Smart Localization of Microphones inside an Automotive Cabin

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ABSTRACT: Interior sound measurements play an important role in vehicle development and refinement. Sometimes hundreds of microphones are installed in an automotive cabin. During test preparation and execution, a lot of time is spent in determining the microphone positions and in tracking cables to the data acquisition channel. A smart acoustic localization approach is presented to automate this process and to realize considerable time gains. It is based on estimating the distance between a microphone and (at least 4) sources by acoustic time-of-arrival measurements, combined with novel algorithms that cope with reflections and non-line-of-sight issues. The method will be illustrated using in-vehicle measurements.

KEY WORDS: vibration, noise, and ride comfort, sensor localization, acoustic modal analysis [B3]

1. Introduction

In the past few decades, the minimization of interior noise levels in automotive cabins has become a very active research area, mainly encouraged by the need for weight reduction and the current trend towards environmentally compatible hybrid-electric powertrain concepts. To improve the interior noise performance, CAE predictions have gained importance, especially in the early development stage when it is still possible to make changes without negatively affecting the vehicle development time. It is obvious that the effectiveness of this approach greatly depends on the accuracy of the predictions made using such a model. Hence, in order to understand the modelling challenges and improve the overall modelling know-how which will be useful in the car development phase, the experimental acoustic characterization of the cabin plays a crucial role. When performing an interior acoustic study, it is important to relate the acoustic responses to the intrinsic system behaviour of the cabin, which can be achieved by Acoustic Modal Analysis (1). Among the various specific challenges of this analysis, the test setup itself is a nontrivial task, as it includes the creation of a geometrical wireframe model that represents the 3D position of the sensors. An acoustic modal analysis test can easily consist of hundreds of degrees of freedom, particularly when the results are used for validating and updating finite element (FE) models. Establishing a precise test geometry then relies on the capability of the operator to instrument exactly at predefined locations. It is clear that in many realistic cases this task can be rather challenging, above all when the structure is complex. In current practice one has to measure the geometry of the test object by hand, which is extremely time-consuming and which typically yields inaccurate results. Automatic methods, which reduce this setup time and which yield more accurate results, are therefore highly sought for. Currently available solutions include photogrammetry (2) and geometry scanning systems (3). Although these solutions are effective, they often require expensive devices, which moreover can only be used by highly specialized technicians or may be impractical to use in complex environment such as an automotive cabin. This paper presents a fast, accurate, cost-effective and entirely acoustic procedure to automatically localize microphones in a car cabin. The microphone localization procedure is based on multilateration: the distances between at least four sources, whose coordinates are known or estimated a priori, and a microphone are utilized to determine the unknown position of the microphone in three-dimensional (3-D) space. Hence the estimation of microphone coordinates comprises two steps (Fig. 1): i) acoustic ranging measurements, ii) multilateration. However, due to the complex structure and obstructions typical of a car cabin (e.g. seats, dashboard, etc.), the direct path may be obstructed, a so-called Non-Line-Of-Sight (NLOS) condition. As a consequence, acoustic range estimates based on the time-of-arrival (TOA) may have an erroneous positive bias, i.e. the signal arrives at a microphone through reflections instead of through the direct (shortest) path. Furthermore, estimating the TOA of the direct path (when it exists) can be rather challenging in such a harsh environment. Reflections of the wavefront may indeed cause the strongest path not to be the first, leading to a possible erroneous detection of the actual first path, because of fading. The (overestimated) NLOS distances may subsequently lead to large 3-D position errors. In this work, we cope with the problem of NLOS identification and discard (IAD): the erroneous NLOS measurements are detected and pruned, so that the microphones are localized using the LOS distances only, hence yielding more accurate 3-D localization results. The paper is organized as follows. Section 2 presents the used
notations, the assumed system model, and the entire microphone localization procedure. In Section 3 the procedure is validated with experimental data from in-vehicle measurements. Finally, in Section 4 some concluding remarks are given.

2. Microphone Localization Procedure

In this section the entire microphone localization procedure is presented. First, the used notations and assumed system model are introduced. Second, the acoustic ranging method is described. Third, once the observed distances are obtained, the 3-D coordinates of the microphones can be determined by solving a non-linear optimization problem, where the microphone coordinates are the optimization variables and where the distance equations involving microphone coordinates form a non-linear objective function. Finally, the IAD NLOS algorithm, which identifies and discards NLOS distances, is briefly presented.

2.1. Notations and Model

Let \( \mathcal{M} = \{1, 2, \ldots, M\} \) be the set of indices of \( M \) sources, whose locations are known \textit{a priori} at \( x_i = [x_i, y_i, z_i]^T, i \in \mathcal{M} \), and let \( \{x_j = [x_j, y_j, z_j]^T\} \) be the unknown location of the \( j \)-th microphone, which we wish to determine. In a reverberant environment, the signal \( r_j(t) \) received by the \( j \)-th microphone can be modelled\(^{(4)}\) as a linear combination of \( N_{\text{path}} \) delayed versions of the signal \( s_i(t) \) emitted by the \( i \)-th source:

\[
r_j(t) = \sum_{l=1}^{N_{\text{path}}} A_l s_i(t - \tau_l) + n_j(t) \quad \text{for } i \in \mathcal{M},
\]

where the amplitude and travel time of the \( N_{\text{path}} \) multipath are \( A_l \) and \( \tau_l \) respectively, \( n_j(t) \) is an additive noise term with a Gaussian distribution with spectral density \( N_0/2 \). The information related to the separation distance between the \( i \)-th source and the \( j \)-th microphone can be hence obtained, under the assumption of synchronous communication system, by measuring the Time-Of-Arrival (TOA) of the direct path:

\[
\text{TOA} \equiv \tau_l = \frac{||x_j - x_i||}{c} \quad \text{for } i \in \mathcal{M},
\]

where \( c \) is the speed of sound in the acoustical medium, and \( || \cdot || \) is the Euclidean norm.

Since each microphone is treated independently, the index \( j \)-th will be omitted in the following in order to streamline the notation.

2.2. Range Estimation Process

Localization performances are highly dependent on the quality of the range measurements. Hence, particular attention has to be paid to this crucial first phase. The TOA can be simply estimated by cross-correlating the received signal with the transmitted signal template. In ideal propagation conditions, TOA estimate is given by the time instant corresponding to the maximum absolute peak of the cross-correlation function (CCF) over the observation interval. Nevertheless, as already reported, in dense multipath scenarios, the TOA estimation actually consists in the correct detection of the first arriving path. In general, while the dominant peaks may correspond to the signal echoes, it is not straightforward to find the correct peak due to the presence of noise and fading. Hence, this ambiguity highlights that TOA estimation in the presence of multipath is not a pure parameter estimation problem, but rather a joint detection-estimation problem\(^{(5)}\). Several approaches have been reported for TOA-based ranging. Many of these approaches are threshold-based TOA estimators\(^{(5)}\); the first threshold crossing event is taken as the estimate of the TOA. In this work, instead, the TOA estimation problem will be considered as a change detection problem without relying on any threshold (Fig. 2).

2.2.1. TOA-Based Range Estimation

Let \( R_{rs}(\tau) \) be the CCF between the transmitted signal \( s(t) \) and the received signal \( r(t) \):

\[
R_{rs}(\tau) = E[r(t)s(t + \tau)] = \sum_{l=1}^{N_{\text{path}}} A_l R_{\alpha l}(\tau - \tau_l) + z(\tau),
\]

where \( R_{\alpha l}(\tau - \tau_l) \), with \( l \in \{1, 2, \ldots, N_{\text{path}}\} \), is the auto-correlation function of the transmitted signal \( s(t) \) and \( z(t) \) is the coloured Gaussian noise of the CCF with auto-correlation function \( R_{zz}(\tau) = N_0 R_{\alpha l}(\tau/2) \). Let \( t_{\text{peak}} \) and \( A_{\text{peak}} \) be the arrival time and amplitude of the strongest arriving path:

\[
\begin{align*}
\{ t_{\text{peak}} &= \arg \max_{\tau} |R_{\alpha l}(\tau)|, \\
A_{\text{peak}} &= \max_{\tau} |R_{\alpha l}(\tau)|.
\end{align*}
\]
Then, the normalized version of the CCF, \( \hat{R}_{rs}(\tau) \), can be represented by
\[
\hat{R}_{rs}(\tau) = \frac{R_{rs}(\tau)}{A_{\text{peak}}} = \sum_{i=1}^{N_{\text{peak}}} \rho_i R_{rs}(\tau - \tau_i) + w(\tau), \tag{5}
\]
where \( \rho_i = A_i/A_{\text{peak}} \), and \( w(t) = (1/A_{\text{peak}}) z(t) \).

### 2.2.2. The TOA Estimator

The TOA represents the transition point between two regions of the CCF: the noisy region consisting of only noise; and the multipath region consisting of the echoes in addition to noise. Such a change point (i.e., the sample correspondent to the TOA) can be detected through an appropriate detection function, which estimates the difference between the two regions.

First of all, in order to remove sign ambiguity of the path amplitude \( \phi(t) \), the square of the normalized CCF, \( \phi(\tau) = |\hat{R}_{rs}(\tau)|^2 \), can be used in the detection process

\[
\phi(\tau) = \left| \sum_{i=1}^{N_{\text{peak}}} \rho_i R_{rs}(\tau - \tau_i) + w(\tau) \right|^2. \tag{6}
\]

Secondly, it is convenient to enhance the process by limiting the detection to a set \( \mathcal{K} \) of candidate peaks only: the local maxima of \( \phi(\tau) \) in the interval \([0, \tau_{\text{peak}}] \):

\[
\mathcal{K} = \left\{ \tau \in [0, \tau_{\text{peak}}] : \frac{d \phi(\tau)}{d \tau} = 0, \frac{d^2 \phi(\tau)}{d \tau^2} < 0 \right\}. \tag{7}
\]

Notice that the detection process will be limited to the interval \( \tau \in [0, \tau_{\text{peak}}] \) since either the first arriving path is the global maximum, or comes prior to the global maximum.

Thirdly, by denoting with \( k = 1, 2, \ldots, K \) the indices of the elements in \( \mathcal{K} \), with \( \tau_k \) the time instants of the \( k \)-th local maximum, with \( \Phi_k \) the values of \( \phi(\tau) \) when \( \tau \in \mathcal{K} \), and focusing our attention on a change in amplitude, it has been observed that the following detection function:

\[
S_k = \frac{\Phi_k}{1/K \sum_{j=1}^{K-1} \Phi_j} \tag{8}
\]
is likely to have a global maximum in correspondence of the transition point (i.e., the local maximum corresponding to the first arriving path):

\[
k_{\text{first}} = \arg \max_k S_k, \tag{9}
\]
hence

\[
\text{TÖA} = \tau_{\text{first}}. \tag{10}
\]

Although the function defined in Eq. (8) has proven to be effective, it cannot be ruled out that erroneous ranging estimates will still occur in practice. In addition, the existence of obstacles in a car cabin (such as the seats, the dashboard, etc.) may also result in NLOS conditions in which the source and the microphone do not share a direct link, i.e., because only reflections of the transmitted signal arrive at the microphone. Therefore, there is a need for removing such erroneous NLOS measurements by the later NLOS IAD algorithm (cf. infra).

### 2.3. Multilateration Process

In absence of noise and NLOS bias, the actual distance \( d_i \) between the microphone and the \( i \)-th source defines a sphere around the \( i \)-th source corresponding to possible microphone locations, i.e.,

\[
(x-x_i)^2 + (y-y_i)^2 + (z-z_i)^2 = d_i^2, \quad \text{for } i \in \mathcal{M}. \tag{11}
\]

The exact location of the microphone is then found at the unique intersection of all the spheres. In practice however, the noisy distance measurements and NLOS bias yield spheres which do not intersect at the same point, resulting in the inconsistent equations

\[
(x-x_i)^2 + (y-y_i)^2 + (z-z_i)^2 = d_i^2, \quad \text{for } i \in \mathcal{M}. \tag{12}
\]
Equation (12) can be solved by minimizing the sum of squared residuals, so that the 3-D localization problem is reformulated as an optimization problem, i.e.

$$\min_{x} \left\{ \sum_{i=1}^{M} \left( \|x - x_i\| - \hat{d}_i \right)^2 \right\}$$  \hspace{1cm} (13)

whose solution yields the final estimate $\hat{x}$ of the target location. In multilateration, at least four LOS sources are required for a unique 3-D solution. If there are more than four LOS anchors, the redundancy of these sources can be utilized in a least square (LS) sense for a more accurate localization result. Minimizing the non-linear expression in Eq. (13) can be achieved by numerical search methods. In this paper, we will use as a solver the Broyden-Fletcher-Goldfarb-Shanno (BFGS) Quasi-Newton method$^6$.

Evidently, the optimization method will be most effective in estimating the location when only LOS distance measurements are combined. Therefore, there is a need for a NLOS IAD algorithm, which is able to identify the NLOS observed distances and discards them in the LS calculation (cf. infra).

2.4. NLOS Identify and Discard Algorithm

The NLOS IAD algorithm$^{7,8}$ is based on the fact that the LS method is extremely sensitive to outliers, i.e. a single outlier (an erroneous NLOS measurement in our case) may already be sufficient to offset the LS estimator$^9$. Such an approach is able to identify the number $L$ of LOS anchors and localize the microphone with them only. To achieve this, the method exploits the redundancy and compares the squared residual of a set of distances to a predefined threshold, $\zeta$:

$$\left( \hat{d}_i^* - \hat{d}_i \right)^2 < \zeta^2$$  \hspace{1cm} (14)

where $d^*_i = \|\hat{x} - x_i\|$ is the estimated distance between the $i$-th source and the microphone at estimated location $\hat{x}$.

Firstly, the algorithm begins by using all the available $M$ distances. It estimates the microphone location, $\hat{x}$, and compares the residuals, as in Eq. (14). If all the $M$ residuals are below the threshold $\zeta$ (as determined by location accuracy requirements), then $L = M$, otherwise the optimization procedure moves to groups of $(L - 1)$ sources. The algorithm stops when all the residuals of the $k$-th set of $M$ sources taken $L$ are below the threshold, or when $L = M^{\text{min}}$, i.e., the minimum number of sources which are necessary for a unique localization result.

3. Experimental Case Study

The method described above has already been proven to be effective with both simulated and real experimental data$^{7,8}$. In this section, the acoustic method is tested by using in-vehicle measurements performed inside a fully trimmed estate car.

The results are then compared with those obtained by means of a state-of-the-art photogrammetric positioning system, the AICON ProCam$^{10,11}$. A total of 38 microphones have been uniformly placed inside the car cabin. As visible in Fig. 3, microphones were present: on the front (FFLR) and rear floor (RFLR), on the front windshield (FWIN), on the rear left (DORL) and right (DORR) door, and on the roof (ROOF). All the microphones were fixed about 10 cm far away from the surfaces. Such microphone positions are typical points used for an acoustic modal test and represent challenging points to be localized.

3.1. Acoustic Method

3.1.1. Sound Sources

For multilateration, the sound source should ideally be a monopole. It means that, within the frequency range of interest, the characteristic length of the source must be smaller than the minimum wavelength, and the source itself must be always omni-directional. For this purpose, a new, compact LMS Qsource volume source has been utilized (Fig. 4). Such a source incorporates an internal sound source strength sensor which outputs a realtime volume acceleration signal, and it emits the noise as a monopole source up to several kHz.

The acoustic centre of the speaker was previously calibrated and assumed stable in the excitation bandwidth.

3.1.2. Experimental Setup and Measurement Arrangement

Eleven sources have been spread over the entire cabin, placed in strategic positions in order to localize the highest number of microphones. Details on their position are reported in Table 1 and Fig. 5$^1$.

During the measurements, the averaged temperature of the environment was constantly recorded in order to calibrate the speed of sound by assuming the temperature was constant throughout the cabin.

The acquisition was performed using an LMS SCADAS front-end together with LMS Test.Lab software. The maximum sampling rate of $f_s = 204.8$ kHz provided by the acquisition system has been used so as to have the highest possible temporal resolution of the CCF. Furthermore, a peak fitting procedure has been used in order to improve the estimation of subsample time delays.
3.1.3. Template signal

In order to have a high resolution, the pulse length must be short to avoid the overlapping of near echoes in the CCF. This is not a trivial requirement, as it is difficult to emit sharp pulses with general purpose loudspeakers. In addition, the signal has to be emitted with the highest power available in order to increase the signal-to-noise ratio (SNR), and to minimize the Cramer-Rao Lower Bound (CRLB)\(^{(12)}\). Nevertheless the SNR is proportional to pulse duration\(^{(13)}\). The contradiction is solved by long duration signals with short duration correlation functions, so when the received signal goes through an appropriate template signal, the output will be a short pulse. A signal commonly used in radar application in order to achieve pulse compression is the linear frequency modulated (chirp) signal, which also prevents standing waves. Hence, such a signal will be used in the ranging phase.

The source is able to emit a broadband sound between 1-20 kHz. A typical power spectral density (PSD) of the source is displayed in Fig. 6.

For this test, a chirp waveform that varied in frequency from 1000 to 20000 Hz was generated in 0.64 s. A high SNR was already obtained with a single measurement, hence no repetition was needed to reduce uncorrelated background noise power by averaging.

The measurement and relocation of 4 roving sources took less than half an hour for one test operator.

3.1.4. Results

In Figs. 7–8, a comparison is made between a localization algorithm where the pruning is not applied (i.e. NLOS distances remain present), versus the considered localization algorithm where erroneous distances are identified and discarded. As visible in Figs. 7–8, the application of the NLOS IAD algorithm is not only effective, but also essential for a correct localization of all microphones.

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Table 1: Source Positions

| # Source | Full Name | Position          |
|----------|-----------|-------------------|
| 01       | DOFL:01   | Door Front Left   |
| 02       | DOFR:02   | Door Front Right  |
| 03       | DORL:03   | Door Rear Left    |
| 04       | DORR:04   | Door Rear Right   |
| 05       | DOFL:05   | Door Rear Left    |
| 06       | DORR:06   | Door Rear Right   |
| 07       | FLOR:07   | Floor Front Left  |
| 08       | FLOR:08   | Floor Front Right |
| 09       | FLOR:09   | Floor Rear Centre |
| 10       | ROOF:10   | Roof Front Centre |
| 11       | ROOF:11   | Roof Rear Centre  |

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Fig. 4: New compact LMS Qsources volume source

Fig. 5: Sources Distribution

Fig. 6: A typical PSD of the sound source
3.2. Photogrammetric Method

The same 38 microphone positions were measured as accurately as practicable by using the AICON 3D ProCam, a state-of-the-art Geometry Scanning system based on inverse photogrammetry. Such a photogrammetric digitizer system uses a roving hand-held probe (Fig. 9) that contacts the microphones. This probe consists of a probe tip on the front, an infrared flash and a CCD camera at the back of the probe. The CCD camera is located behind the black lens in the center of the back of the probe. The user points the probe tip at the required point location and triggers the measurement by pushing a button on the probe while the back of the probe is oriented towards an array of retro-reflective targets (Fig. 9). The image is processed and the coordinates of the probe tip are calculated. Thanks to the integration in the LMS Test.Lab Geometry workbook, a new node is automatically created in the geometric model. The test operator repeats this process for each required point in order to determine the complete geometrical model.

The achieved accuracy depends on the proper setup of the panels (i.e. requiring a skilled test operator), but is 0.4 mm or smaller for a typical setup \(^{(2,14)}\).

Although such a method is an appealing solution for applications where sensors are instrumented on car exteriors (e.g. a car body-in-white modal test), the method appeared less practical for applications where sensors are mounted in car interiors (e.g. acoustic modal analysis as considered in this work). A proper strategy therefore had to be devised and adopted in this work. Due to the size of the car, limited focal length of the infrared camera and the inevitable obstacles (e.g. doors, seats, etc.), two calibrated panels were used and moved several times during the measurement process. A proper measurement plan was needed in order to minimize the repositioning of the panels. A global coordinate system was defined on the roof of the car.

Fig. 7: Localized microphones without the application of the NLOS IAD algorithm

Fig. 8: Localized microphones with the application of the NLOS IAD algorithm

In total, 3 panel locations (right, left and back side) were used to digitize all the microphones. In each case, the panels were positioned about 1.5 m away from the side of the vehicle at a height of about 1 m. For each panel position, the origin of the system and common points in different datasets needed to be measured in order to merge the points and obtain a single coordinate system. Furthermore, two different probe tips (a short and an angled probe) were used in order to avoid some obstacles.

From the start of the digitizing project to the end, the entire task, comprising the relocation of the calibrated panel, the re-definition of the coordinate system, and the measurement of the microphones positions, took about 2 hours for two experienced test operators.

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\(^2\)The results have been obtained by using all the 11 sources and defining a threshold $\zeta = 2$ cm.
3.3. Comparison of Acoustic and ProCam Position Measurement

As displayed in Fig. 10, the mean value of the discrepancies between the ProCam measurements and the acoustically estimated coordinates was around 2.7 cm. Such a value is not representative of a quantitative evaluation of the acoustic method, since it cannot be guaranteed that the ProCam measurements are actually accurate, due to the difficult and impractical use of this method in such an environment. Nevertheless, the discrepancies are sufficient for demonstrating the effectiveness of the approach in a complex scenario, such as a car cabin.

It is worth highlighting that the results obtained by the acoustic method appear to be comparable to those yielded by the photogrammetric method, a process that proved much more difficult, impractical and time-consuming in this environment. Furthermore, the acoustic technique has proven to be much faster than the photogrammetric system: around 0.5 man-hour against around 4 man-hours.

This experiment has qualitatively proven the effectiveness of the NLOS IAD algorithm in complex scenarios such as a car cabin. The approach not only provides reliable 3-D positions, but also drastically reduces the measurement set-up time, exploiting the same hardware used for acoustic tests, unlike more conventional methods which proved much more difficult and impractical in this scenario and require experienced test operators.

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