Introduction to the Issue on Acoustic Source Localization and Tracking in Dynamic Real-Life Scenes

I. Introduction

ACOUSTIC source localization and tracking is a well-studied topic in signal processing, but most traditional methods incorporate simplifying assumptions such as a point source, free-field propagation of the sound wave, static acoustic sources, time-invariant sensor constellations, and simple noise fields.

However, these assumptions may be seriously violated in a range of emerging real-life applications. In these applications, the environment is extremely challenging, with spatially distributed sources, reverberation and reflections, complex noise fields, multiple concurrent speakers, interference signals, and time-varying positions of sources and/or sensors.

Aside from classic source localization scenarios, present-day application areas include audio recording with mobile devices (e.g. cell phones, action cameras, and robots), video conferencing on the go, and recording for 3D reproduction and virtual reality. Underwater acoustic localization is another application, which is characterized by intricate propagation patterns.

This special issue presents recent advances in the development of signal processing methods for localization and tracking of acoustic sources and the associated theory and applications. Addressing the challenges raised by real-life environments, novel methods are introduced representing advances in array processing, speech processing, and the use of data inference tools.

II. Contributions

Fourteen manuscripts were accepted for publication in this special issue, addressing localization and tracking of acoustic sources in underwater environments or in air.

The papers can be roughly categorized in four groups: Papers proposing algorithms using either deep learning, statistical inference or array processing, as well as papers addressing sound propagation modelling. These four groups are summarized below.

A. Algorithms Using Deep Learning

In the paper "Multi-speaker DOA estimation using deep convolutional networks trained with noise signals" a convolutional neural network (CNN)-based supervised learning method for estimating the directions of arrival (DOAs) of multiple speakers is proposed. Multi-speaker DOA estimation is formulated as a multi-class multi-label classification problem, where the assignment of each DOA label to the input feature is treated as a separate binary classification problem. In a comprehensive evaluation, it was empirically shown that for a microphone array with M microphones, $M-1$ convolution layers are required for the best localization performance. The proposed method outperforms the multiple signal classification (MUSIC) and the steered response power with phase transform (SRP-PHAT) in terms of mean absolute error (MAE).

The paper "CRNN-based multiple DOA estimation using ambisonics acoustic intensity features" proposes a neural network to improve DOA estimation in reverberant environments using features derived from acoustic intensities as input to a convolutional recurrent neural network (CRNN). Analysis of the workings of the proposed method using layer-wise relevance propagation is also presented, together with a detailed description of experimental results. It is demonstrated that the technique can identify the sound "attack" times as being most relevant, in agreement with psychoacoustic studies.

In the paper "Sound event localization and detection of overlapping sources using convolutional recurrent neural networks", a CRNN for joint sound event localization and detection (SELD) of multiple overlapping sound events in 3D-space is proposed. The proposed network takes a sequence of consecutive spectrogram time frames as input and maps it to two outputs in parallel. At the first output, the sound event detection is carried out as a multi-label classification task on each time frame producing temporal activity for all the sound event classes. At the second output, localization is carried out by estimating the 3D Cartesian coordinates of the DOA for each sound event class using multi-output regression. The proposed method is able to associate multiple DOAs with respective sound event labels and further track this association over time. The paper also provides a survey of many neural network-based localizers.

B. Algorithms Using Statistical Inference

The paper "Robust ocean acoustic localization with sparse Bayesian learning in non-stationary noise" notes that the matched-field processing (MFP) problem can be thought of as an underdetermined set of linear equations, which suggests a compressed sensing (CS) approach. There is some literature showing that CS implemented using the non-greedy basis

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pursuit (BP) method has significant shortcomings. The insight provided by this paper is to use sparse Bayesian learning (SBL) to learn the sparsity profile (common across snapshots and frequencies) corresponding to the location of present sources. The approach is evaluated using real data: the SWellEx-96 experiment.

The paper "Distributed source localization in acoustic sensor networks using the coherent-to-diffuse power ratio" proposes an approach to acoustic source localization that "learns" the source-to-microphone distances in reverberant rooms. The feature used in the Bayesian learning model is based on the coherent-to-diffuse power ratio (CDR), the power of the coherent signal component to the power of the reverberation, assumed here to be a function only of the relative source-to-microphone distances and not of position. The trained model based on Gaussian processes produces estimates of source-to-microphone distances given estimates of CDRs. The source is then localized by weighted least-squares combining the estimated source-to-microphone distances. The algorithm has been designed to minimize data communication among microphone nodes and is proposed as a viable distributed-sensor processing technique.

The paper "Speaker tracking based on distributed particle filter and iterative covariance intersection in distributed microphone networks" presents a method for tracking an acoustic source using a network of spatially distributed microphone pairs. The method is based on distributed particle filtering, with a modified iterative covariance intersection (MICI) fusion strategy. First, the classical generalized cross-correlation phase transform (GCC-PHAT) function is used to estimate the TDOA at each node. Then, multiple TDOA measurements are sent to the local particle filter at each node to estimate the speakers local state posterior. Finally, the distributed calculation of the local posteriors is implemented via the MICI fusion strategy to obtain a global consistent state estimate at each node. The performance of the method is verified in both simulated and real-life noisy and reverberant environments. It is also shown to be robust to node failures and to unknown network size.

In the paper "Online localization and tracking of multiple moving speakers in reverberant environment" the authors use the direct-path relative transfer function (DP-RTF), an inter-channel feature that encodes acoustic information against reverberation in a robust fashion, and propose an online algorithm well-suited for estimating the DP-RTFs associated with moving audio sources. They also assign the DP-RTFs to audio-source directions. To this end, they adopt a maximum-likelihood formulation and use an exponentiated gradient to efficiently update the source-direction estimates starting from their currently available values. To track multiple speakers, a Bayesian framework is adopted. In this context, the authors propose a variational approximation of the posterior filtering distribution associated with multiple speaker tracking and an efficient variational expectation maximization solver. The proposed online localization and tracking method is evaluated using two datasets that contain recordings in real environments.

The paper "An active acoustic track-before-detect approach for finding underwater mobile targets" considers the challenge of tracking and estimating the size of a single submerged target in a highly reverberant underwater environment using a single active acoustic transceiver. This problem occurs in many applications, ranging from security to environmental research. Considering that the target can be either slow (e.g., a scuba diver) or fast moving (e.g., a shark), the authors avoid assumptions about the targets reflection patterns or motion type, and instead, perform probabilistic tracking using a constraint Viterbi algorithm, whereby detection is performed based on the maximum likelihood criterion, implemented using the expectation-maximization (EM) approach. Based on the tracked path, the authors then evaluate the target's size. To test their approach, they have performed extensive simulations as well as eight sea experiments in different environmental settings to track both a scuba diver and a sandbar shark. The simulation results show a tracking performance that is close to the Cramér-Rao lower bound.

C. Algorithms Based on Array Processing Methodologies

The authors of the paper "An underwater acoustic positioning algorithm for compact arrays with arbitrary configuration" use simple geometrical insight to find a fast approach for localization in a small underwater ultra-short baseline (USBL) array with an arbitrary configuration. In the proposed algorithm, the bearing is estimated by applying a group of linear equations. The method is evaluated by both simulations and real data. Both numerical and experimental results indicate that the proposed algorithm outperforms competing methods in terms of positioning accuracy when some of the array sensors malfunction.

The paper "Direction of arrival estimation for reverberant speech based on enhanced decomposition of the direct sound" presents an approach for speaker localization in reverberant conditions, and compares this to existing approaches. The proposed approach uses a single-term signal-subspace MUSIC to determine a threshold for time-frequency (TF) bins that contain coherent single-source information, and utilize these to extract the DOA of the direct sound of the sources. The proposed direct-path dominance (DPD) test is compared to previous DPD tests, and to other recently proposed reverberation robust methods, using computer simulations and an experimental study, demonstrating its potential advantage. The studies include multiple speakers in highly reverberant environments, therefore representing challenging real-life acoustics scenes.

The paper "Acoustic source localization based on geometric projection in reverberant and noisy environments" deals with the problem of acoustic source localization from the perspective of geometric projection with four types of narrowband power functions and three fusion methods for broadband sources. Interestingly, it is shown that widely-used conventional algorithms (e.g. MUSIC, SRP-PHAT) could be cast into the presented framework based on the projection of the observation signal vector onto a hypothesized steering vector.

The paper "Subspace-based algorithms for localization and tracking of multiple near-field sources" investigates the task of estimating and tracking the DOAs and ranges of multiple near-field (NF) narrowband sources impinging on a uniform linear array (ULA). It presents a simple subspace-based algorithm for the localization of NF sources (SALONS), where the computationally costly eigen-decomposition and spectrum peak search are avoided. In the proposed SALONS algorithm, the DOAs and the ranges are estimated separately using a one-dimensional subspace-based estimation technique. Furthermore, an online algorithm is developed for tracking multiple moving NF sources with possible crossover points of their trajectories. The effectiveness and the theoretical analysis of the presented algorithms are verified through numerical examples.

D. Sound-Field Modelling

The paper “Sparse representation of a spatial sound field in a reverberant environment” proposes a general model for sparse sound field decomposition in a reverberant environment. Numerical and experimental results indicate that this decomposition enables an accurate sound field estimation, inside a region including sources. This study demonstrates that, in a reverberant environment, the reverberating component of the field cannot be treated as a residual, but must be explicitly treated. It was shown that this reverberating component can be modeled as the sum of a small number of plane waves, and a low-rank part.

The paper “Passive source depth discrimination in deep-water” addresses the problem of passive source depth discrimination in ocean acoustics using a horizontal line array. The scope is restricted to low-frequency sources, broadband signals, a deep-water environment, and distant sources at the endfire position. In this context, the environment acts as a dispersive waveguide, and classical source localization methods based on plane waves or any other simplistic wave model should not be used. Instead, a method based on the modal behavior driving the propagation is proposed. It uses the concept of the waveguide invariant, a scalar that summarizes the waveguide dispersion. In deep water, the waveguide invariant largely depends on source depth and thus represents an important input for source depth discrimination. An algorithm is proposed to compute the energy ratio in groups of modal interferences. The input data for the algorithm is a range-frequency intensity, as measured on a horizontal line array. The modal interference groups are defined based on their respective waveguide invariant values that in turn depend on the source depth. This paper proposes a source depth discrimination method that is expressed as a binary classification problem. As long as the sound speed profile features a surface thermocline, the algorithm does not require detailed knowledge about the environment and it allows the classification of sources under two hypotheses, above or under a user-chosen threshold depth.

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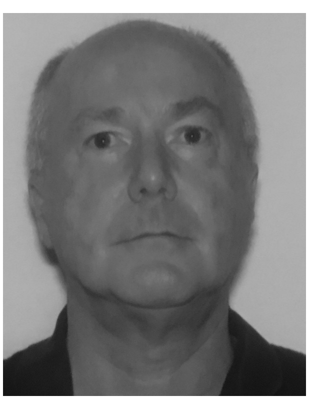

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