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GUEST EDITORS' NOTE
Special Issue on Dereverberation and Reverberation of Audio, Music, and Speech

Reverberation is the phenomenon where the physical and perceived structure of an audio signal is altered due to a multitude of acoustic reflections in a closed space such as a room, a hall, or a car interior. Reverberation is the key acoustic feature that makes one particular concert hall sound better than other ones. It is also used as a trademark by musicians and audio producers, who use reverberation to create a spacious and personal sound in music recordings. On the other hand, higher amounts of reverberation may degrade performance of automatic speech recognition or other audio classification systems, and may cause hearing aid users to encounter problems in localizing and understanding target speakers. It is also one of the main causes for decreased speech intelligibility or speech quality in public announcements systems, acoustic surveillance, or hands-free communication systems.

Since the early 1960s, when electroacoustics research started expanding, the problems of synthesizing as well as removing acoustic reverberation have been attracting the attention of researchers in different scientific communities. Five decades of room acoustics, electroacoustics, psychoacoustics, and signal processing research have paved the way for the results that are presented in this special issue. In the last decade, however, research in reverberation synthesis and dereverberation has been booming. Massive computational power has undoubtedly fostered the rise of reverberation and dereverberation research that we are currently witnessing.

When taking a glimpse of the table of contents of this special issue, one cannot deny that computationally based disciplines such as digital signal processing, machine learning, and computational acoustics have become fundamental to a large share of the ongoing research in dereverberation and reverberation synthesis. The special issue thus aims to cover the state of the art in reverberation and dereverberation research. Several papers in this issue present extensions of work presented at the AES 60th International Conference on Dereverberation and Reverberation of Audio, Music, and Speech held in Leuven, Belgium in February 2016.

Dereverberation removes the effect of reverberation from captured audio signals. A growing tendency can be observed toward multichannel techniques because they exploit both spectro-temporal and spatial characteristics of the received microphone signals. Dereverberation can be classified into reverberation suppression and reverberation cancellation, depending on whether or not the room impulse response or its inverse needs to be estimated. The paper "Multichannel Wiener Filters in Binaural and Bilateral Hearing Aids—Speech Intelligibility Improvement and Robustness to DoA Errors" by Adam Kuklasiński and Jesper Jensen presents a joint dereverberation and noise-suppression technique based on multichannel Wiener filtering. The benefit in terms of speech intelligibility and robustness against steering errors and self noise of binaural and bilateral approaches are being evaluated.

Reverberation cancellation can follow an indirect approach by designing equalization filters based on estimated room impulse responses or a direct approach by blindly inverting the room impulse responses. A popular direct technique is based on multichannel linear prediction (MCLP). The paper "A General Framework for Incorporating Time–Frequency Domain Sparsity in Multichannel Speech Dereverberation" by Ante Jukić et al. generalizes state-of-the-art MCLP-based speech dereverberation methods by incorporating time–frequency domain sparsity of the anechoic speech signal.

An alternative approach to mitigate the detrimental effects of reverberation is to modify the source signal that is being played back in such a way that the impact of reverberation is minimized. The paper "A Speech Preprocessing Method Based on Overlap-Masking Reduction to Increase Intelligibility in Reverberant Environments" by Julian Grosse and Steven van de Par presents two approaches that reduce the detrimental effects of reverberation on speech intelligibility by reducing the amount of overlap-masking due to room reverberation and/or enhancing speech onsets. The paper "Spatial Reverberation and Dereverberation Using an Acoustic Multiple-Input Multiple-Output System"

by Hai Morgenstern and Boaz Rafaely utilizes a spherical loudspeaker array to facilitate reproduction with a controlled reverberation at the listener position. The paper proposes a method for modifying the reverberant sound field in a room using beamforming to robustly change the temporal and spatial attributes of the sound field.

As already suggested in the latter paper, in many applications the control of reverberation goes beyond dereverberation. Indeed, a certain (controlled) amount of reverberation is often desirable, in particular in music applications. The paper "A Machine Learning Approach to Application of Intelligent Artificial Reverberation" by Emmanouil T. Chourdakis and Joshua D. Reiss adopts a machine learning framework to adaptively control reverberation when used as a digital audio effect in multitrack mixing.

Reverberation control and synthesis has recently also become a topic of interest in spatial audio research. The paper "Object-Based Reverberation for Spatial Audio" by Philip Coleman et al. fits in the recent framework of object-based spatial audio. Recent research and standardization efforts in this area have not given full consideration to how reverberation should be dealt with when capturing, processing, and reproducing spatial audio in an object-based format. To this end, the authors introduce, implement, and evaluate a parametric framework coined as the Reverberant Spatial Audio Object (RSAO).

Room modeling is a crucial aspect when it comes to reverberation control and synthesis. The paper "Wave-Based Room Acoustics Simulation: Explicit/Implicit Finite Volume Modeling of Viscothermal Losses and Frequency-dependent Boundaries" by Stefan Bilbao and Brian Hamilton deals with numerical wave-based models for room acoustics. The work considers the finite-volume time domain (FVTD) method and specifically addresses the modeling of viscothermal and relaxation effects.

Evaluation of reverberation and dereverberation schemes is another important research field that gained much interest within the last years.

The paper "Confidence Measures for Nonintrusive Estimation of Speech Clarity Index" by Pablo Peso Parada et al. presents a confidence measure for a nonintrusive estimation method of the clarity index (C50) of reverberant speech. The C50 estimation method uses a bidirectional long short-term memory network that maps multiple features from the reverberant speech signal into the target C50 value. The C50 estimations are found to be more accurate for low C50 values and especially degraded by the presence of babble noise.

The paper "User Preference on Artificial Reverberation and Delay Time Parameters" by Pedro D. Pestana et al. describes a series of two subjective tests to evaluate listeners' preferences with respect to two temporal processing parameters that are typical tools for mixing engineers, i.e., delay time and reverberation decay time. Both parameters are systematically varied and the quality of the different settings is assessed by an expert listener panel.

The paper "Perceptual Evaluation and Analysis of Reverberation in Multitrack Music Production" by Brecht De Man et al. presents the methodology and results of an extensive subjective listening experiment in which the preferred amount of reverberation in professional multitrack music production is evaluated. As relative reverb loudness and reverb decay time are found to be key perceptual attributes, the results of the study may directly be applied to predict optimal parameters for artificial reverberation in digital audio effects.

The paper "Instrumental and Perceptual Evaluation of Dereverberation Techniques Based on Robust Acoustic Multichannel Equalization" by Ina Kodrasi et al. evaluates the performance of several robust extensions of acoustic multichannel equalization by means of instrumental performance measures and subjective listening tests. Analysis of the correlation between the perceptual speech quality and the instrumental measures highlights the need to continue research on developing more accurate instrumental performance measures.

The engineering report "A Rapid Sensory Analysis Method for Perceptual Assessment of Automotive Audio" by Neofytos Kaplanis et al. proposes a new framework for perceptual studies in automotive audio. The paper deals with reproduction of the in-situ soundfield in lab environments, and assessment tools.

The guest editors would like to thank the reviewers, the editor Francis Rumsey, editor-in-chief Bozena Kostek, and copyeditor Mary Ellen Ilich for their valuable advice and help in realizing this special issue.

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