Noise reduction approaches for improved speech perception

Jan Wouters[1], Simon Doclo[2], Thomas Klasen[1,2], Jean-Baptiste Maj[1,2], Marc Moonen[2], Lies Royackers[1], Ann Spriet[1,2], Tim Van den Bogaert[1,2]

[1] Katholieke Universiteit Leuven, Lab. Exp. ORL, Leuven, Belgium
[2] Katholieke Universiteit Leuven, Dept. of Electrical Engineering, Leuven, Belgium

Speech understanding in noisy listening conditions is a big problem for hearing aid users. Several noise reduction signal processing schemes have been investigated for application in hearing aids, but few implemented in commercial devices. Some single microphone strategies have been introduced yielding limited benefit. Configurations of 2 microphones are commonly implemented and demonstrate benefit in some sound scenes. Multi-Microphone systems can take advantage of the spatial information of the desired and the jammer sound sources in addition to spectro-temporal information. At present, adaptive directional microphone systems are available in commercial hearing aids, allowing to adapt to changing jammer directions and to track moving noise sources. The small sizes of these arrays in hearing aids as well as specific signal processing aspects, however, introduce an increased sensitivity to errors in the assumed signal model (microphone mismatch, speech distortion, voice activity detection, room acoustics, …)

In our research we have been focussing on 2 and 3 microphone configurations for behind-the-ear (BTE) devices in combination with robust algorithms economic enough to be implemented in signal processors of hearing aids and cochlear implants. The degree and the robustness of the benefit in realistic, difficult listening environments is of major importance. The performance measures used are speech intelligibility weighted speech-to-noise ratios and distortion on the physical side, and speech reception thresholds on the perceptual side.

In general, the multi-microphone noise reduction approaches studied consist of a fixed spatial pre-processor that transforms the microphone signals to speech and noise reference signals, followed by an adaptive processing stage. In the two-stage adaptive beamformer or generalized sidelobe canceller (GSC) the second stage is an unconstrained adaptive noise canceller that uses the output of the first stage, and is only allowed to adapt during non-speech periods [1]. The performance of this adaptive noise reduction strategy has been compared with an adaptive directional microphone, state-of-the-art in modern commercial digital hearing aids. Improvements in speech reception threshold between 7 and 12 dB have been obtained for the GSC in different realistic multi-source scenes.
These developments have been extensively evaluated with normal hearing, hearing impaired and cochlear implant subjects [2,3]. Results will be given.

Since the fixed and the adaptive stage of the GSC-technique or two-stage adaptive beamformer rely on a-priori assumptions, this may give rise to undesired speech distortion and a reduced noise cancellation. Recently, we developed optimal filtering techniques which are more robust against signal model errors. A first technique is based on singular value decomposition (SVD). Although this approach has a high computational complexity, we were able to show performances better than GSC in difficult listening conditions (speech coming from some non-zero angle in a background of 3 different noise sources in a reverberant room). Secondly, a novel multi-microphone optimal filtering noise reduction scheme (spatially pre-processed speech distortion weighted multi-channel wiener filter) has been developed. The latter technique has an improved robustness of the fixed stage by incorporating statistical knowledge about the microphone characteristics in the design of the adaptive filter, and is computationally less expensive [4,5,6]. Experimental results show increased benefit.

The former monaural adaptive noise reduction techniques have been extended to bilateral systems. First results will be discussed.

[1] J. Vanden Berghe and J. Wouters (1998), "An adaptive noise canceller for hearing aids using two nearby microphones", *Journal of the Acoustical Society of America*, 103, 3621-3626
[2] J. Wouters and J. Vanden Berghe (2001), "Speech recognition in noise for cochlear implantees with a two-microphone monaural adaptive noise reduction system", *Ear & Hearing*, 22, 420-430
[3] J.-B. Maj, J. Wouters, M. Moonen (2004), "Noise reduction results of an adaptive filtering technique for dual-microphone behind-the-ear hearing aids", *Ear & Hearing*, in press
[4] S. Doclo, M. Moonen (2003), "Design of broadband beamformers robust against gain and phase errors in the microphone array characteristics," *IEEE Trans. Signal Processing*, 51, 2511-2526
[5] A. Spriet, M. Moonen, and J. Wouters (2004), "Robustness analysis of multi-channel Winer filtering and generalized sidelobe cancellation for multi-microphone noise reduction in hearing aid applications", *IEEE Trans. Speech and Audio Processing*, in press
[6] A. Spriet, M. Moonen, and J. Wouters (2004), "Stochastic gradient based implementation of spatially preprocessed multi-chanel Wiener filter for noise reduction in hearing aids", *IEEE Trans. Signal Processing*, in press