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Microphone position localization with acoustic pulse sequences

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Abstract

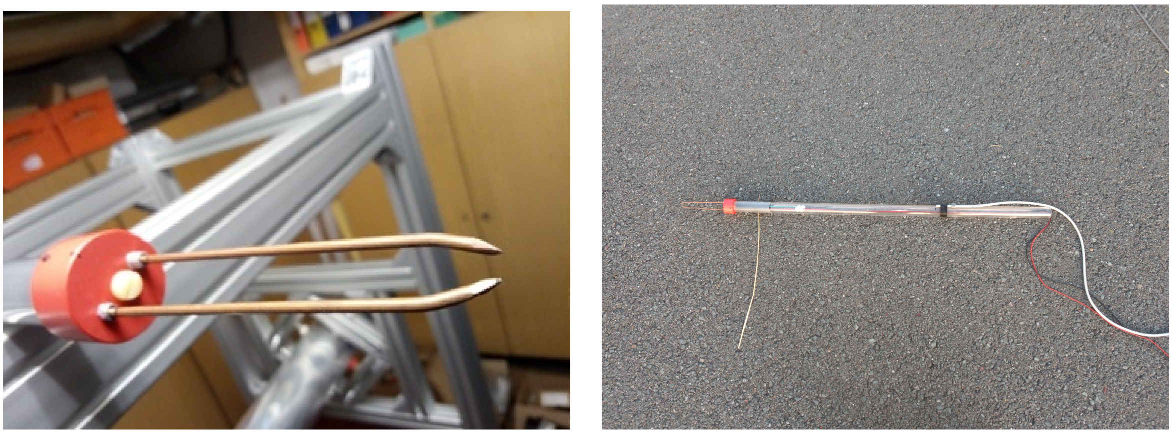
Microphone arrays of complex geometry are used at DLR Engine Acoustics, in order to conduct fly-over experiments with real aircraft or to conduct low-reflection acoustic measurements in laboratories or enclosures, with reverberating walls. The fly-over measurement are performed by adopting a multi-arm spiral array of over 200 microphones, laid across a large (over 20 m) elliptical spot on a surface made of tarmac or concrete. The laboratory measurements are performed by adopting a low reflection barrel, instrumented with an array of 12-microphone rings. The geometric complexity of both microphone distributions requires a time expensive measurement procedure, in order to determine the positions of the sensors in the array. A technique to realize an automated localization of the sensors has been recently implemented and tested. The automated microphone localization follows the work done by Döbler and Heilmann [2] and determines the microphone positions with respect to a sequence of point acoustic pulses distributed on an arbitrary three-dimensional trajectory. An electric-arc generator has been used, in order to introduce the sequence of acoustic pulses in a limited spatial region above the spiral array or in the interior of the barrel. The time-delay distribution between pulse generation and microphone reception can be used to reconstruct both the acoustic-pulse trajectory and the microphone positions, by applying a non-linear optimization technique. In the present paper, we show results for the microphone-position distribution in the laboratory barrel, as determined in an acoustic experiment at the DLR-Berlin inlet parasitic noise test rig.

1 INTRODUCTION

The use of complex microphone arrays for the localization of acoustic sources, implies an accurate measurement of the array geometry. This geometric measurement outputs the microphone coordinates in a given reference frame. It can be quite time consuming, if the number of sensors is large, or if the sensors are distributed in a complex geometry. When working with 250-sensor arrays laid following a multi-arm spiral on the tarmac of a runway or with 60-sensor arrays supported by a flexible cylindrical structure, see Figure 1, one realizes that a remarkable amount of time is required to carry on this geometric assessment of the acoustic array. An automated technique which determines the effective position of the microphones, once the sensors are laid on their measurement positions not only would save the timely direct measurement, but would also be useful to confirm that no sensors have been accidentally moved or remain misplaced after the amplitude calibration. When operating with microphone supports whose geometry has a certain degree of uncertainty, it is of paramount importance to have at hand such an automated geometrical calibration technique, since the array-geometry assessment must be repeated more times during a measurement campaign.

2 METHOD

A technique following Döbler [2] has been implemented at DLR Berlin in order to automatically determine microphone-array coordinates with respect to an arbitrary three-dimensional sequence of acoustic pulses. The acoustic pulses are generated by electric arcs. A high voltage electric arc generator has been realized by incorporating a commercial TASER circuit within a lance supporting two copper electrodes whose pointy ends are kept at a fixed distance, see Figure 1 (a). When attached to a given DC electricity source, the TASER circuit will charge



(a) (b)

Figure 1: Head of the acoustic pulse generator (a) in the free jet laboratory in DLR Engine Acoustics. Acoustic pulse generator with reference microphone on the tarmac of the runway in Cochstedt Airport, (b).

a high voltage electric field between the two electrodes. The voltage between the electrodes grows in time until the dielectric rupture limit of the air is reached and an electric arc between the electrodes discharges the high voltage field. This process repeats in time by following a time sequence which depends on the taser circuit parameters, the characteristics of the air path between the electrodes and the voltage of the input DC current. The electric arc stretches between the electrode tips, causing a sudden variation of the thermodynamic variables in the small volume of air around the electrode tips. The sudden injection of thermal energy associated with the electric arc generates a sudden expansion in the small volume of air, which causes the departure of an acoustic pulse centered in this region. By moving the electric arc generator in the space of interest for our acoustic measurement, one generates a three dimensional trajectory, on which the sequence of acoustic pulses is emitted. This trajectory is different for each measurement while the microphone positions in the acoustic array are fixed. The microphone localization technique makes use of the time-delay distribution between each acoustic pulse emission and the microphones. By assuming that each pulse propagates with spherical wave fronts at a constant speed of sound c, one can construct a non linear optimization algorithm where both the source positions and the microphone positions are the optimization variables and the cost function represents deviations of the time delay distribution from the spherical wave front assumption. A measurement can be set up, where for each emission of an acoustic pulse, the reception times at all different microphones are recorded. Assuming that our microphone array contains m microphones, a measured vector containing the m time delays between acoustic-pulse emission and receptions is evaluated. If the measurement contains n acoustic pulses, the measurement result is a $m \times n$ matrix containing the time-delay vectors as columns. If the matrix is scaled by the speed of sound c and under the hypotheses of constant speed of sound plus absence of wave convection, one obtains a measure of the Euclidean distances between each acoustic-pulse origin and the microphones. We call this matrix R. The determination of R is the objective of a geometrical calibration experiment, which one can run parallel to a microphone array measurement. The m positions of the array microphones x and the n positions of the acoustic-pulse origin x in Cartesian coordinates can be used to analytically express the measured $(R)_X$.

$$R_{ij} = \left(\left(x_{1i} - s_{1j}\right)^2 + \left(x_{2i} - s_{2j}\right)^2 + \left(x_{3i} - s_{3j}\right)^2 \right)^{0.5} \qquad i = 1,m \quad j = 1,n \tag{1}$$

In Eq. 1 the Cartesian components of the microphone positions x_i and of the acoustic-pulse origins s_j are unknown. We can set an initial guess $\left(x_i, s_j\right)_0$ for these unknown variables and evaluate a guess matrix $\left(R_{ij}\right)_0$. By following an iterative procedure, one can minimize the residuals between the guess matrices R and the target matrix $(R)_X$ by moving in the space of variables $\left(x_i, s_j\right)$. In the present work, we adopted a steepest descent method, based on the local gradient of the residuals. The amplitude of the step has been kept constant for reducing amplitude of the residuals. If a growth of the residual amplitude is observed, the algorithm performs a step backwards and reduces the step by a factor 2. The use of the steepest descent method has been suggested by the fact that our cost function

$$|R - (R)_X| \tag{2}$$

allows for an analytical expression of the gradient. Inspired by the work of Döbler and co-authors [2], we operate the residual reduction in two-step iterations, where for each iteration at first the source positions s_j are varied and the microphone positions are kept constant, before a second step where the microphone positions x_i are varied and the s_j are kept constant. The exit condition of the algorithm is the crossing of a maximum threshold value for the cost function 2.

3 EXPERIMENTS

3.1 Low reflection barrel at DLR Engine Acoustics

A measurement campaign was conducted at the free-jet laboratory in the year 2017, including the Engine Acoustic rig for parasitic inlet noise. The measurement regarded aeroacoustic sources at a model-scale turbofan inlet and was performed by using the low-reflection barrel, Figure 2, equipped with a 60-microphone cylindrical array. The array is made of five 12-microphone rings aligned along the ring axes. Each ring structure is made of 12 panels of acoustic foam supported by a back perforate plate and light-weight aluminum profiles. One ring can be easily moved by two persons. The light-weight structure has the drawback of a relatively high deformation and high misplacement risk. A geometrical calibration of the array by using the present technique is necessary, in order to reduce the barrel positioning time and the measurement of each microphone position. A 64-channel acquisition system was used to record the data. The microphone array occupied the first 60 channels, while the last 4 channels have been used for 3 laboratory-coordinate microphones and a reference microphone connected to the acoustic-pulse generator, like in Figure 1 (b) for the more recent measurement. The derivation of the time delay matrix between pulse sequence and microphones in the array has been derived by Cultrera di Montesano [1] and input to the present algorithm. The results of the geometrical calibration are shown in Figure 3. Here the diagram in (a) reports the acoustic-pulse sequence and the laboratory reference system. In the diagrams (b), (c) and (d), the output array of microphone positions is represented in three different views. Unfortunately, during the present measurement no absolute coordinates of the microphones within the laboratory reference system could be taken. Two relative distances within the array (two consecutive microphones in ring 1 and two microphones at the same azimuth in ring 1 and ring 5) showed that, once the speed of sound is defined to set one of the distances,[1] the other distance is within a millimeter from the measured one. In Ref. [1], a parallel implementation of the non-linear optimization algorithm of the present work has been performed. The results obtained with the algorithm in Ref. [1] are in complete agreement with the ones in Figure 3, once similar residual thresholds are chosen and the same matrix $(R)_X$ is used. The coordinate distributions reported in Figure 3 showed no sensitivity to the initial guess for the microphones and sources, which is different for each iteration of the procedure and based on a random distribution of the sources and the microphones. The initial guess only marginally affected the iteration number of the procedure.

3.2 Multi spiral array for fly-over measurements

The fly-over measurement campaign in July 2019, also reported in Ref. [3], featured a large 238-sensor multi-spiral microphone array placed on the head of the runway at Madgeburg Cochstedt Airport, in Germany. The sensor positioning involved the use of 3d-printed microphone holders glued to the tarmac. Measuring the positions of the microphone holders involved the use of an optical system and the work of 2 operators over about 5 hours, this position measurement need to be done only once, after the microphone holders are placed in their positions. The presence of the microphone array and associated acquisition system allowed for geometrical-

[1] In our procedure the speed of sound works as a scaling factor for the microphone and source coordinates. Either an accurate measure of the speed of sound is required in the lab, or an accurate measurement of the distance between two sensors are required.

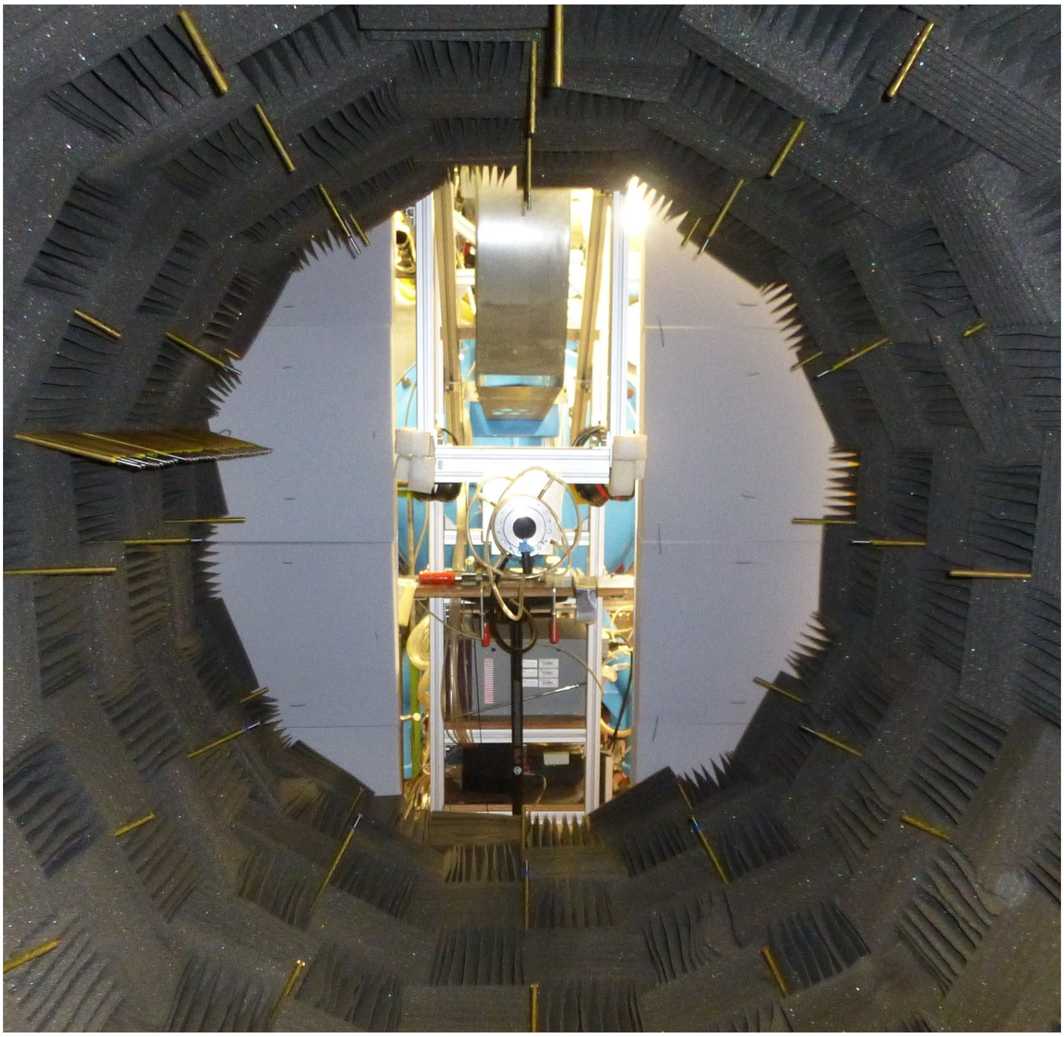
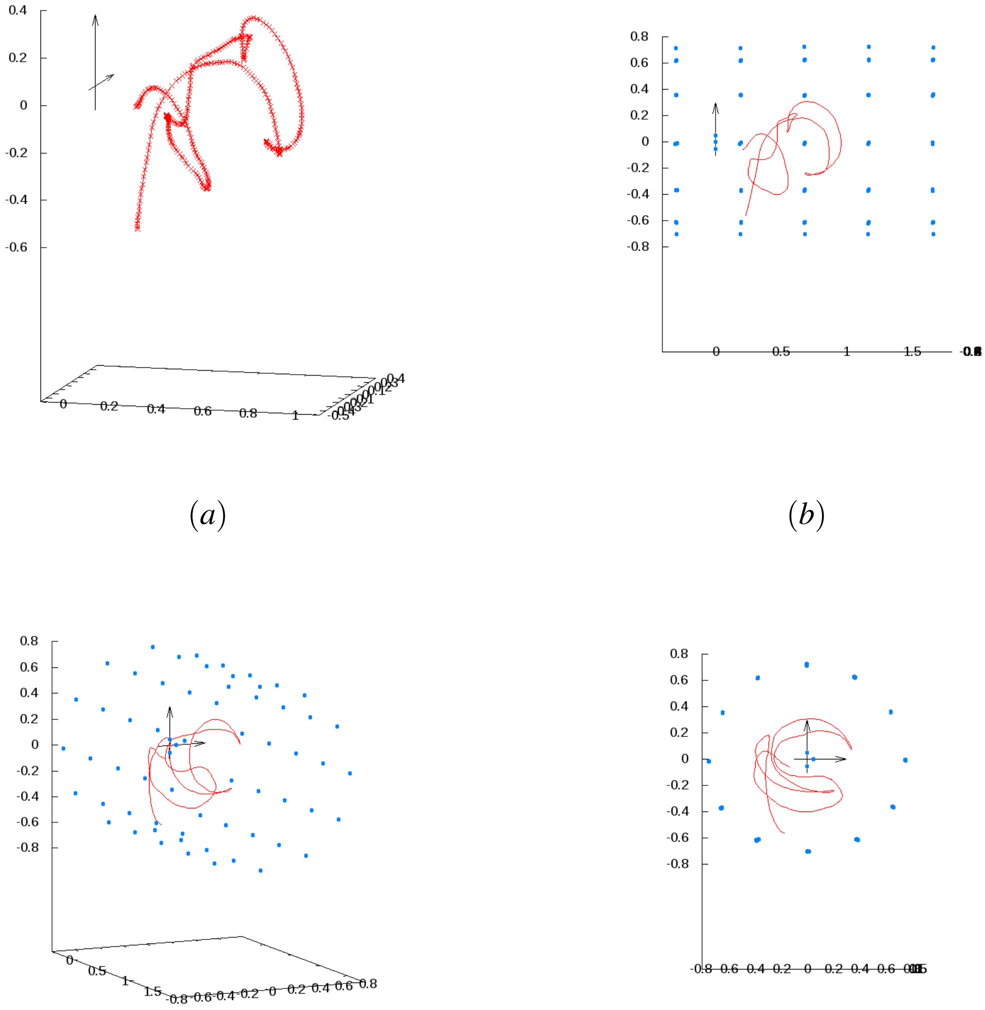


Figure 2: The low-reflection barrel including a cylindrical microphone array and a linear array. The barrel is used at DLR, Engine Acoustics, when measuring in reverberating environments. The light-weight structure makes the barrel portable, meaning the microphone positions are also subject to displacement from the nominal positions.

calibration recordings before and after the fly-over measurements. The acoustic-pulse generator was set similarly as in the experiments in the free-jet laboratory, with one reference microphone bound to the holding lance, see Figure 1 (b) and connected to one of the empty channels in the acquisition system. The distance between the reference-microphone protection grid and the electrode tips measured 0.691 m. As explained in section 2, the objective of the experiment is to determine the times at which the acoustic pulses are emitted and the time delays associated with all the microphones in the array. Compared to the experiment in the previous section, the present experiment presents new challenges. As it is shown in Figure 4, where the recorded signals associated with the first pulse of a measurement are shown for the reference microphone (the closest to the pulse origin) and for a microphone at a much larger distance from the pulse origin, very different signal amplitudes are recorded for different microphones, since the array

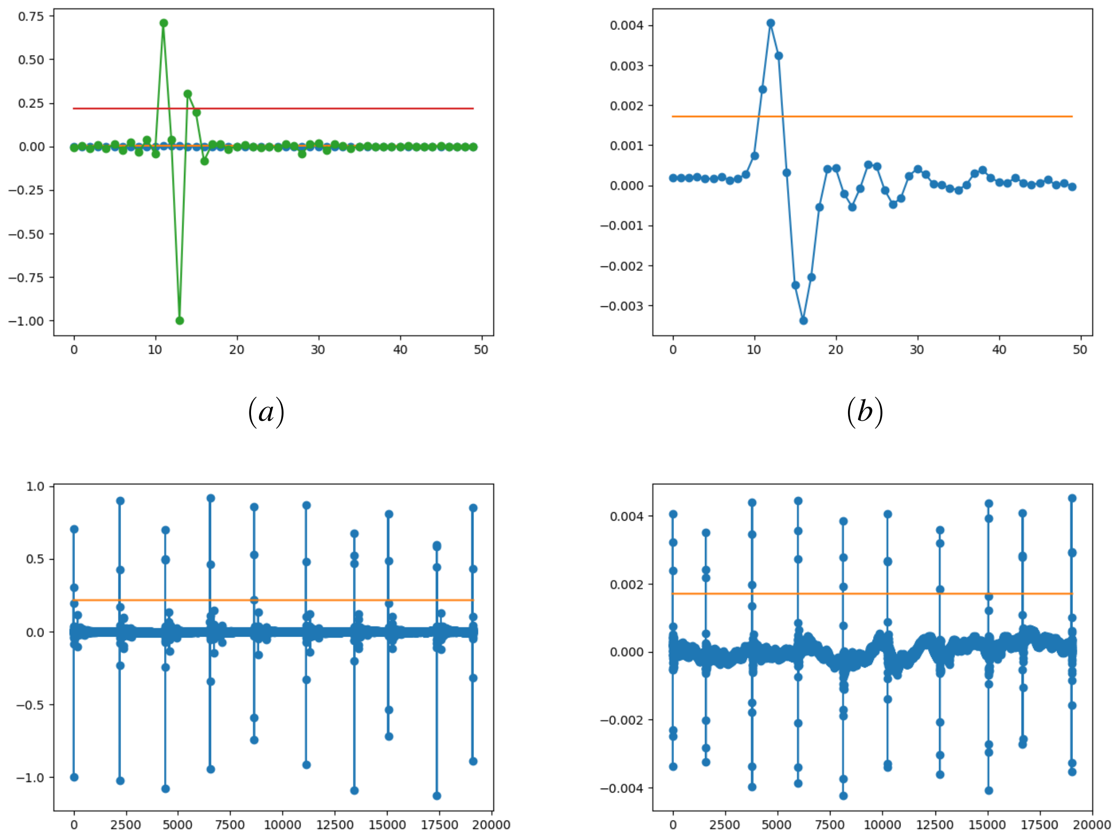


(*a*) (*b*)

(*c*) (*d*)

Figure 3: Results of the source and microphone localization algorithm, in the low-reflection barrel configuration, including 3 reference microphones connected with the jet nozzle in the laboratory.

spans a very large area, compared to the possible movement of the pulse origin. Further analysis of the data is required, in order to reconstruct the matrix $(R)_X$.



(a) (b)

(c) (d)

Figure 4: Record of the first pulse signal for the reference microphone (Channel No. 247) and for a far microphone. The far-microphone signal is reported on its own in the (b) diagram and it appears together with the reference-microphone signal in (a). The diagrams in (c) and (d) report the initial sequences at the reference and at the far microphones, containing 10 pulses each. The horizontal lines show the threshold value set to identify crossings of the signal, or the start of a pulse event in a given channel.

4 CONCLUSIONS

A geometrical calibration technique, similar to the one described in Ref. [2], has been implemented at DLR, Engine Acoustics. The technique makes use of a custom acoustic-pulse generator, based on electric arcs. It allows to determine the coordinates of the sensors in microphone arrays, by evaluating a time-delay matrix and using it as input to a non-linear optimization algorithm, as described in section2. The algorithm outputs the coordinates of the microphones and the curve on which the sequence of acoustic-pulse emission origins lies. The geometrical calibration has been successfully tested with a cylindrical microphone array, mounted in a barrel of low reflection walls, see Figure 3. Measurement performed with a large microphone array are being analyzed, in order to correctly determine a time-delay matrix. This will assess the method in conjunction with large sensor arrays for fly-over measurements.

REFERENCES

[1] F. Cultrera di Montesano. *Automatic and accurate determination of microphone coordinates in an acoustic array*. Master's thesis, Università degli Studi Roma Tre, 2018.

[2] D. Döbler and G. Heilmann. "New method for positioning of microphones." In *Proceedings on CD of the 3rd Berlin Beamforming Conference, 24-25 February, 2010*. GFaI, Gesellschaft zu Förderung angewandter Informatik e.V., Berlin, 2010. ISBN 978-3-00-030027-1. URL `http://bebec.eu/Downloads/BeBeC2010/Papers/BeBeC-2010-15.pdf`.

[3] H. Siller, T. Schumacher, and W. Hage. "Analysis of the jet noise emitted by an aircraft during take-off." In *Proceedings of the 8th Berlin Beamforming Conference, 2-3 March, 2020*. German Aerospace Center (DLR), Institute of Propulsion Technology, Engine Acoustics, Berlin, 2020.