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AERO-ACOUSTICAL SOUND SOURCE TRACKING OF A FLOWED CYLINDER WITH A BEAMFORMING CODE

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ABSTRACT

A cylinder placed in an air flow in a wind tunnel with a defined Reynolds Number (Re) can cause a tonal noise in the backlash area of the cylinder. This phenomenon can be recalculated with an appropriate CFD[1] software package. Also, by placing virtual microphones in a simulation of the flow, it is possible to have information about the frequency domain of the pressure signals gained [1].

With suitable signal processing, the location of sound sources in the flow domain is possible and a MATLAB® code was written to solve this particular task. Hence it is possible to process the microphone signals in a way that the beam patterns occur. This beam pattern consists of one main-beam in the middle for one sound source and smaller side-lobes beside it. This is called a directivity pattern and it points directly to the sound source. By this methodology, sound sources in flow simulations can be tracked.

Additionally, it is important to mention that the mesh for the acoustical simulation must be discretised very finely in time and in space. Hence the biggest cell size in the ROI[2] is 0.1 mm. In this case the ROI is chosen at the cylinder wall and in the backlash region of the cylinder, because a sound origin is located in these regions.

In this paper, an insight to the MATLAB® code is given with examples of feasibility of the location of sound sources. Also, in addition to this, the mathematical theory is shown to include the preventive steps required to realise the MATLAB® code.

1. INTRODUCTION

Beamforming is it known, is a signal processing method that allows for the detecting of sound sources. Generally this method is used in acoustic cameras, but it is possible to reprogram this into a mathematical process using MATLAB®. So, disturbing sound sources like machines can be found [2]. Disturbing sound sources can only be quieten down when the sound sources are known. The beamforming signal processing, presented in this paper, works for medium to large distances and for a wide range of frequencies [3]. A weakness of beamforming is the poor resolution at low frequencies (<800Hz). Here another beamforming method called SONAH [4] can be used.

With a defined number of microphones mounted on an acoustic camera (microphone array) or placed in a FLUENT® simulation this can be realised. The incident wave with its amplitude and direction can be delayed in time for each of the microphone signals to correct the time shifts which are produced by the microphone positions. To have information about the frequency a Fourier Transformation can be achieved. In the frequency domain the time delays from the time domain become phase shifts.

[1] CFD – Computational Fluid Dynamics
[2] ROI – Region of interest

2. THEORY

Consider a plane wave from infinite distance in time domain.

$$p_m(\vec{k_0},t)=\hat{p}e^{j(\vec{k_0}\cdot\vec{r_m}-\omega t)} \quad (1)$$

Where $\vec{k_0}$ is the wave number vector, $\hat{p}$ the pressure amplitude and ω the angular frequency calculated with $2\pi/f$. [5]

Also, Equation (1) computes the pressure of each of the M microphones with position $\vec{r_m}$. The inner term $(\vec{k_0}\cdot\vec{r_m}-\omega t)$ includes the individual time delays and go to phase shifts after the Fourier Transformation. The individual pressure signals p_m can be summed up to a so-called beam b.

$$b(\vec{k_0},t)=\sum_{m=1}^{M} w_m p_m(t-\Delta_t(k_0) \quad (2)$$

This beam is the summation of the in time shifted pressure signals constructively. With w_m as optional weighting factors and the time delay from one microphone to the next Δ_t. The time is t. The beam depends upon the incoming wave with ist direction $\vec{k_0}$.

The time delay between the microphones can be expressed as:

$$\Delta_t(\vec{k_0})=\frac{\vec{k_0}\cdot\vec{r_m}}{c} \quad (3)$$

where c declares the speed of sound in the medium (air for this purpose).

The frequency version of Equation (2) is given by:

$$B(\vec{k_0},\omega)=P_0\sum_{m=1}^{M} w_m e^{-j(\vec{k}-\vec{k_0})r_m} \quad (4)$$

This version is achieved by the discrete Fourier Transformation of Equation (2). Furthermore Equation (4) is the beamformer expression scanning all directions to the incident wave. It is the steered beam, identifiable at the term $\vec{k}-\vec{k_0}$ where $\vec{k}$ belongs to the direction where no wave is coming for this point. The term $exp(j(k-k_0)r_m)$ correlates to the multiplication with the phase factors.

Now introducing the so-called array pattern W - it describes the characteristic of the given microphone array. It is also called two-dimensional directional response pattern [3] and has the form as shown in Figure 2.

$$W(K)=\sum_{m=1}^{M} w_m e^{jK\cdot r_m} \quad (5)$$

Now K is the projection of the wave number vectors given as $k=\vec{k}-\vec{k_0}$.

All these Equations can be applied to microphone signals developed from a FLUENT® flow simulation. For this purpose a benchmark problem "Flowed Cylinder" was used to recalculated the beamforming procedure and locates the sound source in a flow simulation.

3. EXPERIMENTS

For the following analyses and results, a FLUENT® flow simulation was fitted. The simulation was done with a two dimensional Large Eddy Simulation and a very fine discretised mesh. The time step for the flow simulation was 5e-6 seconds and the receiver within FLUENT® were placed where the acoustic camera stands in the experiments to compare the results to each other. Figure 1 shows the in principle assembly of the test case.

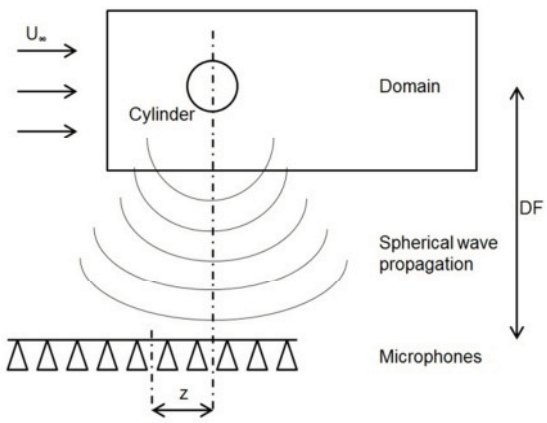

Figure 1: Set up flowed cylinder

The cylinder is flowed from the left with defined velocity U_∞. The microphones are distanced $DF=1m$ from the cylinder and are not in the computational domain. An offset is given by z.

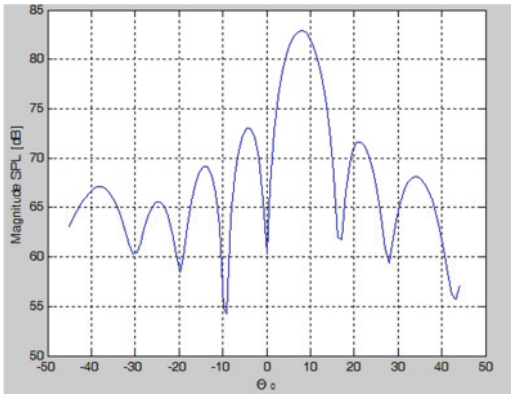

Figure 2: Array pattern for 2kHz

Figure 2 shows the array pattern for a flowed cylinder with microphones arranged as shown in Figure 1. The y-axis shows the sound pressure level (SPL) as a function of the angle θ from the middle of the microphone array to the sound source. The offset is 0.15 m and the distance from the microphones to the middle of the cylinder is 1 m. So the main lobe, visible in the middle, shows to the sound source directly at about 8.5°. Analysing frequency is 2 kHz. Beside the larger main beam in the middle there are so-called side-lobes visible.

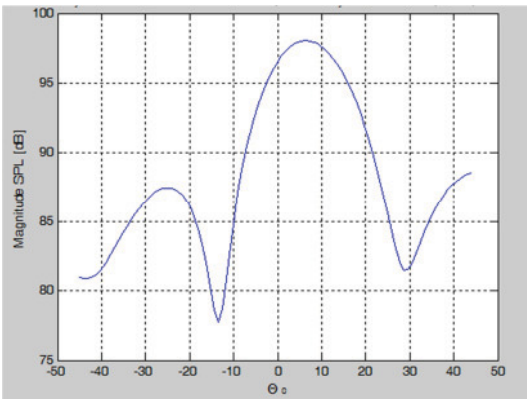

Figure 3: Array pattern for 850Hz

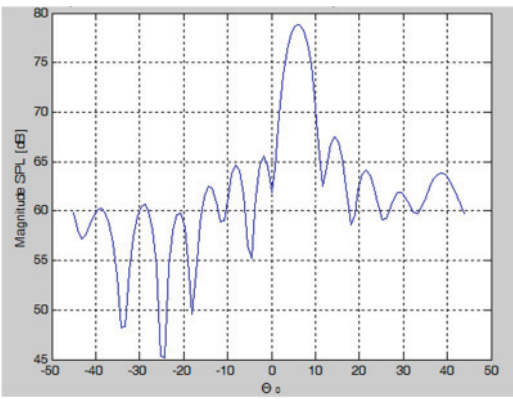

Figure 4: Array pattern for 3kHz

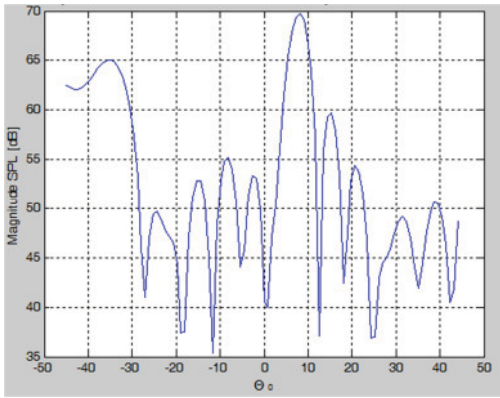

Figure 5: Array pattern for 4kHz

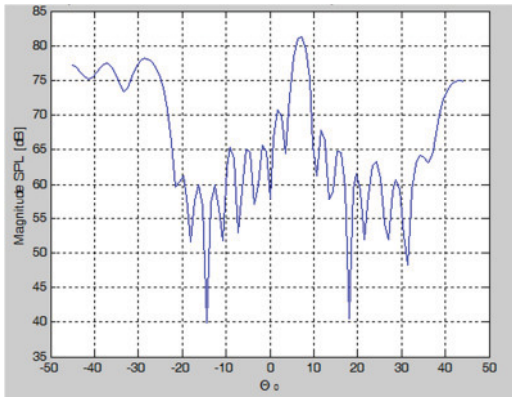

Figure 6: Array pattern for 5kHz

Figure 2 - Figure 6 shows the array pattern for different frequencies. Figure 3 shows the most width of the main lobe and Figure 6 the smallest width of the main lobe. The main lobe is frequency dependent and can be recalculated using Equation (5).

$$\theta_{3dB} = \frac{\lambda}{D} \tag{5}$$

Here the theta 3dB width can be calculatet. The 3dB width is the width 3 dB below the maximum of the main beam. Because the theta 3dB width is frequency dependent the width of the main beam has different widths. λ describes the wavelength calculated with:

$$\lambda = \frac{c}{f} \tag{6}$$

The speed of sound is declared with c and is 340 m/s for the medium air. The frequency is f. Beside this, the array geometry is given by the value D.

An Example for a centred microphone array to the flowed cylinder can also be applied. So the shift from the microphone centre position to the cylinder z is 0mm.

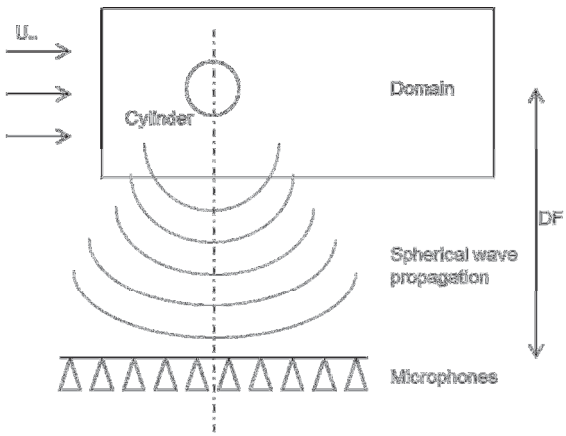

Figure 7: Test Set-Up without centre shift

Now, the main beam, that shows on the sound source directly, is positioned in the middle of the plot at 0°. Figure 8 shows that main beam in the middle of the figure at 0°.

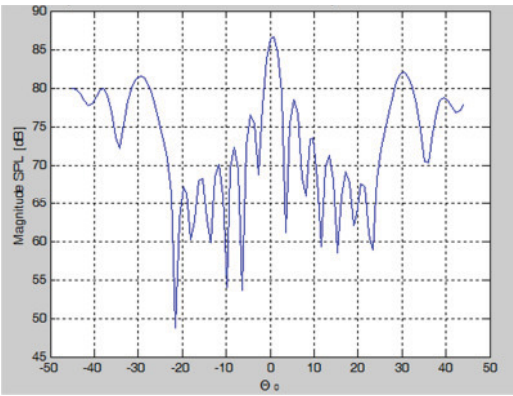

Figure 8: Array pattern 3kHz

Calculating and predicting noise and acoustics using computational fluid dynamic (CFD) software offers a powerful way to compare noise levels to a measurement. So it is possible to make an experiment, e.g. with an acoustic camera or a sound level meter (SLM), to record this sound level and reconstruct this case with FLUENT® and do an acoustic analysis. But it is important to know the limitations of this simulation.

To validate the simulation, measurements with an acoustic camera are done.

Figure 9: Test set-up flowed cylinder [6]

Figure 9 shows the test set-up for the flowed cylinder tests in the wind tunnel. On the right side the microphone array is visible and in the middle of the figure the wind tunnel is behind the cylinder which is visible in front of the black nozzle.

So, what can be expected from this noise prediction? It is possible to get information about receiver files like the acoustic pressure, directivity, spectra and overall sound pressure level (OASPL). There is also information about source strength, contribution from different sources and source classification. [1] FLUENT® is not able to predict absolute noise levels using 2D simulations. 2D results are used only to predict trends. There is an important factor for 2D acoustics calculations, the source correlation length - though in general it is not possible to predict the length of this parameter as this parameter fluctuates from case to case. [6]

4. SIMULATION PARAMETERS

To resolve all eddies for aero-acoustics a very fine mesh is necessary. This demands about 15-20 grid points a wavelength. So the first cell height at the cylinder wall is 1e-5m. At all there are 70 boundary layers at the cylinder wall. Beside this the mesh is more or less divided into four zones with different mesh sizes. The finest mesh zone is at the cylinder wall. Followed by the backlash zone behind the cylinder with a coarser mesh at the end of the backlash. At the outer zone, not at the backlash the mesh is the coarsest.

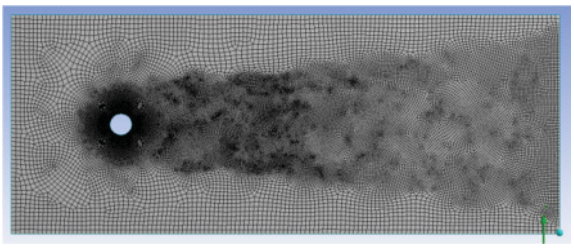

Figure 10: Mesh zones for the flowed cylinder

For Large Eddy Simulations the wall should be resolved accurate to calculate acoustics. So, the Y^+ value should be smaller than five to resolve the viscous sublayer completely. The Y^+ value can be calculated using

$$y^+ = \frac{u \cdot y}{\nu} \tag{7}$$

Where u is the flow velocity, y is the height of the first cell at the wall and ν is the local kinematic viscosity.

5. CONCLUSION

Figure 2 – Figure 6 shows the main lobe at 8.5°. This is for the case of a shift of 0.15 m a correct prediction. So it is possible to detect the cylinder in the flow simulation and proceeding the microphone signals within MATLAB®.

6. OUTLOOK

For future work other high resolution beamforming algorithms should be applied. Aero-acoustic sound sources are often non-correlated sound sources, therefore it possible to separate sound sources using the orthogonal beamforming processing method or locate sound sources more accurately with the DAMAS [7] approach. It should be possible to program and use these algorithms with the acoustic camera as well as MATLAB® program code to process FLUENT® receiver files.

7. REFERENCES

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