Design, implementation and evaluation of a robust multi-microphone noise reduction algorithm for hearing instruments

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Noise reduction algorithms in hearing aids and cochlear implants are crucial for hearing impaired persons to improve speech intelligibility in background noise. Multi-microphone systems exploit spatial in addition to spectro-temporal information of the desired and the noise signals and are hence preferred to single-microphone systems. For small-sized microphone arrays such as typically encountered in hearing instruments, multi-microphone noise reduction however goes together with an increased sensitivity to errors in the assumed signal model such as microphone mismatch (gain, phase, position), reverberation, speech detection errors, etc.

In this presentation we discuss the algorithm design, the low-cost implementation and the real-time evaluation of a robust generalised multi-microphone noise reduction scheme, called the *Spatially Pre-processed Speech Distortion Weighted Multi-channel Wiener Filter (SP-SDW-MWF)* [1]. The structure of this scheme strongly resembles the widely used Generalised Sidelobe Canceller (GSC) [2,3,4], and consists of two parts: a robust fixed spatial pre-processor, generating speech and noise reference signals, and a robust adaptive Multi-channel Wiener Filter (MWF), reducing the residual noise in the speech reference. This generalised scheme encompasses both the GSC and the MWF [5,6] as extreme cases and allows for attractive in-between solutions such as the Speech Distortion Regularised GSC (SDR-GSC).

In the standard GSC, both the fixed spatial pre-processor and the adaptive stage strongly rely on a-priori assumptions (e.g. about the microphone characteristics). When these assumptions are not satisfied, both stages give rise to undesired speech distortion and to a reduced noise reduction performance. Robust solutions have been proposed e.g. by calibrating the used microphone array [7] and by using a Quadratic Inequality Constraint (QIC) [8]. In the SP-SDW-MWF, *robustness against signal model errors* is achieved by incorporating statistical information about the microphone characteristics (gain, phase, position) into the design procedure of the fixed spatial pre-processor [9] and by taking speech distortion explicitly into account in the optimisation criterion of the MWF.

For the *implementation* of the adaptive MWF, we discuss an efficient stochastic gradient algorithm in the frequency-domain, whose computational complexity and memory usage is comparable to the NLMS-based Scaled Projection Algorithm for implementing the QIC-GSC [8].

Experiments have been performed using a 3-mic BTE hearing aid, mounted on a dummy head in an office room, where the desired speech source is positioned in front of the head and 5 babble noise sources are positioned at different angles. Simulation results show that the proposed scheme achieves a better noise reduction performance for a given maximum speech distortion level, compared to the widely studied QIC-GSC. The subjective speech enhancement and robustness performance of this generalised multi-microphone noise reduction scheme is currently being evaluated with normal hearing and hearing impaired persons.

[Simon Doclo is a postdoctoral researcher funded by KULeuven-BOF. This work is supported in part by F.W.O. Project G.0233.01, I.W.T. Project 020540, I.W.T. Project 020476, Concerted

Research Action GOA-MEFISTO-666, Interuniversity Attraction Pole IAP V/22, and is partially sponsored by Cochlear.]

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