Source Localisation with Beamforming in Vehicle Acoustics

S.Guidati[1]

[1] HEAD acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Email: sandro.guidati@head-acoustis.de

Introduction

The basic theory of array technology is known and has been used since the beginning of the 20th 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.

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[4, 5]. 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.

Advanced Array Processing Techniques

In the following a number of advanced processing techniques are described. Like the basic delay-and-sum algorithm, these techniques are well-known for a long time, but have been limited to scientific usage. Progresses in computer technology as well as optimization of the algorithm implementation allow real-time processing, and therefore the use in the industrial context.

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.

Multiband 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.

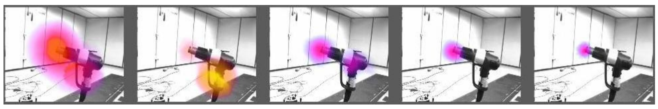

Figure 2: Multiband view of a hot-air gun.

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). 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.

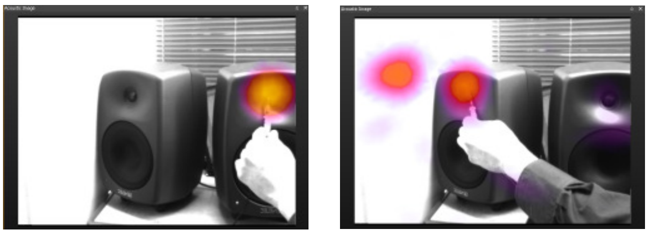

Figure 4: Coherence filtering with additional sensor

Now in addition the acoustic image of the unfiltered microphone signals is calculated. The difference between the energy equivalent values of the unfiltered and filtered acoustic images is calculated and displayed as new acoustic image. Here all sources that are coherent to the reference signal are damped. A typical application is the damping of the dominant noise source in an acoustic image. Secondary sources that have been hidden by ghost sources become visible. The plot dynamic is increased. In the basic example the sensor is again placed at the right source (figure 5, left). The right source disappears, whereas the left source and the mirror source remain. By placing the sensor at the left source only the right source remains (figure 5, right).

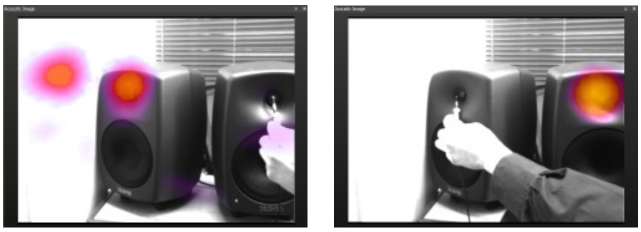

Figure 5: Incoherence filtering with additional sensor

Figure 6 shows the noise radiated by a hot-air gun. There are two major noise sources. In the left picture the motor noise is dominant in a given frequency band. With incoherence filtering the dominant noise source is damped and the nozzle noise (fan noise) is enhanced.

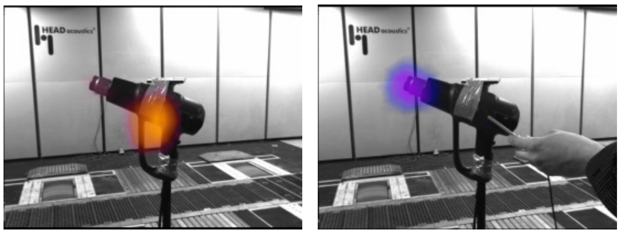

Figure 6: Unfiltered acoustic image of the hot-air gun (left). Incoherence filtering with additional sensor (right).

There is no limitation on the kind of sensor used as a reference as long as it delivers a time signal. Typical examples are microphones, accelerometers, laser vibrometers, etc. The damping effect increases with sensors that are not effected by cross talking (laser, accelerometer). For the use in the high frequency range (above approx. 7kHz) it is also important to use a fixed sensor since even small movements can affect the phase relation between the reference and the array microphones. The procedure can be iterated adding multiple sensors. The additional computation effort is proportional to the number of sensors.

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 some examples for the application of the beamforming technique in vehicle acoustics are shown.

Engine Cylinder Measurement

In idle the main noise sources in a vehicle are the cylinders. Figure 7 shows a time averaged acoustic image of a standard four cylinder engine in idle.

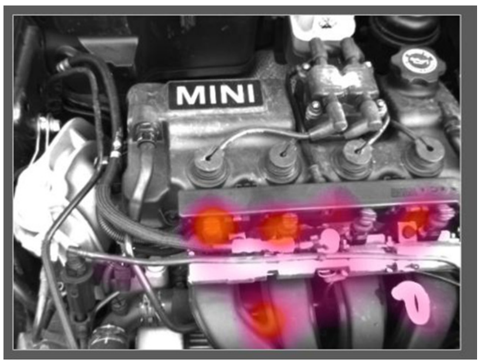

Figure 7: Radiation of a four cylinder engine in idle between 5-7 kHz.

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 8 shows the sound events for the single cylinders in the order of the engine cycle in the frequency range 5-7 kHz.

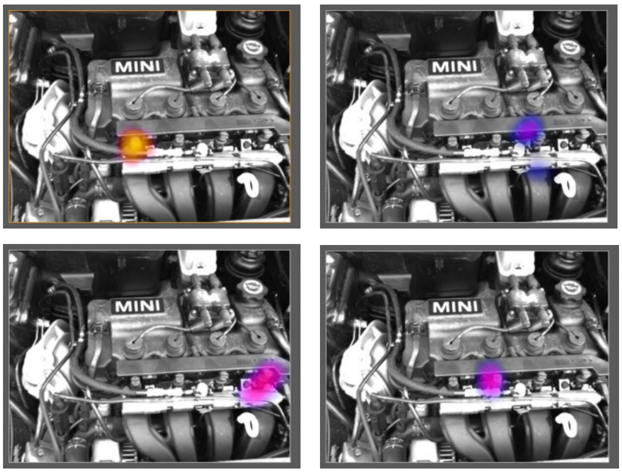

Figure 8: Noise radiation of single cylinders in a four cylinder engine at idle (5-7 kHz).

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 9)

Figure 9: Noise radiation of the third cylinder in the 2-4 kHz frequency band.

Leakage Detection

A typical problem for car manufacturers is the detection of damage e.g. at the door or window sealing. During the drive air is pressed through even small holes due to the pressure gradient between the vehicle exterior and interior. The air flow is turbulent causing either broad band noise or even single tones. Both kinds of noises are annoying for the customer.

To avoid these effects the sealing has to be carefully checked after the completion of the vehicle mounting. One way to perform this test is to place a powerful loudspeaker inside the car. The loudspeaker emits broad band white noise. Now the vehicle exterior is scanned for extra noise emission. Typically a “perfect” car with undamaged sealing is used as reference. The scan can be performed with the use of a simple sound level meter placed in the very near field of the interesting sealing. If the detected levels are significantly higher than the levels detected for the reference car the sealing is damaged.

The use of a sound level meter requires manual interaction of an operator and is therefore time consuming. The test can be performed much faster with the use of a real-time beam forming array. The array is simply moved around the car. The leakage due to a damaged sealing is detected as additional noise source. Once the major leakage is detected and fixed also minor leakages can be identified. With the use of a traversing system the test could be automated. If the test environment is contaminated with additional noise the known loudspeaker signal can be used as reference signal for the coherence filtering described above. Now the additional noise is damped significantly.

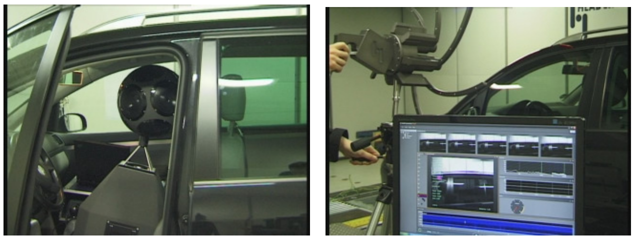

Figure 10: Leakage detection with point source inside the vehicle (left picture) and microphone array (right picture)

Figure 11: Still images taken from a continuous video stream showing the leakage

Another way of leakage detection is shown in the next figure. Here a person inside the vehicle is blowing compressed air against the door sealing. In case that the sealing is undamaged only little noise is radiated to the exterior (figure 12, left picture). In case of a damaged inner sealing the noise is not only radiated from the damage point but also transmitted along the sealing channel and then radiated at both ends.

Figure 12: Leakage detection with compressed air. In case of a damaged inner sealing the noise is radiated at the damage point as well as on both ends of the sealing (right picture)

Beamforming with measured transfer functions

The localization of noise sources in the vehicle interior is important to increase the driving comfort for the customer. The use of the standard beamforming technique often fails due to the problem of reflections and damping. One way to increase the accuracy of the beamforming result is to use measured radiation characteristics instead of the standard free field assumption. For this purpose a small loudspeaker is placed sequentially at any possible noise source location of interest. The speaker is emitting a known broad band signal (e.g. pseudo noise). The array system measures the sound and calculates the transfer function between each speaker position and array microphone.

Figure 13: Beamforming with measured transfer functions. Array placed in the middle of the cabin (left). Speaker postions (red dots) for the measurement of the transfer functions (right).

The transfer functions are stored on hard disk. The position of the speaker is marked on the video image. After the recording of all positions the speaker is removed and the vehicle is driven under the interesting running conditions. With standard beamforming the acoustic image is calculated for a grid of scan positions on a two or three dimensional scan surface. Now the beamforming algorithm is only applied for the previously measured speaker positions using the transfer functions instead of the standard free field model. The transfer functions include all reflections and damping inside the vehicle cabin leading to a higher accuracy of the localisation. In fact, it is also possible to detect sources that are not directly visible by the array (e.g. behind the seat)

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.

Leakage detection for a passenger car is a typical industrial application for real time array processing allowing for interactive or automated search and fast results.

Beamforming using measured transfer functions instead of the standard free field radiation characteristics allows for the localisation of sources in highly reverberant environment.

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, G. Guidati and S. Wagner, A modification of the classical beamforming algorithm for reverberating environments, 7th International Congress on Sound and Vibration, July 4-7 2000, Garmisch-Partenkirchen, Germany, Paper 6-3167.

[4] S. Guidati, G. Guidati and S. Wagner, Beamforming in a reverberating environment with the use of measured steering vectors, 7th AIAA/CEAS Aeroacoustics Conference, May 28-30 2001, Maastricht, Netherlands, Paper AIAA-2000-2166.

[5] 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).