Project objectives

In the context of multi-channel noise cancellation applications the adaptive filter represents a major importance block. Its performance highly influenced the overall performances of the system. Consequently, a first set of objectives are focused on the performance enhancement of the adaptive algorithms used in multi-channel ANC systems. The main performance criteria that have to be taken into account when designs such a system are the convergence rate, the tracking capabilities, the computational complexity, and the numerical robustness. The convergence rate is very important in the acoustic field, where the length of the filter is very high (hundreds or even thousands of coefficients) and it highly influence the performance of the adaptive algorithm. Moreover, the tracking capability is also a major problem since the acoustic environment is variable in time (it depends on the moving objects or bodies). The high length of the filter also influences the computational complexity of the adaptive algorithm. This is the main reason for using LMS algorithms in such applications, instead of LS algorithms. Finally, the numerical stability of the algorithm is very important from the practical point of view. In the present, there isn't an adaptive algorithm that satisfied all the criteria mentioned above. The solution is to make a compromise between these factors, according to the requirements of the application.

The FAP algorithms are involved within the most recent multi-channel ANC systems [1]. The reason behind this is the fact that these algorithms offer a superior convergence rate over the LMS algorithms and a reduced computational complexity as compared to the LS algorithms. Nevertheless, they are facing some numerical stability problems, mainly due to the LS procedure used for solving the set of equations within the affine projections. Even if some solutions were proposed in order to overcome this problem (e.g. Gauss-Seidel method [2]), there isn't an unitary solution for this aspect.

In the context of this research project we aim to provide some new solutions in this direction. First, we will use QRD-LS (*QR-decomposition-based-least-squares*) [3] for solving the set of equations within the FAP. This choice is motivated mainly by two factors: 1) the QRD method offers good numerical properties, and 2) they are suitable for systolic array implementation, which offer an important degree of parallelism and modularity. The potential problems of the QRD-LS procedures are related to the square-root operations due to the Givens rotations. The solution is to used square-root free QRD methods [4], taking into account the fact that they reduce the parallelism capabilities of the algorithm. Moreover, we aim to use the LNS (*LNS – logarithmic number system*) implementations [5], which are very suitable for this type of methods. In the same context, we will try to use some SRF-QRD-LSL (*square-root-free-QR-decomposition-based-least-squares-lattice*) algorithms [6]. Of course, we have to adapt the structure of the algorithm taking into account the multi-channel case, the filtered-x scheme, and the lattice structure of the algorithm.

Another objective is related to the computational complexity of the FAP algorithms used for multi-channel ANC systems. In this context, the most recent solutions proposed the use of GS-PAP algorithms [7], [8]. Even if these algorithms achieve a lower computational complexity as compared to the classical algorithms, they still require a inverse matrix operation which implies a certain computational load and prone to numerical instability. To avoid this aspect we propose to use DCD (*dischotomous* *coordinate descent*) methods [9] instead of the GS approach. These methods could lead to both lower computational complexity and improved numerical stability.

The third direction within the adaptive algorithm framework regards the convergence speed and the tracking capabilities of the algorithms. These two performance criteria are driven by the step-size parameter of the algorithm. Nevertheless, the choice of this parameter reflects a tradeoff between these two factors. A high value of the step-size parameter leads to a high convergence rate and tracking capability, but also to a high misadjustment and stability problems. Ideally, we want a higher step-size in the beginning and in a tracking situation and a lower value for a low steady-state error. Of course, this demand cannot be achieved using a fixed value of the step-size. Consequently, the step-size needs to be controlled. Thus several versions of variable step-size (VSS) affine projection algorithms have been proposed [10]. Unfortunately, these algorithms require the knowledge of too many additional parameters (e.g. input signal power, noise power) which are difficult to know or estimate in a real environment. This aspect motivates us to find non-parametric VSS-FAP algorithms, suitable for real-world multi-channel ANC systems.

The second major area of this project is related to the microphone array. The major problem in this field is to achieve an optimal directivity characteristic. The situation is equivalent to a beamforming system. The most common approaches used in this context are the delay-and-sum and the filter-and-sum methods. Nevertheless, these methods cannot provide very good performance in real acoustic environments and they are highly influenced by some factors, such as signal type (narrow/broad band) and noise type.

Within this project we follow a very recent scientific idea in this field. The goal is to approach the problem of microphone array beamforming from a MIMO (*multiple inputs multiple outputs*) system perspective [12]-[14]. The main goals of this approach is to find some relation with the MINT (*multiple-input/output inverse theorem*) and LCMV (*linearly constrained minimum variance*) filtering, which further allows to propose new acoustic channel estimation algorithms and to combined them with adaptive noise reduction techniques.

As a practical aspect of our research project, we aim to test all of our proposed solutions in fixed-point DSP and FPGA. Besides the theoretical aspects related to the finite precision arithmetic, it remains an essential programming aspect for optimal use of fixed-point resources. Our team has sustained many of its previous research results using fixed-point DSP implementation, accumulating an important experience on this field. In this project we will follow the same trend in order to validate our algorithms and schemes. Additionally, our team members performed some training periods at Motorola Romania, working together with specialists from the firm on research projects. Our final goal is to design a multi-channel ANC system with superior performances, and to introduce it in the technological and commercial circuit. In that sense, the collaboration with Motorola Romania represents an important opportunity in order to validate our research from practical and commercial points of view.

This research project has an important degree of interdisciplinary, using techniques, methods and solutions from complex domains, such as: digital signal processing, adaptive algorithms, acoustics, speech processing, digital signal processors, programming (MATLAB, C, VHDL, and assembler), mobile communications, beamforming systems and MIMO communication systems.

Recently, our research team has established some collaboration (also publishing some scientific papers [2], [7]-[9], [11]) with three well-known researchers: Martin Bouchard (*Senior Member IEEE*, University of Ottawa, Canada), Jacob Benesty (*Senior Member IEEE*, University of Quebec, Montreal, Canada) and Yurij Zakharov (*Member IEEE*, University of York, U.K.). This aspect is very positive for our research and especially for the youngest members of our team. This represents a good opportunity to validate our result and to increase the Romanian research level in the context of the international research.

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Research methodology

The time duration of this project is 36 months, as follows: 2007 – 3 months, 2008 – 12 months, 2009 – 12 months, 2010 – 9 months. The work plan was elaborated according to this calendar. As we have mentioned, the major blocks of a multi-channel ANC system are the adaptive filter and the microphone array. Our objectives approach these two fields in an apparent separate manner, to converge in the final to the overall goal.

In a first scientific objective of the year 2007, we aim to develop the theoretical and conceptual framework in the field of FAP/PAP adaptive algorithms. This stage will continue in the first objective from 2008, by implementing and testing using Matlab environment, StarCore simulator, and VHDL language the analyzed algorithms.

The year 2008 continue with three scientific objectives with a common point: to improve the performance of the FAP/PAP adaptive algorithms. In the first one we aim to improve the numerical stability such that we propose some versions of the FAP/PAP algorithms based on the QRD methods (known for their good numerical robustness). Also, we will try to implement the resulted algorithms using systolic arrays and LNS. In a second objective from this series we aim to decrease the computational complexity of the FAP/PAP algorithms using the DCD methods instead the Gauss-Seidel procedures. Finally, in the third objective we will propose some versions of variable step-size (VSS) FAP/PAP with improved convergence rate and tracking capabilities. In this point of the project some improved versions of FAP/PAP algorithms will be available (FAP/PAP-QRD, FAP/PAP-DCD, FAP/PAP-VSS). Consequently, in a first objective of 2009, we will develop new versions of these algorithms suitable for multi-channel case and filtered-x structure. In the following objective we will implement and test using StarCore DSP and FPGA Virtex.

The third objective of 2009 approach the second part of the project, which is related to the microphone array beamforming. First, we aim to develop the theoretical and conceptual framework in this field by studying the methods of delay-and-sum and filter-and-sum. The last objective of 2009 approaches the problem of microphone array beamforming from a MIMO system perspective. We will try to develop some connection with MINT and LCMV filtering, and to propose new acoustic channel estimation algorithms. During a first objective of 2010, these algorithms will be implemented and tested using StarCore DSP and FPGA Virtex.

The last two objectives of 2010 and also of the project, aim to design and test a multi-channel ANC system with superior performances and to validate it in a teleconferencing application. Based on our relation with Motorola Romania we further aim to integrate this system in a real-world communication system.