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PERSONAL ACTIVE CONTROL SYSTEMS OVER COUPLED NETWORKS

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This work presents the practical implementation of personal active noise control systems using collaborative acoustic nodes. To this end, experimental tests have been carried out in a listening room where two personal active control (PAC) systems have been deployed. Each PAC is composed of a car seat and sound control node (SCN) equipped with: processor, loudspeakers, microphones and network access. The SNC intends to cancel out an external disturbance noise at the listener's ears location (around the headrest). Distributed PAC can be especially suitable instead of massive multichannel active noise controllers since they are versatile, scalable, eventually heterogeneous, and allow to distribute and optimize the algorithm computing cost. On the other hand acoustical coupling between PACs can lead them to unstability under certain conditions, consequently cooperative strategies have to be considered. In this paper we analyze strategies based on information exchanges between nodes following an incremental strategy in order to ensure PAC stability. A practical version of a distributed algorithm based on the filtered-x Least Mean Square (FxLMS) method was developed. It uses the frequency-domain partitioned block technique. Experimental results show that the proposed approach assures PAC convergence in cases where the fully decentralized method diverges. Its performance has been analyzed and tested for several PAC configurations within the listening room.

Keywords: active noise control, personal audio, distributed networks, filtered-x Least Mean Square,

1. Introduction

Active noise reduction techniques have been gaining an increasing importance due to the health problems related to the exposure to high levels of noise from industrial enviroment or traffic noise, among others. Hearing loss, stress, sleep disturbance or aggressive behaviors are some of the adverse effects of noise on human health [1]. Based on the superposition principle of sound waves, active noise control (ANC) systems try to reduce the sound level of a undesired noise at a specific control point [2]. ANC systems commonly use adaptive algorithms [3] to generate the anti-noise signal (with the same amplitude and opposite phase of the noise) from a reference signal that is correlated with the disturbance signal. Based on this and due to the comfort problem of active headphones, ANC headrest systems have been developed to reduce the noise level at listener's ears [4] [5] [6]. These systems usually use two fixed actuators to generate the anti-noise or control signals and two fixed sensors to capture the error signals as well as either a reference microphone signal (feedforward control) [7] or an internally estimated signal (feedback control) [8]. The zones of quiet are generated in a limited areas, specifically at close points around the error sensors with a diameter of about $\lambda/10$ [9] (where λ is the wavelength of the highest frequency of the noise), although out of these zones, noise level may even increase [10]. Generally, ANC headrest systems are designed to be finally installed in the cabin of a public transport (train, airplane, bus, etc.). Therefore, the use of many of these systems working at the same time are required in order to create quiet zones in all the occupants positions. In addition, the seats in those cabins are close to each other, so the overlapping between adjacent quiet zones will be unavoidable. Consequently, the addition of more ANC headrest systems or even the use of these system is reverberant enclosures may cause performance degradation of the ANC application because of the different acoustic channels involved will be acoustically coupled. On the other hand, a group of several ANC headrest systems can be seen as a unique multichannel ANC headrest system controlled by a single centralized processor. These multichannel system require a high computational capacity to capture, process and generate all the signals involved on the control process. However, the centralized multichannel ANC headrest system can be divided into a distributed network composed of two-channel nodes where the computational cost is divided into them. In the context of our work, we define a personal active control (PAC) system as a two-channel distributed ANC system composed of a seat car and a sound control node (SCN). A SCN is an acoustic device composed of two loudspeakers to generate acoustic signals, two microphones to obtain acoustical information, and one processor to individually process that information and to interchange it with the other SCN using a suitable communication network. Note that the selection of the network topology will affect how data is processed by each SCN. The PAC systems use a feedforward control hence the undesired noise is monitored by a sensor that is called reference sensor. As previously commented, the interaction between PAC systems may lead to the unstability of the system. If the PAC systems are located at such a distance that does not allow the acoustic interaction among them (uncoupled system), each SCN may perform a distributed and independent processing by using its local information to achieve the stability of the system (non-collaborative strategy). The problem arises when the level of acoustic interaction increases (coupled system). In those cases, the use of the non-collaborative strategy makes the PAC systems unstable. On the contrary, a collaborative strategy allows that each SCN can update its status by using its local data and assuming some collaboration with its neighbouring SCN to minimize the effects of the acoustic coupling and ensuring the stability of the ANC system. It must not be forgotten that, as well as the interference problem, the characterization of the acoustic environment involves an additional uncertainty to take into account in sound field control [11]. Note that, for a personal audio system, at least 11dB between the different zones is required to provide adequate separation [12]. In [13], a similar ANC application composed of one PAC system was experimentally tested achieving an attenuation of 10 dB around the headrest of a car seat. However, it was demostrated in [14] that the subjective improvement of the ANC system strongly depends on the power spectrum of the controlled sound, and not just the attenuation level achieved. About the proposed algorithmic approach, we consider the most commonly used algorithm for ANC applications, the filtered-x Least Mean Squares (FxLMS) [15] strategy but using a Frequency-domain Partitioned Block technique for the filtering operation (FPBFxLMS) [16]. The reason to use this technique is because most of the common audio cards work with block data buffers. Moreover, hardware devices, such as Digital Signal Processors (DSPs) or Graphics Processing Units (GPUs), in addition to using blocks of samples, work with libraries of frequency-domain operations for a more efficient processing.

In summary, this paper reports a description of the design and implementation of a distributed ANC system developed to create quiet zones in the vicinity of the left and right ears of two listeners. Future applications of the prototype described in this work may be the installation of the PAC systems in cabins of public transport where the separation between actuators and sensors are critical due to the level of acoustic interaction between them. The paper is organized as follows: In Section 2 the prototype description is given. In Section 3 we present both the non-collaborative and the collaborative FPBFxLMS algorithms for a distributed ANC system composed of two SCNs. The experimental results to compare the performance of both algorithms are shown in Section 4. Finally, Section 5 outlines the main conclusions of the present work.

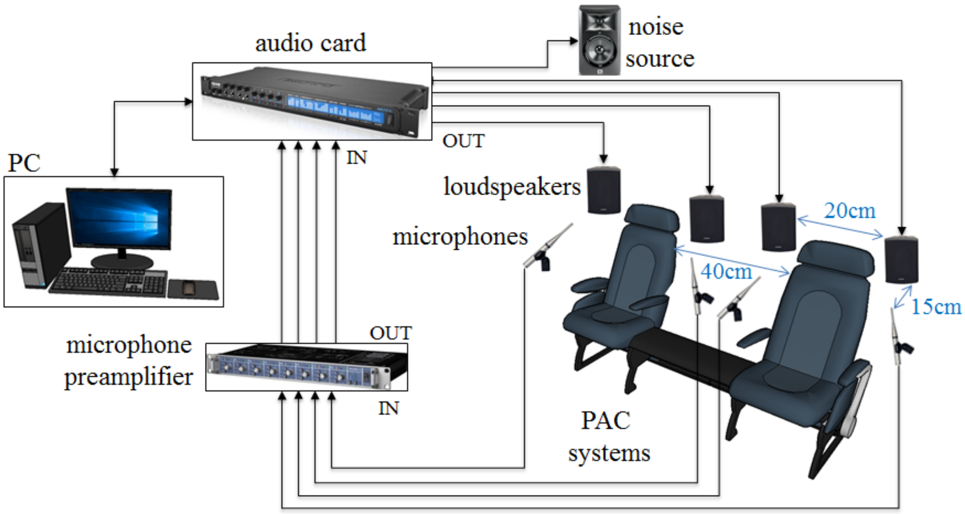
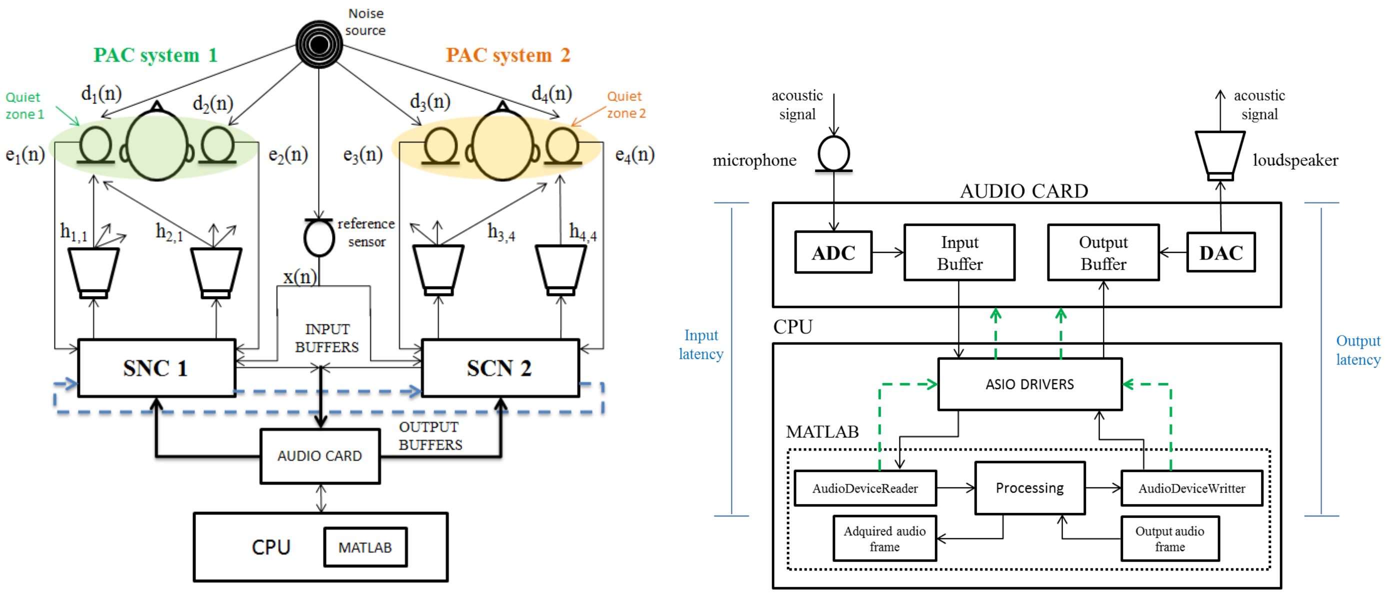


Figure 1: Scheme of the ANC prototype.

2. Description of the PAC systems

The configuration of the distributed ANC application is depicted in Figure 1. Two PAC systems composed of a car seat and a SCN were considered. Each SCN is equipped with two actuators, two sensors and one processor with computation and communication capability. The aim of the PAC systems is to create independent quiet zones by minimizing an external disturbance noise at the sensors's location. The PAC systems have been tested inside a listening room of $9,36$ meters long, $4,78$ meters wide and $2,63$ meters high, located at the Audio Processing Laboratory of the Polytechnic University of Valencia. The SCN actuators consists of two hi-fi autoamplified loudspeakers APART model SDQ5P. The election of these loudspeakers is based on a tradeoff between good low-frequency response, small size, enough sound power level and easy mounting system. The loudspeakers were positioned adjacent to the seat car in the rear of the headset, as shown in Figure 1 and with a separation of 40 cm between SNC loudspeakers. As error sensors, two electrec condenser microphones Behringer model ECM8000 with omnidirectional pattern and suited for measurements applications were selected. Two of them were positioned close to each headrest, at close location around the listener ears (see Figure Figure 1) in order to create the desired personal quiet zones. The microphones were separated 20 cm away from the loudspeakers and the distance between the SNC microphones was 20 cm. To generate the disturbance noise to cancel out by the ANC application, an autoamplifier loudspeaker JBL model LSR305 has been used. A KU 100 dummy head manufactured by Neuman has been used to monitor the impact of the system over a listener placed in the control area. It resembles the human head and has two high-quality microphone capsules built into the ears.

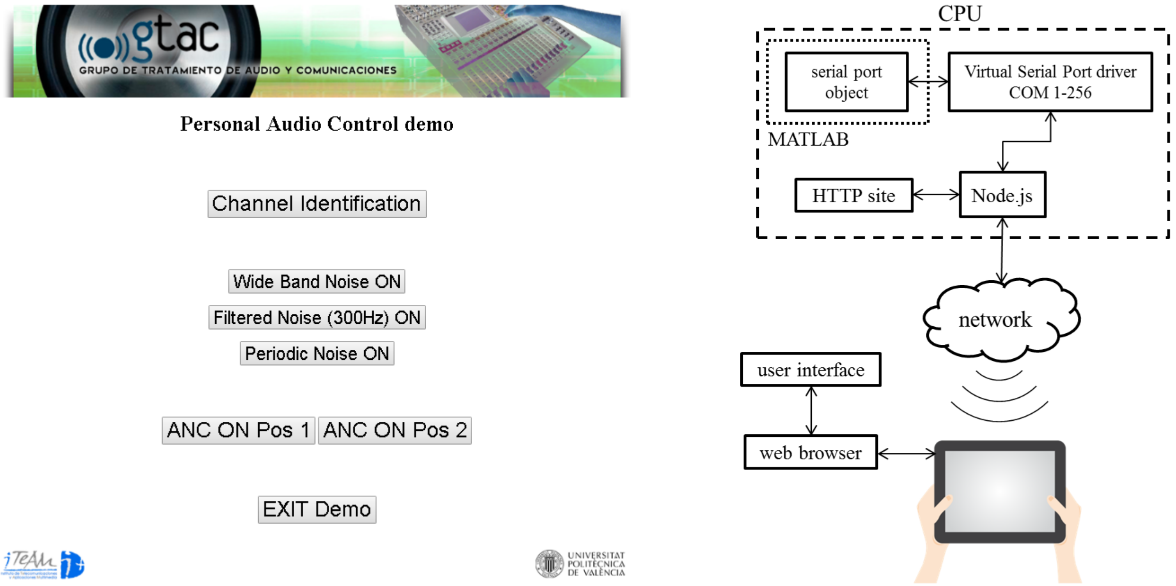
The acoustical signals picked up by the microphones are sent to the inputs of the microphone preamplifier RME OctaMic II to increase the microphone signal to the required line level of the next device of the audio chain. the audio card Motu 16 AVB. The outputs of the audio card are connected to the loudpspeakers of the PAC systems. A diagram of the audio data flow can be seen in Figure 2b. The audio card stores the input data from the sensor of each node in buffers and send them to the CPU through the ASIO drivers. The CPU, with MATLAB® support, executes the audio processing, saves the output data in buffers and sends them back to the audio card through the ASIO drivers, to be reproduced by the loudspeaker of each node. The addition of the recent Audio System Toolbox [17] in the computing environment MATLAB® provides multichannel real-time audio recording, processing and reproduction at low latency. Audio objects based on Object Oriented Programming (OOP) have been optimized for iterative computations that process large streams of audio data. Moreover, ASIO drivers [18] have been incorporated to this software providing a low-latency and high fidelity interface



(a) (b)

Figure 2: (a) Two personal audio control (PAC) systems composed of two-channel sound control nodes (SCN) each of them. Dashed blue lines represent the incremental communication strategy. (b) Audio data flow. The communication of the configuration parameters is represented by dashed green lines.

between MATLAB® and audio card. The communication between the CPU (Intel Core i7 3.07 GHz) and the audio card is performed by ASIO drivers and it is controlled by using the MATLAB® System objects (*AudioDeviceWriter* and *AudioDeviceReader*) provided by the Audio System Toolbox of MATLAB® software. Note that the selection of the frame size in the audio card is critical to determine the latency of the application. The latency is the sum of the time spent on storing data in the input buffers, on processing these data and on sending them to the output buffers as well. A real-time application must satisfy that the time spent to fill up the input buffers (buffering time) was higher than the time spent in data processing (processing time). The buffering time is the ratio between the frame size and the sampling rate of the audio card, fixed at 44.1 kHz as the lowest possible rate because of the limitations of hardware (though the sample rate could be lower due to the developed application). The audio card offers different values of frame size between 8 and 1024 samples. Due to the real-time condition explained previously, we have selected 1024 as a tradeoff between the processing time and the buffering time. Note that, although each PAC should carry its own processing unit as described in the prototype, in the experimental tests presented below, we have focused on the experimentation of both the acoustic and the processing models. Therefore, we have simulated the communications model between PACs using the same central processing unit (CPU), as it has been shown in Figure 2.b. A web interface which involves real-time transfer of data from a web browser to Matlab® has been developed. An important benefit brought by such web interface is that it allows us to control independently each PAC system in real time. For example, it enables to minimize the undesired noise at the right seat control zone while, at the left seat control zone, the noise is present and vice versa. The web interface has been built as an HTTP site with a simple user design (see Figure 3.a) using HTML language. Node.js® [19] is a stable web serving platform running on Javascript® that offers solutions to building servers and web/mobile applications. Thanks to the software Eltima Virtual Serial Port Driver®, it is possible to create up to 254 virtual serial ports (depending on the license) by emulating all the settings of the real ones. By creating a serial port object in Matlab® associated with a virtual serial port, it is possible to communicate with the web interface through Node.js® (see Figure 3.b).



(a) (b)

Figure 3: User interface design (a) and interface communication (b) of the developed web interface

3. Description of the algorithms

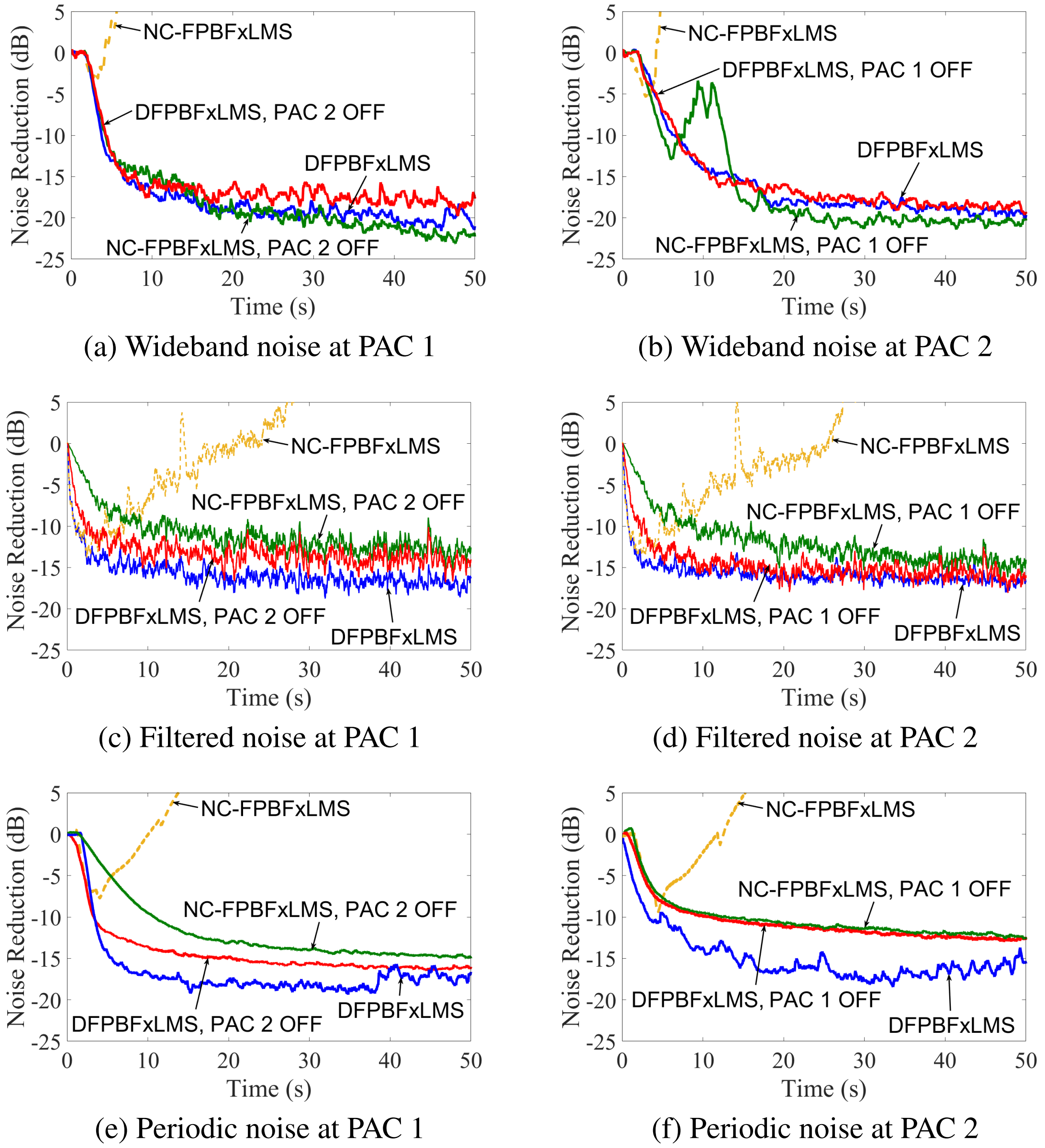
Considering a PAC system composed of a single SCN, (see PAC system 1 in Figure 2a). We aim at cancelling the acoustic noise signal in the sensors location, $d_1(n)$ and $d_2(n)$, designing two adaptive filters at the SCN, $\mathbf{w}_1(n)$ and $\mathbf{w}_2(n)$, where each one contains L coefficients at the discrete time instant n. For simplicity, we consider one disturbance noise signal captured by a monitoring sensor and denoted as the reference signal $x(n)$. The centralized SCN recursive solution for the two adaptive filters is stated as follows

$$\begin{bmatrix}\mathbf{w}_1(n)\\ \mathbf{w}_2(n)\end{bmatrix} = \begin{bmatrix}\mathbf{w}_1(n-1)\\ \mathbf{w}_2(n-1)\end{bmatrix} - \mu \begin{bmatrix}\mathbf{v}_{11}(n) & \mathbf{v}_{12}(n)\\ \mathbf{v}_{21}(n) & \mathbf{v}_{22}(n)\end{bmatrix} \begin{bmatrix}e_1(n)\\ e_2(n)\end{bmatrix}, \tag{1}$$

where μ is the step-size parameter, vector $\mathbf{v}_{ij}(n)$ contain the last L samples of the reference signal $x(n)$ filtered through the estimated acoustic channel $\mathbf{s}_{ij}$ that links the ith actuator with the jth sensor (where $i = j = 1, 2$). $\mathbf{s}_{ij}$ is defined as the FIR filter that models the estimation of the real acoustic channel $\mathbf{h}_{ij}$. The error signals $e_1(n)$ and $e_2(n)$ are the signals recorded at the sensors 1 and 2, respectively. Consider now adding another PAC system close to the previous one (as it is depicted in Figure 2a). We assume that the two SNC have the same computation and communication capabilities, and execute the same algorithm. If the SCNs work independently and they do not collaborate at all, the filter updating equation of the non-collaborative distributed multiple error FxLMS algorithm (NC-DMEFxLMS) ([20]) for the whole network is stated as follows

$$\begin{bmatrix}\mathbf{w}_1(n)\\ \mathbf{w}_2(n)\\ \mathbf{w}_3(n)\\ \mathbf{w}_4(n)\end{bmatrix} = \begin{bmatrix}\mathbf{w}_1(n-1)\\ \mathbf{w}_2(n-1)\\ \mathbf{w}_3(n-1)\\ \mathbf{w}_4(n-1)\end{bmatrix} - \mu \begin{bmatrix}\mathbf{v}_{11}(n) & \mathbf{v}_{12}(n) & 0 & 0\\ \mathbf{v}_{21}(n) & \mathbf{v}_{22}(n) & 0 & 0\\ 0 & 0 & \mathbf{v}_{33}(n) & \mathbf{v}_{34}(n)\\ 0 & 0 & \mathbf{v}_{43}(n) & \mathbf{v}_{44}(n)\end{bmatrix} \begin{bmatrix}e_1(n)\\ e_2(n)\\ e_3(n)\\ e_4(n)\end{bmatrix}. \tag{2}$$

However, if the SCNs are so close that makes the system is coupled, a possible solution to ensure the system stability is by using a collaborative method that allows information exchange between the SCNs in order to minimize the acoustic coupling. In those cases, the filter updating equation of the distributed multiple error FxLMS algorithm (DMEFxLMS) using an incremental strategy on the



(a) Wideband noise at PAC 1 (b) Wideband noise at PAC 2

(c) Filtered noise at PAC 1 (d) Filtered noise at PAC 2

(e) Periodic noise at PAC 1 (f) Periodic noise at PAC 2

Figure 4: Noise reduction obtained by both the decentralized PAC systems and the distributed PAC systems for different disturbance noises.

network communication is described in [20] and is stated for the two SCNs as follows

$$\begin{bmatrix} \mathbf{w}_1^1(n) \\ \mathbf{w}_2^1(n) \\ \mathbf{w}_3^1(n) \\ \mathbf{w}_4^1(n) \end{bmatrix} = \begin{bmatrix} \mathbf{w}_1^2(n-1) \\ \mathbf{w}_2^2(n-1) \\ \mathbf{w}_3^2(n-1) \\ \mathbf{w}_4^2(n-1) \end{bmatrix} - \mu \begin{bmatrix} \mathbf{v}_{11}(n) & \mathbf{v}_{12}(n) \\ \mathbf{v}_{21}(n) & \mathbf{v}_{22}(n) \\ \mathbf{v}_{31}(n) & \mathbf{v}_{32}(n) \\ \mathbf{v}_{41}(n) & \mathbf{v}_{42}(n) \end{bmatrix} \begin{bmatrix} e_1(n) \\ e_2(n) \\ 0 \\ 0 \end{bmatrix}, \tag{3}$$

$$\begin{bmatrix} \mathbf{w}_1^2(n) \\ \mathbf{w}_2^2(n) \\ \mathbf{w}_3^2(n) \\ \mathbf{w}_4^2(n) \end{bmatrix} = \begin{bmatrix} \mathbf{w}_1^1(n) \\ \mathbf{w}_2^1(n) \\ \mathbf{w}_3^1(n) \\ \mathbf{w}_4^1(n) \end{bmatrix} - \mu \begin{bmatrix} \mathbf{v}_{13}(n) & \mathbf{v}_{14}(n) \\ \mathbf{v}_{23}(n) & \mathbf{v}_{24}(n) \\ \mathbf{v}_{33}(n) & \mathbf{v}_{34}(n) \\ \mathbf{v}_{43}(n) & \mathbf{v}_{44}(n) \end{bmatrix} \begin{bmatrix} 0 \\ 0 \\ e_3(n) \\ e_4(n) \end{bmatrix}, \tag{4}$$

where $\mathbf{w}_p^k(n)$ is a local version of the adaptive filter coefficients of each kth SCN of the network (for $k = 1, 2$ and $p = 1, 2, 3, 4$). The details about the frequency-domain partitioned block version of these algorithms can be seen in [21][22].

4. Experimental Results

In this section, we show the experiments carried out to validate the performance of the distributed ANC application. Multiple measurements have been carried out by using different disturbance noises to evaluate the performance of described distributed algorithms in terms of stability and convergence rate. The experimental PAC systems have been deployed in the sound conditioned room described in Section 2. Real acoustic responses between all the loudspeakers and all microphones are identified off-line using adaptive methods and modeled as FIR filters of $M = 1024$ coefficients at a sampling rate of 44.1 kHz. We have considered three diferent types of synthesized noises as disturbance signals: 1) a wideband zero-mean Gaussian white noise with unit variance, 2) a zero-mean Gaussian white noise with unit variance limited to 200 Hz, and 3) a periodic noise of a fundamental tone of 20 Hz and a five harmonics, trying to simulate an engine noise. All the disturbance signals are emitted by a loudspeaker located in front of the PAC sytems. Furthermore, we have considered a block size of $B = 1024$ and a Fast Fourier Transform (FFT) of the adaptive filter with a size of $2B$. A constant step size parameter of $\mu_1 = 3 \cdot 10^{-5}$, $\mu_2 = 6 \cdot 10^{-5}$ and $\mu_3 = 10 \cdot 10^{-5}$ for the three type of noises respectively, as the highest value that ensures the stability of the algorithms, have been used.

Figure 4 illustrates the time evolution of the noise reduction of the ANC system in dB for the Frequency-domain Partitioned Block version of both non-collaborative and collaborative distributed algorithms (NC-FPBFxLMS and DFPBFxLMS, respectively) using the different disturbance noises. As expected, in a coupled environment, the non-collaborative algorithm makes the PAC systems unstable for the three type of noises. However, if both SCNs collaborate, the PAC systems achieve the stability showing a robust and stable performance for all the disturbance noises. In the cases where only one of the PAC systems is trying to cancel the noise while the other one is off, for all the cases, both collaborative and non-collaborative algorithms present similar results achieving a stable behavior. In summary, the performance of the coupled PAC systems improves when there is collaboration between the SNC's.

5. Conclusions

In this paper, a description of the design and implementation of a distributed ANC application based on two PAC systems has been presented. These systems, composed of a seat car and a two-channel SCN, have been mounted inside a sound-conditioned room. The aim of each PAC system is to create a local quiet zone close to the listener's ears. It has been demostrated that the ANC application suffers strong performance degradation when PAC systems are acoustically coupled. In those cases, a collaborative incremental strategy among the SCNs has been used to ensure the stability of the systems. In order to evaluate the performance of the distributed algorithms over the PAC systems, we have implemented a real-time adaptive noise controller based on the audio toolbox provided by Matlab®. Experimental results show that, in acoustically coupled enviroments, the collaborative implementation avoids the unstability of the fully decentralized strategy achieving the convergence of the PAC systems. However, a subjective evaluation method to validate the performance of the application must be considered in future works.

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REFERENCES

1. Cohen, S. and Weinstein, N. Nonauditory effects of noise on behavior and health, *Journal of Social Issues*, **37** (1), 36–70, (1981).

2. Elliott, S. and Nelson, P. Active noise control, *IEEE Signal Process. Mag.*, **10** (4), 12–35, (1993).

3. Haykin, S., *Adaptive filter theory*, Prentice-Hall, Upper Saddle River, NJ, US (2002).

4. Rafaely, B., Elliott, S. J. and Garcia-Bonito, J. Broadband performance of an active headrest, *The Journal of the Acoustical Society of America*, **106** (2), 787–793, (1999).

5. Pawelczyk, M. Adaptive noise control algorithms for active headrest system, *Control Engineering Practice*, **12** (9), 1101–1112, (2004).

6. Cheer, J. and Elliott, S. Design and implementation of a personal audio system in a car cabin, Acoustical Society of America (ASA), (2013).

7. Misol, M., Bloch, C., Monner, H. and Sinapius, M. Performance of active feedforward control systems in non-ideal, synthesized diffuse sound fields, *The Journal of the Acoustical Society of America*, **135** (4), 1887–1897, (2014).

8. Song, Y., Gong, Y. and Kuo, S. A robust hybrid feedback active noise cancellation headset, *IEEE Transactions on Speech and Audio Processing*, **13** (4), 607–617, (2005).

9. Elliott, S., Joseph, P., Bullmore, A. and Nelson, P. Active cancellation at a point in a pure tone diffuse sound field, *Journal of Sound and Vibration*, **120** (1), 183 – 189, (1988).

10. Joseph, P., Elliott, S. and Nelson, P. Near field zones of quiet, *Journal of Sound and Vibration*, **172** (5), 605–627, (1994).

11. Betlehem, T., Zhang, W., Poletti, M. A. and Abhayapala, T. Personal sound zones: Delivering interface-free audio to multiple listeners, *IEEE Signal Processing Magazine*, **32** (2), 81–91, (2015).

12. Druyvesteyn, W. F. and Garas, J. Personal sound, *J. Audio Eng. Soc*, **45** (9), 685–701, (1997).

13. de Diego, M., Gonzalez, A. and Garcia, C. On the performance of a local active noise control system, *1999 IEEE International Conference on Acoustics, Speech, and Signal Processing. Proceedings. ICASSP99 (Cat. No.99CH36258)*, Institute of Electrical and Electronics Engineers (IEEE), (1999).

14. Gonzalez, A., Ferrer, M., de Diego, M., Piñero, G. and Garcia-Bonito, J. Sound quality of low-frequency and car engine noises after active noise control, *Journal of Sound and Vibration*, **265** (3), 663–679, (2003).

15. Widrow, B. and Stearns, S., *Adaptive signal processing*, Prentice Hall, Englewood Cliffs, NJ, US (1985).

16. Lorente, J., Ferrer, M., de Diego, M. and Gonzalez, A. GPU implementation of multichannel adaptive algorithms for local active noise control, *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, **22**, 1624–1635, (2014).

17. Mathworks. audio system toolbox., *https://es.mathworks.com/products/audio-system.html*.

18. Steinberg, asio., *http://www.steinberg.net/nc/en/company/developers/sdk-download-portal.html*.

19. Node.js, *https://nodejs.org/en/*.

20. Ferrer, M., de Diego, M., Piñero, G. and Gonzalez, A. Active noise control over adaptive distributed networks, *Signal Processing*, **107**, 82–95, (2015).

21. Lorente, J., Antoñanzas, C., Ferrer, M. and Gonzalez, A. Block-based distributed adaptive filter for active noise control in a collaborative network, *Signal Processing Conference (EUSIPCO), 2015 23rd European*, pp. 310–314, IEEE, (2015).

22. Antoñanzas, C., Ferrer, M., de Diego, M. and Gonzalez, A. Blockwise frequency domain active noise controller over distributed networks, *Applied Sciences*, **6** (5), 124, (2016).