ADVANCED BEAMFORMING TECHNIQUES IN VEHICLE ACOUSTICS

[1]Sandro Guidati
Ebertstrasse 30a, 52134, Herzogenrath, Germany

ABSTRACT

A problem for the classic beamforming approach is the limited dynamic range of the acoustic mapping. Louder sources are masking quieter sources. This limitation is valid for sources in the same frequency range. With multi band beamforming sources in different frequency ranges can be visualized with the only limitation being the dynamic range of the measurement system.

Coherence filtering techniques allow for the distinction between incoherent sources and the detection of mirror sources. Here, the coherence between an additional sensor placed as reference close to the main source is used to filter the array signals increasing the overall dynamic of the acoustic source mapping. The real sensor can be replaced by a virtual sensor signal as result of the beamforming calculation allowing for a wider range of applications.

Recent advantages in computer technologies allow for a real time processing of the microphone array data resulting in an online visualization of the sound sources.

1 INTRODUCTION

The basic theory of array technology is known and has been used since the beginning of the 20^{th} century[1]. At the introduction, collectors were used to enhance the sound coming from a single direction; the steering had to be done manually. Later the first arrays using microphones were built, principally for submarines. The superposition of the microphone signals was done electronically. With the progresses in computer technology, the array technology also evolved. Nowadays, with 24 bit/low noise A/D converters, PCI Express bus, Quadcore CPUs and fast hard disks and graphic cards, the measurement of a large number of microphones as well as the processing and visualization of high resolution acoustic images is possible online and with very low latency.

2 BEAMFORMING

The basic algorithm used for microphone array source localization is called delay-and-sum or beamforming. The sound radiated by the sources of interest is measured by an array of microphones with nominal omnidirectional measurement characteristic (other measurement characteristics have been used, e.g. spherical beamforming). By superposing the microphone signals the array is steered electronically towards all directions or points of interest. In the basic delay-and-sum algorithm the steering is performed simply by delaying the microphone signals individually to compensate the propagation time between the assumed source position (or direction) and the individual array microphone. The algorithm itself has no constraint for the assumed radiation characteristic used for propagation time calculation. Typically for near-field sources a monopole radiating in free field is assumed. For sources in the far field a plane wave assumption is made. In general the assumed radiation characteristic should match the radiation characteristic of the real sources as well as possible[2, 3]. For example the noise radiated by the trailing edge of an airfoil is best modeled by a dipole[3]. The algorithm has also no constraint on source position. Typically a scan-plane parallel to the array plane is defined on which the strength of potential sources is calculated, but three-dimensional scan surfaces are possible without changing the algorithm.

3 ADVANCED ARRAY PROCESSING TECHNIQUES

Due to the limited aperture of the array and the limited number of sensors, the level difference between two sources that can be localized in a single source map is limited. Every source is mapped with additional ghost sources. The level difference between the real source and its ghost sources is called *plot dynamic*. Typical values are 7-14 dB depending on the frequency and microphone pattern. If the level difference between two real sources is greater than the plot dynamic, the quieter source cannot be distinguished from the ghost sources of the louder source.

3.1 Multi Band Beamforming

This limitation is valid only for a single acoustic image and therefore for the frequency range that is displayed in this image. Sources that radiate noise in non-overlapping frequency ranges can be displayed in separated acoustic images. The usable level difference for those sources corresponds to the dynamic range of the measurement system (up to 100 dB). Figure 1 shows a basic example. A two-way loudspeaker is radiating a broadband noise signal with 46 dB level decay from 1 to 6 kHz. The multiband view displays the cut-off frequencies of each band and the maximum level in that band. The color map is the same for all five images (100 dB yellow, 50 dB blue). One can clearly identify the crossover frequency between the middle and high frequency drivers (2478-3446 Hz).

Figure 1: Multiband view of a two-way loudspeaker. Broadband noise with decreasing level to higher frequencies.

3.2 Coherence Filtering

As mentioned above the level difference between two sources that can be localized is limited (plot dynamic 7-14 dB). One technique to increase the plot dynamic is coherence filtering. An additional sensor is placed as reference in the very near field of the dominant noise source in the acoustic image. The reference signal is measured synchronously along with the array microphones. All signals are Fourier-transformed. Now the complex Fourier coefficients of all array microphones are multiplied with the normalized conjugated Fourier coefficients of the reference signal. The resulting complex values are continuously averaged in time. The averaged complex values are now used for the beamforming. In the acoustic image the referenced source and all other sources that are coherent to the reference are enhanced (up to the correct levels) whereas all other sources are damped. The technique allows identifying reflections, or filtering those parts of radiated noise that are relevant for an observer position of interest (e.g. artificial head in the vehicle interior). The technique is also working for so called virtual reference signals. Here the signal radiated by a point of interested is calculated with the standard delay and sum algorithms. Since the signal is a valid time signal it could be used as virtual reference for the coherence filtering. In the following a basic example is given. Figure 3 shows two sources (loudspeakers) radiating broadband noise. The sources are incoherent. In the acoustic image two sources at the speaker location and an additional source on the wall are displayed.

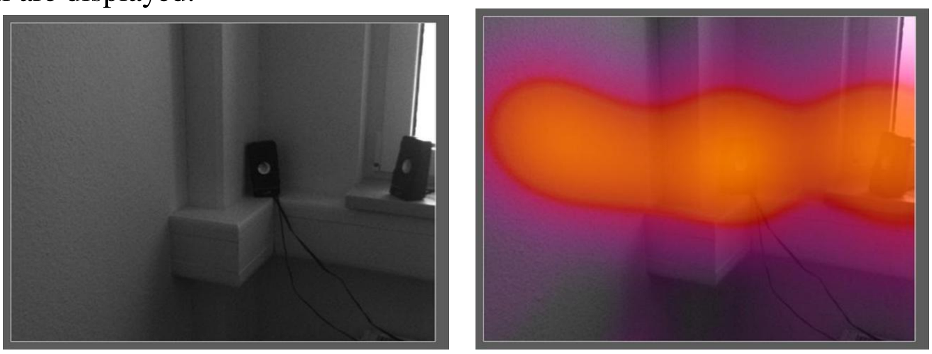

Figure 3: ***Two loudspeakers radiating incoherent broadband noise. At the left side a wall reflection can be seen.***

Now a virtual reference signal (loudspeaker icon) is placed on the position of the left loudspeaker (figure 4, left image). Due to the coherence filtering the right loudspeaker and the wall reflection is damped. By moving the virtual reference point on the right loudspeaker the left speaker source disappears (figure 4, right image), but the reflection still remains visible.

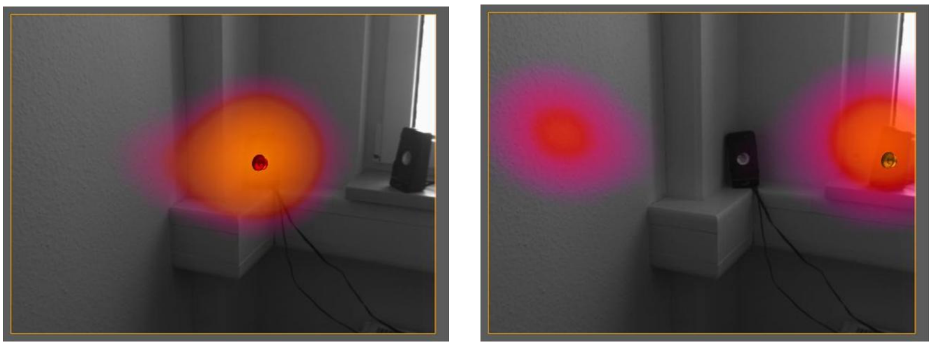

Figure 4: ***Coherence filtering with virtual sensor***

To distinguish between original source and reflections further processing technique is needed. The transfer function between the virtual reference signal and all array microphone signals is calculated and then transformed back into the time domain. Figure 6 shows the magnitude envelope of all resulting impulse responses. As expected the envelope displays two maxima.

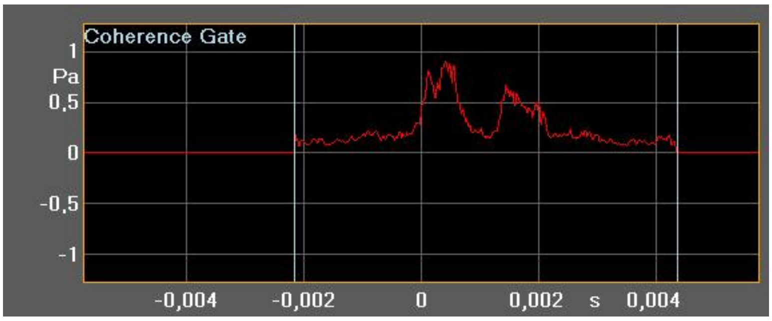

Figure 5: ***Impulse Response between Virtual Sensor and Array Microphones***

Now the impulse responses are gated using a rectangular window, transformed back in the frequency domain and then used for the beamforming. Figure 6 shows the beamforming result in the case that the first maximum in the envelope is gated. The right loudspeaker is visible, whereas the reflection is cancelled.

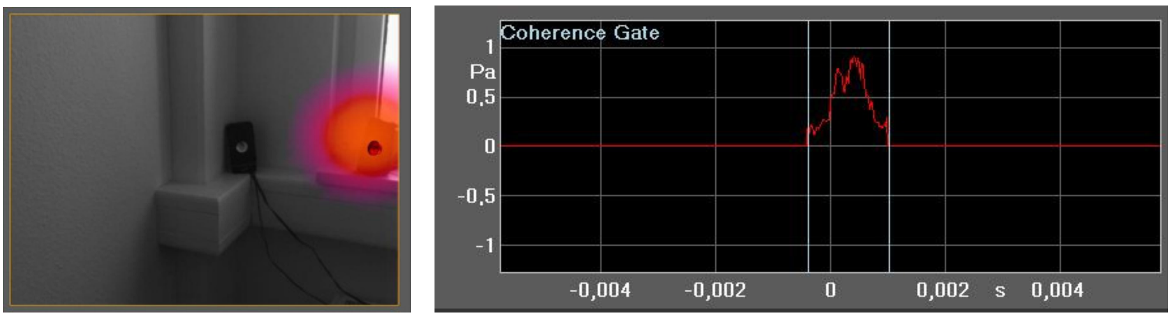

Figure 6: ***Gated Impulse Response (first event)***

Now the second maximum in the envelope is gated. As expected the real source is cancelled and the reflection remains (figure 7). In the case that the virtual sensor is placed on the reflection and not on the original source the envelope would move towards negative values with the original source being always the first maximum.

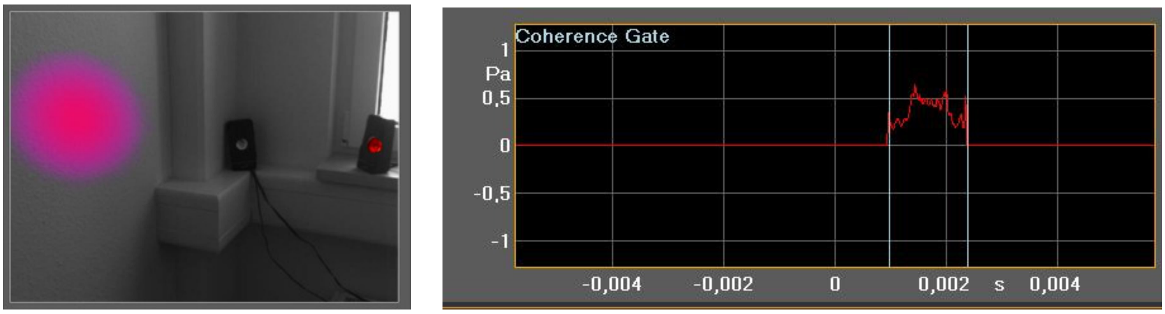

Figure 7: ***Gated Impulse Response second event)***

A standard approach often used in array processing is the principal component analysis. Here a correlation matrix is calculated as outer product of the Fourier transformed array signals. Each eigenvalue of this matrix corresponds to one incoherent source in the signals. By canceling of the single eigenvalues sources can be separated. This techniques is very time consuming but delivers very good results in the case that the source are located in different dynamic ranges. Then the highest eigenvalue corresponds to the loudest source and so on. In the case of incoherent sources that have similar levels the eigenvalues have also similar values. A simple distinction is no longer possible.

4 APPLICATIONS IN VEHICLE ACOUSTICS

Due to the reduced low frequency resolution and the need for a known radiation characteristics (e.g. monopole in free field) not all interesting problems of vehicle acoustics can be solved. Nevertheless, in the following a typical example for the application of the beamforming technique in vehicle acoustics is shown.

4.1 Engine Cylinder Measurement

In idle the main noise sources in a vehicle are the cylinders. Figure 8 shows a time averaged acoustic image of a standard four cylinder engine in idle.

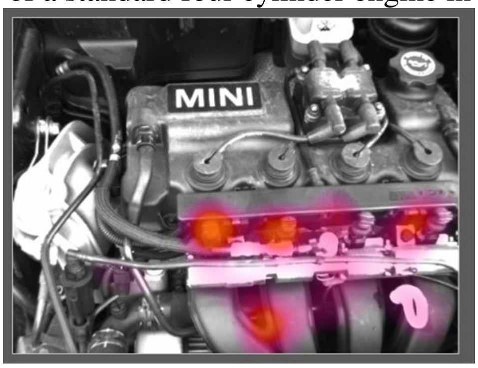

Figure 8: ***Radiation of a four cylinder engine in idle between 5-7 kHz.***

With the use of slow motion and adapted temporal filtering it is possible to detect, visualize and auralize the noise radiation of each single cylinder. Figure 9 shows the sound events for the single cylinders in the order of the engine cycle in the frequency range 5-7 kHz.

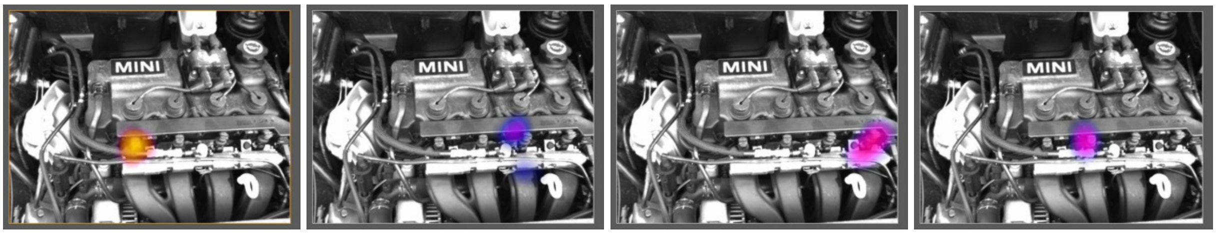

Figure 9: ***Noise radiation of single cylinders in a four cylinder engine at idle (5-7 kHz).***

The first cylinder is dominant. The third cylinder displays the lowest level and only poor localisation. By looking in another frequency band (in the multiband view) one can see that the main acoustic energy of this cylinder is radiated in a lower band (2-4 kHz, figure 10)

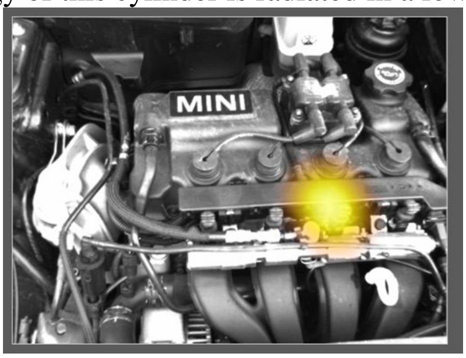

Figure 10: ***Noise radiation of the third cylinder in the 2-4 kHz frequency band.***

5 CONCLUSIONS

In this paper advanced techniques for microphone array processing have been presented. Latest progresses in computer technology allow for applying all techniques online and in real-time with very low latency in combination with frequency-domain beamforming. The possibility to visualize sound sources in real time with high accuracy and immediately observe all effects of modification at the sources offers a new work quality to the acoustic engineer.

Techniques like multi band beamforming and coherence filtering increase the limited dynamic range of the standard beamforming approach and give valuable information about the character and dependency of the various sources. Slow motion and temporal filtering allows for the analysis of very short noise events.

6 REFERENCES

[1] D.H. Johnson and D.E. Dudgeon. Array Signal Processing – Concepts and Techniques, Prentice-Hall, NJ. (1993).EU Project “Quiet City Transport” (QCITY), TIP4-CT-2005-516420

[2] T.F. Brooks and W.M. Humphreys Jr., Effect of directional array size on the measurement of airframe noise components, 5th AIAA Aeroacoustics Conference, May 10-12 1999 Seattle/USA, Paper AIAA 99-1958.

[3] S. Guidati: Entwicklung eines akustischen Messsystems für den Einsatz in aero-dynamischen Windkanälen. PhD Thesis, Library of University of Stuttgart, Signature: Diss. 2005/1987. (2005).