KU LEUVEN

STADIUS
Center for Dynamical Systems, Signal Processing and Data Analytics

Citation/Reference	Alexander Bertrand, Simon Doclo, Sharon Gannot, Nobutaka Ono, Toon van Waterschoot (2015), **Special Issue on wireless acoustic sensor networks and ad hoc microphone arrays** Signal Processing, vol. 107, pp. 1-3, Feb. 2015.
Archived version	Author manuscript: the content is identical to the content of the published paper, but without the final typesetting by the publisher
Published version	http://dx.doi.org/10.1016/j.sigpro.2014.10.001
Journal homepage	http://www.journals.elsevier.com/signal-processing
Author contact	alexander.bertrand@esat.kuleuven.be *+ 32 (0)16 321899*
IR	

(article begins on next page)

Special Issue on wireless acoustic sensor networks and ad hoc microphone arrays

Despite their obvious advantages over single-microphone systems, traditional microphone arrays have their limitations because they usually sample the sound field only locally, typically at a relatively large distance from the sound source(s). Furthermore, due to space and energy constraints, especially in miniature and portable devices, the array is often limited in physical size and in processing power. *Ad hoc microphone arrays,* and in particular their wireless implementation in so-called *wireless acoustic sensor networks* (WASNs), have been introduced to overcome these limitations.

A WASN consists of a set of wireless microphone nodes, which are spatially distributed over the environment, usually in an ad hoc fashion. Due to the wireless communication, the array-size limitations disappear and the microphone nodes can physically cover a much larger area, which vastly increases the amount of spatial information.

However, depending on the application, the algorithm design for WASNs or ad hoc microphone arrays is considered to be very challenging due to several aspects:

- The microphones at different nodes are usually uncalibrated and their relative positions are often unknown, in which case no a priori information about the array geometry is available. In some applications, the microphone positions can even change during operation of the algorithm, and microphones can be added or removed.
- There are often stringent *communication bandwidth constraints* when exchanging data between microphone nodes. Usually, there is insufficient bandwidth to broadcast tens or hundreds of microphone signals from the microphone nodes to a fusion center or other nodes (for example, the wireless link of currently commercially available binaural hearing aids only allows wireless transmission of a single microphone signal). Therefore, algorithms for WASNs should use the available communication resources as efficiently as possible.
- Often a *distributed in-network processing* is envisaged such that the audio signals are jointly processed by the nodes, rather than in a central device. This significantly reduces the communication cost and it avoids the need for a dedicated and power-hungry central processing device, which is often unavailable.
- As all the microphone nodes in the WASN have their own local clock, there will be a *sampling rate mismatch* between the different microphone signals, which may significantly affect the performance of coherent signal processing techniques as used in traditional multi-channel acoustic signal enhancement algorithms.

This special issue aims to provide new insights in the signal processing theory, modeling, algorithms, and implementation aspects related to wireless acoustic sensor networks and ad hoc microphone arrays and their use in various applications.

In the sequel, we briefly outline the main contribution of the 15 papers that appear in this special issue.

The special issue starts with a review paper by S. Markovich-Golan and his co-authors, entitled "*Optimal distributed minimum-variance beamforming approaches for speech enhancement in wireless acoustic sensor networks"*, which addresses the algorithmic challenges arising in the distributed realization of minimum variance (MV) beamformers in WASNs. The authors review three optimal distributed MV-based algorithms, and discuss their underlying relations and differences. Furthermore, although the original algorithms were designed for fully-

connected network topologies, the authors explain how they can be modified to operate in partially-connected WASNs.

In the paper "*Classification of reverberant audio signals using clustered ad hoc distributed microphones*", S. Gergen and his co-authors discuss automatic classification of audio signals contaminated by reverberation and interfering sounds. The mismatch between test and anechoic training data is mitigated by using the spatial distribution of microphones of ad hoc microphone arrays. In the proposed algorithm, clusters of microphones that either are dominated by one of the sources or contain mainly signal mixtures and reverberation are estimated in the audio feature domain. Information is then shared to create one feature vector for each cluster to classify the source dominating this cluster. The proposed method allows for a robust classification of simultaneously active sources in reverberant environments.

The paper of S. Burgess et al. entitled "*TOA sensor network self-calibration for receiver and transmitter spaces with difference in dimension"* addresses the problem of finding both source and node positions using time of arrival (TOA) measurements when there is a difference in dimensionality between the subspaces spanned by the sources and the microphones. The authors designed an algorithm for the degenerate and overdetermined case, and tested it in indoor acoustic experiments with multiple microphones and speakers, demonstrating a good localization performance.

The paper *"Distributed Localization using Acoustic Doppler"* by D. Lingren et al. deals with the problem of localizing a moving sound source based on Doppler shift observations obtained in a wireless acoustic sensor network. The work proposes a novel motion model specifically aimed at sensor networks, and a numerical optimization procedure to solve the associated localization problem. Monte Carlo simulations as well as real-world experiments are carried out to validate the proposed method, and a computational complexity analysis is included to illustrate its applicability to practical sensor network implementations.

In the paper *"Localizing Multiple Audio Sources in a Wireless Acoustic Sensor Network",* A. Griffin et al. propose a grid-based method for the localization of multiple sound sources by means of a wireless acoustic sensor network. The authors consider a WASN setup in which each sensor node contains a microphone array, and only transmits direction-of-arrival (DOA) estimates to a central processing node, thereby heavily reducing bandwidth requirements. A new model for the DOA estimation error obtained in this setup is derived, and the results of extensive realistic simulations are reported to show the superiority of the proposed method to state-of-the-art methods in localization accuracy as well as computational complexity.

In the paper "*Cooperative Integrated Noise Reduction and Node-specific Direction-of-Arrival Estimation in a Fully Connected Wireless Acoustic Sensor Network*", A. Hassani and his co-authors describe a distributed algorithm in which each node of a WASN is tasked to estimate the direction-of-arrival (DOA) of a source with respect to its local array. Using a specific cooperation strategy in which the nodes exchange pre-processed signals, the authors demonstrate that each node is able to improve its node-specific DOA estimate, without knowing or estimating the relative geometry between them.

The paper *"Active Noise Control Over Adaptive Distributed Networks"* by M. Ferrer et al. is presumably one of the first papers considering a distributed implementation of an active noise control (ANC) system. The authors define a distributed system architecture for ANC based on a wireless acoustic sensor network in which each node consists of microphones, loudspeakers, and a processing unit with communication capabilities. A distributed implementation of the state-of-the-art Multiple Error Filtered-x Least Mean Square (MEFxLMS) adaptive filtering algorithm is derived, and a performance analysis in terms of the WASN parameters is provided. Simulations of realistic scenarios confirm the potential performance of the distributed ANC system.

In the paper *"Analysis of the average performance of the multi-channel Wiener filter for distributed microphone arrays using statistical room acoustics",* T. C. Lawin-Ore and S. Doclo derive analytical expressions for the

average performance of the multi-channel Wiener Filter in a homogeneous noise field, by exploiting the statistical properties of the acoustical transfer functions between the desired source and spatially distributed microphones. Experimental results show that the derived analytical expressions can be used to easily compare the performance of different microphone configurations, without having to measure or numerically simulate a large number of acoustical transfer functions.

In the paper *"Distributed Estimation of the Inverse of the Correlation Matrix for Privacy Preserving Beamforming"*, Zeng and Richard Hendriks consider a privacy preserving scenario where users in the network want to perform distributed target source estimation with a WASN, without revealing the actual source of interest to other entities in the network. The estimation of the inverse noise (or noise plus target) correlation matrix in the so-called minimum variance distortionless response (MVDR) beamformer is structured as a consensus problem that can be realized in a distributed fashion via the randomized gossip algorithm. The proposed method makes it possible to compute a distributed MVDR beamformer without revealing the steering vector to any of the other entities in the network, and providing privacy about the actual source of interest.

The paper *"Ad Hoc Microphone Array Calibration: Euclidean Distance Matrix Completion Algorithm and Theoretical Guarantees"*, by Taghizadeh et al., addresses the problem of ad hoc microphone array calibration where only partial information about the distances between the microphones is available. The paper proposes a Euclidean distance matrix completion algorithm for recovering the distance matrix, enabling ad hoc microphone calibration from partial noisy measurements of pairwise distances, by using a low-rank matrix completion approach. A theoretical bound of the microphone calibration error is developed and the proposed algorithm is evaluated using real recording.

In the paper *"Audio Coding in Wireless Acoustic Sensor Networks",* A. Zahedi et al. propose a distributed source coding (DSC) approach to audio coding in a wireless acoustic sensor network. The rate-distortion function (RDF) for Gaussian sources is derived, which is considered an upper bound for the RDF of more realistic audio signals since Gaussian sources are shown to provide the lowest coding gain. An acoustical model based on the room impulse response (RIR) is used to design practical coding schemes which are validated using real audio measurements. Simulation results using audio measurements obtained in a standard listening room show that joint coding and dereverberation can be achieved.

The paper *"On the effect of compression on the complexity characteristics of wireless acoustic sensor network signals"*, by N. Tatlas et al., considers the application of audio compression algorithms at the sensor node level in a WASN. More specifically, the effect of data compression on signal complexity is investigated by employing four widely used audio compression algorithms in combination with different entropic and information measures. Numerical results obtained for a variety of environmental monitoring sounds suggest that it is indeed possible to compress audio while maintaining the complexity characteristics of the sound signal. This result has important implications for reducing bandwidth usage in WASNs by means of local audio compression.

In the paper *"Hybrid Digital-Analog Transmission for Wireless Acoustic Sensor Networks"*, M. Rüngeler et al. consider the wireless transmission of an acoustic signal to a central fusion node in a WASN and propose the use of Hybrid Digital Analog (HDA) transmission systems. In contrast to purely digital transmission, it is shown that the quality of the HDA transmitted signal is not limited by the inherent quantization noise in the source encoder. The received audio SNR improves with increasing radio channel quality. Especially for WASN with many transmitters, which may not be able to adapt to the radio channel quality, HDA transmission systems can show their full potential.

The paper *"A combined hardware-software approach for acoustic sensor network synchronization", by* Schmalenstroeer et al., presents an approach for synchronizing a WASN using a two-stage procedure. First the clock frequency and phase differences between pairs of nodes are estimated employing a two-way message exchange protocol. The estimates are further improved in a Kalman filter with a dedicated observation error

model. In the second stage, network-wide synchronization is achieved by means of a gossiping algorithm which estimates the average clock frequency and phase of the sensor nodes. While these steps are done in software, the actual sampling rate correction is carried out in hardware by using an adjustable frequency synthesizer.

The last paper in this special issue is written by S. Miyabe and his co-workers, and is entitled *"Blind compensation of interchannel sampling frequency mismatch for ad hoc microphone array based on maximum likelihood estimation"*. The sampling frequency mismatch among different recording devices is one of the problems in an ad-hoc microphone array. By assuming that the sources are not moving and have stationary amplitudes, the authors propose a blind method to estimate the sampling frequency mismatch based on maximum likelihood estimation as well as an efficient resampling method involving the modification of short-time Fourier transform (STFT) analysis with a non-integer frame shift. The effectiveness of the proposed method is evaluated in experiments.

We would like to express our sincere gratitude to all the authors for contributing to this special issue, as well as the many reviewers for helping us in assessing the papers and giving constructive comments to the authors. We would also like to thank the Editor-in-Chief of Signal Processing, Prof. Björn Ottersten, for giving us the opportunity to compile this special issue and for handling the review process for the papers co-authored by the guest editors. Finally, we are grateful to the entire staff of Signal Processing for their efforts and guidance in compiling this special issue.

We hope that the many ideas and techniques described in this special issue will inspire many researchers working in similar fields, and that they may help advancing this flourishing and exciting topic of WASNs.

Alexander Bertrand, Simon Doclo, Sharon Gannot, Nobutaka Ono, and Toon van Waterschoot.