Threedimensional Beamforming considering the Effects of covered Microphones

O. Jaeckel, D. Döbler, R. Schröder, A. Meyer[1]

[1] *Gesellschaft zur Förderung angewandter Informatik e.V. (GFaI e.V.), 12489 Berlin-Adlershof, Germany, Email: jaeckel@gfai.de, doebler@gfai.de, schroeder@gfai.de, meyer@gfai.de*

Introduction

During the past years, the method of acoustic beamforming using microphone arrays has found a wide application range in various fields of research and industrial problems. While most academic and commercial systems currently used are still based on planar array geometries, there are also some systems available that utilize transparent [1] or solid [2] threedimensional microphone arrays. In the case of our Acoustic Camera [3] [4], we use two acoustically transparent microphone arrays with a special distribution of 48 or 120 microphones over the surface of a sphere.

The calculation method here is still the classical delay-and-sum beamforming, with the difference with respect to the 2D-case that the simple virtual measurement plane for the computation of the time delays between each individual focus point on that plane and all the microphones is now replaced by the complete surface of a 3D-model of the device under test. That means the beamformer now has to compute all the time delays between every microphone and each point lying on the surface of that 3D-model, which on one hand substantially increases calculation times. On the other hand, this use of CAD-models in the 3D-case is a necessity and a well-founded decision, because the detailed knowledge of the complete object geometry is an inevitable precondition for a successful acoustic 3D-beamforming and generally for beamforming onto depth structured objects which needs a correct focus information [5].

But even if the problem of the correct focus distance determination for each individual point at the model surface has been solved, there remains a further problem which is the main topic of this paper. Said problem refers to surface areas of the object from which the emitted soundwaves can not reach the microphone array directly. Comparable to a similar situation in an optical system, the beamformer is faced with the problem of fully or partially shaded regions of the 3D object.

The paper introduces and discusses two possible approaches to handle this problem. A first method that is already implemented will be described in more detail in the following. The first method is a simplified algorithm that is well suited if the array is relatively small compared to the object's size. To show its advantages and limitations, practical measurement examples for 3D mappings will be given.

A second possibility will be presented also that is still a matter of ongoing research. The second method discussed intends to actually determine all the microphones that actually have "free sight" onto each 3D- model point.

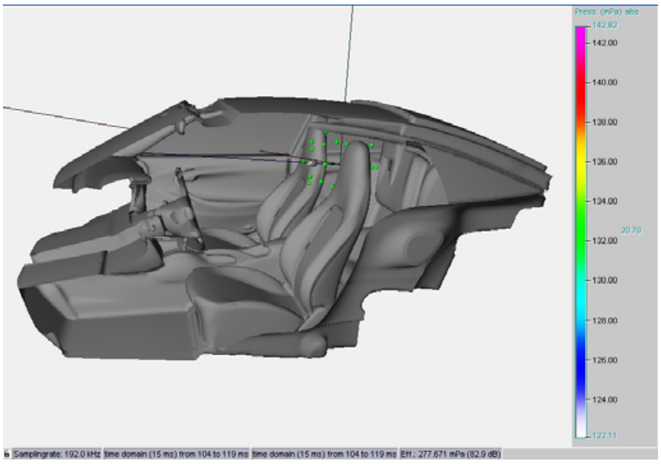

Figure 1: Typical measurement setup for 3D-beamforming in a car interior. The array (green dots) is placed in the middle between both seats.

Determination of Acoustic Visibility

An instructive example for the difficulties with 3D-beamforming is shown in Figure 1 and Figure 2. In Figure 1, a real measurement setup of a 48 channel transparent microphone array inside a car is shown, where the array position has already been correctly fitted geometrically with respect to the 3D-CAD-model of the car's interior.

When standard beamforming is performed in this situation, the result now may look like that shown in Figure 2. In this case, the location of the dominant sound source is found correctly, it was a small speaker in the left corner emitting simulated clicks of a virtual "relais" of the car's blinker.

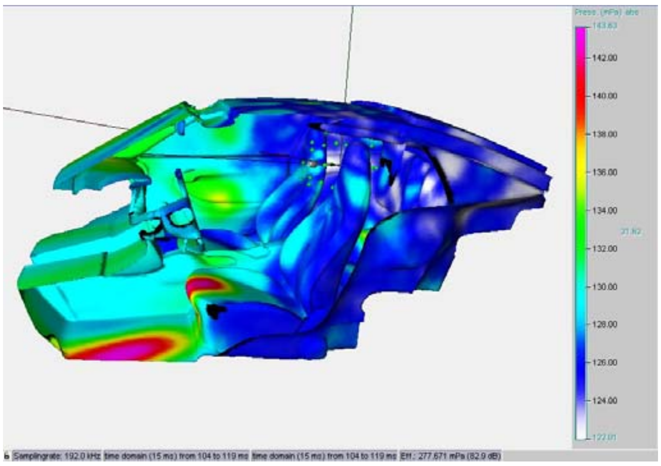

Figure 2: Standard 3D-beamforming simply maps onto the complete model surface without taking any covered microphones or acoustic shading into account.

This speaker just gives the driver an acoustic feedback of the left indicator light's function because the actual blinker circuitry nowadays is of course implemented purely electronically and not with electromechanical relais.

The problem with the standard 3D-beamforming is that it is completely ignoring the actual topological and acoustical relations between array and model. It simply maps onto the complete model surface, wether the microphone array has a free acoustic path to a certain point on that surface or not. This will cause problems with the interpretation of the 3D-maps, especially when the coloured model is graphically rotated by the user on the computer screen, and in more complicated cases with more sources it can give users completely misleading results. In Figure 2, for example, even the outer surfaces of both front seats are colour coded (in blue) with some sound pressure level value, while this is not realistic because the microphone array will be shaded by these seats themselves.

To correctly handle this situation would, in theory, require a complete simulation of the whole acoustic field inside the car, including all frequency modes, structural vibrations, all the energetic transfer paths, the material properties, acoustic diffractions, reflections and absorptions, a task which is still far out of reach at present. To simplify the task, we treated the acoustic path problem like a comparable problem in optics. Our main goal was to determine the "acoustic visibility" of every point on the model surface with respect to the array position. We developed two methods to achieve this.

The first approach to accomplish a more realistic 3D beamforming map is relatively straightforward. It excludes all points on the model surface from the beamforming calculations that do not have a free linear path to the geometric center of mass of the array. This simplification will give good results as long as the array aperture is small enough compared to the geometric extent of the object's surface. The computational load is mainly determined by the complexity of the 3D model. Usual 3D object models consist of N = 200000 to 300000 triangles, hence it must be tested for the same number of straight lines from each triangle's center to the array barycenter if there is an intersection with another plane (other triangles in the model). The algorithm has a complexity of $O(N^2)$when N is the number of triangle centers in the model. In this case, the problem reduces to a ray tracing task that can be solved quite efficiently by applying standard algebraic methods and optimizations known from the field of computer graphics. This first method is already implemented in a prerelease of our new beamforming software. Application examples will be given in the next section.

The second method is computationally more intensive but more general. While the first approach will fail when the geometric dimension of the array is too large, the second method actually tries to figure out which detailed subsection of the microphone array has free paths to each model triangle. This approach better reflects the true situation for complex model surfaces with convex as well as concave parts: now some points on the surface will only be acoustically visible by a certain subset of the microphones, while others may be shaded completely, and some still may be completely visible. The algorithmic complexity of this second method is $O(N^2M)$, where N is again the number of triangles and M is the number of microphones. Here, the ray tracing problem as known from the first method must be solved completely for each individual microphone.

While the second method is now able to solve the problem of partial visibility, at the same time it introduces new problems related to the beamforming method itself. In algorithm 1, either the complete array is used for a certain model point or no beamforming is done at all for that point. With algorithm 2, however, the situation is much more intricate: For all the points with partial visibility, the beam-former will no longer consist of the whole array, and a completely different number of microphones may be used for the actual beamforming calculations. Practically, in some extreme cases we are performing a beamforming with a different subarray for each focus point on the model surface! A minimum number of microphones is necessary that can be specified by the user. On the other hand, the problem of energy normalization of the resulting beamforming sound pressure levels of all the model points in relation to each other remains unsolved. E.g., a subarray with only half the channel number will not only have a different array gain, it will also show a totally different threedimensional array pattern and contrast behaviour. Additionally, the remaining subarrays for each point resulting from the visibility calculations are seldom well behaved and optimized array geometries, but rather more or less stochastic microphone clusters depending on the topology of the model itself. Because these problems are very complex and they are still under ongoing research, the following application examples refer to the first method only.

Measurement Examples

In Figure 3, the car interior example from above is shown with the first algorithm for the determination of the shaded model regions. Only those parts of the surface are coloured onto which the array center has free sight.

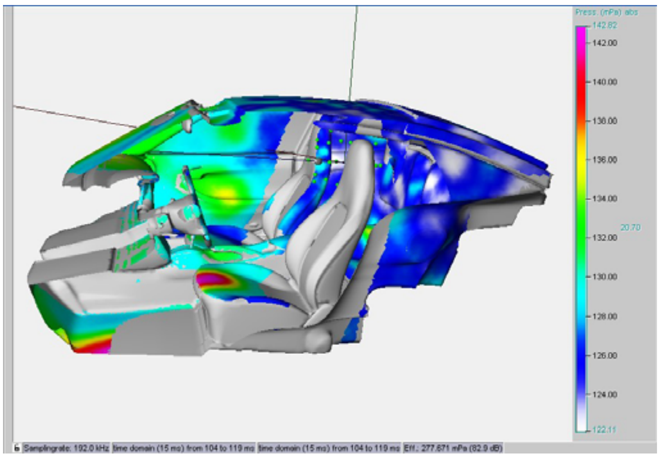

Figure 3: Beamforming inside a car as in Fig. 2, but now with an algorithm that determines the free acoustic paths from every surface point to the array. Shaded regions are excluded from the mapping.

As a second example, the opposite case of a planar array outside a given 3D model structure is demonstrated. In Figure 4, where a complete beamforming map onto a combustion engine is shown, the user could get the misleading impression that there are some sound sources located to the right of the motor, in the vicinity of the crank shaft. But this seemingly nice 3D-visualisation is again completely misleading.

In the acoustic reality, no clear statements about sound sources in this region of the motor can be made at all for the given array position. To find true sources there would require an additional measurement taken from another array position relative to the motor.

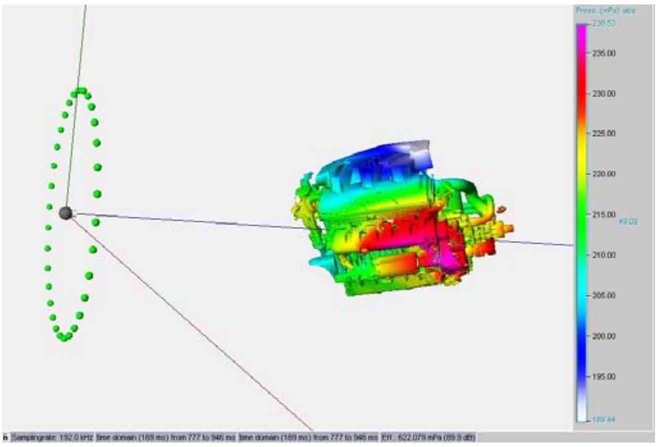

Figure 4: Conventional beamforming with a planar array onto a 3D motor model. Again the beamformer simply maps onto the complete surface.

In Figure 5, the 3D result for the motor example calculated with the first of the two visibility algorithms is much more plausible from the user's point of view. Now the covered parts of the motor model as seen from the array position are explicitely excluded from the beamforming calculation. Only the region of free sight is coloured. This gives the user a clear indication that another measurement would be necessary to cover the right side of the motor. Even if the model is rotated on the screen no further fake emissions occur.

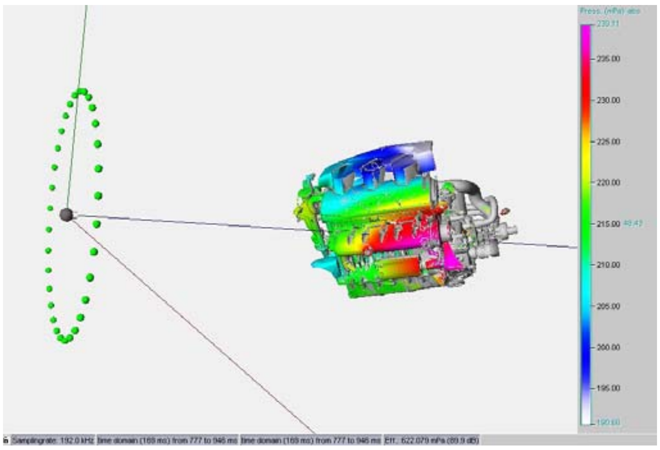

Figure 5: Beamforming of the example in Fig. 4 with taking the acoustically shadowed model parts into account.

Summary and Outlook

Two algorithmic methods to handle the problem of covered microphones in 3D beamforming applications were presented in this paper. The first method is based on an optical ray tracing approach and determines all straight lines from the array centre of mass to each individual point on the 3D model surface with no obstacles (other model triangles) in between. This method is already efficiently implemented and first practical 3D measurement experiences clearly demonstrate that is is very useful.

The second algorithm is an extension of the first one and takes all the individually covered microphones into account. This method can, in principle, also calculate partially shadowed model regions. However, there are still a lot of problems to be solved with this method. Especially the questions of sound pressure level normalisation and erratic array pattern behaviour are not sufficiently explored yet. As this is an ongoing research work, we will try to find appropriate solutions or other approaches for these complex problems in the future.

References

[1] Meyer, A.; Döbler, D.: Noise source localisation within a car interior using 3D-Microphone arrays. Proc. of 1st Berlin Beamforming Conference, Berlin, 2006.

[2] Juhl, P.M.; Petersen, S.O.; Hald, J.: Localizing sound sources in 3-D space using spherical harmonic beamforming. Proc. of Internoise 2005, Rio de Janeiro, Brazil.

[3] Meyer, A.; Döbler, D.: Dreidimensionale akustische Kartierung mit kugelförmigen Mikrofonarrays. DAGA 2007, Stuttgart.

[4] Heilmann, G.; Meyer, A.; Döbler, D.; Mayer, S.: Car interior measurements using 3D-Microphone arrays. Proc. of Internoise 2007, Istanbul, Turkey.

[5] Döbler, D.; Heilmann, G.; Schröder, R.: Investigation of the depth of field in acoustic maps and its relation between focal distance and array design. Proc. of Internoise 2008, Shanghai, China.