In-Situ Control of Microphone Responses in an Array

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Summary:
A method to perform measurements of microphone responses directly in an array is proposed. It targets especially arrays built with MEMS Microphones and does not require high instrumentation effort. The method was successfully tested for calibration of two types of planar arrays in reverberant environments. Presented experimental results illustrate the application.

Keywords: MEMS microphone array, in-situ calibration

Introduction

Microphone arrays are widely used in acoustic measurements. An example can be found in acoustic cameras, where a suitable combination of single sensor signals allows to localize sound sources with the help of beam forming or to reconstruct the sound field by use of acoustic near field holography. The processing algorithms rely on exact matching between the microphones or at least on the knowledge of their amplitude and phase responses to compensate the differences, therefore extensive research was devoted to methods of calibrating microphone arrays in respect of positions of single sensors and their responses [1], [2].

The calibration task becomes especially challenging when the single microphones cannot be removed from the array after the assembly, e.g. when the array is built of miniature MEMS devices soldered onto printed circuit boards [3]. The approach proposed here allows to calibrate amplitude responses of the sensors and to control the actual condition of the microphone array. The method was developed especially for use with arrays based on low-cost MEMS Microphones and was successfully tested with planar arrays in reverberant cites.

Description of the Proposed Method

The approach targets the case when the microphones cannot be removed from the array for calibration, so the acquisition of sensor responses is performed directly by the measurement system itself in a dedicated firmware branch. In this way the influence of the whole acquisition chain microphone-preamplifier-ADC is taken into account.

To cope with the practical situation of reverberant environment excitation characterized by a distinguished first wavefront is used. The subsequent processing concentrates on this first wavefront to effectively suppress the influence of scattered waves which would arrive later. Knowing the geometry of the array and the anticipated sound pressure distribution on its elements, measurement and calibration of the microphone responses can be carried out. A suitable excitation can be produced e.g. by hands clapping or balloon popping. In our tests the most practical source turned out to be a loudspeaker driven with a rectangular voltage pulse.

Systems operated at typical audio sampling frequencies (about 40 kSps) cannot acquire all necessary details of the short wavefront. Hence the following processing is performed on interpolated signals. It is important, however, that they do not contain aliased frequencies.

After the high-pass filtering the peak amplitude of the first wavefront and its arrival time is defined from the interpolated signal. If at least one of the array microphones was previously calibrated, it can be used as the reference. Otherwise the average amplitude response over the array is used to obtain the relative amplitudes of all microphones. After performing several measurements the averaged amplitude responses are taken as the calibration factors.

Interpolation of original signals can be carried out by the zero padding in frequency domain. Good results were obtained also using interpolation by convolution with the Lanczos kernel of length 7 or higher, which can be performed on systems with very limited hardware resources.

Experimental Results

Experimental verification of the proposed calibration method was carried out with two different measurement systems based on MEMS microphone arrays. The presented results were gathered with a system containing 128 MEMS microphones arranged in a rectangular 16 by 8 matrix and digitized at f_s = 48 kSps [4]. An example of a recorded microphone signal is given in Fig. 1. It was obtained using a loudspeaker (60 W, 4 Ohm 5¼´´ Woofer of JBL GTC5210) positioned 3 m away from the array and driven by a rectangular voltage pulse of 15.5 V amplitude and 0.25 s duration.

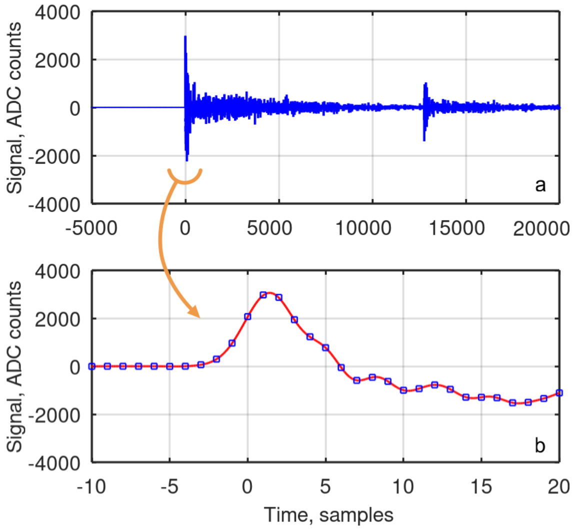

Fig. 1. Microphone signal recorded at 48 kSps (blue line and points). Red line – interpolation of the first wavefront.

Fig. 2 shows the amplitude calibration factors for all 128 sensors in the array calculated as relative responses. The variation of the microphone sensitivity is specified with ±1 dB, which is equivalent to approximately ±12% on the amplitude scale. It can be seen that all sensors lie well within the anticipated range.

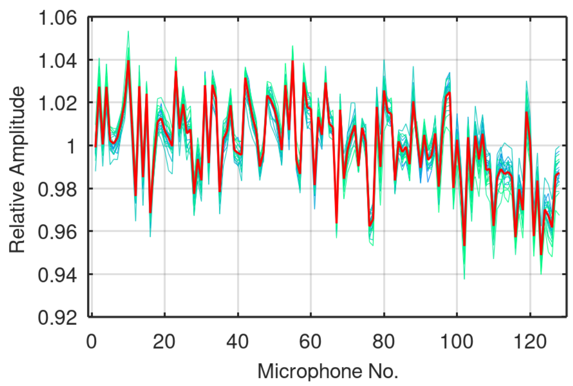

Fig. 2. Amplitude responses of 128 microphones in the array. Results of 20 measurements as blue-green lines, averaged values are shown in red.

The distances between the loudspeaker and the microphones differed over the array area by 0.5% at maximum. This variation should have a barely detectable effect on the amplitudes in Fig. 2. The apparent decrease of amplitudes at higher microphone numbers is attributed to the fact that the boards carrying microphones 97 to 128 originate from a different production batch.

The arrival time of the first wavefront is plotted in Fig. 3 for an exemplary measurement (only two of the array lines are shown for the clarity of representation). The green curve shows the theoretically anticipated arrival time based on the point source model for the given geometry (taken speed of sound is c = 344 m/s). Note the sub-sample time resolution which is possible due to the use of interpolated signals. Experiment agrees well with the theoretical model.

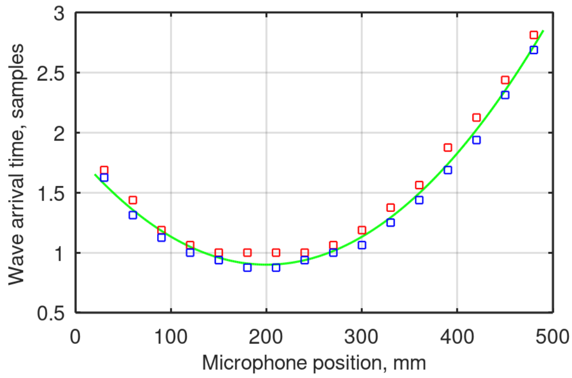

Fig. 3. First wavefront arrival time (squares) compared with the theoretical curve for 32 microphones.

Irregularities in the time response plot can be used for control of the system condition: experiments showed for example that an obstruction of a microphone port with a dirt particle can be clearly detected as a shift in the plot.

Conclusions

The presented approach allows to perform measurements of microphone responses directly in the array for their control or calibration in reverberant environments. Additionally performed comparisons with microphone calibration in a standing-wave tube confirm the validity of the results.

References

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