The scientific importance and relevance of the domain

Noise enters speech communications systems in many ways. In traditional wire-line telephone calls, one or both parties may be speaking within an environment having high levels of background noise (i.e., calls made from public telephone). Similarly, cellular, or wireless, telephones permit users to place calls from virtually any location, and it is common for such communications to be degraded by noise of varied origin. In room teleconferencing applications, in which the acoustical characteristics of the environment are generally assumed to be controlled (i.e., quiet), it is not uncommon for heating, ventilation and air-conditioning systems to contribute substantial levels of noise. Noise originates not only from acoustical source, however. Circuit noise, generated electrically within the telephone network, is still prevalent throughout the global telecommunications system.

When present at small or moderate additive levels, noise degrades the subjective quality of speech communications. Listening tests broadly show that people grow less tolerant of, and less attentive to, listening materials as the signal-to-noise ratio (SNR) of the material decreases. This phenomenon is known as listener fatigue. When SNR of speech material is very low (e.g. less than 10dB) the intelligibility of speech is affected.

Even traditionally low levels of noise can present a problem, especially when multiple speech channels are combined as in conferencing or bridging. In multiparty, or multipoint, teleconferencing, the background noise present at the microphone(s) of each point of the conference combines additively at the network bridge with the noise processes from all other points. The loudspeaker at each location of the conference therefore reproduces the combined sum of the noise processes from all other locations. This problem becomes serious as the number of conferencing points increases. Consider a three-point conference in which the room noise at all locations is stationary and independent with power P. Each loudspeaker receives noise from other two locations, resulting in a total received noise power of $2P$, or 3 dB greater than that of a two-point conference. With N points, each side receives a total noise power that is $10\log(N–1)$ dB greater than P. For example, in a conference with 10 participating locations, the received noise power at each point is about 10 dB greater than that of the two-party case. Because a 10 dB increase in sound power level roughly translates to a doubling of perceived loudness, the noise level perceived by each participant is twice as loud as that of the two-party case. The benefits of noise reduction processing for cases such as this are clearly evident.

Today, noise reduction processors appear in variety of commercial products, including cellular telephone handsets; cellular hands-free, in-the-car telephone adjuncts; room teleconferencing systems; in-network speech processors, such as bridges and echo cancellers; in-home telephone appliances, including speakerphones and cordless phones; and hearing aid and protection devices.

A variety of approaches have been proposed to reduce noise for purpose of speech enhancement. Included are: classic (static) Wiener filtering; dynamic comb filtering [1], in which a linear filter is adapted to pass only the harmonic components of voiced speech as derived from the pitch period; dynamic, linear all-pole and pole-zero modeling of speech [1], in which the coefficients of the (noise-free) model are estimated from the noisy speech; short-time spectral modification techniques, in which the magnitude of the short-time Fourier transform is attenuated at frequencies where speech is absent [2]; and hidden Markov modeling [3], a technique also employing time-varying models for speech but where the evolution of model coefficients is governed by transition probabilities associated with model states.

All the above noise reduction methods share the property that they operate on a single channel of noisy speech. They are blind techniques, in the sense that only the noise-corrupt speech is known to the algorithm. Thus, in order to enhance the speech-signal-to-noise ratio, the algorithms must form bootstrap estimates of the signal and noise. When multiple channels containing either the same noisy speech source or noise source alone are available, a wide range of spatial acoustic processing technologies applies. Included are adaptive beamformes and adaptive noise cancellers. Adaptive noise cancellation methods are coherent noise reduction processors, exploiting the phase coherency among multiple time series channels to cancel the noise, whereas the noise reduction methods listed above are incoherent processors. This class of adaptive noise reduction techniques is often combined with acoustic echo cancellation and microphone-array processing (beamforming) [4]. These techniques will be approached during this research project. The main motivation is related to the importance and impact of the multi-channel system applications within the modern communication systems.

In this context, active noise control (ANC) systems are being increasingly researched and developed [5]. These systems work on the principle of destructive interference between an original primary disturbance sound field measured at the location of a number of error sensors (typically microphones) and a secondary sound field that is generated by a number of control actuators (typically loudspeakers). In ANC systems a common approach is to use finite impulse response (FIR) filters as adaptive controllers, which try to identify the plant impulse responses between sources. The plant impulse responses each have a delay, corresponding to the propagation delay between an actuator and an error sensor. In control theory, it is well known that delays are a nuisance for control applications and influence the performances of adaptive FIR filtering algorithms. Basically, the larger the delay, the slower the adaptation of the algorithms used to train the adaptive FIR filters in the ANC system, because a smaller step size has to be used in algorithms. This means that a large delay can reduce both the initial convergence speed of the ANC system and the ability of system to quickly adapt to varying conditions (i.e., tracking). To avoid this problem, the delay compensated or modified filtered-x structure for ANC systems using FIR adaptive filtering was introduced [6]. This structure eliminates the plant delay by computing an estimate of the primary field signals, which are unaffected by the changes of the adaptive FIR filter coefficients. This is done by using the knowledge of the actuator signals and the plant models. By filtering the actuator signals with the plant models, estimates of the actuators contributions at the location of the error sensors are obtained. By subtracting these contributions from the measured error sensors signals, estimated of the primary filed signals are obtained. The delay compensated modified filtered-x structure then performs a commutation of the plant and the adaptive filters, so that the adaptive filters directly try to predict the estimated primary field signals. The resulting system behaves like a classical adaptive filtering system, without

any delay between a change to the filter coefficients and its effect on the error signals being minimized. This allows for faster convergence (through a larger step size in the adaptation algorithms) and a better tracking.
For ANC using adaptive FIR filters, the multichannel filtered-x least-mean-square (FX-LMS) algorithm [6] is the most commonly used algorithm. The drawback of the FX-LMS algorithm is the slow convergence speed, especially for the broadband multichannel systems. Although it converges faster than the FX-LMS algorithm, the delay compensated or modified FX-LMS (MFX-LMS) algorithm [7] also suffers from the same slow convergence problem, especially for multichannel systems. Recently, some fast affine projection (FAP) algorithms have been introduced for multichannel ANC [8], [9], as an interesting alternative to the FX-LMS and MFX-LMS algorithms. They can provide a significantly improved convergence speed at a reasonable additional computational cost. In the field of adaptive filtering, it is well known that FAP algorithms present a good tradeoff between convergence speed and computational complexity. Even if they can't provide the same convergence speed as recursive least-squares (RLS)-based algorithms, they demonstrate a much improved convergence speed compared to the stochastic gradient descent algorithms like the FX-LMS and MFX-LMS algorithms, without the high increase of the computational load or the numerical instability often found in RLS-based algorithms, especially for multichannel systems [8], [9]. Yet, the additional computational cost or the potential numerical instability in some recently proposed FAP algorithms for ANC can prevent the use of those algorithms for some applications in this field.
These algorithms include a set of linear equations to be solved (the number of these equations is much smaller than the length of the adaptive filter). This set of equations is often solved by the use of a built-in recursive least-squares (RLS) algorithm or some more efficient built-in fast-RLS algorithm inside the FAP algorithm. In recent years, other schemes have been investigated to replace the RLS-based algorithms inside the FAP algorithm, to improve the numerical stability and also possibly reduce the computational load. The Gauss-Seidel inversion scheme is one of those schemes that were successfully applied to the FAP algorithm [10]. The resulting algorithm, named MFX-GSFAP algorithm exhibits a lower computational complexity than the MFX-FAP-RLS algorithm, for the same convergence speed. As an attempt for further complexity reduction, a pseudo-affine projection algorithm named GS-PAP algorithm was derived [11]. Recently, the GSPAP algorithm was extended to the case of multichannel ANC systems, which led to the so-called MFX-GSPAP algorithm [12]. However, this algorithm still requires at least one inverse matrix computation. This can be very complex for large matrices and prone to numerical instability. Consequently, it is highly desirable to improve the performances of these algorithms in terms of the computational complexity and numerical stability. The project objectives regarding this aspect will be detailed in the next section.
Microphone arrays consist of sets of microphone sensors that are spatially arranged in specific patterns. These systems have already played, and will continue to play an important role in applications like audio-bridging and teleconferencing where distant or hands-free signal acquisition is required [13]. One of the most important functionalities of microphone arrays is to extract the signal of interest from its observations corrupted by noise, reverberation, and competing source. The typical method for this is to form a beam and point it to a desired direction. As a result, signals from this so-called look direction are reinforced, while signals from all the other directions are attenuated. In the case of an array of microphones, beamforming is achieved based on manipulating the signals from the microphone outputs. Many algorithms have been developed for this type of applications. The simplest one is the delay-and-sum beamformer, which was originally investigated in the underwater acoustics and radar antenna areas. The basic idea is to delay (or advance) each microphone output by a proper amount of time, so that the signal components from the desired source are synchronized across all sensors. These delayed (or advanced) signals are then weighted and summed together. Since they add up together coherently, the desired signal components are reinforced. In contrast, the other sources and noise are suppressed or even eliminated as they are added together destructively. The weighting coefficients can be either fixed or adaptive determined. The latter leads to an adaptive beamformer [14]. The advantage of adaptive beamforming over nonadaptive beamforming can roughly be interpreted as follows. Suppose that we have an array consisting of N microphones, and correspondingly N signals at their outputs. At a single frequency, a total of $N – 1$ nulls can be formed in the directivity patterns. If we adjust the coefficients adaptively by taking into account the signal and noise characteristics, the $N – 1$ nulls can be properly designed and placed so that noise and interference can be better rejected.
The above technique was developed for narrowband signals. Although it serves as the basis for any array beamforming, this technique is not very useful for acoustic applications since speech is a typical broadband signal. Consequently, the directivity pattern of the delay-and-sum beamformer would not be the same across a broad frequency band. If we use such a beamformer, then noise and interference signals coming from a direction different from the beamformer's look direction will not be uniformly attenuated over its entire spectrum. These spectral tilt results in a disturbing artifact in the array output.
One way to overcome this problem is to use harmonically nested subarrays [15]. Every subarray is designed for operating at a single frequency. However, such a solution requires a large array with a great number of microphones, and the array geometry is unusual. Another way to circumvent this problem is to perform narrowband decomposition and design narrowband beamformers independently at each frequency. The broadband beamformer will be equivalent to applying a FIR filter to each microphone output and then summing the filtered signals together. The filter-and-sum algorithm has been intensively studied in the last years [16]-[18]. As mentioned earlier, for an array consisting of N microphones, the delay-and-sum structure can only produce $N – 1$ nulls at a single frequency. By applying FIR filters of length L to N channels, we can produce $N – 1$ nulls at $L – 1$ different frequency. Therefore, this technique offers more flexibility in rejecting noise and interference than the delay-and-sum beamformer. Similar to the delay-and-sum case, the filters' coefficients can also be determined either in a nonadaptive way or adaptively. With adaptive algorithms, the nulls can be properly designed and placed at directions and frequencies for better interference and noise attenuation.
Although so many efforts have been devoted to microphone array processing, the performance of most microphone array beamformers in practical acoustic environments still cannot meet expectation. Apparently, the potential of microphone arrays has

not been fully realized and expected. The reasons behind this are very sophisticated and have not been fully understood thus far. Therefore, further research in this area is indispensable. Recent researches try to approach these systems by a MIMO perspective [19]-[21]. The major goal is to extend some methods and algorithms from this field to the microphone array beamforming. We aim to continue the research in this direction, and we detail the related objectives in the next section.

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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 algoritms used in multi-channel ANC systems. The main performance criteria that have to be taken into account when design such a system are the convergence rate, the tracking capabilities, the computational complexity, and the numerical robusteness. 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 capabilities 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. Finnaly, 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 this factors, according to the requirements of the application.

As was mentioned in the previous section (11.1.), the FAP algorithms are envolved 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 offer 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 used 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 porpose 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 perfomance criteria are driven by the step-size parameter of the algorithm. Nevertheless, the choice of these 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 need 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 aditional parameters (e.g. input signal power, noise power) which are difficult to know or estimate in a real environment. This aspect motivate 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 interdisciplinarity, 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, assembler), mobile communications, beamforming systems and MIMO communication systems.
Recently, our reseach 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*, Universitaty of York, U.K.). This aspect is very positive for our research and especially for the youngest members of our team. This represent 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

he 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 in the previous sections (11.1 and 11.2) 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.