Applied Acoustics 129 (2018) 229–238

Contents lists available at ScienceDirect

Applied Acoustics

journal homepage: www.elsevier.com/locate/apacoust

On the array configuration and accuracy of remote in-ear level sensing for in-vehicle noise control applications

Mirjana Adnadjevic [a,*], Bart Goossens [b], Dick Botteldooren [a]

[a] Department of Information Technology, Ghent University, Technologiepark-Zwijnaarde 15, B-9052 Ghent, Belgium
[b] Department of Telecommunications and Information Processing, Ghent University, St. Pietersnieuwstraat 41, B-9000 Ghent, Belgium

ARTICLE INFO

Article history:
Received 30 January 2017
Received in revised form 20 July 2017
Accepted 1 August 2017
Available online 10 August 2017

Keywords:
Virtual microphone
Microphone selection
Regularization
Optimization
Sparse signal processing

ABSTRACT

Measuring in the ear of the operator of a vehicle may be useful in several applications such as noise assessment, adaptive communication, or active noise control. However, wearing in-ear microphones is not very comfortable. This work proposed a design methodology for an array-based-virtual-microphone for remote in-ear level sensing. To extend the sensing capabilities of current virtual microphone methods, a method that besides robust filter design also includes microphone placement optimization is proposed. Motivated by the recent progress in sparse signal processing, an unconstrained optimization is proposed which keeps the fidelity of in-ear level while reducing the number of initial active microphones through the log-sum-sparsity-induced regularization term. It has been compared to several screening methods which are optimizing for system diversity and are often used for optimal sensor placement. To study the importance of microphones being locally placed around the ears, several methods to maximize the system redundancy are also evaluated. The selection approach based on the sparsity coding theory achieves the best remote sensing performance with the least number of microphones. It results in the array configuration which balances well between the local (around the head) and global placement (throughout the vehicle roof). Sensitivity of the design to the natural head movements of 20 and 45 degrees is assessed, showing an improved system robustness by accounting for those variations in the design.

1. Introduction

The quality of the working environment and in particular acoustic comfort has been a topic of concern for many years. Indeed, excessive noise exposure may lead to occupational noise-induced hearing damage and/or lower job efficiency [1,2]. In addition, the trend in moving the noise control from simple level control to sound quality engineering is already seen in consumer products and vehicles [3,4] and is of increasing interest for the work environment as well. Numerous active noise control algorithms have been proposed for this purpose [5–7]. They are mostly tuned for local control in a small area around the operator, as global control is often not efficient. The latter is due to the complex sound field in the enclosed space resulting in high hardware costs [8]. However, the main constraint of traditional local control approaches is the limited zone of control located around the error sensor. For best performance, the error microphone should be therefore placed in the ear of the operator, assuring the controllability of the actual exposure. Since wearing in-ear microphones or microphones very close to the ear is usually impractical, virtual microphone techniques are proposed that can shift the zone of control to the desired location which is at a distance from the physical sensors [9]. The use of virtual in ear microphones is not limited to the noise control. It can be for example applied for remote monitoring of in-ear noise exposure at the work floor. Also, when reproducing desired sound in the enclosed space, such as speech or music, feedback systems for enhancing quality of experience could be envisaged. Moreover, the same microphone arrangement may be used for recording speech produced by the operator, thus leading to an integrated audio-system. The concept of virtual sensing, as a generalization of virtual microphone concept, is also exploited in related industrial applications where the sensor placement at the desired locations is an issue [10–13].

Current virtual microphone approaches are mainly based on the filter design stage during which remote microphone signals are combined to reproduce the field at the virtual locations. In an experimental setup, this is done by recording simultaneously the field at the array of microphones and at a set of microphones temporarily placed at the virtual microphone positions. Otherwise, the

* Corresponding author.
E-mail address: mirjana.adnadjevic@ugent.be (M. Adnadjevic).

http://dx.doi.org/10.1016/j.apacoust.2017.08.004

field is obtained through the numerical modeling. After training, microphones are removed from the virtual locations and the derived filters are applied to the remaining microphones to estimate the field at the virtual locations. Different virtual microphone algorithms have been derived depending on the filter design method used. Some of them are based on Kalman filtering and state-space modeling [14,15], while others use FIR and IIR black-box modeling, or different derivatives of the adaptive LMS technique [9]. In the framework of noise control, these algorithms have been evaluated through number of theoretical and experimental studies for acoustic ducts [16], small enclosed spaces [17], but also complex acoustic enclosures such as a mock up of an aircraft [18]. The main drawback of these current methods is the sensitivity to errors in the estimation, resulting in a poorer controllability of the noise at the virtual location [18]. While most of the research has been focusing on the robust filter design to mitigate this issue, no further explanation on the selection of remote microphone placements for this type of application was provided.

For virtual sensing in related industrial applications, sensors are usually selected prior to the filter design phase from the pool of available sensors by utilizing different observability metrics. The most common ones are the scalar properties of the Fisher-information-matrix, the Popov-Belevich-Hautus (PBH) criterion, or simply the covariance of the optimization error [10]. More specifically, for the virtual sensing in vibro-acoustical enclosures, where the low frequency field is dominated by distinctive modes, the authors in [19] propose a spatially averaged modal observability metric. It has been shown that observability of the modes relevant for the virtual location significantly contribute to the overall virtual sensing accuracy and consequently to the filter design [19]. The importance of the sensor placement to accurately sense the low frequency modal acoustical field was the motivation for this paper to move current virtual microphone methods for noise control application in vehicle-type enclosures from simple filter design to include sensor placement optimization.

Regardless of observability metric used, the disadvantage of the aforementioned observability screening is the high computational cost due to its combinatorial nature [20]. The optimization is characterized as mixed integer/convex problem that is usually efficiently solved by guided random search methods, such as the genetic algorithm (GA) [21]. However, other alternative solutions started to arise, since the method was shown not to scale well to larger system requirements [20]. An increased interest in sensor placement research field was shown for convex relaxation approaches that promote sparse solution with regards to the number of active sensors in use [20,22–25]. This principle guarantees near-global performance with less sensors and within short calculation time [20]. The method is, therefore, seen to be suitable for the case under study as, unlike modal observability screening, it does not require the knowledge of the system modal behavior. Moreover, an extensive screening is replaced by the regularizer which minimizes the system hardware requirements. To authors' knowledge, no such method was derived, or a applied in the field of virtual sensing.

In this paper, the possibilities for using an array of microphones positioned at the roof of the agricultural cabin for primary noise field estimation at the ear of the driver are investigated. The placement at the roof is considered unobtrusive regarding the comfort and esthetics. The hardware costs of such a system increase with the number of microphones and could be considerable. In addition to the microphones, the signal processing power needs to be increased to allow performing the microphone virtualization in real time. Moreover, there is a risk of over-fitting to the source locations used during the identification phase. To tackle this, an experimental method for selecting optimal microphone locations, starting from the measurements at a multitude of potential locations, is proposed. Several selection criteria are presented and compared for their achieved sensing performance.

The main focus of the paper is on the selection method which minimizes active microphones through regularized unconstrained optimization and is related to the sparsity principles mentioned upfront. The method is based on the theory behind the Re-Weighted Zero Attraction LMS algorithm (RZA-LMS) [26] where the regularizer is modified (m-RZA-NLMS) to fit the case under the study. For the comparison, a diversity metric is exploited for sensor placement in a manner similar to [27,28], i.e. in a greedy fashion. Several selections are based on minimization or maximization of this metric on different signals of the optimization process respectively. Maximization of such a metric in many cases leads to obtaining the system with the least redundancy and, therefore, optimal sensor placement [27,28]. The metric was also minimized to study the possibility of improving the robustness of the system to the potential variations by closely placed microphones.

The structure of the paper is as follows: Section 2 describes the proposed virtual microphone design in terms of system model, filter design and selection criteria; Section 3 contains the details about the measurement setup and procedure; Section 4 proofs the design through validation measurements, provides an analysis for the effect of the number of microphones and microphone selection on sensing performance. It also discusses the sweet spot of the best selection for the cases when head rotations were excluded and included in the design. Sections 5 and 6 discuss and conclude the paper respectively.

2. Virtual microphone design

The proposed virtual microphone design is seen as a part of the machine prototyping phase. In this phase, a microphone configuration, which complies with driving comfort requirements, is placed throughout the cabin. The number of microphones placed in this way should widely exceed the number of a excited modes in a same-size-rectangular-shaped room for the targeted frequency range. At the location of the operator, a head and torso simulator (HATS) is seated, containing two reference microphones at the ears. The HATS is needed to consider the presence of the head and body in the virtual microphone design. The microphone selection criteria is then applied to derive an optimal array configuration which also matches the desired hardware and computational constraints in terms of number of microphones. The configuration is optimized on both ears and for different cabin excitations. The latter can be generated by operating the machine in different regimes or by stimulating through spatially distributed vibration or loudspeaker exciters.

The virtual microphone filter design follows the microphone selection. A method for the optimal filter designs is applied here and should also hold for the operational regimes of interest. Filters are derived for each ear independently for the final tuning of the proxy measurements.

2.1. General system model

The proposed virtual microphone relies on an experimental model identification with the measurement sensor noise included, but it can also be used with data from numerical models. Combining multiple signals from the remote microphones for an estimation of the signal at a virtual location is a Multiple Input Single Output (MISO) identification problem. For the case under test, the system model is schematically drawn in Fig. 1. The signal at the ear can be estimated as a sum of the FIR filtered microphone signals.