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Editorial

Simon Doclo

Department of Electrical Engineering (ESAT-SCD), Katholieke Universiteit Leuven, Kasteelpark Arenberg 10, 3001 Leuven, Belgium
Email: simon.doclo@esat.kuleuven.be

Søren Holdt Jensen

Department of Communication Technology, Institute of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7A, 9220 Aalborg, Denmark
Email: shj@kom.aau.dk

Philippe A. Pango

Gennum Corporation, P.O. Box 489, Station A, Burlington, ON, Canada L7R 3Y3
Email: philip_p@gennum.com

Søren K. Riis

Oticon A/S, Kongebakken 9, 2765 Smoerum, Denmark
Email: skr@oticon.dk

Jan Wouters

Exp. ORL, Department of Neurosciences, Katholieke Universiteit Leuven, O&N, Herestraat 49, bus 721, 3000 Leuven, Belgium
Email: jan.wouters@med.kuleuven.be

Digital signal processing for hearing aids was initiated as a topic of research in the mid-late 1980s. However, it was not until 1995 that the technology matured to a level where small-size and low-power consumption allowed the market introduction of hearing aids with full digital signal processing capabilities.

Today, 83% of hearing aids sold worldwide are digital. Advanced packaging technologies enable hearing aids that fit completely in the ear canal, and the introduction of truly programmable platforms has allowed the development of advanced digital signal processing algorithms that provide the hearing-impaired user a natural sound picture with increased speech intelligibility and comfort.

Modern cochlear implant systems are capable of far more advanced processing than before. Whereas cochlear implants adopted digital technology prior to hearing aids, it is only until very recently that they have integrated some specialized algorithms such as adaptive noise reduction. A cochlear implant needs, in addition, a speech processing strategy that converts the acoustical signal into electrical signals to be applied to the electrodes placed in the cochlea. The design of such sound processing strategies poses additional signal processing challenges, but at the same time builds on knowledge acquired through physiological and psychophysical studies.

This special issue on DSP in Hearing Aids and Cochlear Implants gathers 15 articles. It reflects aspects of the multiple disciplines necessary for the treatment of hearing impairment. Indeed, the included papers address a variety of methods and algorithms, all related to the research in signal processing for hearing aids and cochlear implants. It is clear from the submissions that through the years, the inclusion of perception in signal processing and the development of psychoacoustically motivated signal processing algorithms are becoming more and more relevant and important, as in other domains of audio processing.

The papers in this issue are organized according to the topic of research, since some of these contributions are applicable to both hearing aids and cochlear implants. The most frequent themes are speech intelligibility, speech enhancement and noise suppression (6 papers), and new signal processing developments in filterbanks and compression algorithms implementation (4 papers). Furthermore the issue presents 5 contributions from various research domains such as auditory scene analysis for classification of input sounds, a new cochlear implant processing strategy, a versatile research platform for cochlear implant research, a new wireless link between the external and internal cochlear implant parts, and blind source separation.

"Signal processing in high-end hearing aids: state of the art, challenges, and future trends" (V. Hamacher et al.) provides a discussion of signal processing in modern hearing aids. The authors distinguish between two types of algorithms: those that aim at compensating the hearing loss and improving hearing ability and those that aim at compensating side effects of hearing aids. The former category comprises, for example, amplification strategies, noise reduction, and directional (beamformer) systems, whereas the latter comprises, for example, acoustic feedback cancellation and automatic control of the signal processing in the hearing aid. For each signal processing component a discussion of future trends is given.

In "An improved array steering vector estimation method and its application in speech enhancement" (Z. L. Yu and M. H. Er), a multimicrophone speech enhancement method is presented. This method is an extension of the transfer function generalized sidelobe canceller (TF-GSC), developed by Gannot et al., where the acoustic transfer functions between the desired speech source and the microphone array are estimated and used in the design of the fixed beamformer and the blocking matrix of the GSC. Instead of using one of the microphone signals as the reference signal, this paper proposes to use an optimal combination of all available microphone signals as the reference signal. Hence, by increasing the signal-to-noise ratio of the reference signal, the accuracy of the estimated acoustic transfer functions is improved.

In "An auditory-masking-threshold-based noise suppression algorithm GMMSE-AMT[ERB] for listeners with sensorineural hearing loss" (A. Natarajan et al.), a new noise suppression algorithm for hearing aid applications is described. The algorithm is based on an approach that uses the auditory masked threshold (AMT) in conjunction with a modified generalized minimum mean square error estimator (GMMSE) to adjust enhancement parameters based on the masked threshold of the noise across the frequency spectrum. The new algorithm also establishes a framework for customization of the AMT estimation to individual subjects with hearing loss. The representation of cochlear frequency resolution is achieved in terms of auditory filter equivalent rectangular bandwidths (ERBs). The estimation of the AMT and spreading functions for masking is implemented in two ways: with normal auditory thresholds and normal auditory filter bandwidths and with the elevated thresholds and broader auditory filters characteristic of cochlear hearing loss.

In "Speech enhancement with natural sounding residual noise based on connected time-frequency speech presence regions" (K. V. Sørensen and S. V. Andersen), a low-complexity single-microphone speech enhancement method is presented. To achieve natural sounding attenuated background noise, this method uses a generalized spectral subtraction attenuation rule in time-frequency regions where speech is present and a constant attenuation rule in regions where speech is absent. The proposed speech detection technique provides smoothly connected time-frequency regions in a perceptually functional way and enables a new bias compensation method for minimum-statistics-based noise estimation. Listening tests show that the proposed method produces a higher mean opinion score than minimum mean-square error log-spectral amplitude (MMSE-LSA) speech enhancement methods.

The paper "A block-based linear MMSE noise reduction with a high temporal resolution modeling of the speech excitation" (C. Li and S. V. Andersen) proposes a method for single-channel speech enhancement. The method is based on an all-pole model of speech production and estimates the clean speech spectral envelope and LPC residual separately by a frequency-domain version of the linear minimum mean-square error estimator. Objective performance measures show that the proposed method compares advantageously to known methods for speech signals in white noise at an SNR of 10 dB.

In "The effects of noise on speech recognition in cochlear implant subjects: predictions and analysis using acoustic models" (J. J. Remus and L. M. Collins), the reduction of speech recognition performance in the presence of noise is discussed for patients with cochlear implants. In the paper, listening tests using normal-hearing subjects are conducted on noisy consonant and vowels processed by two different acoustic models of cochlear implant processors. An extensive analysis of the results is given along with statistical models for predicting patterns of vowel and consonant confusion based on the processed speech tokens in the listening test.

"Sound classification in hearing aids inspired by auditory scene analysis" (M. Büchler et al.) presents a systematic evaluation of classifiers and features for speech and music classification in a hearing-aid application. The considered features comprise, for example, amplitude modulation and harmonicity, and a comparison between performance of simple classifiers and complex classifiers like hidden Markov models is given. It is illustrated that good performance can be obtained even with simple classifiers in many situations, but also that most classifiers yield poor performance for speech in noise.

In "Multichannel dynamic-range compression using digital frequency warping" (J. M. Kates and K. H. Arehart), a novel multichannel dynamic-range compressor system using digital frequency warping is described. The frequency-warped filter is realized by replacing the filter unit delays with all-pass filters, and the warped compressor is shown to have substantially reduced group delay in comparison with a conventional design having comparable frequency resolution.

In "A low-power two-digit multi-dimensional logarithmic number system filterbank architecture for a digital hearing aid" (R. Muscedere et al.), the implementation of a filterbank for digital hearing aids using a multidimensional logarithmic number system (MDLNS) is addressed. By exploiting various properties of the MDLNS, an improved design for a two-digit 2D MDLNS filterbank implementation reducing the power and area by over 2 times from the original design is presented.

"An intrinsically digital amplification scheme for hearing aids" (P. Blamey et al.) suggests a new intrinsically digital amplification scheme for hearing aids. Contrary to some existing amplification strategies like linear amplification and compression, the suggested method is not a digital "reimplementation" of an established technique from an analog hearing aid. The new amplification strategy is based on statistical analysis of the signal and aims at maximizing the dynamic range in each frequency band in a multiband hearing aid. The method has been implemented on a commercially available DSP. A comparison to existing schemes indicates improved audibility of sound in narrow frequency bands.

In "Effects of instantaneous multiband dynamic compression on speech intelligibility" (T. Herzke and V. Hohmann), instantaneous multiband dynamic compression based on an auditory filterbank is investigated. Instantaneous envelope compression is performed in each frequency band of a gammatone filterbank, which provides a combination of time and frequency resolution comparable to the normal healthy cochlea. The gain characteristics used for dynamic compression are deduced from categorical loudness scaling. By means of speech intelligibility tests, the instantaneous dynamic compression scheme is compared with a linear amplification scheme, which uses the same filterbank for frequency analysis, but employs constant gain factors that restored the sound level for medium perceived loudness in each frequency band.

In "A psychoacoustic "NofM"-type speech coding strategy for cochlear implants" (W. Nogueira et al.), a new signal processing technique is described for cochlear implants. The scheme is based on the ACE strategy, as applied in devices of Cochlear, but uses a psychoacoustic masking model in addition to determine the essential components in the input audio signal. They have been able to show with cochlear implant users that improvements in speech understanding are obtained when a small number of channels is stimulated within the same cycle.

The paper "SPAIDE: a real-time research platform for the Clarion CII/90 K cochlear implant" (L. Van Immerseel et al.) describes a platform for tests and experiments with cochlear implants of the Advanced Bionics Corporation. It facilitates advanced research on sound processing and electrical stimulation strategies with the Clarion CII and 90 K implants. SPAIDE allows for real-time sound capturing, sound processing, application of stimulation strategy, and streaming of the outcome to the implant. This experimental platform is being used by different research groups.

In "Ultra wideband transceivers for cochlear implants" (T. Buchegger et al.), the practical implementation of an ultra wideband (UWB) transceiver for cochlear implants is described. A UWB link for a data rate of 1.2 Mbps and a propagation distance up to 500 mm as well as transmitters with step recovery diode and transistor pulse generators are proposed. Moreover, two types of antennas and their filter characteristics in the UWB spectrum are discussed.

In "Design of low-cost FPGA hardware for real-time ICA-based blind source separation algorithm" (C. Charoensak and F. Sattar), a real-time implementation of a modified version of Torkkola's convolutive blind source separation algorithm is described. A discussion of the tradeoffs between separation performance and efficient implementation is given and it is shown how the algorithm can be mapped to an efficient implementation on a low-cost FPGA platform.

Simon Doclo
Søren Holdt Jensen
Philippe A. Pango
Søren K. Riis
Jan Wouters

Simon Doclo was born in Wilrijk, Belgium, in 1974. Simon Doclo received the M.S. degree in electrical engineering and the Ph.D. degree in applied sciences from the Katholieke Universiteit Leuven, Belgium, in 1997 and 2003, respectively. Currently, he is a postdoctoral fellow of the Fund for Scientific Research - Flanders, affiliated with the Electrical Engineering Department, Katholieke Universiteit Leuven. During 2005 he was a Visiting Researcher at the Adaptive Systems Laboratory, McMaster University, Canada. His research interests are in microphone array processing for acoustic noise reduction, dereverberation and sound localization, adaptive filtering, speech enhancement, and hearing aid technology. Dr. Doclo received the first prize "KVIV-Studentenprijzen" (with E. De Clippel) for his M.S. thesis in 1997, a Best Student Paper Award at the International Workshop on Acoustic Echo and Noise Control in 2001, and the EURASIP Signal Processing Best Paper Award 2003 (with M. Moonen). He was Secretary of the IEEE Benelux Signal Processing Chapter (1998–2002) and serves as a Guest Editor of the EURASIP Journal on Applied Signal Processing.

Søren Holdt Jensen received the M.S. degree in electrical engineering from Aalborg University, Denmark, in 1988, and the Ph.D. degree from the Technical University of Denmark in 1995. He has been with the Telecommunications Laboratory of Telecom Denmark, the Electronics Institute of the Technical University of Denmark, the Scientific Computing Group of the Danish Computing Center for Research and Education (UNI-C), the Electrical Engineering Department of Katholieke Universiteit Leuven, Belgium, the Center for PersonKommunikation (CPK) of Aalborg University, and is currently an Associate Professor at the Department of Communication Technology, Aalborg University. His research activities are in digital signal processing, communication signal processing, and speech and audio processing. Dr. Jensen is a Member of the Editorial Board of EURASIP Journal on Applied Signal Processing, an Associate Editor of IEEE Transactions on Signal Processing, and a former Chairman of the IEEE Denmark Section and its Signal Processing Chapter.

Philippe A. Pango joined Gennum Corporation, Burlington, Ontario, in 1999. As a Senior ASIC Design Engineer and Team Leader, he contributed to the hardware design of several hearing-aid DSPs, including the company's first digital audio processor. Philippe's responsibilities spanned from Matlab modeling (Matlab) to actual hardware implementation of common hearing-aid functions, such as time-domain and frequency-domain filterbanks, multirate filters, directionality algorithms, wide dynamic-range compression, and oversampled D/A converters. Since 2002, Philippe has been a Senior Algorithm Developer at Gennum's Advanced Development Group, Audio and Wireless Division. His responsibilities now include the validation of Gennum's programmable platform, the design and real-time implementation of the company's hearing-aid noise reduction algorithms, and the design of bidirectional noise reduction algorithms for Bluetooth wireless headsets. Philippe's interests include low-power DSP platforms, real-time firmware implementation, speech enhancement, and psychoacoustic models.

Søren K. Riis received his M.S.E.E. and Ph.D. degrees from the Technical University of Denmark in 1994 and 1998, respectively. He joined Nokia Networks, Denmark, in 1998 working as a System Designer on intelligent network solutions. In late 1998 he moved to Nokia Mobile Phones, Denmark, to work as a Specialist in audio signal processing with focus on low-complexity speech recognition systems for mobile applications. Since 2002, he has worked as Competence Manager of DSP and embedded SW design in Oticon. His interests include real-time audio signal processing, statistical signal processing, machine learning, and efficient implementation on dedicated embedded systems.

Jan Wouters was born in Leuven, Belgium, in 1960. He received the physics degree and the Ph.D. degree in sciences/physics from the Katholieke Universiteit Leuven, Leuven, Belgium, in 1982 and 1989, respectively. From 1989 till 1992, he was a Research Fellow with the Belgian NFWO (National Fund for Scientific Research) at the Institute of Nuclear Physics (UCL Louvain-la-Neuve and KULeuven) and at NASA Goddard Space Flight Center, USA. Since 1993 he has been a Professor at the Neurosciences Department, Katholieke Universiteit Leuven. His research activities are about audiology and the auditory system, signal processing for cochlear implants, and hearing aids. Dr. Wouters received in 1989 an Award of the Flemish Ministry, in 1992 a Fullbright Award and a NATO Research Fellowship, and in 1996 the Flemish VVL Speech Therapy-Audiology Award. He is member of the International Collegium for Rehabilitative Audiology and of the International Collegium for ORL, a Board Member of the NAG (Dutch Acoustical Society), an author of about 100 articles in international peer-review journals, and a reviewer for several international journals.