Applied Acoustics 139 (2018) 93–100

Contents lists available at ScienceDirect

Applied Acoustics

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

Acoustic source localization with microphone arrays for remote noise monitoring in an Intensive Care Unit

Markus Müller-Trapet[a,*], Jordan Cheer[a], Filippo Maria Fazi[a], Julie Darbyshire[b], J. Duncan Young[b]

[a] Institute of Sound and Vibration Research, University of Southampton, University Rd, Southampton SO17 1BJ, United Kingdom
[b] Nuffield Department of Clinical Neurosciences, Level 6, West Wing, John Radcliffe Hospital, Oxford OX3 9DU, United Kingdom

ARTICLE INFO

Keywords:
Noise source localization
Microphone arrays
Intensive Care Unit

ABSTRACT

An approach is described to apply spatial filtering with microphone arrays to localize acoustic sources in an Intensive Care Unit (ICU). This is done to obtain more detailed information about disturbing noise sources in the ICU with the ultimate goal of facilitating the reduction of the overall background noise level, which could potentially improve the patients' experience and reduce the time needed for recovery. This paper gives a practical description of the system, including the audio hardware setup as well as the design choices for the microphone arrays. Additionally, the necessary signal processing steps required to produce meaningful data are explained, focusing on a novel clustering approach that enables an automatic evaluation of the spatial filtering results. This approach allows the data to be presented to the nursing staff in a way that enables them to act on the results produced by the system.

1. Introduction

High noise levels in Intensive Care Units (ICUs) have been reported as a possible contributing factor to patients' poor physiological recovery [1]. It is thought that these high noise levels can increase the risk of disturbed sleep patterns, hallucinations and periods of delirium [2,3]. While the World Health Organization (WHO) recommends that sound levels in patient areas should remain below 40 dB(A) [4], this limit is often exceeded in ICU environments [5,6].

A number of strategies have been attempted to reduce sound levels in ICUs. Building design and materials [7,8], reducing patients' perception of noise with earplugs or headphones [9,10] and staff education [11], have all shown variable effectiveness, as has the manipulation of alarm thresholds and volumes [12]. A recent review of noise-reducing measures comes to the conclusion that when patients experience lower noise levels through the use of earplugs, the risk of delirium can be reduced [13].

To better identify the contributing noise sources and explore ways of reducing them, a more detailed understanding of the distribution of acoustic sources over time and space is necessary. Currently, studies use individual sound level meters at discrete positions in the ICU to record the level variation over time [14]. While this offers insight into the periods over which the overall noise activity is higher, it cannot provide information about the distribution of specific sources in time or space. An additional problem is that in order to obtain representative data, sound level meters have to be installed close to the patients' beds. This often leads to spurious unrealistically high peak levels when staff come into contact with the audio equipment or the supporting structure.

A solution to the problem of using local sound level meters lies in the application of spatial filtering methods such as beamforming to discriminate acoustic sources in space from a remote location [15]. The use of array signal processing methods allows the ICU environment to be scanned for sources with an (almost) arbitrary resolution in a non-intrusive way.

Microphone arrays in combination with spatial filtering methods have found widespread usage in recent years. Typical scenarios where beamforming is used include industrial and environmental noise source identification as well as automotive and aeroacoustic applications [16–20]. In most of these cases, a manual evaluation of the results in terms of source maps per third-octave band is usually necessary, hence requiring an expert to make sense of the data. Automatic source localization is a problem often encountered in speech signal processing [21,22], but the approaches are not always generally applicable because of underlying assumptions about the array geometry or the number and type of source signals.

In this paper, a microphone array system designed for the task of automatic source localization is presented and described in detail, with a specific focus on the array signal processing. A modified formulation

* Corresponding author at: National Research Council, 1200 Montreal Road, Ottawa, ON K1A 0R6, Canada.
E-mail address: markus.mueller-trapet@nrc.ca (M. Müller-Trapet).

https://doi.org/10.1016/j.apacoust.2018.04.019
Received 23 October 2017; Received in revised form 16 February 2018; Accepted 16 April 2018
Available online 24 April 2018

of the standard Delay-and-Sum beamforming algorithm is presented to increase the computational performance. A deconvolution algorithm usually employed for aeroacoustic measurements is applied here in the context of the ICU environment, leading to increased spatial selectivity. For an automatic evaluation of the beamforming result, a novel clustering approach is described in detail, enabling a display of the results that is easy to understand for non-acousticians.

The outline of the paper is as follows: in Section 2 the setup of the audio hardware of the array system is described. The design of the microphone array is presented in Section 3. The beamforming strategies including the data clustering approach are described in detail in Section 4 together with simulated example data. In Section 5 selected results of measurements made with the system installed in the adult ICU of the John Radcliffe Hospital in Oxford are shown. The paper finishes with concluding remarks in Section 6.

The descriptions in this paper will be useful for replicating the presented array system, and the clustering approach helps to automatically evaluate the result of beamforming calculations and make it presentable to non-technical personnel. Practical considerations for the array system as well as the restrictions imposed by the environment are addressed specifically.

2. Hardware setup

To provide insight into the practical setup, a general plan of the ICU environment and the location of the hardware is shown by a schematic diagram in Fig. 1. This includes the location of the beds, microphones and cables.

As a cost-effective means to capture audio signals, electret microphones were used. To obtain the best result under the circumstances prescribed by the application, two steps had to be taken. Firstly, the microphone capsules were connected with an electric circuit, which lowers the phantom voltage of 48 V down to the supply voltage of approximately 4 V, enables longer cables between the microphones and the pre-amplifiers and provides a balanced signal for improved

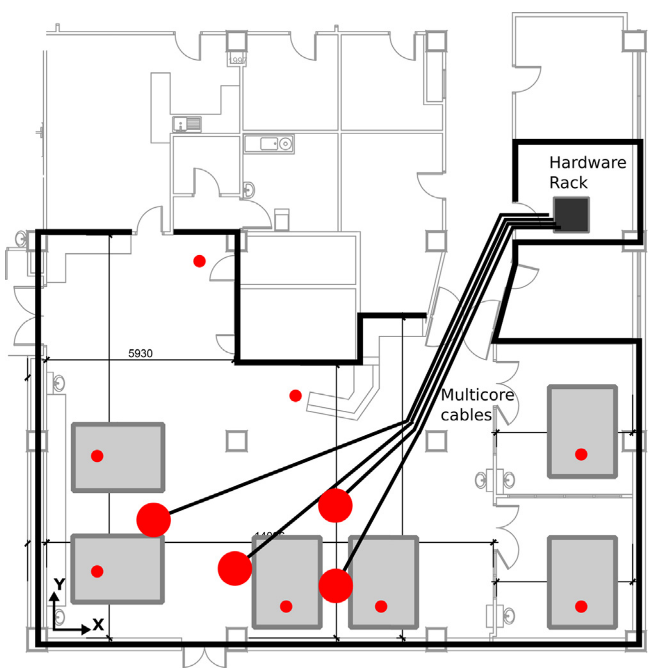

Fig. 1. Schematic of the floor plan of the ICU at John Radcliffe Hospital including the location of the beds (grey squares), the microphones (arrays: large red circles; individual: small red circles), the multi-core cables (black lines) as well as the hardware rack. (For interpretation of the references to colour in this figure legend, the reader is referred to the web version of this article.)

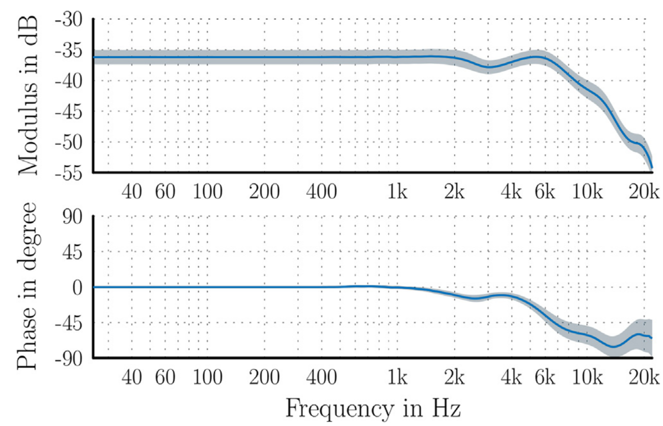

Fig. 2. Microphone sensitivity in dB re V/Pa: average result, with standard deviation across all microphones represented by shaded region.

interference rejection. Secondly, all microphones were calibrated over the entire frequency range to match the level and phase response for an optimum beamforming result. It has been shown that the phase response has to be matched for all array microphones to obtain beamforming results of high quality [23]. The broadband calibration was carried out in the small anechoic chamber at the ISVR by measuring the frequency response of a Genelec 8010A [24] with each microphone and comparing this to the response measured with a *Brüel & Kjær* $\frac{1}{2}$" free-field measurement microphone with a flat frequency response. The average sensitivity response in dB re V/Pa is plotted in terms of the modulus and phase as a function of frequency in Fig. 2. The shaded area represents the standard deviation across all microphones. The calibrated responses were combined with the measured pre-amplifier gains and inverted to obtain equalization filters that are applied to the audio input data in the frequency-domain to convert digital signal levels into sound pressure levels before any further processing takes place.

To avoid spending time developing bespoke hardware, off-the-shelf audio equipment was chosen to provide the microphone pre-amplifiers and digital conversion. To process the microphone signals and provide phantom power, a total of nine Focusrite OctoPre MKII [25] were used, each converting the input voltage of eight microphones into a digital ADAT stream. The eight ADAT streams for the 64 array microphones were then combined into one MADI stream with the RME ADI-648 [26]. For an additional eight microphones distributed in the ICU, a second MADI stream was provided by the Ferrofish A16-MKII [27]. Both MADI inputs were fed to the RME MADI FX sound card [28] to provide digital audio for further processing.

The connection between the hardware rack and the microphones was achieved with five 16-channel multi-core cables of 30 m length each. Four of these cables were used to connect to the four sub-arrays (see Section 3) and the fifth combined the signals from the eight individual microphones distributed in the ICU (see the schematic in Fig. 1).

3. Array design

As the microphone system was not supposed to interfere in any way with the daily operation of the medical personnel or the patients, the ceiling was chosen as the location of the microphones. This means that a planar array configuration had to be used. Due to the large spatial extent of the ICU (approximately 12 m× 8 m), four clusters of 16 microphones were distributed throughout the ICU to cover all of the patients' beds (see Fig. 1). For an additional, separate level monitoring, eight individual microphones were also installed in the ceiling above the beds and the nurses' station. The individual microphones were used for stationary level monitoring above the beds, including beds in two separate rooms that cannot be covered by the array system.