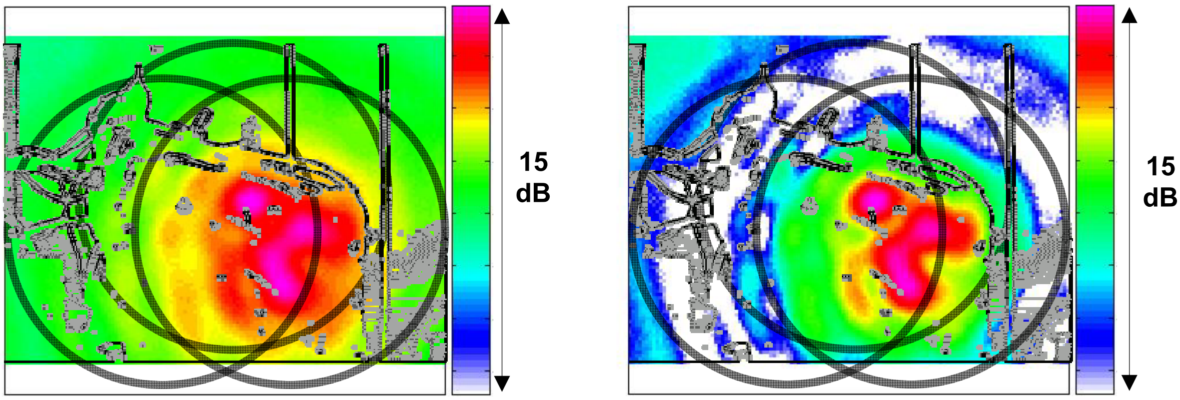

Fig. 2. Acoustic photos of synchronized multiple measurements with a synthetic aperture of 3 arrays: Delay-and-Sum Beamforming (left); Delay-and-Sum Beamforming with removing the auto power (right).

Within the chosen dynamic range of 15dB the modified beamforming algorithm produces an acoustic photo with a higher dynamic range in the main source area (up to 3dB) and also reveals the structure of a ring-like array pattern. Thus, detected pressure levels at the edge can be identified as aliasing artefacts. In the beamforming calculation the overall dynamic range is increased by up to 10dB. The analysis at lower frequencies yields no significant improvements because of the higher correlation of the microphone signals.

4 CONCLUSIONS

The technique of excluding the auto-spectra in the CSM is applied on the delay-and-sum beamforming process in the time domain as it suggested by Dougherty in [2]. The results show that the algorithm works well with broad band signals and particularly in noisy environments. In these cases the amount of signal energy (second part of Eq. (4)) is higher relative to the phased added signals than with tonal signals or in acoustically optimized environments, because up to high frequencies the signals are more and more uncorrelated and averaged out. This effect is more pronounced when analysing multiple measurements of quasi-stationary signals. In combination with synchronized multiple measurements proposed in [3] the algorithm also yields improved results. The amount of signal energy in the acoustic picture increases with a high number of microphones whereas the correlation of the signals

decreases. Thus, time domain beamforming used by the *Acoustic Camera*, principally a broad band method, is extended effectively by removing the auto power in the microphone channels. Ghost image levels are reduced and sound sources are detected much easier in many automotive applications.
Further investigations have to quantify the effect on the absolute level when eliminating the auto power in dependence on the measured signal.

REFERENCES

[1] R. P. Dougherty, *Advanced Time Domain Beamforming.* 10th AIAA/CEAS Aeroacoustics Conference, 2004.

[2] Gesellschaft zur Förderung angewandter Informatik (GFaI): *Acoustic Camera and NoiseImage-Software.* Berlin, http://www.acoustic-camera.com

[3] M. Kern, H. Opfer, *Extension of the Beamforming Method by Synchronized Multiple Measurements.* 1st Berlin Beamforming Conference (BeBeC), 2006.