



ACOUSTIC ARRAY CORRECTIONS FOR COHERENCE LOSS DUE TO THE WIND TUNNEL SHEAR LAYER

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ABSTRACT

Out-of-flow acoustic array measurements in open jet wind tunnels are hampered by the presence of the turbulent shear layer. Signal coherence between pairs of microphones in the array is reduced because sound from a wind tunnel model passes through the shear layer. As a result, the spatial resolution of the beamforming images is significantly lower than without wind, and integrated sound levels are too low. In this paper, a method is discussed to counteract the effects of this shear layer induced coherence loss. This method features the use of one or more miniature loudspeakers, mounted non-intrusively on a wind tunnel model. By instantaneously measuring the unsteady time delay between such a speaker and each of the microphones, corrections can be made to counteract the coherence loss. Such a method for improving phased array results is already applied successfully in the field of radio astronomy, where a well-known bright star is used to define corrections for the turbulence in the atmosphere. This paper presents an implementation of this “Guide Star” method for acoustic arrays. An application to measurements on an A320 model in the open jet of the DNW-LLF is presented. Provided that a number of conditions are fulfilled, the method proves to work well.

1 INTRODUCTION

Out-of-flow acoustic array measurements in open jet wind tunnels are hampered by the presence of the turbulent shear layer [1]. Signal coherence between pairs of microphones in the array is reduced when sound from a wind tunnel model propagates through the shear layer. This coherence loss depends progressively on the turbulence level in the shear layer, on distance between the microphones, and on frequency.

For aeroacoustic array measurements, the consequence is that the microphone cross-spectra have lower levels than they would have without coherence loss. As a result, auto-spectra dominate the cross-spectral matrices. For most wind tunnel applications, auto-spectra must be discarded in the beamforming algorithm in order to get useful acoustic images [2]. On the other hand, when auto-spectra are discarded, essential information is lost about absolute values. Furthermore, even when auto-spectra are discarded, the spatial resolution is much lower than without wind. Dependent on frequency and turbulence levels, there is a maximum array dimension, above which the spatial resolution does not increase anymore.

The coherence loss through the shear layer can be explained by the fact that, due to the turbulence in the layer, the time delay from acoustic source to microphone varies with time, and also with microphone position¹. Both in time and in space, there is limited correlation of the time delay variation. The time-dependency (unsteadiness) of the time delay induces the phenomenon of spectral broadening [3], the additional spatial dependency causes loss of coherence.

In [4], it is proposed to place a known reference source (viz. a loudspeaker) on the wind tunnel model, and to measure instantaneously the unsteady time delay to each of the microphones. Then, corrections can be made to counteract the coherence loss. Afterwards, the effects of the reference source have to be filtered out of the acoustic array results. A similar method for phased arrays is applied successfully in the field of radio astronomy, where a well-known bright star is used to define corrections for the turbulence in the atmosphere. In that discipline, the method is called “Self-calibration”.

When the time delay variations between the reference source and all array microphones are measured, then they are known at the points where the acoustic rays intersect the shear layer. Through interpolation, we can approximate the time delay variation in a continuous (but bounded) region on the shear layer. When the acoustic ray from a point on the scan grid to a microphone in the array intersects the shear layer somewhere in this region, we can use the interpolated time delay variation to make a correction for the microphone signal. This should correct for spatial loss of coherence. The assumption is made that the time delay variation uniquely depends on the point of intersection between acoustic ray and shear layer, in other words, that the dependence on the angle of intersection can be discarded.

This paper presents an implementation of this “Guide Star” method. In Section 2 an outline of the method is given. In Section 3 some practical implementation issues are treated, and in Section 4 experimental results are discussed.

¹ In addition to the time delays, also the spectral amplitudes vary strongly [3], [4], but the amplitude variations are neglected in this paper. Time delay variations are assumed to have the strongest impact.

2 OUTLINE OF THE GUIDE STAR METHOD

2.1 Ideal case

Generally speaking, beamforming expressions to estimate source powers A of scan points $\vec{\xi}$ can be written as the following double summation:

$$A = \sum_{(m,n) \in S} w_m^* C_{mn} w_n. \quad (1)$$

Herein C_{mn} are microphone cross-spectra, N is the number of microphones, w_n are weight factors and S is a set of possible m,n -combinations. The asterisk denotes complex conjugation. For wind tunnel measurements the diagonal is usually removed from the cross-spectral matrix, which means that S consists of all microphone combinations, except those with $m = n$.

The weight factors w_n are chosen such that the array output is unity for a unit point source in $\vec{\xi}$. Suppose that g_n are the (synthesized) acoustic pressures at the microphones, induced by such a unit point source. Then, for the cross-spectrum we have

$$C_{mn} = g_m g_n^*, \quad (2)$$

and, consequently, we should have

$$A = \sum_{(m,n) \in S} w_m^* g_m g_n^* w_n = \sum_{(m,n) \in S} (g_m^* w_m)^* (g_n^* w_n) = 1. \quad (3)$$

For the widely used Conventional Beamforming method, Eq. (3) is fulfilled by choosing

$$w_n = g_n / \left(\sum_{(m,n) \in S} |g_m|^2 |g_n|^2 \right)^{\frac{1}{2}}. \quad (4)$$

For the unit sources and the weight factors we can write:

$$g_n = |g_n| \exp[2\pi i f \Delta t_n], \quad (5)$$

$$w_n = |w_n| \exp[-2\pi i f \Delta t_n], \quad (6)$$

where i is the imaginary unit, f the frequency, and Δt_n the time delay between $\vec{\xi}$ and the microphone location \vec{x}_n .

2.2 Distorted case

Due to the turbulence in the shear layer, Δt_n varies with time and space [3], [4]. We write this as

$$\Delta t_n(t) = \overline{\Delta t_n} + \varepsilon(t, \vec{y}_n), \quad (7)$$

where \vec{y}_n is the shear layer intersection point of the acoustic path from $\vec{\xi}$ to \vec{x}_n . The microphone pressures and cross-powers of a unit point source are therefore given by

$$p_n = |g_n| \exp\left[2\pi i f \left(\overline{\Delta t_n} + \varepsilon(t, \vec{y}_n)\right)\right], \quad (8)$$

$$C_{mn} = |g_m| \cdot |g_n| \exp \left[2\pi i f \left(\overline{\Delta t_n} - \overline{\Delta t_m} + \varepsilon(t, \bar{y}_n) - \varepsilon(t, \bar{y}_m) \right) \right]. \quad (9)$$

In the Conventional Beamforming method, the time delay variations $\varepsilon(t, \bar{y}_n)$ are not included, in other words, we have just

$$w_n = |w_n| \exp \left(-2\pi i f \overline{\Delta t_n} \right). \quad (10)$$

For the source power estimate, Eq. (1), this gives

$$A = \sum_{(m,n) \in \mathcal{S}} |w_m| \cdot |w_n| \cdot |g_m| \cdot |g_n| \exp \left[2\pi i f \left(\varepsilon(t, \bar{y}_n) - \varepsilon(t, \bar{y}_m) \right) \right]. \quad (11)$$

Due to variations of $\varepsilon(t, \bar{y}_n) - \varepsilon(t, \bar{y}_m)$, we will find $|A| < 1$ after averaging over time.

The time-averaged value of Eq. (11) gets closer to zero when the frequency increases, and when the distance between the microphones increases. In other words, in those cases there is an increasing loss of coherence.

2.3 Correction

A possible way to correct for this loss of array performance is to measure the time delay variations $\varepsilon(t, \bar{y}_n)$ by using a loud source (equivalent of bright star) in a reference location $\bar{\xi}_0$. Thus, values for $\varepsilon(t, \bar{y}_{0,n})$ are obtained at locations $\bar{y}_{0,n}$ where the acoustic paths from $\bar{\xi}_0$ to \bar{x}_n intersect the shear layer. If a scan point $\bar{\xi} \neq \bar{\xi}_0$ is considered, then the intersection points \bar{y}_n are not the same as $\bar{y}_{0,n}$. However, assuming that there is enough spatial correlation, $\varepsilon(t, \bar{y}_n)$ can be obtained by interpolation, using the values of $\varepsilon(t, \bar{y}_{0,n})$ in neighbouring points $\bar{y}_{0,n}$.

Using the thus obtained time delay variations, we can update the weight factors w_n :

$$w_n = |w_n| \exp \left[-2\pi i f \left(\overline{\Delta t_n} + \varepsilon(t, \bar{y}_n) \right) \right]. \quad (12)$$

With this, we should regain $A=1$ for the source power estimate, by which the effects of coherence loss are counteracted. This Guide Star method ignores the fact that the time delay variations are dependent on the angle at which the acoustic path crosses the shear layer. Therefore, the best results are expected with scan points $\bar{\xi}$ close to the reference source location $\bar{\xi}_0$.

3 PRACTICAL IMPLEMENTATION ISSUES

3.1 Reference speaker

In the previous chapter, it is explained that adding a reference speaker to a model in an open jet wind tunnel may lead to improved beamforming results. Such a reference speaker should meet the following requirements:

- (a) The speaker must be non-intrusive: it should not modify the aerodynamic and aeroacoustic behavior of the model.
- (b) Its level should be much higher than the other noise levels, otherwise no time delay information can be extracted from the microphone signals.
- (c) Its frequency content should be sufficiently low in relation to the expected time delay variations.

At DNW miniature piezo-loudspeakers are available that can be built in a wind-tunnel model non-intrusively, so that requirement (a) is satisfied. These speakers produce high level tonal sound, so that (b) may also be fulfilled. Because of their small size, the speakers can not produce low frequency tones at high levels. The lower limit is approximately 5000 Hz. An experimental investigation should show if this is low enough for fulfilling requirement (c).

3.2 Measurement accuracy of time delay variations

Time delay variations between a microphone and a speaker can be experimentally determined using a speaker that produces a tone with frequency f_0 . Let $t_j, j = 0, \dots, B-1$ be a sequence of sample times, $p_{n,j}$ sampled microphone pressures, and q_j sampled values of the driving signal. The phase delay variation φ is then evaluated by

$$\varphi = \arg(s_1 s_2^*) - \overline{\arg(s_1 s_2^*)}, \quad (13)$$

where

$$s_1 = \sum_{j=0}^{B-1} p_j \exp[-2\pi i f_0 t_j] \text{ and } s_2 = \sum_{j=0}^{B-1} q_j \exp[-2\pi i f_0 t_j]. \quad (14)$$

The time delay ε can then be calculated by

$$\varepsilon = \varphi / 2\pi f_0. \quad (15)$$

This procedure, however, assumes that ε is constant within the considered time block. This means that the block size B should be chosen sufficiently small, in order to capture the variations of ε . On the other hand, B has to be chosen sufficiently large, in order to eliminate noise in the expression of s_1 in Eq. (14). In other words, B should be chosen carefully.

3.3 Computation time

Compared with Conventional Beamforming, the computation time needed for the Guide Star method increases from typically a few minutes to about a full day. This is because the beamforming process has to be applied on each individual time block, whereas Conventional Beamforming uses the averaged cross-spectra.

4 EXPERIMENTAL RESULTS

4.1 Set-up

During a SILENCE(R) test campaign in December 2005 with an Airbus A320 1:7 scale model in the open test set-up of the DNW-LLF, measurements were performed with four miniature piezo loudspeakers mounted flush on the model. These loudspeakers were located at positions close to expected trailing edge noise sources (see Fig. 1). Three of them were mounted on the left wing, which was above the microphone array. The fourth loudspeaker was mounted on the right wing. The loudspeakers were driven each with a different frequency: 5000 Hz (1), 5500 Hz (2), 6000 Hz (3), and 6300 Hz (4).

Measurements were done at a range of wind tunnel speeds: $M = 0$, $M = 0.083$, $M = 0.120$, $M = 0.148$, $M = 0.177$, and $M = 0.206$. At each tunnel speed, microphone data were acquired

for 60 s, at 122880 Hz sample frequency. During the first half of the acquisition time, the loudspeakers were off. After 30 s, they were switched on.

4.2 Loudspeaker spectra and directivity

The average microphone spectra at several wind speeds are shown in Fig. 2. The spectra are obtained using 20 s of measured data, which are processed with FFT blocks of 1024 samples, using Hanning windows and 50% overlap. This resulted in 4800 averages, and a frequency resolution of 120 Hz. A block size of 1024 seems a reasonable starting point to meet the requirements of Section 3.2. Fig. 2 shows that the Guide Star method outlined in Section 2 will probably be difficult for wind speeds above $M = 0.15$, because the protrusion of the speaker tones is too little.

Another complicating factor is the directivity of the speakers. Two-dimensional SPL-maps of the array microphones, at the frequencies of the speakers, are shown in Fig. 3. These results are without wind; with flow the results are similar, though not equal. Since there appear to be large differences of SPL between different microphones, it can be well possible that the Guide Star method can be applied to only a portion of the array. For each test configuration and flow speed, this portion may be different.

4.3 Time delay variations

By using the method of Section 3.2, time delay variations were calculated between each of the speakers and each of the microphones. In Fig. 4, the time delay variations between speaker 4 (6300 Hz) and the microphones are shown, for a fixed time. In Fig. 5, the time delay variations between the same speaker and 7 centrally located microphones are plotted, but now as a function of time. Figs. 4 and 5 show that the time delay variations have a significant correlation length, both in time and in space. Therefore, there are possibilities for a successful application of the Guide Star method.

Fig. 5 also shows that the time delay varies more than the period of the 6300 Hz tone (1.6×10^{-4} s). This may lead to erroneous results. To calculate the variations more solidly, a lower frequency tone would be preferred, say at 3000 Hz. But that would mean that, approximately, a 10 dB higher speaker level is required to have a good signal to noise ratio (see Fig. 2). With the current speakers, this is impossible.

4.4 Typical result

In Fig. 6 a typical result at 8000 Hz (1/3-octave band) and $M = 0.120$ is shown. The left hand side shows the result of Conventional Beamforming (with the speakers off); the right hand side shows the result of the Guide Star method. The small yellow disk in the maps shows the location of the 6300 Hz speaker.

The Conventional Beamforming map shows a quite large source area around the speaker location. By the Guide Star method it is shown that the most important source location is the flap tip, which is some 10 cm downstream of the speaker. The source area is much smaller now, and the peak level is about 2 dB higher.

At locations further away from the speaker, the source levels decrease by using the Guide Star method. This may be due to the assumption that the time delay does not depend on the intersection angle between the sound ray and the shear layer.

5 CONCLUSION

Time delay variations due to sound propagation through the turbulent shear layer in an open jet wind tunnel induce loss of coherence, and therefore loss of array performance. In order to correct for this, the Guide Star method was implemented. It was shown that this method is able to improve the beamforming results close to the reference speakers. However, quite a number of conditions need to be fulfilled before the method can be applied successfully. This, and also the long processing times, implies that the software is less appropriate for standard application. Nevertheless, it can be an interesting tool for detail studies.

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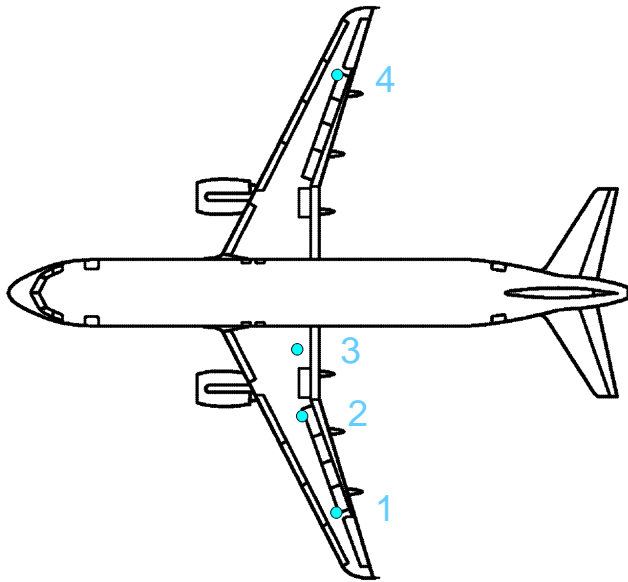


Fig. 1. Locations of piezo-loudspeakers on A320 model

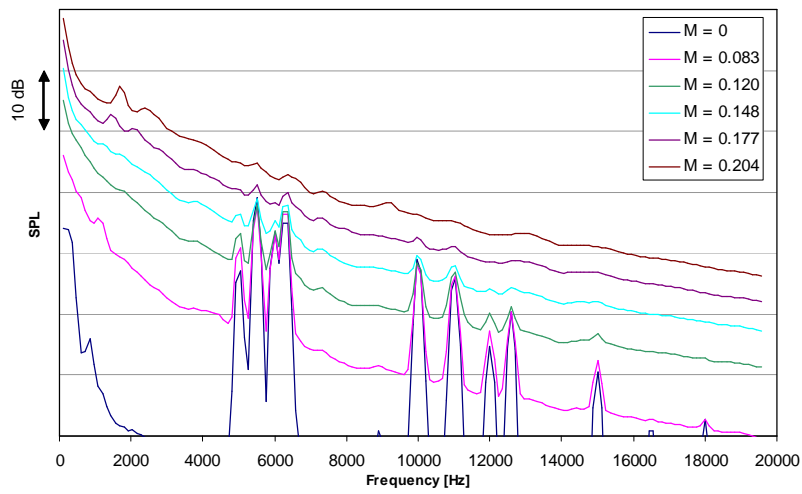


Fig. 2. Average microphone spectra (speakers on)

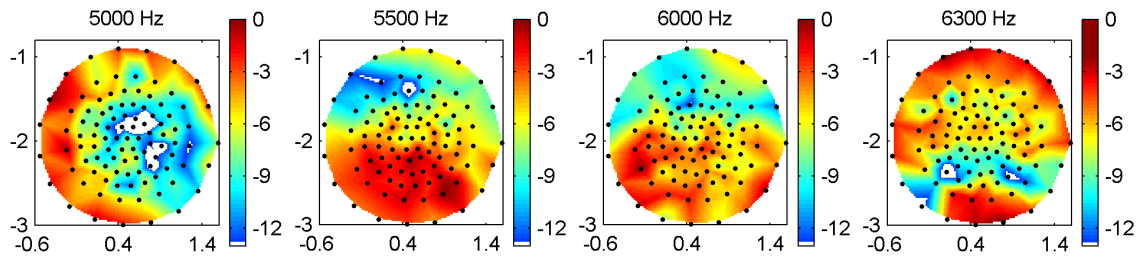


Fig. 3. Directivity of speakers (dB values), no flow

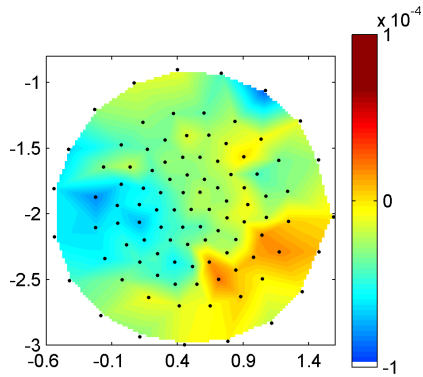


Fig. 4. Time delay variation vs microphone position, speaker 4, $M = 0.120$, time fixed

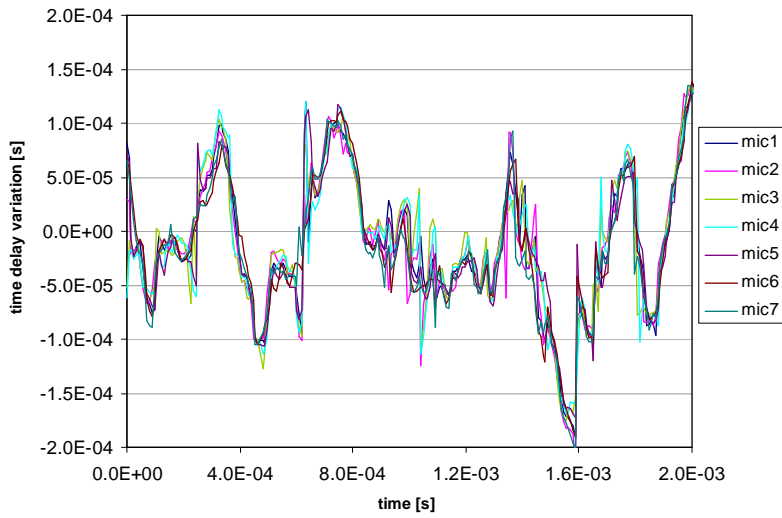


Fig. 5. Time delay variation vs time, speaker 4, $M = 0.120$, central microphones

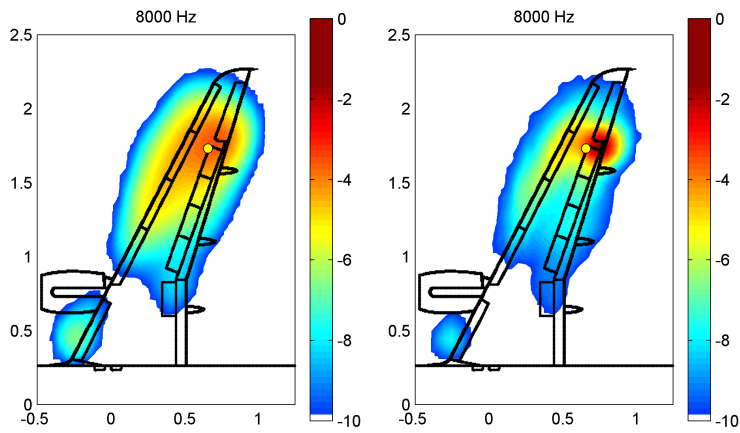


Fig. 6. Typical result at 8000 Hz and $M = 0.120$ wind speed; left hand side: Conventional Beamforming, right hand side: Guide Star method.