MICROPHONE POSITIONING USING ACOUSTIC LOCATION

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ABSTRACT

Part of the procedure in acoustic measurements is to collect the exact location of the microphone during the measurement. In simple measurement setups that include only one microphone, the coordinates of the microphone can be acquired manually. In contrast, when many microphones are in use or their movement is motorized the precise location has to be taken with an automated method. Such positioning systems most commonly represent an additional part of equipment or en extension of the measurement system. The study presents the use of high frequency loudspeaker drivers with known location to acquire the position of the microphones. The drivers are in a non planar configuration and an impulse response measurement is carried out between each driver and the microphones. The respective distance can be calculated based on the time of arrival of direct sound. A test measurement setup is presented together with the implemented routines.

1. INTRODUCTION

For room acoustic measurements the microphone location represents an important information that has to be reported. One of the reasons are the requirements given for its location, such as the minimal distance from the sound source in case of reverberation time measurements following standardized measuring procedures [1]. Furthermore, acoustic parameters, such as sound strength, are measured at a defined distance from the source [1]. In some advanced measurement methods [2–5] the information about the microphone position is required to extract the acoustic parameters in the post-processing of the acquired data. Another important reason is to be able to reproduce measurement results.

The most basic approach to determine the position of the microphone is using a ruler, i.e. manually measuring its distance from the thee Cartesian planes. Although this is a relatively simple task that can be largely facilitated by the use of a laser distance measurer, it can be often challenging to keep the measurement error low, especially if flat and perpendicular room boundaries can not be used as reference. In any case, the manual approach is unacceptably time consuming if many microphone coordinates have to be acquired.

This occurs when measurements are automatized and the movement of microphone(s) is motorized. In such measurements setups, the motion systems can include position tracking (e.g. robot arms [3]) that require robust and advanced mechanical components that come with a financial cost. Alternatively, if there is no tracking requirement, more options are available to move the microphone including simple and imprecise movement principles (e.g. based on cables).

The design of automated room acoustic measurement systems was also the motivation to conduct this study. In fact, as part of previously conducted sound filed characterization measurements the microphones are moved around the room [4, 5]. To facilitate this measuring process an acoustic measurement method was developed to determine the coordinates of the microphone in the room.

The method named microphone positioning system is introduces. In section 2, the basic theoretical background is presented, in section 3, the used positioning algorithm is explained, in section 4, the experimental setup is presented and in section 5, the results and achieved measurement accuracy are discussed. The conclusions of the study are summarized in section 6.

2. PRINCIPLES OF MICROPHONE POSITIONING

To determine the coordinates of an object the therms *localization* and *positioning* are used. To some degree they have a different meaning across diverse fields of applications [6] which introduces quite some confusion. In psychoaeoustics, localization [7] is reserved for the ability of the human auditory system to identify the location of sound sources. In more technical acoustic fields, the same therm is used to identify the location of sound sources using sophisticated equipment, such as microphone arrays. As in our case the microphone does not emit any sound and to avoid further confusion in this article the *positioning* therm is exclusively used to refer to the process of determining the position, i.e., the coordinates of microphones located indoor.

From the technological perspective, indoor positioning technologies mostly rely on electromagnetic or acoustic waves [8, 9] in various frequency ranges. Complex as well as simples devices can be used for positioning, such as distance measurers that measure the time of propagation of an ultrasonic or laser pulse reflected from a surface,

Many positioning technologies assume that the element, which coordinates are to be located, emits sound [10, 11]. This is the opposite of the presented approach that is based on emitting sound from high frequency drivers located at known postpositions distributed over the room's volume, while the microphone is used as receiver. This is convenient as no additional sensors have to be attached to the microphone.

The positioning method is based on intersecting spheres centered around loudspeakers with known locations. The radius of those spheres is determined from the time of arrival of direct sound extracted from impulse response measurements. Therefore, it is important that the timing of the measured impulse response is not affected by delays in the signal chain, such as latencies due to the DA or AD conversion. Those errors have been avoided by measuring also a reference impulse response of the measurement system excluding its acoustic part.

The time of arrival was then computed as the delay of the peaks in the impulse response in relation to the reference impulse response as presented by an example on Image 1. The absolute maximum value of the response was considered as the time of arrival. A threshold relative to the maximum amplitude of the impulse response was used to determine the exact time of the peak.

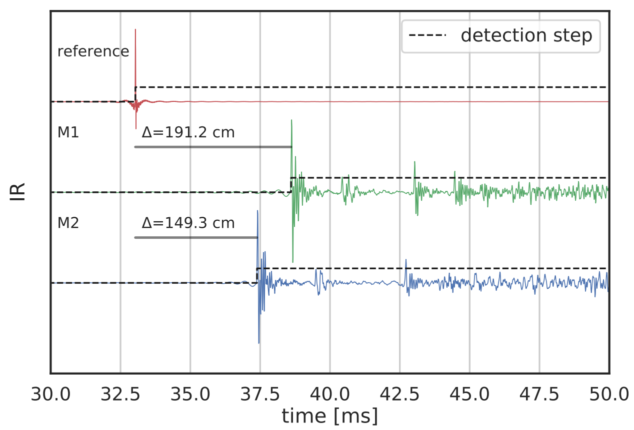

Figure 1. Impulse response measured by excluding the acoustic part (reference) and for two microphones positioned at two locations in the room. The direct sound propagation distance is measured as the time delay of the response peaks in relation to the reference.

Based on the time difference Δt the propagation distance d_M is obtained as

$$d_M = \Delta t \cdot c. \tag{1}$$

c is the speed of sound that can be estimated [12]

$$c = (331.3 + 0.606\,\vartheta)\ \text{m/s}. \tag{2}$$

ϑ is the air temperature in degrees Celsius.

In typical conditions in room acoustics the temperature ranges between 10 and 30 °C, corresponding to speed of sound variation between 337.4 m/s and 349.5 m/s. One way to estimate c is to track the air temperature during the measurement and follow eq. (2). Alternatively, and as it was implemented in the positioning system, the value of c is determined by minimizing the positioning error with an optimization routine.

Spherical surfaces are contracted from each of the microphone-loudspeaker measurement pairs. The intersection of two such spheres determines a circle in space. With the addition of another spherical surface the intersection of the three are two point in space. By properly setting the positions of the loudspeakers in the room, one of the intersection points lays outside the room boundaries, meaning that a three loudspeaker system can be used for microphone positioning.

In the presented measurement setup one additional loudspeaker has been used, with the main intention of achieving an over-determined positioning system. In such system, the four spherical surfaces do not intersect in a point but in a volume confined by the spheres. The size of this volume increases due to measurement error, for instance if the spheres radius is estimated based on erroneous value of c.

3. POSITIONING ALGORITHM

The positioning algorithm is based on generating a rectangular grid of points inside the room. The points are equidistant with the spacing of $\Delta L_0 = 16$ cm and spanning over the entire room's volume. In each grid point the distance to the four loudspeakers $d_{E,i}$ is calculated and its deviation from the measured distance, $d_{M,i}$, used to construct the error function

$$Err = \frac{1}{4}\sqrt{\left(\sum_{i=1}^{4} d_{E,i} - d_{M,i}\right)}. \tag{3}$$

In the next step, the point, in which the value of Err is the smallest, is chosen as the central point to generate a new grid with of half the previous spacing. Six grid points in each space direction are generated. The error function following equation (3) is then calculated again and the grid point with its smallest value is chosen. This is an iterative positioning algorithm with the number of repetitions N as a parameter. After the $N-$th iteration the coordinate of the chosen grid point is declared as the position of the microphone. In the conducted experiment $N = 6$ was chosen leading to the accuracy of the positioning system of $\Delta L_0/N = 5$ mm.

As already discussed in section 2, the speed of sound is temperature dependent. To determine its exact value, the fact that the positioning system is over-determined is exploited. For this purpose a measurements with six microphones distributed around the room has been carried out. For this dataset the positioning algorithm has been executed for 1000 values of c equally distributed between 337.4 m/s and 349.5 m/s. The one which returning the smallest sum Err from the six measurements was $c = 343,3$ m/s. The value corresponds to 19.8°C if calculated by eq. (2).

Apart erroneous c values some other factors may influence the determined position using the presented algorithm. One of them is a low sampling frequency that directly defines the resolution of identifying the peak of the impulse response. In the measurements the sampling frequency $Fs = 48$ kHz has been used meaning that determining the peak with one sample error corresponds to a $Fs \cdot c = 7$ mm distance error. Another potential error can arise if the impulse response is narrow band frequency filtered leading to a less articulated peak that limits the precision of its detection.

4. EXPERIMENTAL SETUP

Four high frequency drives of 1 inch diameter have been located in known positions in the room (coordinates listed in table 1) as shown in Figure 3. Apart not positioning the microphones in the same plain no special strategies have been used to choose their position.

loudspeaker	x	y	z
S1	2.130 m	1.810 m	0.810 m
S2	0.200 m	0.690 m	0.020 m
S3	0.865 m	3.830 m	0.810 m
S4	0.060 m	2.300 m	2.110 m

Table 1. The coordinates of the loudspeakers in the room.

It has been surprisingly challenging to manually measure the coordinates of the loudspeaker with high accuracy. In fact, it is difficult to establish the center of the membrane hidden inside the loudspeaker's housing. This is visually presented in Figure 3. It has been estimated that in practice the coordinates of the loudspeakers can only be measured with a 1 cm accuracy. As a consequence, the performance of the positioning system has not been compared to manually measured coordinates. In stead, the motion of the microphones has been set to follow a well defined trajectory, toward which the accuracy of the positioning system has been tested.

The impulse response is obtained as an inverse Fourier transform of the frequency response which is measured in the conventional way (see e.g. [13, 14]). Each measurement is repeated twice to improve the signal to noise ratio. Based on the coherence [13] the quality of the measurement was monitored.

The reproduced signal was an exponential sine sweep of the duration of 1 s with the frequency span between 2 kHz and 20 kHz. The sweep was reproduced on each of the four loudspeakers independently and recorded using one or two microphones. Furthermore, an additional input channel short-wired to the output has been recorded.

To conduct the measurements, the following equipment was used:

- 2 measurement microphones (Peavey, PVR 2),
- high frequency loudspeakers (1 inch tweeters),
- 2 power amplifiers (Apart Audio, CHAMP-2),
- sound card (Presonus, AudioBox 1818VSL),
- dedicated microphone stand with a rotating arm,

Figure 2. Determining the coordinates of loudspeakers with the ruler. Due to the hidden loudspeaker's membrane the positioning accuracy is estimated to 1 cm.

- personal computer – HP, Elitebook 8540p (Linux).

A calibration measurement in six microphone positions was first performed to determine the sound speed c as presented in section 3. The value $c = 343.1$ m/s has been been chosen and used for the microphone positioning.

A set of two measurements has been performed. In each the microphone stand has been fixed in a different position in the room and its arm holding the microphone rotated in the horizontal plain. As such, the microphone followed a circular trajectory as shown in Figure 2. Confining the microphone is the basis to estimate the error of positioning system. In fact, all the microphone positions would ideally lay on the same circle in space.

5. RESULTS AND DISCUSSION

Two measurement sets have been performed conducted for the microphones moving along two circles in the horizontal plain. The information about the space coordinates of each circle, its radius and number of microphone positions is given in table 2. The radius and the center coordinates of the circles was obtained using a lest mean square fitting algorithm on the microphones positions as obtained by the positioning algorithm.

The results for the two measurement sets are graphically shown in Figure 4. The deviation of the determined positions (points) from the circle line is slightly visible on the zoomed view.

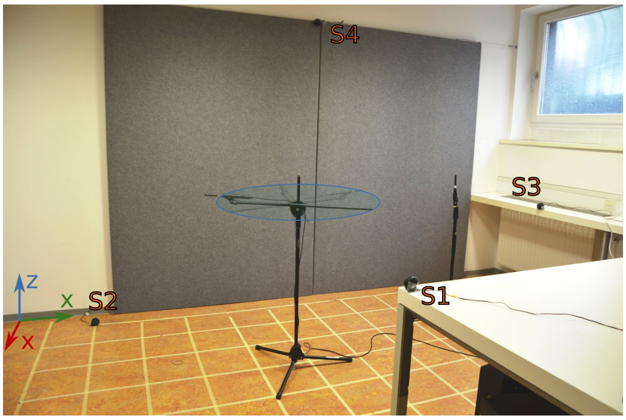

Figure 3. A photography of the room with the indication of the coordinate system (in the corner) the position of the four loudspeakers (S1, S2, S3 and S4) and the microphone stand with the boom arm that enabled the rotation of the microphone in a s horizontal plain.

measurement set	I	II
N	17	10
C_x, C_y, C_z [m]	1.15, 1.77, 0.98	0.62, 2.34, 0.26
r [m]	0.49	0.25
max err. [mm]	3.0	4.2
average err. [mm]	1.4	2.0

Table 2. Number of measurements (N), the center coordinate of the circle (C_x, C_y, C_z), the radius of the circle (r) and the errors of positioning the microphones in relation to the circle. The values are separately given for measurement sets I and II.

The error of the microphone positioning was estimated as the distance of the obtained coordinate to the circle. Its maximum and average value for the two measurement sets is given in table 2. Based on the results, the maximum error was only 4.2 mm which is less as the estimated accuracy for the manual positioning of the loudspeakers. For both measurement sets the average error is approximately half smaller than the maximum error.

During the measurement set I an additional microphone was located at a fixed position in the room. In this case the positioning algorithm should return the same coordinates for all 17 measurements. The coordinates obtained using the positioning system based on 17 measurements has moved up to 5 mm from its average value.

The performed measurements indicate that with the presented microphone positioning procedure the expected precision of determining the coordinates can be expected to be at least 5 mm. This is very close to the estimated precision of 1 cm for determining the loudspeakers coordinates and close to the 0.7 cm sampling error. Furthermore the values is comparable with the data available in literature [6].

The design of the conducted experiment enables that errors systematically displacing all microphone positions into a certain direction would not been detected. In fact, none of the microphone position has been directly compared to its coordinates as measured by a more precise methods. Although no reason is foreseen for such behavior, further tests would be in place to exclude this possibility.

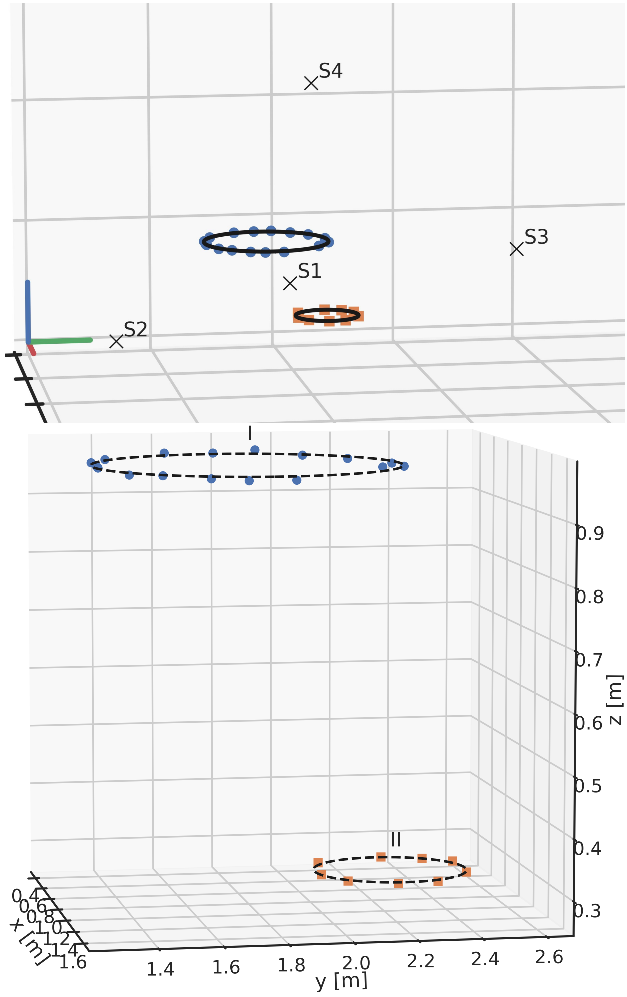

Figure 4. The determined microphone coordinates using the positioning algorithm (circles) for measurement sets I and II. The drawn line represents the best fitting horizontal circle in relation to which the error is estimated. The bottom image is a zoomed version exposing the measurement error of the method.

6. CONCLUSIONS

An acoustic microphone positioning method was presented based on overlap of spheres surrounding loudspeakers at known positions in the room. The radius of the spheres is defined by the time of arrival of direct sound from each loudspeaker. With this approach the microphone coordinates are determined purely acoustically without the need of attaching additional sensors to the microphone. The only parameter directly affecting the method is the speed of sound which has been determined based on an optimization algorithm on a small measurement set.

The obtained results of positioning have been measured for circular movement of the microphones. The maximum positioning error of 5 mm can be expected by the method and only half that value in average. The error is comparable in size to the values reported by the literature [6]. Although additional verification and testing of the method would be in place, we conclude that the accuracy of the method is sufficient for most room acoustic applications.

In the future, the method will be incorporated into automated measurements in room acoustics in which the microphones' movement is motorized. In those conditions all microphones location will be retrieved with a single positioning measurement which is a clear advantage of the method.

As part of future development of the method, special focus will be given into further increasing the number of loudspeakers and therefore over-determining the measuring system. This might enable the optimization of some input parameters of the system, such as the loudspeakers coordinates which exact value could be determined based on an optimization algorithm.

7. ACKNOWLEDGEMENTS

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