



BEAMFORMING WITHIN THE MODAL SOUND FIELD OF A VEHICLE INTERIOR

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ABSTRACT

Physically the conventional beamforming method (CBF) is based on a decomposition of the incident-wave field into plane or spherical elementary waves whose relative phases are interpreted to mean that the directions of incidence and source locations can be identified and classified. A central assumption of the CBF is its application in the acoustic free field. This assumption is particularly affected in cavities such as a vehicle interior, in which to a certain frequency range (below the Schroeder frequency) modal influences dominate the sound field. Over the past years the beamforming in modal sound fields, however, has reached a new level. Studies have shown that modern beamforming methods allow reliable sound source localization inside rooms, in particular in a vehicle interior, in a range above the Schroeder frequency. Furthermore, developments such as the use of generalized cross correlation and automated mode detection to take into account modal sound field effects of a vehicle interior have raised the localization quality of the beamforming below the Schroeder frequency to a new level. A detailed presentation of results will show the potential of these methods individually, and also in combination using the example of a real vehicle.

1 INTRODUCTION

In the field of automotive engineering there is a lively international competition within each vehicle development sub-discipline. Depending on the vehicle segment, the requirements and customer profile, the development priorities differ for each competitor. In the premium segment, the part NVH, which covers all areas of the acoustic and vibration properties of an automobile has moved closer to the focus of the customers and therefore also the focus of the companies in recent years. Consequently, the effort of the developers is constantly increasing and they take advantage of new, improved analytical methods for practical application. It is the same with the beamforming, the localization and classification of sound sources. The beamforming offers the opportunity to locate sound sources both outside and inside the vehicle, which is why it is often termed source localization. The localization of sound sources

within the vehicle turns out to be much more difficult than locating them outside. This is mainly due to the fact that the basic assumptions, under which the beamforming is applied are affected in a closed space. These are in particular the assumption of an acoustic free field and the assumption of a monopole characteristic for the sound sources. In addition, a vehicle interior has a highly complex sound field. On the one hand it is a "small room" (volume approximately 3 m^3), which has a complex geometry. This leads to a complex modal structure. On the other hand, the surface is covered with materials with different absorption properties, which can cause very different diffraction and attenuation effects. In this acoustically demanding environment, the application of beamforming is still a challenge. The best known beamforming algorithm "delay and sum", which is often referred to as Classical Beamforming (CBF), performs very poor under these conditions with respect to parameters such as dynamics and resolution. In the past, therefore, a sound source mapping in a vehicle compartment was limited in performance. However, over the past years the beamforming in modal sound fields, especially in the vehicle interior, has reached a new level.

The scope in which the beamforming inside a room is performed must be first differentiated into two frequency ranges. These frequency ranges are separated by the so-called Schroeder frequency, which in room acoustics represents the transition from a modal sound field (below) to a statistical sound field (above). Studies have shown that modern beamforming algorithms as the MUSIC algorithm, the Functional beamforming or variants of Robust Adaptive Beamforming (RAB) methods provide a reliable detection inside a room from around the Schroeder frequency and above (statistical sound field), particularly in a vehicle interior [1]. The frequency range below the Schroeder frequency is consequently dominated by modal sound field effects (modes). Despite the use of modern beamforming algorithms, limitations of the localization precision below the Schroeder frequency are to be expected.

In this context, essentially there are two approaches that can improve the localization results among the modal sound field conditions of a vehicle interior. On the one hand it is the combination of generalized cross-correlation techniques (GCC) with modern beamforming algorithms, which are able to improve the localization precision under these conditions significantly [2]. On the other hand there is a new approach, the automated mode detection (AMD), which will take into account the modal sound field influences embedded in the beamforming process. Implemented as an upstream mode filter, it is thereby possible to obtain significantly better localization results below the Schroeder frequency in subsequent beamforming.

2 CONVENTIONAL BEAMFORMING

The conventional beamforming (CBF) is a valid method for the spatial localization of acoustic signals in free field. The output of the CBF can be expressed in the frequency domain, its performance can be formulated as:

$$b(\mathbf{g}) = \mathbf{g}' \mathbf{C} \mathbf{g} \quad (1)$$

where \mathbf{g} is the array steering vector (\mathbf{g}' conjugate-complex) and \mathbf{C} the cross spectral matrix (CSM). Due to the formulation of the CBF it is readily apparent that any components which are correlated to the sound source contained in the CSM are considered equally [3]. Adaptive beamforming algorithms show significant advantages over the CBF in terms of dynamics and resolution. Furthermore, studies show that especially some advanced beamforming algorithms are able to provide accurate localization results under the influence of a reactive sound field [1].

In order to enhance the performance of these algorithms and use them as a valid tool for the detection of acoustic signals in reactive sound fields, an additionally modification of the applied signal processing is advantageous. This is realized through the combination of advanced beamforming algorithms with GCC used as a preprocessing step for CSM.

From the structure-borne sound acoustics and speech processing methods are known, this techniques may detect the presence of a radiating source and estimate the signal travel time difference at physically separated sensors, when energy of this source is received at the sensors. A well-known variant of the weighting factor formulation of the GCC, called Smoothed Coherence Transform (SCOT), can be defined as:

$$\psi_{SCOT}(f) = \frac{1}{\sqrt{\varphi_{x_1x_1}(f) \varphi_{x_2x_2}(f)}}, \quad (2)$$

with $\varphi_{x_1x_1}(f)$ and $\varphi_{x_2x_2}(f)$ being the auto power density spectra of the two input signals x_1 and x_2 [4]. Applied to the CSM this definition is to be modified to:

$$\mathbf{C}_{nk\ SCOT} = \frac{\mathbf{C}_{nk}}{\sqrt{(\mathbf{A}_k \cdot \mathbf{A}_n)}}, \quad (3)$$

with the cross correlation matrix \mathbf{C}_{nk} related to the CSM and the row vector \mathbf{A} with the diagonal entries of \mathbf{C} (the auto-correlation) [2]. Originally developed for two sensors the transformation applied to the cross spectral matrix acts as a pre-whitening or respectively correlation filter [5]. By an appropriate weighting of the matrix entries with regard to their correlation to the desired signal, this technique suppresses the correlated components and can therefore increase the robustness of the beamforming towards modal sound field influences significantly.

3 BEAMFORMING IN A MODAL SOUND FIELD USING THE GCC

These signal processing techniques are an essential part of the improvement potential of the beamforming in modal sound fields. A general investigation of the potential of this technique, which is later to be expanded to the case of a vehicle interior, is therefore initially shown for the application of a scale model room (SMR).

3.1 Inside a scale model room

The sound field of an enclosed space can be characterized by its reverberation time (see Figure 1) [6]. Below the Schroeder frequency (f_s) modal influences dominate the sound field, thus ensuring a strong, dynamic sound pressure distribution [6].

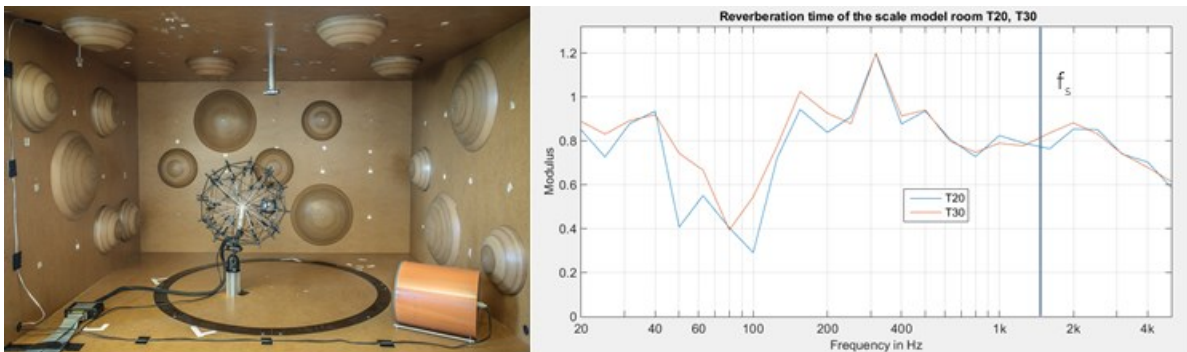


Figure 1: Experiment set-up inside the SMR (left), reverberation time of the SMR (right)

The conventional beamforming “Delay and Sum” is particularly disturbed by the modal influences (modes) inside the room below the Schroeder frequency. Because the transfer functions of the room boundaries to the respective microphone positions assume complex shapes the free field condition, as one basis assumption of the beamforming, is affected [6]. Figure 1 shows a scale modal room with a volume of 1.71 m^3 ($1.2 \text{ m} \times 0.95 \text{ m} \times 1.5 \text{ m}$), in which the subsequent investigation is performed.

Via the cylindrical loudspeaker shown in the front a sweep signal between 20 Hz and 20 kHz over a period of 23.78 s is radiated. This signal impinges at the channels of the microphone array. For the analyzed room, a Schroeder frequency of about 1400 Hz can be estimated from the measured reverberation time of 0.892 s (see Fig 1). Under these conditions the performance with respect to the localization accuracy of advanced beamforming algorithms in combination with the GCC compared to the CBF is investigated. Figure 2 shows the results of this comparison.

The sound source (loudspeaker) is located at the marked position. Here the three-dimensional mapping (relative sound pressure distribution) of the CBF and the mapping of the MUSIC algorithm in combination with the SCOT method between 20 Hz and 500 Hz and a dynamic range of 3 dB is exemplary compared.

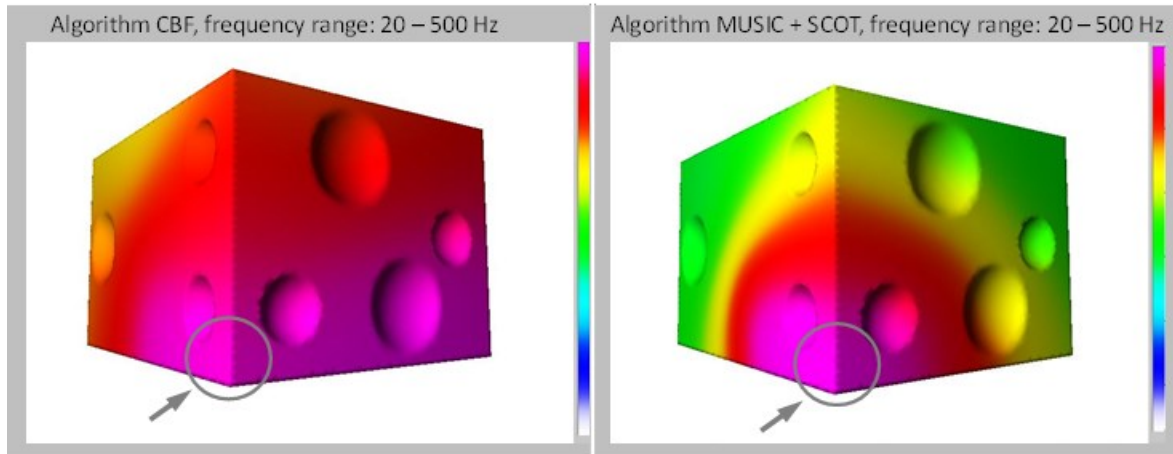


Figure 2: 3D-beamforming map (20 – 500 Hz, dynamics 3 dB), Localization result of the CBF (left), Localization result of the MUSIC + SCOT (right)

Advanced beamforming algorithms like the MUSIC algorithm combined with SCOT show far better results in terms of source localization precision compared to CBF when applied in a reactive modal sound field. In the case of the Conventional beamforming (left), it is no longer possible to identify the source position inside the room, whereas in the right image, the source is clearly located in one place, namely the loudspeaker in the corner. This example illustrates the effectiveness of this technique in conjunction with the beamforming and further leads to the conclusion that this could also result in an improvement of the localization quality inside a vehicle interior.

3.2 Inside a vehicle interior

Initially, the spatial conditions of the passenger compartment are to be compared with those of the scale model room. This comparison is conducted based on the frequency dependent reverberation times of these rooms. Figure 3 illustrates the reverberation times of the two rooms.

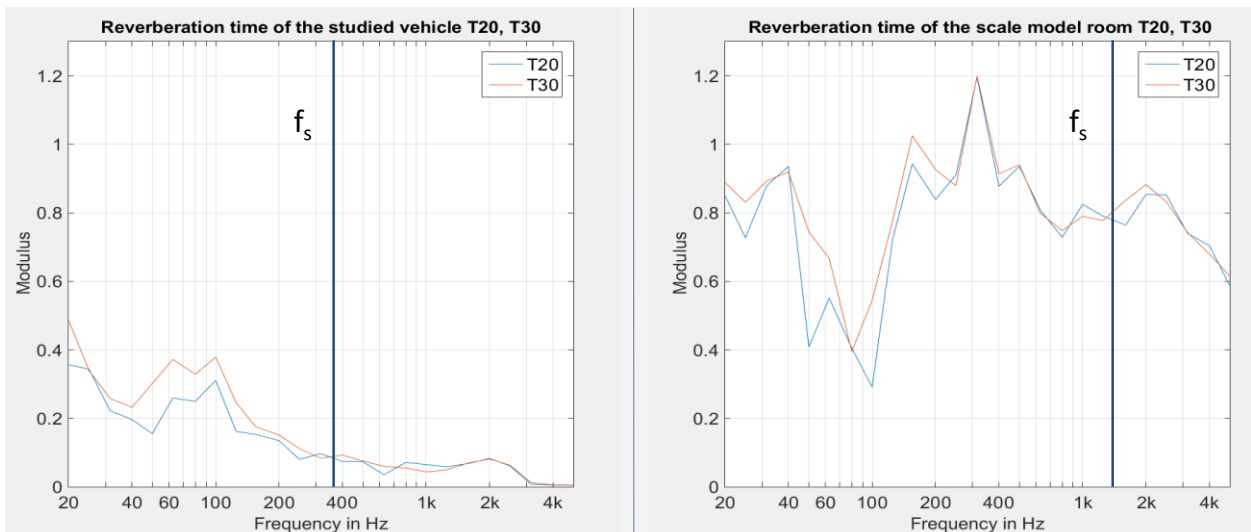


Figure 3: Comparison of the reverberation time of the studied vehicle (left) and the SMR (right)

The much shorter reverberation time of the considered passenger compartment of approximately 0.111 s (factor 8 less than inside the model room) illustrates the relatively very rapid energy loss, which is caused by properties of the surface materials. With a volume of 3,210 m³ (and the measured reverberation time of 0.111 s) the Schroeder frequency of vehicle compartment can be determined to about 372 Hz, which is about a factor of 4 lower than that of the scale model room. In addition, the estimated number of modes of the vehicle interior can be estimated theoretically from [6] to 15 modes compared to the scale model room with theoretically up to 588 modes. Recognizing the conditions under which the beamforming has been performed inside the scale model room and the conditions for the application of the beamforming inside the vehicle interior, it is readily appreciated that the beamforming inside the examined vehicle interior performs better or at least comparable.

Nevertheless it is to be expected that inside the passenger compartment a distinctive modal sound field within below the Schroeder frequency is formed, which affects the beamforming result. The comparison (Fig.4) illustrates the influence of the GCC on advanced beamforming algorithms, carried out by the example of the functional beamforming ($\nu = 300$). From the loudspeaker which is integrated at the marked position in the vehicle, again a sweep signal from 20- 20 kHz and a duration of 23.78 s is radiated. The beamforming maps show an analyzed frequency range of 20- 500 Hz.

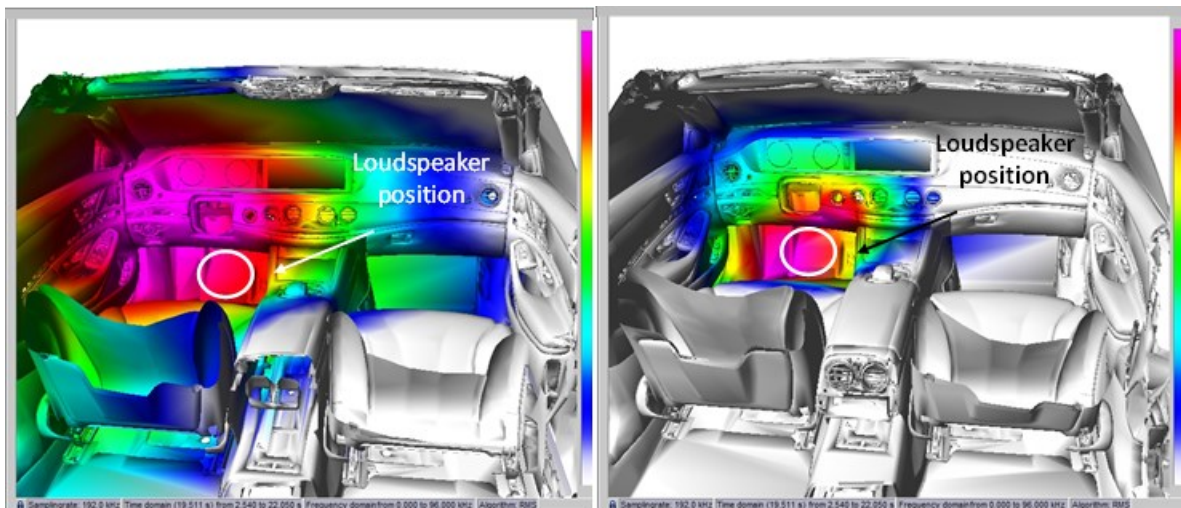


Figure 4: Comparison of the beamforming results (20 – 500 Hz, dynamics 7 dB) using the functional beamforming inside the vehicle without GCC (left) and with GCC (right)

It can be seen that with the GCC as well the location of the radiation is localized more precisely, as also the signal components correlated to the windscreen, the door and the A-pillar can be significantly reduced by using this technique. The previously mentioned room acoustical conditions on the one hand lead to a lower Schroeder frequency. On the other hand, the more convenient acoustic conditions also lead to the fact that a higher dynamics (7 dB) is to be achieved through the beamforming within the vehicle interior, as within the SMR (3 dB).

4 MODE DETECTION

The procedure of the (automatic) mode detection and its consideration in the beamforming process is intended to improve the results of the source location in a vehicle interior as a physical approach. Unlike the signal theoretical approach of the GCC, the approach of mode detection demands knowledge of the wave propagation inside the investigated cavity. Recognizing this characteristic room property of the respective mode distribution this approach is ideally independent of the excitation signal and location of excitation.

To excite the modal sound field of the vehicle interior it is sonicated with the same slow sweep signal as in the SMR. In order to investigate the robustness of the approach regarding the number and positions of the array microphones in the following, three different measurement setups are configured:

- One array (48 channels) above the center armrest
- One array (80 channels) in the back
- Both arrays simultaneously

By detection of the sound field at different positions of the microphone arrays (Fig. 5) the sound pressure is sampled at a up to 128 positions for the duration of the signal and an additional, sufficient decay time.



Figure 5: Positioning of the microphone arrays inside the vehicle interior

To obtain information about the mode field of the vehicle interior from the array data, a parametric model of the measured transfer functions is formed. In this case, a so-called "all-pole" model [7], which transfer function has the form:

$$H(s) = \frac{b_0}{a_0 + a_1s + a_2s^2 + a_ns^n}, \quad (4)$$

with a_i , b_j being the coefficients of the differential equation. Based on this model once a vector of frequency bins is present, that approximation is determined which, at the same time having the least deviation over all frequency bins, so over all microphone channels. Therefore, an identical, common, rational transfer function for all microphone channels is

obtained. After the poles and therefore also the modes of the studied vehicle interior are known from this rational transfer function, an inverse filter is built from this information, which will not be discussed in more detail at this point. Rather, here be important how precisely the proposed model is to determine the modes of the vehicle cavity and what impact the location and number of microphones has on the mode detection. As a reference for this comparison, the numerically determined modes of the FE-Model of the vehicle are used. Table 1 presents this comparison:

Table 1: Comparison of the mode determination regarding the number and positions of the array microphones

Attributes	Array 48 (front)	Array 80 (back)	Arrays simultaneously
No. of Modes from Reference	29	29	29
Determined Modes	23	23	23
Modes not determined	3	3	3
Untruhly determined Modes	3	3	3
Amount of the averaged deviation of the determined modes[Hz]	3,1	3,1	3,1
Rate of determination [%]	79	79	79

4.1 Beamforming results

The following results show the performance of the automatic mode detection (AMD), according to the described procedure on the same vehicle and under identical conditions as in the previous investigations. Well aware that the localization result could be further improved by the application of advanced Beamforming algorithms and the application of the GCC, the performance of the automatic mode detection is demonstrated on the basis of CBF to show its full capabilities.

Figure 6 shows the results of the comparison of the beamforming without using the AMD (left) and with AMD (right) based on the CBF (3 dB dynamics). The beamforming maps show again an analyzed frequency range of 20- 500 Hz.

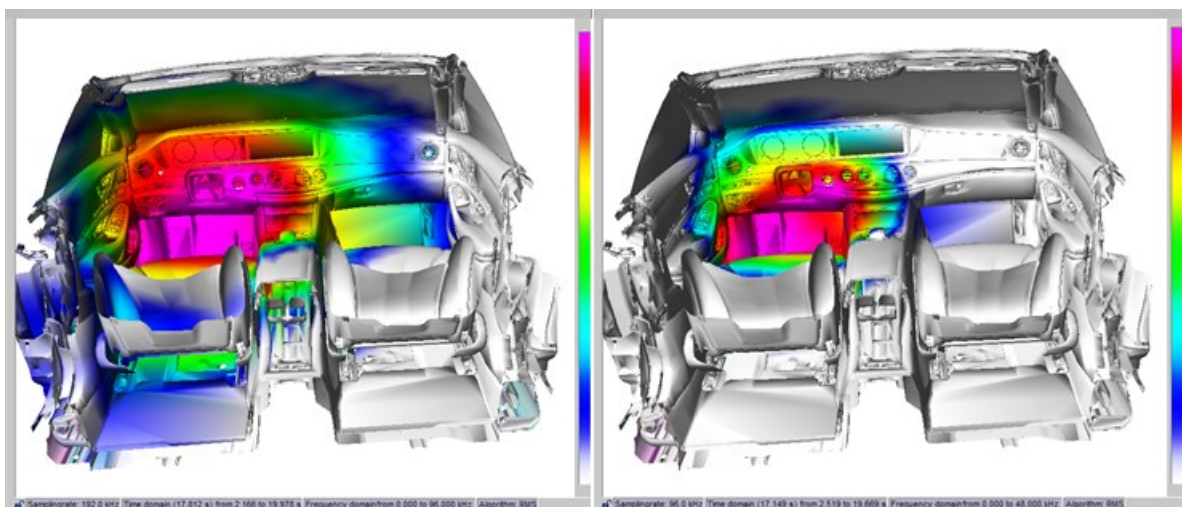


Figure 6: Results of the Beamforming without using the AMD (left) and with AMD (right)

By applying the AMD a significant improvement in localization quality is recognizable. In contrast to the application of GCC, which improvement essentially results in the sharpening of the main lobe and an improvement of the dynamic range. When applying the AMD, as expected there is no sharpening effect to the main lobe. In the case of AMD the result of the beamforming benefits of the desired reduction of modal interfering influences, which can be seen among others by the strong reduction of the reflections on the windshield and the instrument panel.

5 SUMMARY AND FUTURE WORK

The present study addresses two approaches for the improvement of the beamforming regarding its localization ability inside the modal sound field of a vehicle interior. A signal theoretical approach, which is based on the processing of the cross spectral matrix using generalized cross correlations and a physical approach that determines a common rational transfer function of all microphone channels from a parameterized model to create an inverse filter. Both are able to improve the beamforming result inside a room significantly. The GCC-approach is especially advantageous when dealing with highly correlated signals from multiple sources or, as shown here for the suppression of correlated shares of one source. By the integration of the mode detection in the beamforming process, it is possible to locate sources inside a room more precise regardless of the type of beamforming. By the presented approaches it has been shown that it has become possible to repeal two major limitations of the beamforming, namely the failure of a reliable localization regarding correlated signal components and the inadequate localization quality in a range below (depending on the aperture) the Schroeder frequency.

The consideration of mode detection in the beamforming process offers further potential for improvement, for instance, the realization of a proper localization quality while exciting the passenger compartment with several sources, as in this case comb filter-like effects are to be expected.

6 REFERENCES

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