# AAFAA - Advances in Adaptive Filtering for Acoustic Applications

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*Abstract*— **One of the most studied areas for adaptive filtering is the one related to acoustic applications. Both the simplicity of the configuration and the immediate methods to evaluate the performance contributed to the popularity of this binomial pair: adaptive filtering and acoustic applications. Lately, a new step in acoustic application has been made, including the acoustic Multiple Input Multiple Output (MIMO) setups. In this context, we have to compute the inverse of very large matrices, we have to reduce (as much as possible) the echo, even during double-talk periods, and we have to do it with reduced costs. With such configurations, including also stereophonic systems, new approaches as the dichotomous coordinate descent (DCD) iterations were introduced.**

**The objective of the special session Advances in Adaptive Filtering for Acoustic Applications is to identify novel forms of adaptive algorithms, designed for the new acoustic applications reality from the performance point of view, nevertheless efficient from the implementation complexity perspective. In other words, the four included papers present the theoretical models, the efficient implementation solutions and an example of real application.**

***Keywords-adaptive filtering; acoustic applications; VR-RLS; DCD-RLS; FPGA***

## I. INTRODUCTION

Today, distances are not important anymore since video and audio connections are used by doctors to investigate patients, by teachers to present the courses, by business men to close deals, etc. The only constraint is the quality of these connections. This is the reason why new and evolved forms of adaptive algorithms are proposed nowadays in the classical acoustic configurations, such as acoustic echo canceller (AEC) and adaptive noise cancelation (ANC). In the dynamic acoustic environments, and with users having increasing expectations, the Least Mean Squares (LMS) family with fixed step size and the Recursive Least Squares (RLS) family with fixed forgetting factor are not providing the expected results anymore. In this context, several modified versions of these classical adaptive algorithms were proposed in the last years. Researchers presented the variable step size (VSS) approach for the LMS family, as well as the variable forgetting factor (VFF) for the RLS family. Improved performances were reported compared with the classical algorithms, especially in terms of stability and tracking capabilities when the environment changes or when double talk situation occurs [of course, without using a double talk detector (DTD)].

However, the cost of these improved performances proved for the new proposed adaptive algorithms is their increased implementation complexity. Even if the computational power for the new digital signal processors (DSP) or field programming gate arrays (FPGA) increased lately, we are still obliged to pay attention to this aspect. More and more processing power is included in the current acoustic systems. We discus usually about multiple input multiple output (MIMO) acoustic setups, with stereophonic approach.

The objective of the special session Advances in Adaptive Filtering for Acoustic Applications is to present a newly proposed variable regularized (VR) RLS algorithm, and to explain how we can avoid in this context computing the inverse of a very large matrix (usually 1024x1024) by applying the dichotomous coordinate descent (DCD) iterations method. Also, some solutions for efficient FPGA implementation are provided. Finally, a real acoustic application example is described, pointing out the benefits brought by this new adaptive algorithm.

## II. INCLUDED PAPERS

The recursive least squares algorithm is very popular in many applications of adaptive filtering, especially due to its fast convergence rate. However, the computational complexity of this algorithm represents a major limitation in some applications that involve high length adaptive filters, like acoustic echo cancellation. Moreover, the specific features of this application require good tracking capabilities and double-talk robustness for the adaptive algorithm, which further implies an optimization process on its parameters. In case of most RLS-based algorithms, the performance can be controlled in terms of two main parameters, i.e., the forgetting factor and the regularization term. The goal of the first paper in the session [1] is to outline the influence of these parameters on the overall performance of the RLS algorithms and to present several solutions to control their behavior, taking into account the specific requirements of echo cancellation application.

The authors of [1] are Camelia Elisei Iliescu, a Ph. D. Student, and Prof. Constantin Paleologu, one of the leaders of the adaptive algorithms research group inside Telecommunications Department of University Politehnica of Bucharest, Romania.

The second paper of the session [2] discusses about the ANC scenario, which belongs to the interference cancellation class. It employs an adaptive filter to estimate a perturbation signal, which corrupts a primary acoustic source. In most of the corresponding applications, the goal is to imitate an original speech signal. This paper proposes the use of a low complexity RLS adaptive algorithm for the ANC procedure. The combination between the RLS method and the DCD iterations offers good performance with acceptable arithmetic costs. Simulation results are provided in order to demonstrate the validity of the ANC system based on the RLS-DCD adaptive algorithm.

The authors of [2] are student Roxana Mihaescu, Lecturer Cristian Stanciu – acoustic applications implemented on FPGA, and Prof. Lucian Stanciu – signals, circuits and systems. They are all members of Telecommunications Department of University Politehnica of Bucharest, Romania.

The third paper of the session [3] presents the main elements proposed for an efficient implementation on FPGA of the novel VR-RLS algorithm. The followed performance directions are the overall processing speed and the amount of used hardware resources. It also keeps focus on this adaptive algorithm performance in the scenario of AEC, from the finite precision implementation degradation point of view.

The authors of [3] are Assoc. Prof Cristian Anghel – 14 years of experience on FPGA implementation for telecommunications equipments and acoustic systems, and Prof. Silviu Ciochina – one of the founders of the adaptive algorithms research group and the Head of Telecommunications Department of University Politehnica of Bucharest, Romania, for the last 12 years.

The fourth paper of the session [4] addresses a real world example of acoustic application. In many trials, multimedia materials brought as evidence, in video or audio format, could make the difference between “guilty” or “not guilty” verdicts. Most multimedia content is stored in a digital form nowadays; therefore, with so many free editing software at anyone’s disposal, it is very easy to be forged. In other situations the critical evidence (even if recorded), can be heavily masked by other signals and declared inappropriate. This paper is a contribution to the multimedia forensic domain, presenting the impact of the acoustic environment on a computer software based on adaptive filtering, which can be used to recover a speech signal drowned in loud music. The results help to decide if placing a microphone in a certain room could be useful or not given the proposed solution for recovering the speech is to be used afterwards.

The authors of [4] are Assistant Professor Robert Alexandru Dobre, Prof. Mihnea Udrea, Prof. Dumitru Stanomir, and Prof. Cristian Negrescu - the Dean of Faculty Electronics, Telecommunications and Information Technology inside University Politehnica of Bucharest, Romania.

## III. Conclusions

The special session Advances in Adaptive Filtering for Acoustic Applications along with ICN 2017 includes four papers. The session starts with a highly theoretical research work, presenting an improved version of a classical adaptive algorithm. Then it continues by explaining, also in a theoretical manner but with strong concerns on the implementation side, an alternative approach for the matrix inversion problem. The third paper brings into discussion very practical aspects about the FPGA implementation of this new proposed algorithm. Finally, the forth paper presents the behavior and the obtained performances when using this algorithm in a real acoustic application.

In conclusion, one can observe a complete, multi-layer, and multi-approach study proposed by our research group.

## References


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