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Introduction to the Issue on Speech Processing for Natural Interaction With Intelligent Environments

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Introduction to the Issue on Speech Processing for Natural Interaction With Intelligent Environments

WITH the advances in microelectronics, communication technologies, and smart materials, our environments are transformed to be increasingly intelligent by the presence of robots, bio-implants, mobile devices, advanced in-car systems, smart house appliances, and other professional systems. As these environments are integral parts of our daily work and life, there is a great interest in a natural interaction with them. Also, such interaction may further enhance the perception of intelligence. "Interaction between man and machine should be based on the very same concepts as that between humans, i.e., it should be intuitive, multi-modal and based on emotion," as envisioned by Reeves and Nass (1996) in their famous book *The Media Equation*. Speech is the most natural means of interaction for human beings and it offers the unique advantage that it does not require carrying a device for using it since we have our "device" with us all the time.

Speech processing techniques are therefore developed to support either explicit interaction through message communications, or implicit interaction by providing valuable information about the physical ("who speaks when and where") as well as the emotional and social context of an interaction. However, intelligent environments are characterized with distant microphone(s), resource constraints, and large variations in acoustic condition, speaker, content, and context, all being significant challenges to speech processing.

To achieve the goal of natural interaction, a broad range of topics are to be addressed. We roughly group them into four clusters: 1) multi-microphone front-end processing and joint optimization with automatic speech recognition (ASR), 2) ASR in adverse acoustic environments and for low-resource and distributed computing infrastructure, 3) speaker diarization, affective computing and context awareness for interaction, and 4) cross-modal analysis of audio and visual data for smart spaces.

Much progress has been made, yet there is still a long way to go. More importantly, synergy across fields is demanded. This issue aims to bring together researchers and engineers to present latest developments in the given fields in one place and hopefully will stimulate cross-fertilization. The clusters of topics are well represented by the 12 papers in this issue and certainly, several papers fall into multiple clusters.

The first two papers deal with microphone array speech processing. Yoon *et al.* incorporate an acoustic model combination method with a hidden Markov model (HMM)-based mask estimation method for multichannel source separation. The acoustic model combination method is used to reduce the mismatch between training and testing conditions of an ASR after applying source separation. The paper by Yu and Hansen presents a time-frequency domain blind beamforming approach, without a prior knowledge of the array shape, for ex-

tracting the desired speech from a noisy musical environment, along with the evaluations conducted in both real in-vehicle and simulated noisy environments.

The succeeding five papers are concerned with the robustness to environmental distortions and reverberation in single-channel scenarios, and the constraints resulting from limited computational resources available in embedded devices. The paper by Tan and Lindberg addresses both issues by proposing a frame selection algorithm which emphasizes the reliable regions, thus improving the ASR robustness. As hardly any frame is selected for non-speech regions, the method may also serve as a robust voice activity detector (VAD) and a scalable source coding scheme for distributed speech recognition (DSR). Borgström and Alwan present a low complexity algorithm for determining improved speech presence probabilities using HMM-based inference to exploit the temporal correlation present in spectral speech data. The algorithm is applied to soft-decision enhancement and further to noise-robust ASR. The contribution by Ichikawa *et al.* covers robustness towards reverberation resulting from the large distance between the source and the microphone. The authors propose to compute dynamic features in the linear-logarithmic hybrid domain for distant-talking ASR in reverberant environments. The paper by Astudillo *et al.* presents an uncertainty propagation approach for the advanced front-end of the ETSI DSR standards. The advantage of the method lies in the fact that the uncertainty is determined in the domain where most speech enhancement methods operate by using self similarity measures. Fukuda *et al.* present a statistical-model based noise-robust VAD algorithm using long-term temporal information and harmonic-structure based features in speech. The algorithm works well both as a VAD and a robust preprocessing method for ASR.

The three papers in the third cluster address the problem of context acquisition from audio or audio-visual signals. Schmalenstroer and Haeb-Umbach present a low-latency diarization system for smart homes that conducts joint speaker segmentation, localization, and identification, supported by face identification. Speaker positions obtained through a blind beamforming method are combined with speaker change information to improve speaker identification. Stafylakis *et al.* derive a new approach to the Bayesian information criterion (BIC), which combines the strengths of the global and local BIC, and apply the resulting segmental BIC for speaker diarization. The contribution by Wöllmer *et al.* incorporates long short-term memory and dynamic Bayesian networks for incremental recognition of the user's emotional space. The automatic estimation of human affect from the speech signal makes virtual agents more natural and human-like.

The final two papers explore cross-modal analysis. Shivappa *et al.* develop a scheme to fuse audio and visual cues to track multiple persons in an intelligent meeting room equipped with

multiple cameras and microphone arrays. The scheme performs comparably to particle filters and is robust to calibration errors. Tracking is a key step towards natural interaction. The paper by Naqvi *et al.* presents a multimodal approach to blind source separation (BSS) of moving sources. A full 3-D visual tracker based on particle filtering is implemented to provide velocity and direction of sources; a beamforming algorithm is used when sources are moving and a BSS algorithm is performed when sources are stationary.

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