Multi-Microphone Speech Enhancement Using LCMV Beamformers

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Microphone array algorithms emerged in the early 1990s as viable solutions to speech processing problems. Although more than 20 years have elapsed, the adaptation of beamforming methods to speech processing remains an open issue. These difficulties may be attributed to the characteristics of the wide-band and nonstationary speech signal and to the very long, typically time-varying, room impulse responses (RIRs) relating moving speakers and microphones in acoustic enclosures.

In this talk, we will focus on spatial processors ("beamformers") based on the linearly constrained minimum variance (LCMV) criterion, and its special case, the minimum variance distortionless (MVDR) beamformer. The implementation of the LCMV beamformer in the short-time Fourier transform (STFT) domain and its structuring as a generalized sidelobe canceller (GSC) facilitate the application of the presented algorithms to speech signals in real acoustic environments.

We will show how the powerful LCMV criterion can be applied to various related problems, e.g. speech enhancement and desired speakers' extraction in multiple competing speaker environment. Special attention will be given to blind estimation techniques of the GSC components and to the efficient design of its various blocks. We will also elaborate on the relative transfer function (RTF) and its importance in speech processing.

The last part of the talk will be dedicated to the exploration of novel *distributed microphone array* architectures. A distributed microphone array is comprised of multiple sub-arrays (nodes), each of which consists of several microphones, a signal processing unit and a wireless communication module. New and fascinating algorithmic challenges arise by these new distributed structures. In this part of the talk we will briefly explore performance bounds of such distributed microphone arrays and introduce novel algorithms for solving the optimality criteria under the limited communication bandwidth constraints between nodes.

The presentation will be accompanied by processed audio files demonstrating the algorithms' performance.

Sharon Gannot received his B.Sc. degree (summa cum laude) from the Technion – Israel Institute of Technology, Haifa, Israel in 1986 and the M.Sc. (cum laude) and Ph.D. degrees from Tel-Aviv University, Israel in 1995 and 2000 respectively, all in electrical engineering. In 2001 he held a post-doctoral position at the department of Electrical Engineering (ESAT-SISTA) at K.U.Leuven, Belgium. From 2002 to 2003 he held a research and teaching position at the Faculty of Electrical Engineering, Technion-Israel Institute of Technology, Haifa, Israel. Currently, he is an Associate Professor at the School of Engineering, Bar-Ilan University, Israel, where he is heading the Speech and Signal Processing laboratory. Prof. Gannot is the recipient of Bar-Ilan University outstanding lecturer award for 2010.

Prof. Gannot is currently an Associate Editor of IEEE Transactions on Speech, Audio and Language processing. He served as an Associate Editor of the EURASIP Journal of Advances in signal Processing between 2003-2012, and as an Editor of two special issues on Multi-microphone Speech Processing of the same journal. He also served as a guest editor of ELSEVIER Speech Communication journal and a reviewer of many IEEE journals and conferences. Prof. Gannot is a member of the Audio and Acoustic Signal Processing (AASP) technical committee of the IEEE since Jan., 2010. He is also a member of the Technical and Steering committee of the International Workshop on Acoustic Signal Enhancement (IWAENC) since 2005 and the general co-chair of IWAENC held at Tel-Aviv, Israel in August 2010. Prof. Gannot will serve as the general co-chair of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA) in 2013.