ELSEVIER

Speech Communication 39 (2003) 1–3

SPEECH COMMUNICATION

www.elsevier.com/locate/specom

Editorial

Special Issue on Speech Processing for Hearing Aids

With the widespread usage of digital hearing aids and increased demands placed on their performance, advanced speech processing techniques are playing a more important role in modern hearing aid systems. Speech processing for hearing aids encompasses a range of topics including signal acquisition, transfer, amplification, transmission, measurement, filtering, parameter estimation, separation, detection, enhancement and classification. Speech processing techniques for hearing aids are also highly applicable to and benefited by the characterization and modeling of hearing-impaired auditory systems. We believe that speech processing in hearing aids can be primarily divided into three application areas. The first area is devoted to using advanced speech signal processing techniques to characterize and compensate for various hearing impairments. An example of such a technique is the use of perceptual-model-based multi-band compression and amplification to compensate for the loss in loudness, and the use of spectral contrast enhancement to compensate for the loss of frequency selectivity. Because hearing-impaired individuals experience more difficulty understanding speech in noise than normal-hearing people, the second area of speech processing research in hearing aids is effective target signal enhancement and noise reduction. The third area focuses on the real-world use of hearing aids; addressing issues such as flexibility, convenience, feedback cancellation, and artifact reduction. Because of the limitations imposed by the hardware requirements, computational speed, power supply and other practical factors, the development and implementation of speech processing techniques for hearing aids has been a challenging and active area of research over the past decade.

The intention of this special issue is to stimulate and guide the development of new and improved hearing aids by providing a high quality forum for scientists and engineers interested in hearing-aid design to learn about recent developments. In addition, it is hoped that the issue will attract a broad audience in the speech processing community. The call for papers for this issue resulted in a large quantity of excellent submissions from around the world. After two rounds of careful reviews from approximately forty experts in related fields, eleven papers, organized into four groups, were selected to be included in this special issue.

Three papers in the first group focus on speech processing algorithms based on characterizations of auditory systems and comparisons between hearing-impaired people and normal-hearing people. The paper by Lisa G. Huettel and Leslie M. Collins investigates the differences between normal and impaired auditory processing on a frequency discrimination task by analyzing the responses of a computational auditory model using signal detection theory. Hearing impairments that were simulated can be characterized by a threshold shift, damage to the outer hair cells, and impaired neural synchrony. Two kinds of detectors are designed and implemented in the paper. Although a simple hearing aid is simulated in this paper for further analysis and evaluation, the proposed approach could be used to evaluate the design of more complex hearing aids. The second paper in this group is by Thomas Fillon and Jacques Prado. In this paper, the authors present an implementation of the EMSR (Ephraim and Malah Suppression Rule) on

0167-6393/02/$ - see front matter
PII: S0167-6393(02)00054-7

an ERB (Equivalent Rectangular Bandwidth)-scale analysis and make a comparative study for noise reduction between the classical implementation and ERB-based implementation. It is well known that auditory filters of hearing-impaired people are generally broader than normal. Spectral contrast enhancement helps compensate for this reduced frequency selectivity. The paper by Jun Yang, Fa-Long Luo and Arye Nehorai compares three spectral contrast enhancement algorithms emphasizing practical considerations related to the implementation complexity of each algorithm.

The three papers in the second group deal with the separation of signal sources and related parameter estimation using adaptive and neural signal processing techniques. The paper by Michael Kleinschmidt and Volker Hohmann presents a new approach for estimating the long-term speech-to-noise ratio in individual frequency bands on the basis of automatic speech recognition techniques. The proposed approach uses a model of auditory perception as a front end, physiologically and psychoacoustically motivated sigma–pi cells as secondary features and a linear or nonlinear neural network as a classifier. The paper by Scott C. Douglas and Xiaoan Sun addresses the blind separation of speech, that is, the cocktail party problem. Two algorithms for separating mixtures of multiple speech signals on the basis of density-based multi-channel blind deconvolution using natural gradient adaptation are proposed in this paper. Theoretical analyses and simulation results have demonstrated the effectiveness of the proposed algorithms in separating real-world speech signal mixtures. Although blind source separation is highly applicable to noise reduction, practical considerations in implementing digital hearing aids require algorithms with great computational efficiency and adaptation speed. To address these issues, the paper by Jörn Anemueller and Birger Kollmeier proposes a constrained frequency domain approach for adaptive separation of acoustic sources under anechoic conditions. Implementation details are given and performance of the proposed approach is evaluated experimentally and via simulation. This work can serve as a starting point for the use of blind source separation techniques in hearing aids under real reverberant acoustic conditions.

In the third group, three papers discuss the use of multi-microphone-based directional processing techniques in hearing aids. Using spatial domain differences between the target signal and noise, a system with directionality can achieve noise reduction and speech enhancement in various environments, particularly in speech-like noisy environments. Furthermore, the combination of directional processing with binaural processing can provide both the spatial-filtering benefits of the microphone array and the natural benefits of binaural listening to sound localization ability and speech intelligibility. The paper by Douglas R. Campbell and Paul W. Shields presents a two-microphone sub-band adaptive processing scheme. Intelligibility testing is described and results of an assessment of the proposed scheme using ten normal-hearing listeners and signals from a real reverberant environment are presented. The results support the hypothesis that speech processing using the proposed scheme can provide a statistically and practically significant improvement in speech intelligibility. In the paper by Thomas Wittkop and Volker Hohmann, different binaural signal processing strategies for noise reduction are derived and assessed. Moreover, a measure for the general diffusiveness or coherence of the sound field is employed to classify the situation and to switch off particular processing strategies if necessary. Subject testing suggests a higher quality to the processed signal than the unprocessed signal under the diffuse cafeteria noise condition. The paper by Bernard Widrow and Fa-Long Luo outlines principles and practices related to the application of microphone array processing techniques to hearing aids including algorithms, real-time implementation, configuration, processing mode, geometry shape, combination with binaural processing, directivity pattern, frequency responses, compensation for mismatch and misplacement of microphones. A real microphone array device is also presented in this paper to demonstrate that an appropriately configured and connected array can help hearing aids by improving the signal-to-noise ratio, reducing the effects of reverberation and reducing feedback.

Two papers on adaptive feedback cancellation are contained in the final group. In the paper by Hsiang-Feng Chi, Shawn X. Gao, Sigfrid D. Soli and Abeer Alwan, the authors propose a band-limited adaptive feedback cancellation algorithm using normalized filtered-X LMS techniques. Theoretical analysis and computer simulations illustrate the advantages of the proposed approach over wideband feedback cancellation approaches. In addition, the first section of this paper can serve as a good introduction to feedback problem and related cancellation methods used in hearing aids. Because the feedback paths in hearing aids are always time varying, robust and adaptive feedback cancellation schemes are highly desirable. To achieve this goal, the paper by Boaz Rafaely, Ngwa A. Shusina and Joanna L. Hayes presents a scheme that incorporates an adaptive feedback cancellation filter integrated with a robust compensation filter. A limit on the gain of the compensation filter that guarantees robust stability is derived and an adaptive algorithm with improved robustness is proposed in this paper. The effectiveness of this proposed scheme is demonstrated by a simulation example with time varying feedback paths.

As we conclude this overview, we would like to thank Professor Abeer Alwan, Editor-in-Chief of *Speech Communication*, for her kind support and efficient coordination during all phases of the work to make this special issue possible. We would also like to express our sincere gratitude to the reviewers listed below for their timely and insightful comments on the submitted manuscripts, with special thanks to Professor Leslie Collins who also offered useful suggestions in writing this overview.

Fa-Long Luo
Bernard Widrow
Chaslav Pavlovic

List of Reviewers

Futoshi Asano
Mohammed Bahoura
Louis D. Braida
David Burshtein
Douglas R. Campbell
Olivier Cappe
Shi-Huang Chen
Hsiang-Feng Chi
Leslie M. Collins
Brent Edwards
Deniz Erdogmus
Simone Fiori
Nedelko Grbić
Johan Hellgren
Volker Hohmann
Gitte Keidser
Jim Kates
Nam Soo Kim
Martin Kompis
Rafal Krolikowski
Thomas Lotter
Rainer Martin
Heinz Mathis
Peter Nordqvist
Boaz Rafaely
Marion Schabert
Fernando Mira da Silva
Pavel Sovka
Christopher J. Struck
Kari Torkkola
Peter Vary
Marcel Vlaming
Jun Yang
Ming Zhang
Andreas Ziehe