ARRAY ANALYSIS FOR AIRCRAFT FLY-OVER MEASUREMENTS

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ABSTRACT

Microphone Arrays installed on the ground are used to identify and to quantify noise sources at aircrafts. This is an important tool to assess novel treatments, to establish an empirical noise data base for noise prediction tools and to validate acoustic source models used by computational aero acoustics.

An array can comprise several hundred microphones and is widely used for stationary applications in the automotive industry and in wind tunnel facilities. The focus of this presentation is on a specialised array methodology for moving objects, which are far away from the array, e.g. the altitude of an aircraft is about 200 meters for those measurements. The beam-forming map exhibits side lobes, which make the localisation and quantification of sources rather difficult. Therefore an Inverse Source Density Modelling is introduced to remove the side lobes. Care has to be taken to mitigate the frequency shift due to the Doppler effect and to be capable of processing badly conditioned matrices. The success of this approach is discussed based on measurement data.

1 INTRODUCTION

Microphone Arrays are used to identify noise sources at objects of interest, e.g. cars, trains, helicopters and aircrafts. The array processing consists in scanning a grid of focus points on the object. For each focus point the microphone signals are superimposed to give a maximum output for the actual focus points. To achieve this, the microphone signals were delayed, weighted and finally summed. This procedure gives a map with the array output magnitude over the grid of focus points and is called beam-forming map. The name beam-forming denotes the fact, that the spatial array characteristic is utilised with the main beam pointing to the focus point.

In many applications the spatial resolution is insufficient due to the limited number of microphones. Side lobes may give considerable output to other focus points than the actual one. The side lobes and therefore the beam-forming maps as well depend on the array geometry. Thus, for real quantification of noise sources, an additional processing step is required. A modelled transfer matrix, based on free field Greens' functions, is established to derive the beam-forming maps, where a sinusoidal monopole source is placed at the focus points. This source density model (SDM) is applied in the frequency domain for all frequencies of interest.

Special care has to be devoted to the matrix inversion, which has usually a very low condition number. Furthermore when moving objects are considered, the side lobes experience a frequency shift due to the Doppler effect. The coupling of frequencies leads to an increased demand of computational resources and requires the frequency leakage to get accounted for.

EADS Innovation Works, formerly Daimler Benz, has a long history dealing with beam-forming measurements for cars and trains. The SDM method was first applied and validated by the pioneering work of Brühl during the DEUFRAKO project [1] in the mid 90's and published later in 2000 [2]. Brühl achieved a very accurate source power estimation and localisation with the SDM method for moving objects by consideration of the frequency-coupling. At that time the application comprised 24 microphones, 130 focus points, 130 source points and 25 frequency bins, which were analysed by a desktop computer of type Sun Sparcstation 10 with 64 MB memory. The processing time was below 12 hours.

For more than four years, Airbus has worked on aircraft fly-overs measurement techniques, and has integrated EADS algorithm into a dedicated software environment. Simultaneously, EADS is working on the improvement of the method for aircrafts and the author proposes in this paper investigations focused on frequency coupling. Measurement data used are provided by Airbus, gained from tests in December 2008. Now the software has been modified slightly to benefit from a desktop PC with Quad Core CPU and 16 GB RAM. Test cases with 221 microphones, 5751 focus points, 2087 source points and 33 frequency bins were treated with a processing time less than 8 hours.

The results show very good side lobe suppression. Broad band noise and pure tonal sources are found at distinct locations. The assumption of uncorrelated sources has proofed to be feasible.

2 BEAMFORMING

For moving objects the time delay and Doppler effect is corrected in the time domain of the microphone signals p_m for every sample time *n* according to Eq. 1.

$$q_m(\vec{x}_f;n) = r_m(n) M_m(n) p_m(t_0 + n\Delta + r_m(n)/c) \qquad (1)$$

r_m is the distance between the focus point $\vec{x}_f$ and microphone $\vec{x}_m$, whereas M_m denotes the Doppler factor, see Eq. 2. *c* is the speed of sound, $\vec{v}_0$ is the velocity vector of the aircraft and $\vec{e}_m$ is the unit vector pointing from the focus point to microphone *m*.

$$M_m(n) = \left(1 - \vec{e}_m(n) \cdot \vec{v}_0 \,/\, c\right)^2 \qquad (2)$$

Data averaging is an important mean to increase the signal to noise ratio. Spatial averaging is performed summing over all *M* microphones.

$$a(\vec{x}_f;n) = 1/M \sum_{m=1}^{M} q_m\left(\vec{x}_f;n\right) \qquad (3)$$

For temporal averages the signals $a(\vec{x}_f;n)$ are transformed into the frequency domain $a(\vec{x}_f;f)$ and several FFT blocks are power averaged. The over-line in Eq. 4 denotes ensemble average.

$$m(\vec{x}_f;f) = \mathrm{Re}\left\{\overline{a^H(\vec{x}_f,f) \cdot a(\vec{x}_f;f)}\right\} \qquad (4)$$

The signal to noise ratio would be largely improved if the cross spectral matrix is utilised, see Ref. [3][4]. Here however the matrix is build up by the already delayed and Doppler corrected signals $q_m(\vec{x}_f;f)$. Therefore all the elements of the steering vector [3][4] are one giving the following equation.

$$m\left(\vec{x}_f;f\right) = 2\big/M(M-1) \overline{\sum_m \sum_{n>m} \mathrm{Re}\left\{q_m^H\left(\vec{x}_f;f\right) \cdot q_n\left(\vec{x}_f;f\right)\right\}} \qquad (5)$$

Spatial and temporal averaging is combined in a single equation. Furthermore auto spectra are excluded [3]. This is reasonable, because the intrinsic information of the source location is contained in the time delay between different microphones, $n \neq m$. Also note that the symmetry properties of the cross spectral matrix are exploited.

3 SOURCE DENSITY MODELLING

A second analysis step based on the beam-forming projection is introduced to remove the side lobes or the array characteristics from the beam-forming map. Cleaning of the map is the basis for reliable source quantification.

For that purpose a transfer function matrix H between the source points $\vec{x}_s$ and focus points $\vec{x}_f$ is established assuming monopole sources at the source points (Eq. 6). Matrix notation is used in Eq. 6. Please note that the measured frequency f deviates from the source frequency f' due to the Doppler effect. The amplitude at frequency f has to be interpolated onto the neighbouring frequency bins. It follows from Eq. 6 that the source amplitude Q could be calculated from the amplitude of the beam-forming signal a by matrix inversion. But due to the bad matrix conditioning this approach failed.

$$a(\vec{x}_f\,, f) = H\left(\vec{x}_f\,, \vec{x}_s\,; f\,, f'\right)\cdot Q\left(\vec{x}_s\,; f'\right) \tag{6}$$

Brühl [1][2] introduced the constraint of uncorrelated sources which leads to a formulation in terms of auto power spectra of the source amplitudes and beam-forming amplitudes.

$$\left|a(\vec{x}_f\,, f)\right|^2 = \left|H(\vec{x}_f\,, \vec{x}_s\,; f\,, f')\right|^2 \cdot \left|Q(\vec{x}_s\,; f')\right|^2 \tag{7}$$

This equation was iteratively solved for $\Psi = |Q|^2$ under the constrained of positive source power Ψ. 150 iterations were sufficient.

4 RESULTS

The fly-over of a test aircraft was measured with an array comprising 221 microphones on the ground. The data were recorded with a sampling rate of 102.4 kHz. The array lay out was build by five arms, starting all at the array centre. The altitude of the aircraft was about 130 meter above the ground. A time window of 40000 samples with the reference position at 60° was chosen (Fig.1).

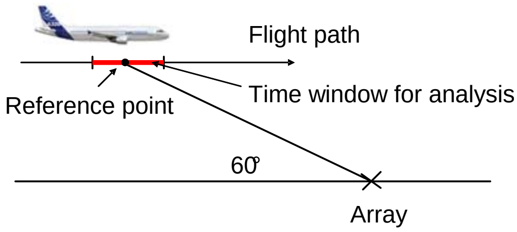
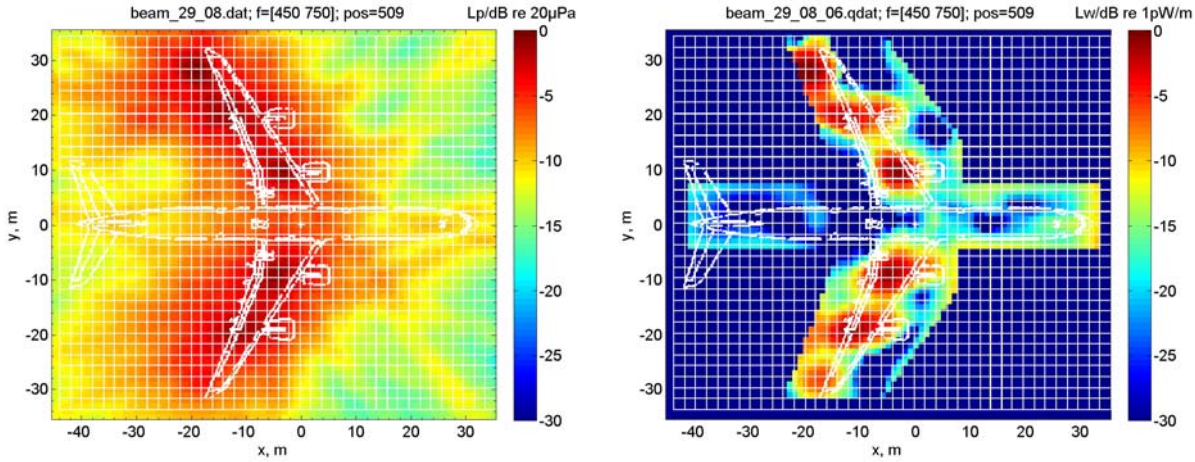


Fig. 1. Selection of analysed time window

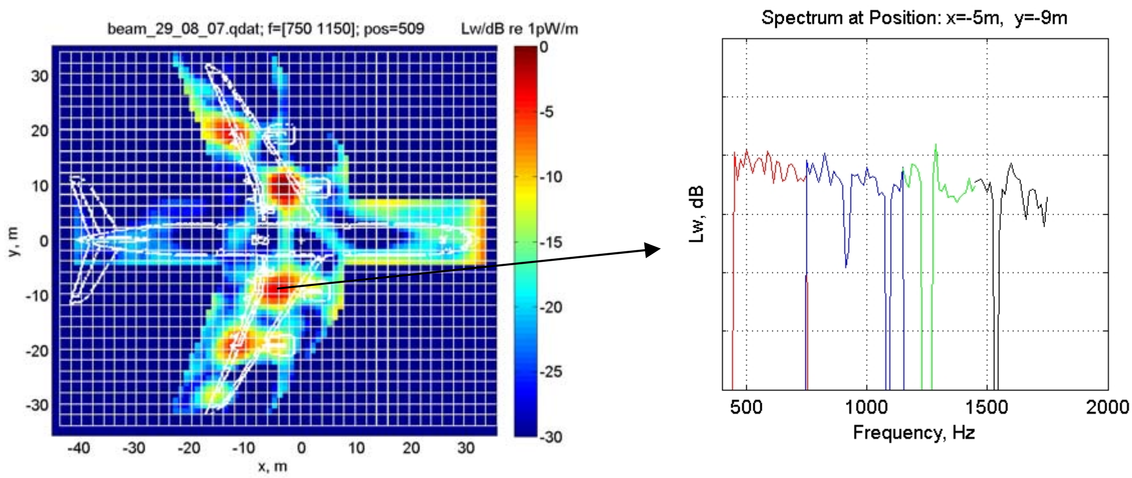
The frequency resolution of the Fourier transformation was 12.5 Hz. Due to the high resolution the Doppler shift exceeded several times the frequency resolution. A Hanning window was used for the Fourier transformation as well as for the frequency interpolation of the SDM method. In order to achieve a reasonable estimate of the broad band spectra, the overlap was set to 87.5 %, resulting in 32 averages.

The SDM calculation was performed for the same focus points as used for the beam-forming, but the source domain was reduced to an area around the aircraft. With a resolution of 1 meter, the number of focus points reached 5751 and the number of source points 2087 respectively. The frequency range of the sources were analysed in four frequency bands individually comprising at most 33 frequency bins for one calculation. Fig. 2 clearly shows the advantage of the SDM method in terms of spatial resolution. Left figure depicts the case with pure beam-forming whereas the right figure shows the result of the SDM analysis applied to the beam-forming map of the left figure. All maps are given with a 30dB dynamic range.



a) beam-forming map *b) SDM map*

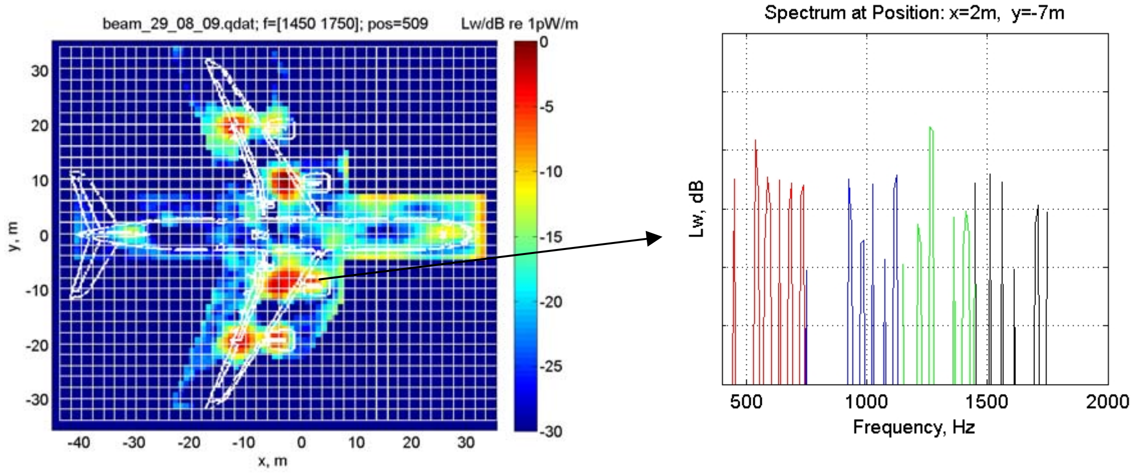
Fig. 2. Array results at 60° in a frequency range of 450..750 Hz



a) SDM map 750..1150 Hz *b) Source spectra at x=-5m, y=-9m*

Fig. 3. SDM result, a) map b) Spectrum at single point with dominating brad band noise

Great progress was achieved in discriminating tonal sources from broad band sources by means of source spectra for individual points on the map. Clear broad band sources are identified downstream of the engines, see Fig. 3. Tonal sources are found just upstream of those points close to the engine intake as shown in Fig. 4.



a) SDM map 1450..1750 Hz *b) Source spectra at x=2m, y=-7m*

Fig. 4. SDM result, a) map b) Spectrum at single point with dominating tonal noise

5 CONCLUSIONS

With the SDM method the quantification of sources seems to be feasible for tonal sources as well as for broad band sources even in the presence of a Doppler frequency shift. Furthermore broad band source areas and tonal sources areas can be distinguished by spectral observations. The assumption of uncorrelated sources has proven to be reasonable.

The current method finds solutions for any frequency bin separately. Time domain approaches are proposed to treat the space and time domain at once.

It is further recommended to take benefit from the various properties of noise sources, e.g. broad band or tonal characteristics, correlation etc..

To the authors mind it seems not feasible to extend the current methodology to correlated sources because of the dramatic increase in complexity. Instead a break down of the complexity into smaller chunks is more attractive, e.g. one single solution for one focus point.

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