Noise and Vibration: *Emerging Methods*
NOVEM 2009
5-8 April 2009 Oxford

SOURCE LOCALIZATION USING ACOUSTIC VECTOR SENSORS: A MUSIC APPROACH

Jelmer W. Wind[1*], Emiel Tijs[2] and Hans-Elias de Bree[3]

[1]Institute of Mechanics Processes and Control Twente
University of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands
Email: j.w.wind@ctw.utwente.nl

[2] Microflown Technologies BV
PO BOX 300, 6900 AH Zevenaar, The Netherlands
Email: tijs@microflown.com

[3] HAN University, Department of Vehicle Acoustics
Ruitenberglaan 26, 6826 CC, Arnhem, The Netherlands
he.debree@han.nl

ABSTRACT

Traditionally, a large array of microphones is used to localize multiple far field sources in acoustics. We present a sound source localization technique that requires far less channels and measurement locations (affecting data channels, setup times and cabling issues). This is achieved by using an acoustic vector sensor (AVS) in air that consists of four collocated sensors: three orthogonally placed acoustic particle velocity sensors and an omnidirectional sound pressure transducer.

Experimental evidence is presented demonstrating that a single 4 channel AVS based approach accurately localizes two uncorrelated sources. The method is extended to multiple AVS, increasing the number of sources that can be identified. Theory and measurement results are presented. Attention is paid to the theoretical possibilities and limitations of this approach, as well as the signal processing techniques based on the MUSIC method.

1 INTRODUCTION

Far field sound source localization is a field with many civil and military applications, both in air and underwater. As long as the distance from the source to the sensor is more than a few times the maximum acoustic wavelength the source is considered to be in the acoustic far field. The approach used to solve the problem depends on the one hand upon the type of acoustic transducer used (capturing sound pressure or particle velocity), their number their spacing and their orientation. On the other hand, the acoustic problem is of relevance, e.g. the number of sources and the type of signal (impulse e.g. explosions; quasi static e.g. jet noise; harmonic, e.g. propeller or combustion engine driven vehicles). Depending on all of these parameters, a variety of signal processing techniques can be applied to solve the problem.

1.1 Localization based on sound pressure sensors

In most cases sound sources are localized using only sound pressure transducers (in both air and underwater). A sound pressure transducer is omni-directional, i.e. it has no directional sensitivity.

Directional sensitivity can be created by a number of spaced sound pressure transducers built in an array. The frequency range is limited due to the upfront selected spacing. As a rule of thumb the spacing of the microphones must in the order of half a wave length of the sound field to be considered. Furthermore, the signal strength of all sensors is equal such that that directional information can only be found in the mutual phase of the signals. The number of sensors affects the signal to noise ratio. Both the intrinsic self-noise and the ability to separate noise sources improves when more sensors are used. The layout of the array determines the directional behavior. A line symmetry is found if the sound pressure sensors are placed on a line; a mirror symmetry is found if the sound pressure sensors are positioned in a plane. A 3D field can be observed only when the sensors are placed in a 3D configuration, e.g. a sphere.

1.2 Literature on vector sensor approaches

Acoustic vector sensors are used only to a limited extent and the number of corresponding research publications is limited. For underwater purposes, tri-axial accelerometers are used to estimate of the acoustic particle velocity, resulting in vector based signal processing techniques to be used for underwater purposes.

In air, pressure gradient microphones are used to some extent. A pressure gradient microphone can be seen as two closely spaced pressure microphones, leading to an inherent limitation in their bandwidth, etc [1].

A wideband sound source localization technique using a distributed acoustic vector sensor array was published in [2] and a wideband algorithm was presented for finding the bearing of a single acoustic source using a single AVS located on the ground. A source's 3-D position can be determined using wideband, closed-form equations that combine bearing estimates from several arbitrary AVS locations. As to the authors' knowledge, the concept was never tested at that time. In [3] this is experimentally proven using measurement data from a helicopter.

Narrow banded algorithms using acoustic vector information were reported in [1]. Their technique was demonstrated to be able to localize 3-5 simultaneous speech sources over 4 seconds with 2-3 microphones to less than one degree of error.

In this paper, a range of broad banded AVS are used to locate a larger number of sound sources. A simulation study shows that the number of sources in 3D that can be separated is $4n-2$, with n the number of AVS. If the number of sources is larger, then there are more unknowns than equations [4]

2 MUSIC

Multiple signal classification (MUSIC) is widely used to determine the direction in which multiple wavefronts are passing an array of sensors. It uses both measurement data and a model of the waves to calculate the *MUSIC spectrum*. This is a function incidence angle, containing a sharp peak at the incidence angles of the incoming wavefronts, so the incidence angles can be read easily from a plot of the MUSIC spectrum. The method can also be used in 3D, where there are two incidence angles.

The method has been developed by Schmidt (see e.g. [5]). This section briefly discusses the background of MUSIC.

2.1 The data model

In the frequency domain, the measurement data of m sensors is a linear combination of the k incident waves and noise. The multiple signal classification approach begins with the following model for the measured signals, which are represented by the m-vector $\mathbf{x}$

$$\begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_m \end{bmatrix} = \begin{bmatrix} & & & \\ a(\theta_1) & a(\theta_2) & \cdots & a(\theta_k) \\ & & & \end{bmatrix} \begin{bmatrix} f_1 \\ f_2 \\ \vdots \\ f_k \end{bmatrix} + \begin{bmatrix} e_1 \\ e_2 \\ \vdots \\ e_k \end{bmatrix} \tag{1}$$

$$\mathbf{x} = \mathbf{A}\mathbf{f} + \mathbf{e} \tag{2}$$

The incident waves are represented by the complex quantities $f_1, f_2, \cdots, f_k$. The noise is represented by the m-vector $\mathbf{e}$.

The elements of the vector $\mathbf{x}$ and the matrix $\mathbf{A}$ are also complex in general. The columns $\mathbf{a}(\theta)$ are known functions of the arrival angles and the aim of the MUSIC method is to identify the correct angles $\theta_1, \theta_2, \ldots, \theta_k$, where each θ_i is a scalar angle in $2D$ problems, but it is a vector containing two angles to pinpoint a source location in $3D$ problems. The function $\mathbf{a}(\theta)$ which is relevant to this article is derived in appendix A.1

2.2 The S-matrix

The $m \times m$ covariance matrix of the $\mathbf{x}$ vector $\mathbf{S}$ is

$$\mathbf{S} \equiv \overline{\mathbf{x}\mathbf{x}^*} = \mathbf{A}\overline{\mathbf{f}\mathbf{f}^*}\mathbf{A}^* + \overline{\mathbf{e}\mathbf{e}}^* \tag{3}$$

where $\bar{\cdot}$ denotes time-averaging and $\cdot^*$ denotes the Hermitian transpose. If the noise is white then

$$\overline{\mathbf{e}\mathbf{e}}^* = \sigma_0^2 \mathbf{I}_m \tag{4}$$

where $\mathbf{I}_m$ is the $m \times m$ identity matrix and σ_0^2 is the variance of the noise. Combining equations 3 and 4 gives

$$\mathbf{S} = \mathbf{A}\mathbf{P}\mathbf{A}^* + \sigma_0^2 \mathbf{I}_m \tag{5}$$

where $\mathbf{P} = \overline{\mathbf{f}\mathbf{f}}^*$ is the $k \times k$ correlation matrix of the incident waves.

2.3 The signal and noise subspaces

When the number of incident waveforms k is less than the number of array elements m then $\mathbf{APA}^*$ is singular: it has a rank less than m. In geometrical language, the measured vector $\mathbf{x}$ can be visualized as a k-dimensional space termed the *signal space*. The space orthogonal to the signal space is termed the *noise space*. Both spaces can be identified from the eigenvalue decomposition of $\mathbf{S}$.

$$\mathbf{S} = \begin{bmatrix} \mathbf{V}_S & \mathbf{V}_N \end{bmatrix} \begin{bmatrix} \mathbf{\Lambda}_S & \\ & \mathbf{\Lambda}_N \end{bmatrix} \begin{bmatrix} \mathbf{V}_S & \mathbf{V}_N \end{bmatrix}^* \tag{6}$$

Here, the $m \times k$ matrix $\mathbf{V}_S$ and the $m \times (m-k)$ matrix $\mathbf{V}_N$ are orthogonal bases for the signal and noise spaces respectively. The noise-eigenvalues in $\mathbf{\Lambda}_N$ are theoretically all equal to σ_0^2 and considerably smaller than the signal-eigenvalues in $\mathbf{\Lambda}_N$.

2.4 Calculating a solution

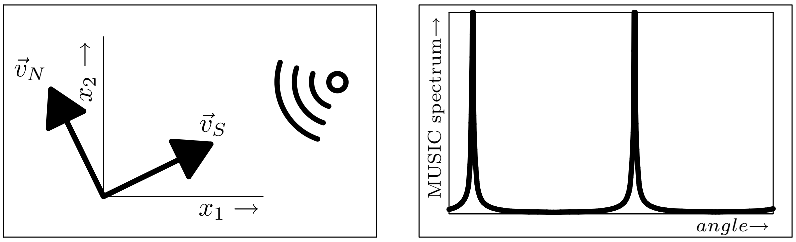
Identifying the noise sources is now equivalent to finding a set of directions $\theta_1, \theta_2, \ldots, \theta_k$ such that the space spanned by $a(\theta_1), a(\theta_2), \ldots, a(\theta_k)$ is equal to the signal space of the measurement data. Equivalently, the space spanned by $a(\theta_1), a(\theta_2), \ldots, a(\theta_k)$ must be orthogonal to the noise space of the measurement data.

The MUSIC algorithm tests if this orthogonality is there for one angle θ at a time. The *music spectrum* is given by

$$p(\theta) = \frac{1}{\left\| \mathbf{V}_N^H \mathbf{a}(\theta) \right\|} \tag{7}$$

It can be seen that this value tends to infinity if $\mathbf{a}(\theta)$ and $\mathbf{V}_N$ are exactly orthogonal and that it has some positive value otherwise. Also note that this definition is the square root of the definition commonly used in the literature.

2.5 An example



(a) example: setup (b) Music spectrum

Figure 1: Source localization using the velocity vector

Figure 1(a) shows an example where the measured signals are the acoustical particle velocity in two directions. Since the source is located in $\vec{v}_S$ direction, the signal space is the entire line along this vector. The noise space lies along $\vec{v}_N$.

The MUSIC spectrum of the example in figure 1(a) is depicted in figure 1(b). The MUSIC method is not necessary to localize a single noise source but it does illustrate a number of important points. First of all, it can be seen that the music spectrum tends to infinity at the correct angle of 26 degrees. At this angle, the modeled sensor response lies exactly in the signal space because no noise has been added in this example. Furthermore, there is a second peak at 206 degrees. The modeled sensor response also lies in the signal space for this angle, even though there is no source present but the peak disappears if a pressure sensor is added to the setup.

2.6 Discussion

Given the general explanation of the MUSIC method in this section, it may be concluded that it is different than other methods such as beamforming or general blind source separation methods in a number of important ways.

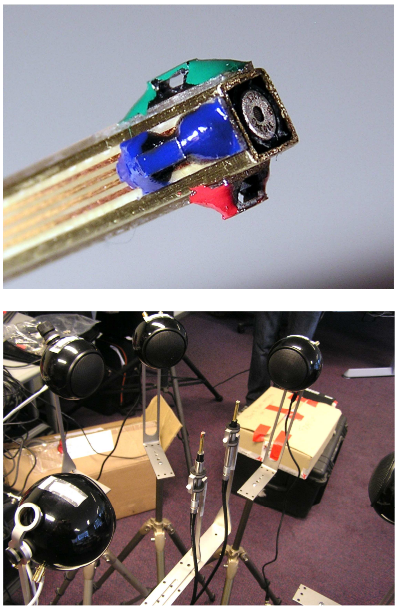
- The sources identified by MUSIC are not the *most uncorrelated* or *most independent*, they are merely vectors in the signal space. Contrary to certain blind source separation methods, the sources can therefore be identified correctly if the sources are correlated. Although some correlation is allowed, the source locations can not be identified for fully coherent sources.
- The Music spectrum should not be interpreted as a spatial distribution of source strengths. Instead, the k largest peaks are unbiassed estimators of the k source locations [5].

3 EXPERIMENTAL VALIDATION

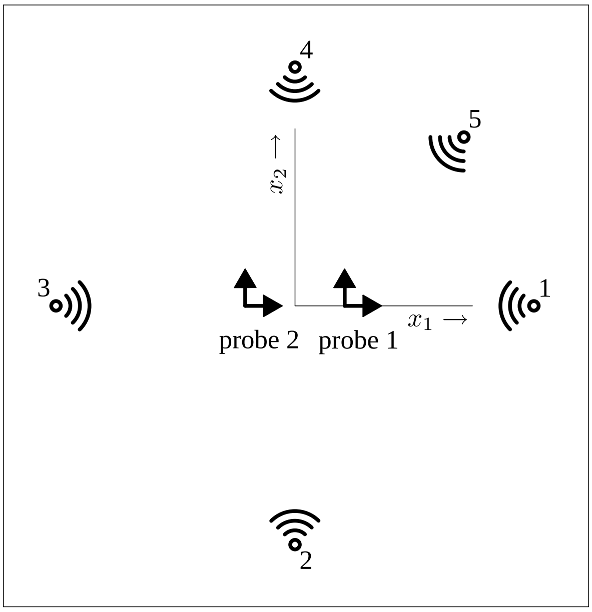
3.1 Experimental setup

The experimental setup consists of two AVS probes of the type Microflown USP, as well as five spherical noise sources arranged as depicted in figure 2(b). All sources are positioned in one plane and arranged in a circle with a radius of 0.24m. The distance between the sensors is 0.1m. The measurement has taken place in an office room with no actions taken to prevent any reflections. Photographs of the measurement setup and the AVS are depicted in figure 2(a).

Since the sources are arranged in a plane, a 2D source localization can be performed. Consequently, the out of plane particle velocity is not used in the MUSIC method. Finally, a frequency of 2151Hz is used throughout for all cases presented in the next section.



(a) Top: AVS head, bottom: setup

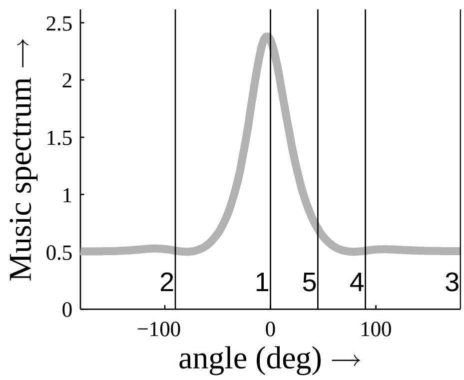


(b) setup (overview)

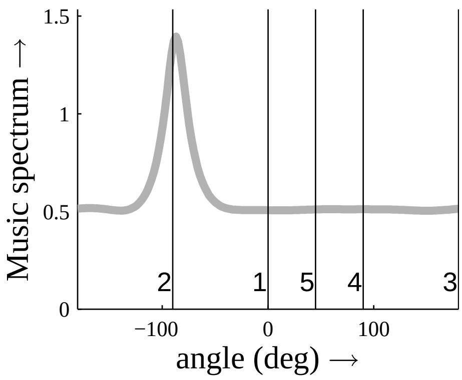
Figure 2: Setup

3.2 Results

First, one sound source is used at a time. This case is used to study the sensitivity of the technique to errors. The AVS probes may for instance not be aligned correctly or the sensor calibration values might deviate. Also, spherical waves have been used in both the experiment and the calibration. For simplicity it is assumed that the distance from the source to the sensor is the same in both cases, but in reality the distances may be slightly different.

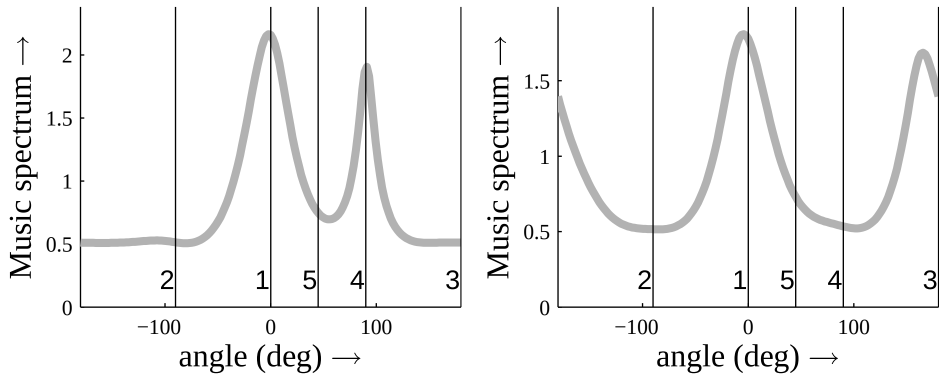


(a) source 1



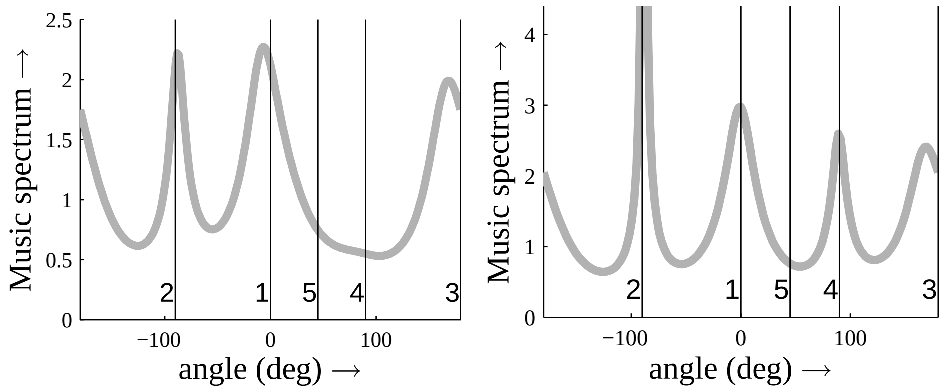
(b) source 2

Figure 3: MUSIC applied to one source



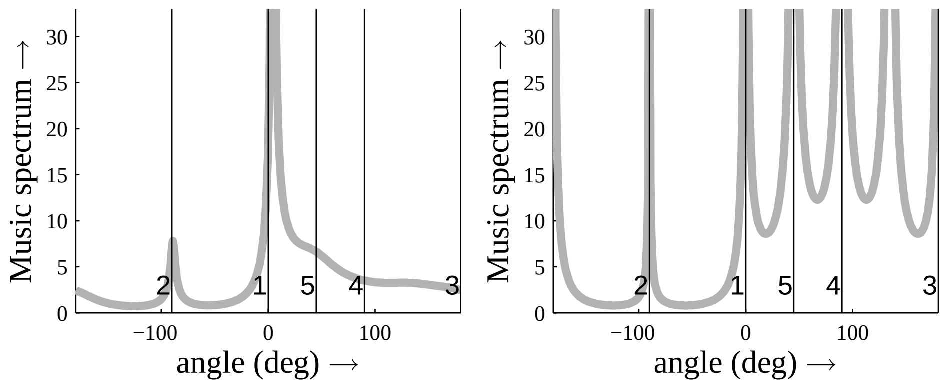
(a) sources 1 and 2 (b) sources 1 and 3

Figure 4: MUSIC applied to two uncorrelated sources



(a) sources 1,2 and 3 (b) sources 1,2,3 and 4

Figure 5: MUSIC applied to three and four uncorrelated sources



(a) sources 1-5 (measurement) (b) sources 1-5 (simulation)

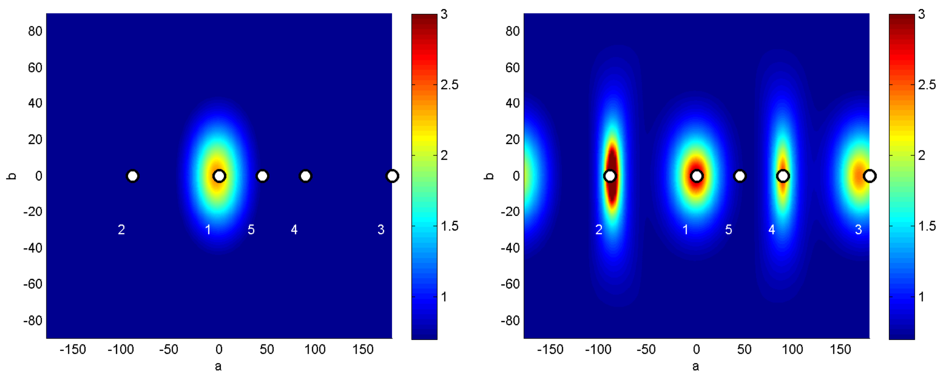
Figure 6: MUSIC applied to five uncorrelated sources

Figures 3(a) and 3(b) show the MUSIC spectrum of sources one and two respectively. The vertical lines indicate the true location of sources 1-5. Both sources are localized within an accuracy of 5°. Since there is only one source, the source location can also be found by inspecting the measurement data directly. The errors are present in the measurement data itself and are caused by slight misalignments of the probes or by acoustical reflections.

The second case consists of using two sound sources with uncorrelated white noise. Figure 4(a) shows the Music spectrum for the case where source 1 and 2 emit sound and figure 4(b) shows the case where sources 1 and 3 emit sound. The peaks are clearly visible. An error of 12 degrees is present in the location of source 3. By inspecting measurement data from source 3 only, it is found that this error is caused by misalignments of the probes, possibly combined with acoustical reflections.

The cases of three and four sources also provide accurate results (see figure 5). Finally, the case of 5 sources is presented in figure 6(a). This the largest number of sources that can theoretically be identified by the current 2D approach using 2 AVS probes. Figure 6(a) indicates that the five sources can not be identified accurately using the current measurement data. Figure 6(b) shows a simulation of the same case. In the simulation, the MUSIC spectrum is much clearer and the sources can be identified accurately. This implies that the source locations identified by Music are sensitive to noise. Judging from the experimental result in figure 6(a), the noise sensitivity is too high for the current measurement data.

This section is ended with the results for the three-dimensional case (see figure 7). It can be seen that both the azimuth and the elevation can be identified quite accurately, although it should be noted that the out-of-plane velocity has not been used.



(a) source 1 (b) sources 1-4

Figure 7: 3D representation of the MUSIC spectrum based on a 2D measurement

4 CONCLUSIONS

This article presents an experimental validation of a noise source localization technique using Acoustic Vector Sensors (AVS) that can be used in air. From literature it is known that a single AVS can find two sources and if the number of AVS is n, then $4n - 2$ sources can be found. The MUSIC method is used to determine the location of several uncorrelated noise sources.

The experimental results indicate that noise sources can be localized with a satisfactory accuracy even if only a straight-forward acoustical model is used. The sources have been positioned in a plane. Up to 4 sources can be identified accurately using 2 Acoustic Vector Sensors, capturing 4 signals each.

The method has proven to be robust to calibration and modeling errors. The main challenge is to control the alignment of the sensor or sensors. Future research will consider methods to calibrate the alignment of each sensor in-situ by placing sources at known locations.

In the future the method will be tested using more AVS probes, more sources in a three-dimensional environment. Since the method is independent of frequency for one AVS, the method will also be tested for the localization of infrasonic booms (f<20Hz). The determination of the source strengths will also be considered.

A APPENDIX

A.1 Plane wave model

This section derives the plane-wave model used in the Music method. A plane acoustical wave in the frequency domain has the following form

$$p = p_0 \exp\left(ik\ \vec{d} \cdot \vec{x}\right) \tag{8}$$

Where $k = \omega/c_0$ are wave number at frequency ω and with the speed of sound c_0. p and p_0 denote the acoustical pressure at point $\vec{x}$, and the wave strength. $\vec{d}$ is a unit vector pointing towards the source. It follows that the sensor output of an AVS probe is

$$\mathbf{s}(\vec{d}) = \begin{Bmatrix} p \\ v_1 \\ v_2 \\ v_3 \end{Bmatrix} = \begin{Bmatrix} 1 \\ -d_1/z_{pl} \\ -d_2/z_{pl} \\ -d_3/z_{pl} \end{Bmatrix} p_0 \exp\left(ik\ \vec{d} \cdot \vec{x}\right) \tag{9}$$

$$z_{pl} = \rho_0 c_0 \tag{10}$$

Where z_{pl} is the impedance of a plane wave. The model may be extended to multiple AVS probes by creating a longer vector containing each of modeled sensors, one below the other.

REFERENCES

[1] Satish Mohan, Michael E. Lockwood, Michael L. Kramer, and Douglas L. Jones. Localization of multiple acoustic sources with small arrays using a coherence test. *Journal of Acoustic Society of America*, pages 2136–2147, 2008.

[2] Malcolm Hawkes and Arye Nehorai. Wideband source localization using a distributed acoustic vector-sensor array. *IEEE Trans. Sig. Proc*, pages 1479–1491, 2003.

[3] Emiel Tijs Tom Basten, Hans-Elias de Bree. Localization and tracking of aircraft with ground based 3d sound probes. *in the proceedings of ERF33, Kazan, Russia*, 2007.

[4] Bertrand Hochwald and Arye Nehorai. Identifiability in array processing models with vector-sensor applications. *IEEE Transactions on Signal Processing*, 44(1):83–95, 1996.

[5] R. Schmidt. Multiple emitter location and signal parameter estimation. *IEEE Transactions on Antennas and Propagation*, 34(3):276–280, 1986.