EURASIP Journal on Applied Signal Processing 2003:11, 1053–1055

Editorial

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The commercial application of advanced acoustic communication systems has become feasible in recent years due to the vast increase of available computational power. In new and future acoustic communication systems, it is expected that people will want to create virtual acoustic communication links that give conversation partners the impression of being in the same acoustic environment. Besides providing quality and robustness, these future acoustic communication systems should exploit the growing computer power to design more flexible systems in which an acoustic interface is built, which on the one hand, acquires speech and sound perfectly, and yet allows people to move freely around without wearing or holding a microphone. On the other hand, sound that is reproduced at the human's ears should sound such that, ideally, the local acoustic environment is masked and remote or virtual environments can be created in the human's perception. The amount of signal processing involved in future acoustic communication systems grows exponentially due to the demand for more and more advanced systems. For this reason, there is an increased interest in more sophisticated algorithms that can deal with multiple source signals, multiple microphones, and multiple loudspeakers running in real time on one or more digital signal processing cores.

The aim of this special issue is to highlight innovative research in signal processing for acoustic communication systems, thus paving the way for future developments in the field. This was the philosophy behind the decision to prepare a special issue of the EURASIP Journal on Applied Signal Processing devoted to this area. Out of sixteen submitted papers, nine have been finally selected by the Guest Editors, taking into account the evaluation via standard international peer-review process. The selected papers cover a wide range of acoustic communication systems ranging from double-talk detectors to blind signal separation. Double-talk detectors are vital to the operation and performance of acoustic echo cancellers. There is a need for an extension of double-talk detection to multidelay block frequency domain adaptive filters, which have been introduced to make a proper selection between processing delay and complexity. In their paper, Benesty and Gänsler define an extended cross-correlation vector in the frequency domain which fits very well with an echo canceller based on the multidelay block frequency domain structure. Since background noise is able to cause low speech intelligibility and hence low overall system performance, the next two papers deal with speech enhancement systems. Lotter, Beniem, and Vary introduce in their contribution two short-time spectral amplitude estimators for speech enhancement with multiple microphones, while Guérin, Le Bouquin-Jeannès, and Faucon present in their paper a two-microphone speech enhance system, dedicated to remove noise in a hands-free car kit.

In their contribution, Gannot and Moonen present a novel approach for multimicrophone speech dereverberation, which is an important subject when two people are far apart in a highly reverberant room since they cannot have a conversation easily. This reverberation is due to strong echoes and decreases the intelligibility of recorded speech. An essential requirement of acoustic communication systems using array processing techniques is the ability to locate and track audio sources. One of the techniques is based on time-delay estimation. Doclo and Moonen present two adaptive algorithms for robust time-delay estimation in acoustic environments where a large amount of additive background noise and reverberation is present. A beamformer is a processor used in conjunction with an array of sensors to provide a versatile form of spatial filtering. The objective of a beamformer is to estimate the signal arriving from a desired direction in the presence of noise and interfering signals. Cohen, Gannot, and Berdugo present a novel approach for real-time multichannel speech enhancement in environments of nonstationary noise and time varying acoustic transfer functions. The proposed system integrates adaptive beamforming, identification of acoustic transfer functions, soft signal detection, and multichannel postfiltering. The next two papers deal with blind signal separation (BSS) techniques. BSS has been proposed to recover source signals from their measurements (the microphone signals). These techniques are termed blind as the acoustic transfer functions from the sources to the microphones are unknown and there are no reference signals to compare the recovered source signal against. Despite of its high complexity and low convergence properties, independent component analysis is a concept that is frequently used for BSS. Saruwatari, Kurita, Takeda, Itakura, Nishikawa, and Shikano describe a new method of BSS combining subband independent component analysis and beamforming which can solve the slow convergence properties. Since many conventional BSS algorithms can hardly be implemented in real time due to high-computational complexity, this is the topic of the paper by Yin, Sommen, and He. In their contribution, they aim at reducing the computational complexity by proposing a new mixing model for multispeaker-multimicrophone environment. Araki, Makino, Hinamoto, Mukai, Nishikawa, and Saruwatari give an interpretation of BSS from a physical point of view by showing that BSS is equivalent to two sets of adaptive beamformers.

In the coming years, it is expected that acoustic communication systems will become even more important both in research and industry. We hope that this special issue will further stimulate work on signal processing in this area.

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Kees Janse received his B.S. degree in electrical engineering in 1976 from the Polytechnical School in Vlissingen, the Netherlands. He joined the acoustics group of Philips Research Laboratories in 1978, where his work focused on measuring techniques and signal analysis, for example, evaluating loudspeakers with the aid of the Wigner distribution. In 1984, he joined the project center of Philips Research, where his work focused on the design of geographic data bases for car navigation systems. In 1988, he joined the radio data transmission group (and later on the digital signal processing group) of Philips Research and since then he has been working in the field of hands-free signal processing, including acoustic echo cancellation, acoustic feedback suppression, acoustic noise reduction, and dereverberation of speech signals. He has written and presented papers and holds a number of patents in his field.

Walter Kellermann is a Professor of communications at the Chair of Multimedia Communications and Signal Processing of the University of Erlangen-Nuremberg, Germany. He received the Dipl.-Ing. (univ.) degree in electrical engineering from the University of Erlangen-Nuremberg in 1983, and the Dr.-Ing. degree from the Technical University Darmstadt, Germany, in 1988. From 1989 to 1990, he was a Postdoctoral Member of the technical staff at AT&T Bell Laboratories, Murray Hill, NJ. From 1990 to 1993, he was with Philips Kommunikations Industrie, Nuremberg, Germany. From 1993 to 1999, he was a Professor at the Fachhochschule Regensburg, and in 1997, he became Director of the Institute of Applied Research of the Fachhochschule Regensburg. In 1999, he cofounded DSP Solutions, a consulting firm in digital signal processing, and he joined the University of Erlangen-Nuremberg as a Professor and Head of the Audio Research Laboratory. Dr. Kellermann authored and coauthored five book chapters and more than 35 papers in journals and conference proceedings. He served as a Guest Editor for various journals and presently serves as an Associate Editor of IEEE Transactions on Speech and Audio Processing. His current research interests include speech signal processing, array signal processing, and adaptive filtering and its applications to acoustic human/machine interfaces.

Marc Moonen received the Electrical Engineering degree and the Ph.D. degree in applied sciences from the Katholieke Universiteit Leuven, Leuven, Belgium, in 1986 and 1990, respectively. Since 2000, he has been an Associate Professor at the Electrical Engineering Department of Katholieke Universiteit Leuven, where he is currently heading a research team of sixteen Ph.D. candidates and postdocs, working in the area of signal processing for digital communications, wireless communications, DSL, and audio signal processing. He received the 1994 KULeuven Research Council Award, the 1997 Alcatel Bell (Belgium) Award (with Piet Vandaele), and was a 1997 "Laureate of the Belgium Royal Academy of Science." He was the Chairman of the IEEE Benelux Signal Processing Chapter (1998–2002), and is currently a EURASIP AdCom Member (European Association for Signal, Speech, and Image Processing, 2000). He is Editor-in-Chief for the EURASIP Journal on Applied Signal Processing (2003), and a member of the editorial board of Integration, the VLSI Journal, IEEE Transactions on Circuits and Systems II, and IEEE Signal Processing Magazine.

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