GTAC Annual Research Report 2016/2017

Head of the Group research report

The Audio and Communications Signal Processing Group (GTAC) started its activities in 1998, and nowadays is composed of 6 senior researchers (PhD) and 4 junior researchers. GTAC research focuses in signal processing algorithms for sound and wireless communication applications, in particular related to active noise control systems, sound quality perception, spatial audio rendering and multi-channel audio filtering in the area of sound processing, and related to efficient MIMO receivers, multi-user communications and coordinated multi-point systems in the area of wireless communications. GTAC has also carried out a tight collaboration with the Interdisciplinary Computation and Communication Group (INCO2) of the Universitat Politécnica de Valencia. Both groups have worked together in several projects since 2009 when the Regional Government distinguished the extended INCO2-GTAC group (together with the HPCA group of the Universitat Jaume I) with the "Prometeo" Excellent Research award, which was validated again in 2014 for a new period of five years.

During 2016/2017 the group has continued its work on the development of sound control applications over any kind of acoustic networks. A network of acoustic devices have specific constraints on communication and data processing, but in contrast, it allows for more flexible and versatile deployments in order to render an improved experience to the listener. The GTAC main lines of action have been articulated through the execution of research projects, the presentation of one PhD thesis and the publication of several results in scientific journals and conference proceedings. The most important results of the GTAC research production are summarized in the following. We encourage the reader to visit our webpage: **www.gtac.upv.es** where he will find a full list of projects and papers related to acoustic networks and other research lines.

1.- Project activities

The GTAC research during the course 2016/17 has been mainly related to the implementation of sound processing applications on wireless acoustic sensor networks (WASN):

- Development of sound signal processing algorithms for Smart Sound Processing

We have developed algorithms for cancelling acoustic noise in particular, and for controlling any sound field in general, in order to be implemented in acoustic sensors networks. We have also developed equalization algorithms for narrow band signals that provide different equalization profiles depending on the zone or the specific location of the listener, but avoiding the interference with other zones or listeners. Moreover, the requirement of real-time data acquisition and processing has led to the refinement of previous adaptive algorithms, which now have been designed to work by blocks of samples. Another severe constraint in the acoustic field is the power limitation of the electroacoustic systems, which has motivated the study and design of distributed algorithms incorporating strategies of effort control to not

Figure 1. Prototype of sound field control system for two listeners. Location: GTAC listening room.

saturate the loudspeakers. As a result, we have built a prototype of a multi-ambient sound field control system consisting of two car seats where active noise control is applied to each headrest as it can be seen in Figure 1.

Regarding the problem of cancelling the crosstalk (interference) when two or more loudspeakers are emitting, a crosstalk canceller application has been developed for a network of two Android devices and two wireless loudspeakers. The application includes a setup stage of acoustic channel estimation and a new protocol for the synchronization of the WiFi link between the Android devices and the two Bluetooth links with both wireless loudspeakers. The capacity of tracking the acoustic channel when the listener moves from his position has been studied through simulations, and it will be implemented in the two-mobile network soon.

Modelling, design and analysis of wireless acoustic sensor networks (WASN)

The creation of personal sound zones has traditionally been understood as creating one or more zones of silence (dark zones) and a zone of sound (bright zone) inside the same room by using an array of loudspeakers. The approximations to this problem are based mainly on the control of the sound field at a certain frequency (plane wave). We have proposed a new system model that includes convex optimization algorithms to extend the problem of creating personal sound zones to the case of broadband signals such as the sound signals. We intend to develop techniques that allow for a parametric classification of the WASNs in order to efficiently apply specific sound processing algorithms.

Regarding the practical aspects of a WASN, we have studied the synchronization limitations as well as the latency in data acquisition and processing capabilities of different platforms based on Texas Instrument DSP, Raspberry architectures (both low-cost) and Android-based devices.

International collaborations of the GTAC in the field of smart sound processing:

- With the group of Professor Marc Moonen, Head of ESAT-STADIUS, Stadius Center for Dynamic Systems, Signal Processing and Data Analytics, KUL, Leuven, Belgium. This collaboration has allowed deepening the estimation of parameters for distributed networks applied to sound field control, and has resulted in a joint research under D-NOISE project granted by the EU-H2020 FET programme. Prof. Moonen and part of his team visited iTEAM on June 2017, whereas the GTAC PhD student Christian Antoñanzas spent 3 months (March-May 2017) at the KUL doing a research stay.

- With the group of Professor Patrick Naylor, Head of the Speech and Audio Processing Group of the Imperial College (London, UK). The collaboration started thanks to the stay as a visiting professor of GTAC member Gema Piñero from April to July 2016 to the Imperial College. Her work focused on adaptive methods to estimate and track the acoustic channel for a crosstalk canceller based on a network of acoustic nodes.

1.1.- Ongoing projects

Name of the project: *SMART SOUND PROCESSING FOR THE DIGITAL LIVING (SSPRESSING)*

Webpage of the project: *http://sspressing.upv.es*

Summary of the project: *The main challenge for researchers, independently from their field, is to contribute to scientific advances that have an impact on the quality of life of citizens. The "smart" concept, so widely used, includes in itself this goal and, in this sense, the concept smart sound processing is here included to define the intelligent generation and/or capture of sound signals by means of signal processing/computing devices potentially heterogeneous, distributed and of massive use. This approach, together with network intelligence at different levels, is able to manage data, to make decisions and to configure capture/reproduction/computing devices.*

The project is carried out in collaboration with three research groups of the University of Alcalá de Henares, University of Oviedo and University of Jaen respectively. It started on January 1st 2016 and will finish on December 31st 2018.

Funding entity: *MINISTERIO DE ECONOMIA INDUSTRIA Y COMPETITIVIDAD (TEC2015-67387-C4-1-R)*

Name of the project: *High Performance Computing and Communication for Engineering Applications.*

Webpage of the project: *http://www.inco2.upv.es/prometeo (Spanish only)*

Summary of the project: *This project focuses on the design, development and implementation of products, systems, programs and algorithms that make use of advanced techniques of last generation architectures, advanced computing and efficient communications and have applicability in the field of Engineering. The project takes advantage of the experience of the INCO2 (Interdisciplinar Computation and Communication Group) acquired in the development of a previous Excellent Science PROMETEO project (2009 Call) entitled "High Performance Computing on Current Architectures in Multiple Signal Processing Problems".*

Figure 2. Kick-off meeting of the SSPRESSING project held in Valencia, June 2016.

The current project continues the research in this field combining High Performance Computing and Communication with the development of sustainable applications. As the main thematic lines of the project we can highlight:

- Distributed control systems for sound applications.
- Detection and decoding algorithms in MIMO communications systems.
- Useful Matrix Algorithms in Engineering and its organization in the form of libraries.
- Sustainable Computing Tools for Numerical Linear Algebra.
- Source detection, tracking and separation algorithms.

Funding entity: Regional Government – Call for Excellent Science projects (PROMETEOII/2014/003).

Name of the project: DISTRIBUTED NETWORK OF ACTIVE NOISE EQUALIZERS FOR MULTI-USER SOUND CONTROL (D-NOISE)

Webpage of the project: https://www.d-noise-fet.eu

Summary of the project: The project is aimed at developing a novel wireless acoustic sensor network (WASN) for multi-user active noise equalization without the need of using headsets. This system is formed by multiple active noise equalizers (ANEs), which act as a node in a WASN-type set up and cooperate to simultaneously solve their node-specific noise equalization problems. By doing so, the proposed system can simultaneously ensure the auditory comfort of multiple users that are regularly exposed to high noise levels. This is radically different from the current multi-user systems for ANE where each node aims at ensuring the auditory comfort of each passenger without cooperation.

To implement the proposed multi-user ANE, the project will rely on distributed algorithms derived within the EU-FET HANDiCAMS project. These algorithms build on node-specific cooperation rules to allow the cooperation of devices interested in different signal processing tasks and let them attain superior performance as compared to the case where they would operate on their own. In particular, through these node-specific cooperation rules, these algorithms yield a system that overcomes the two main limitations of the existing technologies. First, the spatial diversity of all the microphone signals can be leveraged and hence superior levels of noise attenuation can be attained in the active noise equalization problem of each node. Second, there is no performance degradation due to the acoustical interaction between ANEs of different users, which allows to perform simultaneous ANEs in scenarios with a high density of users.

Due to its competitive advantages, the current levels of noise reduction can be attained with much simpler and cheaper ANEs, which will make the technology more accessible and will enhance the commercialization of ANEs on grand-scale. Thus, the proposed project is expected to constitute a strong seed for the creation of jobs, which will encourage the entrepreneurship in the FET research world.

Funding entity: *European Commission (EU FET Innovation Launchpad programme).*

2.- Research results

2.1.- Featured Journal publications

Distributed Affine Projection Algorithm Over Acoustically Coupled Sensor Networks. *M. Ferrer, A. González, M. de Diego, G. Piñero, IEEE Trans.s on Signal Processing, available online http://ieeexplore.ieee.org/document/8013848/, 2017.*

Abstract*: In this paper we present a distributed affine projection (AP) algorithm for an acoustic sensor network where the nodes are acoustically coupled. Every acoustic node is composed of a microphone, a processor and an actuator to control the sound field. This type of networks can use distributed adaptive algorithms to deal with the active noise control (ANC) problem in a cooperative manner, providing more flexible and scalable ANC systems. In this regard, we introduce here a distributed version of the multichannel filtered-x AP algorithm over an acoustic sensor network that it is called distributed filtered-x AP (DFxAP) algorithm. The analysis of the mean and the mean-square deviation performance of the algorithm at each node is given for a network with a ring topology and without constraints in the communication layer. The theoretical results are validated through several simulations. Moreover, simulations show that the proposed DFxAP outperforms the previously reported distributed multiple error filtered-x least mean square (DMEFxLMS) algorithm.*

DOI: 10.1109/TSP.2017.2742987.

GPU-Based Dynamic Wave Field Synthesis Using Fractional Delay Filters and Room Compensation. *J.A. Belloch, A. Gonzalez, E.S. Quintana-Ortí, M. Ferrer, V. Välimäki, IEEE Trans. on Audio, Speech, and Language Processing, vol. 25, no. 2, pp. 435-447, 2017.*

Abstract*: Wave field synthesis (WFS) is a multichannel audio reproduction method, of a considerable computational cost that renders an accurate spatial sound field using a large number of loudspeakers to emulate virtual sound sources. The moving of sound source locations can be improved by using fractional delay filters, and room reflections can be compensated by using an inverse filter bank that corrects the room effects at selected points within the listening area. However, both the fractional delay filters and the room compensation filters further increase the computational requirements of the WFS system. This paper analyzes the performance of a WFS system composed of 96 loudspeakers which integrates both strategies. In order to deal with the large computational complexity, we explore the use of a graphics processing unit (GPU) as a massive signal co-processor to increase the capabilities of the WFS system. The performance of the method as well as the benefits of the GPU acceleration are demonstrated by considering different sizes of room compensation filters and fractional delay filters of order 9. The results show that a 96-speaker WFS system that is efficiently implemented on a state-of-art GPU can synthesize the movements of 94 sound sources in real time and, at the same time, can manage 9216 room compensation filters having more than 4000 coefficients each.*

DOI: 10.1109/TASLP.2016.2631338

Blockwise Frequency Domain Active Noise Controller Over Distributed Networks. *C. Antoñanzas, M. Ferrer, M. de Diego, A. Gonzalez, Applied Sciences, vol. 6, no.5, 124, 2016.*

Abstract*: This work presents a practical active noise control system composed of distributed and collaborative acoustic nodes. To this end, experimental tests have been carried out in a listening room with acoustic nodes equipped with loudspeakers and microphones. The communication among the nodes is simulated by software. We have considered a distributed algorithm based on the Filtered-x Least Mean Square (FxLMS) method that introduces collaboration between nodes following an incremental strategy. For improving the processing efficiency in practical scenarios where data acquisition systems work by blocks of samples, the frequency-domain partitioned block technique has been used. Implementation aspects such as computational complexity, processing time of the network and convergence of the algorithm have been analyzed. Experimental results show that, without constraints in the network communications, the proposed distributed algorithm achieves the same performance as the centralized version. The performance of the proposed algorithm over a network with a given communication delay is also included.*

DOI: 10.3390/app6050124

Adaptive Filtered-x Algorithms for Room Equalization Based on Block-Based Combination Schemes. *L. Fuster, M. de Diego, L.A. Azpicueta-Ruiz, M. Ferrer, IEEE Trans. on Audio, Speech, and Language Processing, vol. 24, n.10, pp. 1732-1745, 2016.*

Abstract: *Room equalization has become essential for sound reproduction systems to provide the listener with the desired acoustical sensation. Recently, adaptive filters have been proposed as an effective tool in the core of these systems. In this context, this paper introduces different novel schemes based on the combination of adaptive filters idea: a versatile and flexible approach that permits obtaining adaptive schemes combining the capabilities of several independent adaptive filters. In this way, we have investigated the advantages of a scheme called combination of block-based adaptive filters which allows a blockwise combination splitting the adaptive filters into nonoverlapping blocks. This idea was previously applied to the plant identification problem, but has to be properly modified to obtain a suitable behavior in the equalization application. Moreover, we propose a scheme with the aim of further improving the equalization performance using the a priori knowledge of the energy distribution of the optimal inverse filter, where the block filters are chosen to fit with the coefficients energy distribution. Furthermore, the biased block-based filter is also introduced as a particular case of the combination scheme, especially suited for low signal-to-noise ratios (SNRs) or sparse scenarios. Although the combined schemes can be employed with any kind of adaptive filter, we employ the filtered-x improved proportionate normalized least mean square algorithm as basis of the proposed algorithms, allowing to introduce a novel combination scheme based on partitioned block schemes where different blocks of the adaptive filter use different parameter settings. Several experiments are included to evaluate the proposed algorithms in terms of convergence speed and steady-state behavior for different degrees of sparseness and SNRs.*

DOI: 10.1109/TASLP.2016.2583065

2.2.- Featured Conference Proceedings

Channel estimation for crosstalk cancellation in wireless acoustic networks, *G. Piñero, P. Naylor, 42th Int. Conf. on Acoustics, Speech, and Signal Processing (ICASSP 2017), New Orleans, USA, 2017.*

Abstract: *In this paper we deal with the estimation of the room impulse response (RIR) between each loudspeaker and each microphone of a wireless acoustic network of two nodes when used to implement a crosstalk canceller. The nodes of the network are commercial devices connected via standard wireless links, presenting low computational requirements and non-ideal synchronization between them. Moreover, the nodes can exchange information, but they cannot share their signals due to the high throughput and perfect synchronism that would be required. The proposed scheme adaptively estimates the global impulse response between the source signals and the recorded signal at each node of the network, and afterwards estimates the corresponding RIRs between each loudspeaker and the node's microphone. This scheme does not need any additional synchronism between loudspeakers. Simulations show that proportionate-type affine projection algorithms obtain good performance for order N = 4, being their cost affordable in commercial devices.*

DOI: 10.1109/ICASSP.2017.7952223

Collaborative method based on the acoustical interaction effects on active noise control systems over distributed networks. *C. Antoñanzas, M. Ferrer, M. de Diego, A. González, 42th Int. Conf. on Acoustics, Speech, and Signal Processing (ICASSP 2017), New Orleans, USA, 2017.*

Abstract: *This paper focus on the implementation of an active noise control system over a network of distributed nodes acoustically coupled. We have considered the distributed Multiple Error filtered-x Least Mean Square (DMEFxLMS) algorithm that allows collaboration between nodes following an incremental strategy. To reduce the computational network requirements, an alternative strategy which brings together only the acoustically coupled nodes has been used. However, a rule to define the subsets of nodes that are required to collaborate is needed. A collaborative condition based on the analysis in the frequency domain of the eigenvalues of the acoustic path matrix is proposed. Results demonstrate the ability of the introduced method to define the subsets of nodes that collaborate.*

DOI: 10.1109/ICASSP.2017.7952227

On the feasibility of personal audio systems over a network of distributed loudspeakers. *G. Piñero, C. Botella, M. de Diego, M. Ferrer, A. Gonzalez, 25th European Signal Processing Conference (EUSIPCO 2017), Kos, Greece, 2017.*

Abstract: *Personal audio reproduction systems deal with the creation of personal sound zones within a room without the necessity of using headphones. These systems use an array of loudspeakers and design the required filters at each loudspeaker in order to render the desired audio signal to each person in the room as free of interferences as possible. There are very interesting proposals in the literature that make use of circular or linear arrays, but in this paper we study the problem considering a network of distributed loudspeakers controlled by a set of acoustic nodes, which can exchange information through a network. We state the model of such a distributed system by considering the electro-acoustic paths between the loudspeakers and each microphone, and try to provide a minimum signal- to-interference-and-noise ratio (SINR) to each zone, but constraining the emitted power of the loudspeakers to a maximum value (avoiding annoying feedback effects). We make use of optimization techniques to decide if, given a distribution of the loudspeakers and a location of the personal sound zones within the room, the system will be feasible. Simulations are done to support the use of the proposed optimization techniques.*

http://www.eurasip.org/Proceedings/Eusipco/Eusipco2017/papers/1570341655.pdf