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EDITORIAL

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Advances in adaptive filtering theory and applications to acoustic and speech signal processing

Markus Rupp[1*], Walter Kellermann[2], Abdelhak Zoubir[3] and Gerhard Schmidt[4]

The advancement of adaptive digital systems, in particular adaptive filters with applications to acoustic echo and noise control, but also to digital transmission systems, has been influenced by the work of Eberhard Hänsler and his research group members who contributed to this area over several decades. On the occasion of his 80th birthday, this issue of the EURASIP Journal of Advances in Signal Processing (JASP) is dedicated to Dr. Hänsler to acknowledge his outstanding contributions, both as an eminent researcher and as a gifted educator.

Eberhard Hänsler studied electrical engineering at Technische Hochschule Darmstadt, Germany (now Technische Universität Darmstadt), from 1956 to 1961, specializing in control theory. From 1961 to 1963, he was with the Research Institute of the German Post Administration, where he worked on problems of information storage. From 1963 to 1968, he was a research and teaching assistant at the Department of Electrical Engineering of Technische Hochschule Darmstadt, where he received the doctoral degree for his work on optimal filtering in 1968. From 1968 to 1974, he was with IBM Research, both in Zurich, Switzerland, and in Yorktown Heights (IBM Thomas J. Watson Research Center), NY, USA, working on reliability problems of communication networks. In 1974, he became a full professor at Technische Hochschule Darmstadt, Germany.

In 1989, Eberhard Hänsler organized the first International Workshop on Acoustic Echo Control in Berlin, Germany. This biennial workshop became the leading forum in the field and developed into a global event with venues in the USA, Israel, and Japan, in addition to many places throughout Europe. In 2001, the seventh edition, now called IWAENC (International Workshop on Acoustic Echo and Noise Control), returned to Germany—again organized by its initiator, Eberhard Hänsler. The last one was held in Antibes, France, in 2014 and the next one is planned in 2016 in Xi'an, China, (see http://www.iwaenc2016.org/).

During his time at Technische Universität Darmstadt, Professor Hänsler authored and co-authored several books. One of the early books, entitled *Statistische Signale: Grundlagen und Anwendungen,* printed by Springer in 1991, 1997, and 2001, demonstrates the dedication of Eberhard Hänsler to education. His latest research book, entitled *Acoustic Echo and Noise Control,* printed by Wiley in 2004, covers those topics of speech and audio processing on which he and his research group have been working for the last thirty years.

Especially on the acoustic research topics of echo and noise reduction—both in terms of academic research and in terms of its industrial applications—Eberhard Hänsler was one of the leading researchers worldwide. Many of the algorithms developed in his group found their way into applications and products.

A key recipe for this success is certainly his emphasis on experimental work, alongside with sound theoretical analysis. While many others in this field were satisfied with measuring improvements in form of MATLAB simulations, he always pushed his group to implement new ideas in the form of real-time demonstrators and prototypes so that the acoustic sensation was perceivable.

Eberhard Hänsler received many awards, including the EURASIP Technical Group Award in the year 2000.

On this special occasion of his 80th birthday, we are also proud to present the following greetings by Simon Haykin:

Adaptive Signal Processing

By Simon Haykin

In historic terms, the topic of adaptive signal processing—also frequently referred to as adaptive filtering—started with the least-mean-square (LMS)

* Correspondence: mrupp@nt.tuwien.ac.at
[1]TU Wien, Wien, Austria
Full list of author information is available at the end of the article

algorithm, which was first described in a conference paper co-authored by Widrow and Hoff (1960). The LMS algorithm operates in an iterative manner, which therefore differentiates itself from the Wiener filter. In the same year, the Kalman filter so named in honour of Rudi Kalman, also operates in an iterative manner.

With the emergence of (digital) computers at that item, iterative processing proved to be preferable with respect to batch processing as in the Wiener filter. Subsequently, it was found that the recursive least squares (RLS) algorithm is indeed a special case of the Kalman filter. The two of them, the LMS and RLS algorithms and their respective extensions dominated the adaptive filtering literature. The LMS algorithm follows a stochastic gradient procedure, whereas the RLS algorithm is based on the matrix inversion lemma. Naturally, from an engineering perspective, robustness plays a key role in practical applications when an adaptive filtering algorithm operates in the presence of uncertainties. What is truly remarkable to note is the fact in mathematical terms, the LMS algorithm is more robust than the RLS algorithm in light of the H_{oo} optimization theory due to Zames (1981). That theory was described many years after formulation of these two adaptive filtering algorithms.

Again from an engineering perspective, I find it remarkable that Professor Eberhard Hänsler opted for the normalized version of the LMS algorithm as the tool for the design of a telecommunication system which is aimed at the control of acoustic echoes that represent uncertainties in the system. Moreover, the system was built to have the capacity to adjust the step-size parameter in the normalized LMS algorithm from one iteration to the next.

In light of this brief historical background, it is an honour to raise my hat to my good friend Eberhard for devoting much of his professional life to the supervision of graduate students with emphasis on practical applications, which became his passion over the course of many years.

Simon Haykin

For this special issue, we received 26 articles, among which 12 were accepted. The topics range from fundamental research in adaptive filters to applied algorithmic advances in hands-free telephony and modern cellular networks.

The first article, entitled "Adaptive Filters: Stable but Divergent", by M. Rupp not only provides a historical overview of the past fifty years of adaptive filter algorithmic advances but also shows that in terms of stability, there are still open problems. The well-known concept of "stability in the mean square sense" although very suitable to apply in this algorithmic context turns out to be problematic for many algorithms; a more robust stability metric, however, is not always easy to apply.

The second paper, "Filtered-X Least Mean Fourth (FXLMF) and Leaky-FXLMF Adaptive Algorithms", by, A.M. Al-Omour, and A. Zidouri, is dedicated to an analysis of FXLMF and the LFXLMF algorithms starting from an LMF ('least mean fourth') criterion, and stability ranges for their step size parameter and the leakage factor were derived. Simulations demonstrate the favourable behaviour of these algorithms with noise as input and for low SNR scenarios especially when combined with a convex combination adaptive filtering scheme.

The third article entitled "Asymptotic Equivalent Analysis of the LMS Algorithm Under Linearly Filtered Processes", by M. Rupp, sheds new light on well-established concepts for stability in the mean square sense based on the independent assumption and allows to separate correlation and density properties of the input sequence when designing adaptive algorithms of LMS type. For very large filters, perturbation effects disappear and the results agree with those obtained without the independent assumption.

With the paper "The Generalized Frequency-Domain Adaptive Filtering Algorithm as an Approximation of the Block Recursive Least-Squares Algorithm", by M. Schneider and W. Kellermann, the authors first provide a rigorous derivation of the well-known GFDAF algorithm, thereby generalizing and correcting previous derivations, and showing that the GFDAF can be viewed as an approximation of the block recursive least-squares (RLS) algorithm. The resulting formulation is then successfully applied to pruned transform domain loudspeaker-enclosure-microphone models as used for massive multichannel AEC, with dozens of loudspeakers and microphones.

The article "A Bayesian Network Approach to Linear and Nonlinear Acoustic Echo Cancellation", by C. Huemmer, R. Maas, C. Hofmann, and W. Kellermann, presents a general Bayesian approach to linear and nonlinear acoustic echo cancellation (AEC), where a latent state vector in a state-space model captures all relevant information of the unknown system. Using probabilistic graphical models, the normalized least mean square algorithm (with fixed and adaptive step size), the Hammerstein group model and a numerical sampling scheme for nonlinear AEC can be represented in a unified way to serve as a powerful framework for future algorithmic development.

The article "An Overview on Optimized NLMS Algorithms for Acoustic Echo Cancellation", by C. Paleologu, S. Ciochina, J. Benesty, and S.L. Grant, presents several schemes for controlling adaptive filters to assure robust and reliable convergence. Due to the "non-parametric" nature of the proposed schemes, no additional features (knowledge) are required. Simulations show the good performance and the practical applicability of the proposed solutions.

The contribution "Features for Voice Activity Detection: A Comparative Analysis", by S. Graf, T. Herbig, M. Buck, and G. Schmidt, presents an overview about a multitude of features that have been proposed for voice activity detection. After introducing several feature candidates out of six groups (power, pitch, modulation, etc.), a performance comparison using large databases was performed.

The contribution "Signal Processing Techniques for Seat Belt Microphone Arrays", by V. K. Rajan, M. Krini, K. Rodemer, and G. Schmidt, describes a new type of microphone that can be weaved in security belts. Due to the proximity to the speaker's mouth, such microphones show several advantages for automotive hands-free and speech dialog systems. However, since belt microphones are not installed on fixed positions, also a variety of drawbacks arises. The paper presents an overview about the pros and cons and shows dedicated signal enhancement schemes for this microphone type.

The article "A Low Complexity Reweighted Proportionate Affine Projection Algorithm with Memory and Row Action Projection", by L. Liu, S.L. Grant, and J. Benesty, proposes a reweighted proportionate affine projection algorithm (PAPA), derived from a family of sparseness measures, which demonstrate performance similar to other PAPA derivatives but with lower computational complexity. The sparseness of the channel is taken into account to improve the performance for dispersive system identification, while the memory of the filter's coefficients is combined with row action projections to significantly reduce computational complexity.

The article entitled "Robust and Adaptive Diffusion-based Classification in Distributed Networks", by P. Binder, M. Muma, and A.M. Zoubir, describes a robust and adaptive distributed hybrid classification algorithm that is based on the diffusion principle. Here, information is diffused through the network via local communication between neighbouring nodes. The algorithm can deal with non-stationary features that may also contain outliers. It is useful, e.g. for speech source labelling and object labelling in camera networks as it can answer the question: who sees what?

The article entitled "Distributed Gram-Schmidt Orthogonalization with Simultaneous Elements Refinement", by O. Sluciak, H. Strakova, M. Rupp, and W. Gansterer, describes a distributed adaptive solution for the problem of Gram-Schmidt orthogonalization. The proposed algorithms offer a trade-off between number of messages that need to be exchanged and accuracy as well as learning rate.

The last article entitled "Adaptive Techniques in Advanced 4G Cellular Wireless Networks", by B. Hätty, H.J. Dressler, H. Kröner, G. Schnabl, M. Schopp, and A. Splett, treats time-variant communication systems as they emerge in the context of 4G cellular networks. Appropriate linear or nonlinear adaptive techniques which are able to track various network changes are described for wideband linear power amplifiers, multiple-input multiple-output antenna systems, heterogeneous networks, and self-organizing networks, including their corresponding realization and performance aspects.

Author details
[1]TU Wien, Wien, Austria. [2]University Erlangen-Nuremberg, Erlangen-Nuremberg, Germany. [3]Technische Universität Darmstadt, Darmstadt, Germany. [4]Kiel University, Kiel, Germany.

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