REAL TIME SOURCE LOCALISATION FOR INDUSTRIAL APPLICATIONS

S. Guidati HEAD acoustics GmbH, 52134 Herzogenrath, Germany

1 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.

1.1 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.

1.2 Time-domain beamforming vs. frequency-domain beamforming

The simplest way to perform beamforming is calculation in the time domain. The time signals recorded by the microphones are shifted (delayed) to compensate the propagation time between the assumed source and the individual microphone and then summed. The resulting new time signal is segmented into blocks. For each block the signal energy is calculated. The energy calculated for all scan points (on a plane or a three-dimensional surface) is displayed in the acoustic image. The differences in propagation time between the microphones and scan points are often very small. Depending on the sample rate there is a granularity for shifting the time signals, affecting the precision of the beamforming results. For typical sampling rates (32 kHz, 48 kHz) this effect is not negligible for sources that radiate noise in the higher frequency range. One way to overcome this effect is to sample the microphone signals at a higher sampling rate (e.g. 192 kHz). This increases the amount of data to be recorded and processed, making online evaluation almost impossible. The other way is to resample the time signals to a higher rate.

Another way to perform beamforming is calculation in the frequency domain. The recorded time signals are divided into overlapping blocks. Each block is transformed in the frequency domain. To minimize wrap-around errors during the following operations the time signals are windowed before the transformation. The delay in the time domain corresponds to a phase shift in the frequency domain. The correct phases must be calculated for each microphone and frequency. The advantage compared to time-domain beamforming is that this operation is equivalent to a high-order resampling. As a result there is no granularity for the shift operation. For each frequency the Fourier coefficient of all microphone signals are phase-shifted (corresponding to a complex multiplication) and then summed. The energy equivalent value is calculated over the frequency area of interest and finally displayed as acoustic image.

Apart from errors due to sample granularity in the time domain and wrap-around in the frequency domain, the results of the two approaches are equivalent. Most of the techniques described in the following can be performed online when executed in the frequency domain but require significantly more computation time when using time-domain beamforming. Therefore in the following, only frequency-domain beamforming is used.

2 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.

2.1 Multiband beamforming

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.

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 shows a multiband view of a hot-air gun. Here one can identify the two major sources: broadband aeroacoustic noise coming out of the nozzle and narrow band noise (second illustration) generated by the electric motor.

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

Beamforming performed in the frequency domain has the great advantage that the acoustic images are first calculated by means of the Fourier transform for each single frequency and then summed for frequency ranges. Therefore it costs no additional computation effort to calculate a number of acoustic images with non-overlapping frequency ranges, and the increase of computation effort for overlapping frequency ranges is almost negligible. By displaying a number of acoustic images covering a wide frequency range an instantaneous overview of the complete frequency range and all sources radiating in this range can be achieved.

In time-domain beamforming each frequency range would require an individual bandpass filtering of all array microphone signals.

2.2 Coherence/incoherence 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.

Now a reference sensor is placed in the near field of the right source. As expected the left source disappears (figure 4, left). Additionally the source at the wall disappears and is identified as a potential mirror source. This can be verified by placing the sensor at the left source (figure 4, right). The wall source remains and is therefore coherent to the left source.

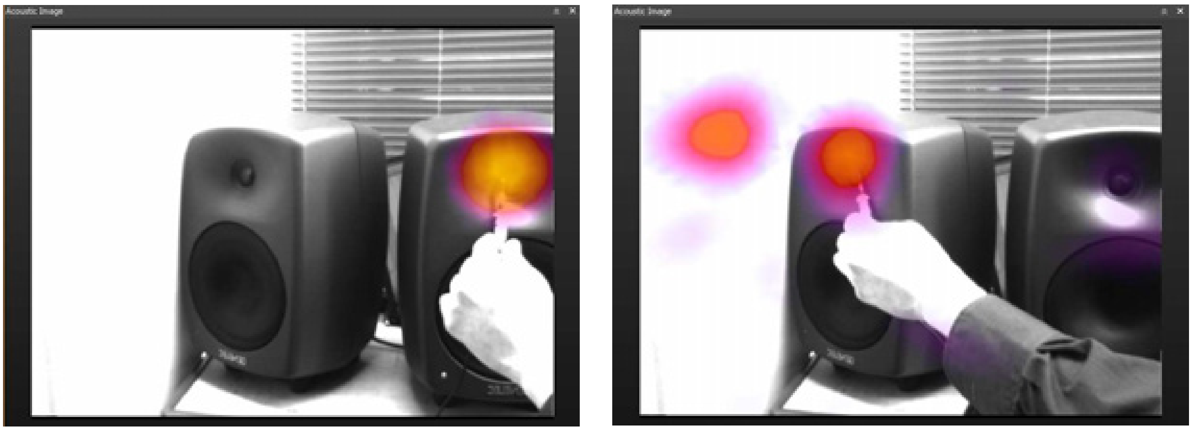

Figure 4: Coherence filtering with additional sensor.

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).

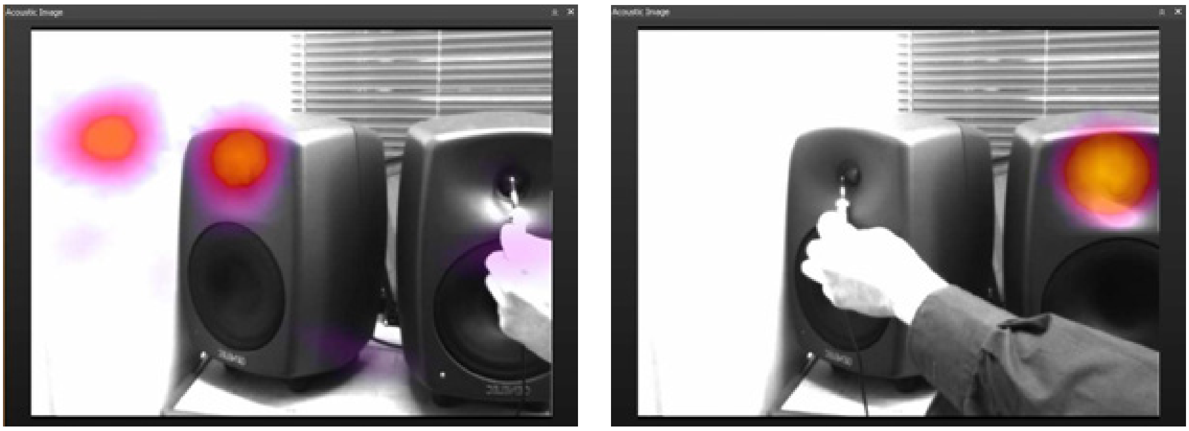

Figure 5: Incoherence filtering with additional sensor.

Figure 6 shows the noise radiated by the hot-air gun. As mentioned above 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.

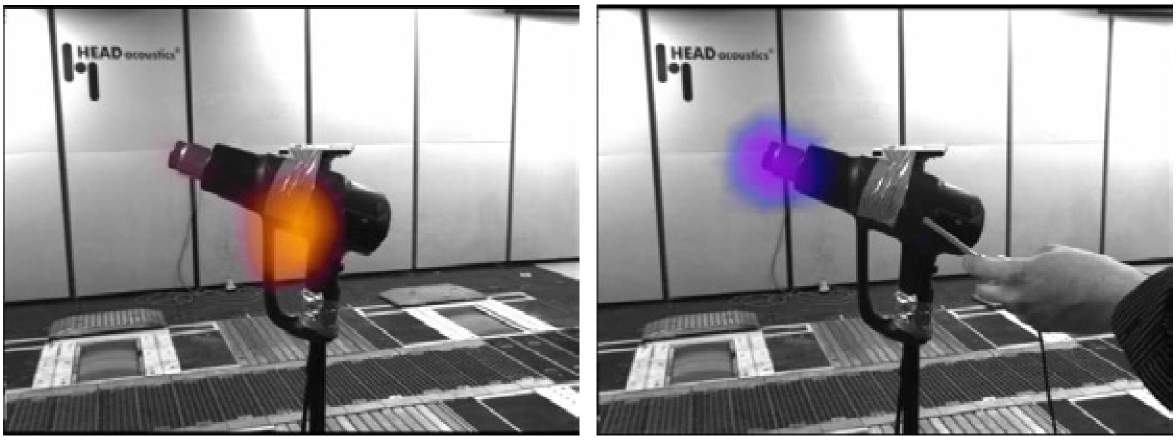

Figure 6: Unfiltered acoustic image of the hot-air gun (left). Incoherence filtering with additional sensor (right).

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 procedure can be iterated adding multiple sensors. The additional computation effort is proportional to the number of sensors.

In time-domain beamforming the described technique would require a transformation in frequency domain, the multiplication with the complex conjugate of the reference spectrum and finally the inverse transformation back to time domain.

2.3 Industrial Application Example

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

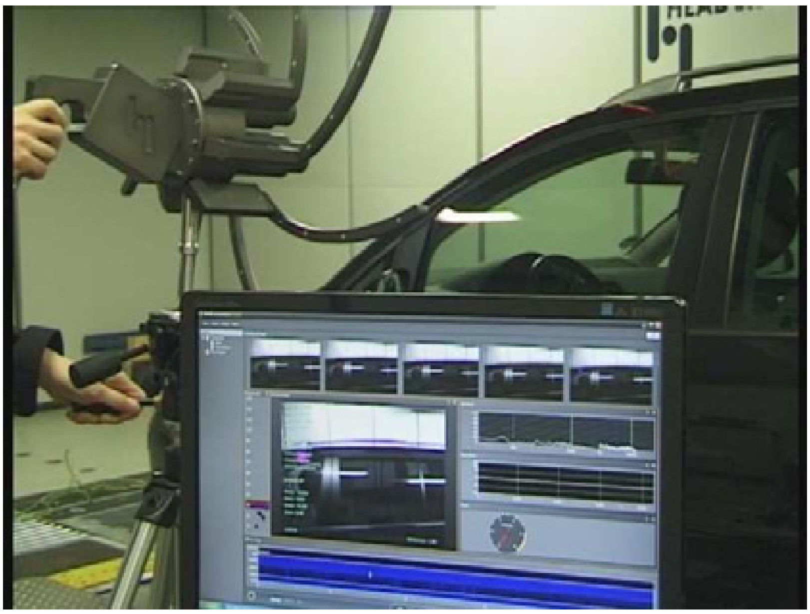

Figure 8: Online evaluation. The software (foreground) displays online the current source map while the array is moved around the vehicle.

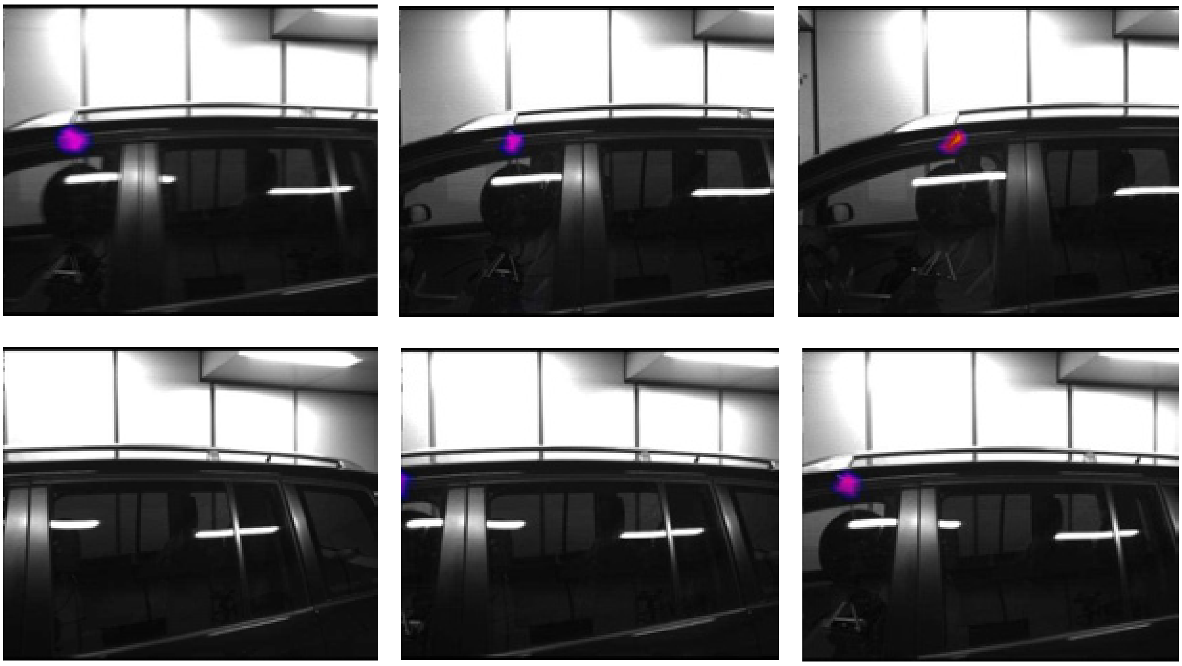

Figure 9: Leakage detection (damaged door sealing) with a real time beam forming array (single snap shots of a continuous video recording.

3 CONCLUSION

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. Classical time-domain beamforming would require considerably more computation time. 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.

The combination of microphone array technology with optical detection of three-dimensional source distributions increases the accuracy of localization and quantification of the sources.

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.

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