AN ADAPTIVE ALGORITHMS COMPARISON FOR REAL MULTICHANNEL ACTIVE NOISE CONTROL

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ABSTRACT

A comparative study of the multichannel Affine Projection (AP), the Fast Transversal Filter (FTF), the filtered-X LMS (FXLMS) and the Recursive Least Squares (RLS) algorithms is presented for active noise control (ANC) systems. This study is based on simulations using real data and laboratory experiments, and is focused on: their computational cost, their convergence properties, their stability and their ability to create quiet zones around listener ears. The performance of the AP algorithm in the real system suggests its use in ANC systems as an alternative to the classical multichannel FXLMS since it provides meaningful attenuation levels, lower convergence time and similar computational cost.

1. INTRODUCTION

The classical filtered-X LMS (FXLMS) [1] algorithm is the most widely used adaptive filtering strategy applied to ANC systems. However, it is well known that recursive least squares (RLS) algorithms [2] produce a faster convergence speed than stochastic gradient descent techniques, such as the FXLMS algorithm, but requires a higher computational cost, minimizing a weighted sum of the past squared error signals. A previous work proposed a multichannel RLS algorithm for ANC and evaluated its performance by means of simulations for a random noise source, [3]. To overcome the computational complexity of the RLS algorithm various schemes have been developed like the fast RLS algorithm and the fast transversal filter (FTF) algorithm [4]. A multichannel FTF algorithm was proposed in [3]. On the other hand, the Affine Projection (AP) algorithm [5] can improve the convergence speed of FXLMS algorithm avoiding the high computational complexity and the instabilities of the RLS algorithms. A multichannel AP algorithm was proposed in [6].

In this work, the multichannel FXLMS, RLS, FTF and AP algorithms have been implemented in an ANC system. Since multichannel ANC systems for local control intend to create a *quiet zone* around a listener head, as first approach it would be meaningful to know the sound levels around the listener head, in addition to the traditional convergence and computational cost studies. Therefore, as a previous step to the practical implementation of a real multichannel ANC system on a DSP card, the attenuation levels at the listener ears within an actively controlled area have been simulated in the present work using real data from a 1:2:2 ANC system, see Figure 1. Furthermore, the multichannel algorithms which exhibited better performance have been evaluated in a real-time system. Those algorithms were the multichannel AP algorithm and the classical multichannel FXLMS. A Bruel & Kjaer mannequin head and torso, illustrated in Figure 2, and a moving platform have provided the measurement of the quiet zones.

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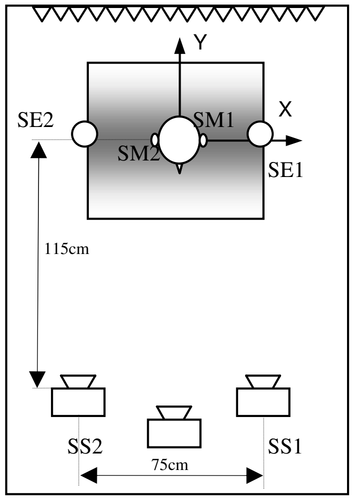


Figure 1: Scheme of the 1:2:2 ANC system

2. MULTICHANNEL ANC ALGORITHMS

A general feedforward multichannel ANC system consisting of J secondary sources, K error sensors and I primary signals [7] will be considered.

The multichannel algorithms evaluated in this work process as inputs the error signals, $e_k(n)$, and a version of the reference signal $x_i(n)$ filtered through an estimation of the secondary paths $\mathbf{h}_{j,k}$, $v_{i,j,k}(n)$, to update the adaptive filters coefficients within each new sampling period. A complete description of the different multichannel algorithms can be found in [3, 6, 7]. However, a new approach have been given in the algorithms implementation with regard to the temporal index of the reference signal and its filtered version. First of all, the filtered reference signal $v_{i,j,k}(n)$ is defined as:

$$v_{i,j,k}(n) = \mathbf{h}_{j,k}^T \mathbf{x}'_i(n-1) \quad (1)$$

being $\mathbf{h}_{j,k} = [h_{j,k,1}, h_{j,k,2}, ..., h_{j,k,M}]^T$ and $h_{j,k,m}$ is the value of the mth coefficient of the FIR filter that models the plant between y_j(n) and the kth error sensor, being M the filter length. $\mathbf{x}'_i(n) = [x_i(n), x_i(n-1), ..., x_i(n-M+1)]^T$.

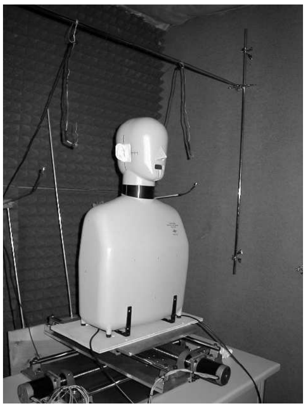


Figure 2: Photograph of the mannequin and platform used in the measurement process.

Equation (1) contrasts with similar descriptions of multichannel algorithms, for instance the FXLMS algorithm described in [7], and it can be easily seen that the filtered reference signals, $v_{i,j,k}(n)$, have been delayed by one sample in order to derive them from the same iteration that the reference signal implicitly presented at the kth error signal,

$$e_k(n) = d_k(n-1) + \sum_{j=1}^{J} \mathbf{h}_{j,k}^T \mathbf{y}_j(n-1), \qquad (2)$$

where $d_k(n)$ and $y_j(n)$ are the primary sound field at the kth error sensor and the jth secondary signal, respectively, at time n. $\mathbf{y}_j(n) = [y_j(n), y_j(n-1), ..., y_j(n-M+1)]^T$. Consequently, there exists higher correlation between the error signals and the filtered reference signals and the algorithms behaviour significantly improves in practice. This approach has been applied to all the multichannel algorithms considered in this work.

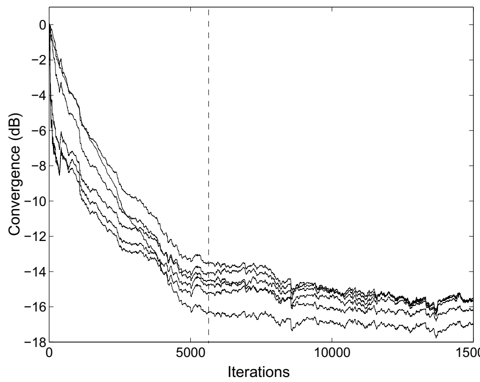


Figure 3: Convergence curves for random noise measured at SE1. From top to bottom at the dashed line position: FXLMS, AP2, FTF, AP3, AP5, RLS.

3. ANC SYSTEM DESCRIPTION

A practical ANC system with two secondary sources (SS1 and SS2), two error sensors (SE1 and SE2) and one primary source, has been implemented inside a wooden listening room, see Figure 1. Simulations have been performed using real acoustic channels measured using an in-house software [8, 9]. A 4100 type Bruel & Kjaer acoustic mannequin provided with two high precision microphones (SM1 and SM2) was used to estimate the plants corresponding to the measuring zone. The mannequin was moved by means of a mobile platform inside of a square zone, see Figure 2.

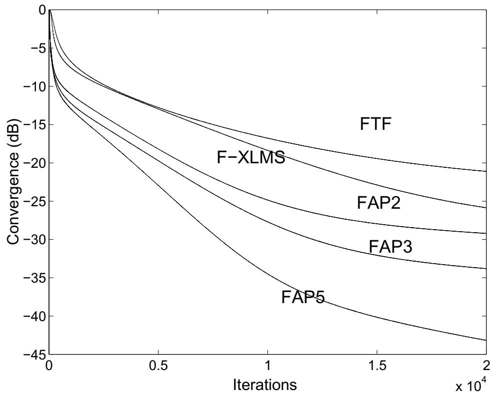


Figure 4: Convergence curves for multitone noise measured at SE2.

4. SIMULATION RESULTS

FTF, RLS, FXLMS and AP algorithms have been simulated using real data for a 1:2:2 and a 1:1:2 ANC systems in this study. The mannequin has been virtually moved inside the desired controlled zone (see Figure 2) in order to measure the resulting acoustic field once the adaptive algorithm has converged. Different primary noises have been used: a multitone signal (10 harmonics of fundamental frequency 20 Hz), random noise and a 80 Hz single tone. The initialization variables of the different algorithms have been empirically chosen by trial and error. Adaptive filters of 50 and 14 coefficients (for the single tone primary signal case) have been used.

Figure 3 shows the performance of the different algorithms for the first 15000 iterations with random noise as reference signal measured at the first error sensor (corresponding sampling ratio of 500 Hz) in the 1:2:2 ANC system previously described. These curves, called *convergence curves* [2], are obtained by plotting the instantaneous estimated power at each error sensor divided by its initial value in decibels. As can be seen from Figure 3, the RLS and FTF algorithms exhibit slower convergence speed at the beginning but finally achieve higher attenuation levels. However, the convergence performance of the AP algorithms is good from the first iterations achieving finally lower attenuation levels. Figure 4 illustrates the convergence curves for the multitone signal for the same 1:2:2 system. A poor performance is shown for the FTF algorithm as it can be observed, this is due to the use of a rescue variable that assures the algorithm stability. On the whole, the FXLMS and AP algorithms exhibit good stability in practice when the initialization variables have been conveniently chosen from the adaptation beginning.

Table 1 shows the average attenuation levels at both error sensors after algorithm convergence. The final attenuation noise levels for random noise are very similar for all the al-

		Random noi.	Multi. noi.
Algor.	Flops/ite.	Atten.	Atten.
FXLMS	2887	16.24	36.66
RLS	215496	17.09	
FTF	10975	16.29	22.31
AP(2)	10214	16.05	39.33
AP(3)	19265	15.50	52.55
AP(5)	48065	15.70	64.79

Table 1: Number of floating point operations per iteration and average attenuation levels in dB at both error sensors for the different algorithms and reference signals.

gorithms. However, different performances are shown for the multitone signal. It must be noted that the RLS algorithm became unstable when such a large number of iterations was used and a reference signal different to random noise was applied. That is why the corresponding cell in table 1 is empty. As it was aforementioned, the FTF algorithm is protected with a rescue variable which causes poorer results in the multitone signal compared with the other algorithms, but assures algorithm stability. As it was expected, the RLS algorithm is the most computationally demanding, but at the same time the fastest one if stability is assured, see Figure 3.

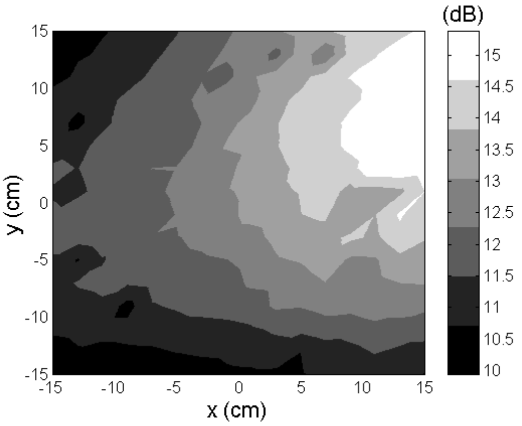


Figure 5: Simulation of 1:2:2 ANC system. Attenuation measured for random noise at SM1 using the FXLMS algorithm.

We now investigate the ability of the considered multichannel algorithms to create *quiet zones* around the listener ears noise levels at the mannequin microphones have been measured. Figures 5 and 6 show the attenuation noise levels at microphone SM1 in the controlled zone. The quieter area is close to the error sensor 1. In terms of the final attenuation levels, it is not observed meaningful differences between the algorithms. Moreover the quiet zones shapes are very similar.

Finally, simulations with a 80 Hz single tone where then performed in a 1:1:2 ANC system. This configuration was chosen to compare the results with experimental ones. Figure 7 illustrates the attenuation levels achieved at microphone SM1 using the AP algorithm with projection order $N = 2$ and adaptive filters of 14 coefficients. The selected secondary source was SS1, see Figure 1. The FXLMS and FTF algorithms provided very similar quiet zones.

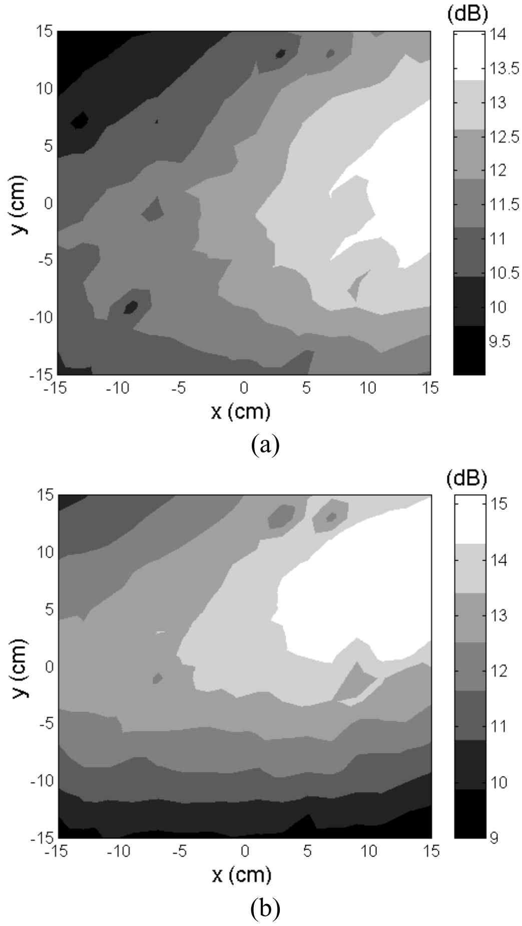


Figure 6: Simulation of 1:2:2 ANC system. Attenuation measured for random noise at SM1 using: (a) the FTF algorithm and (b) the AP algorithm for the affine projection order N=5.

5. EXPERIMENTAL RESULTS

The best way to complete this study is to test the algorithms behaviour in a real practical system. A 1:1:2 ANC system and a 80 Hz single tone disturbance signal were chosen because of computational limitations of the processing facility.

Figure 8 shows attenuation noise levels recorded at SM1 mannequin microphone with the FXLMS and the AP (N=2) algorithms. It can be easily observed that very similar quiet zones were provided by both algorithms. An average attenuation noise level over 30 dB was achieved in the controlled zone with both algorithms, slightly higher to the average attenuation of 25 dB predicted by the simulations, see Figure 7. It must be noted that the transducer positions for this experiment were not exactly the same than those used in simulations. The error microphones were placed closer to the monitored area and the secondary source was moved slightly towards its corresponding error sensor. This fact explains the improvement of the final attenuation levels compared to the simulations and also the changes observed in the shape of the quiet zones.

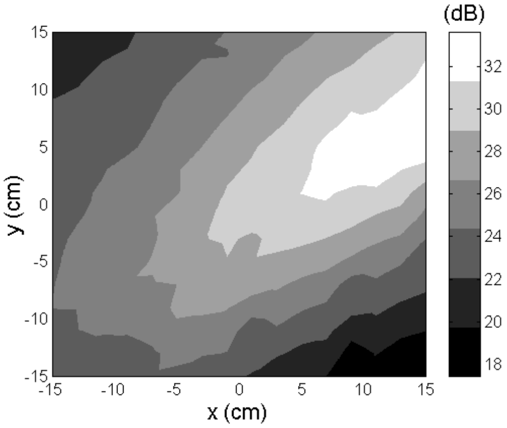
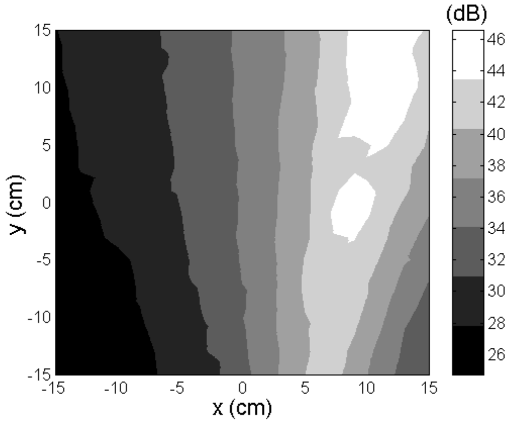


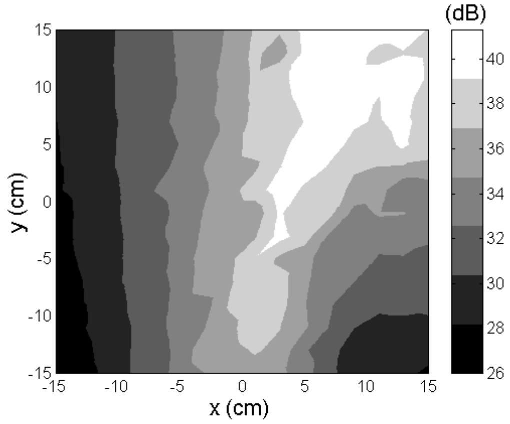
Figure 7: Simulation of 1:1:2 ANC system (SS1). Attenuation measured for a single tone at the microphone SM1 using the AP algorithm for N=2.

6. CONCLUSIONS

In this work, a performance comparison of different multichannel adaptive algorithms for noise control has been carried out by means of simulations using real data and experimental results. Meaningful attenuation levels at mannequin microphones are obtained. Among the algorithms tested, the AP algorithm has shown the best performance. It achieves high attenuation levels, is robust in practical cases and exhibits a meaningful versatility in terms of convergence speed and computational cost. Furthermore, computationally efficient versions of this algorithm can be developed for multichannel ANC applications. Real experiments validate this statement.



(a)



(b)

Figure 8: Real 1:1:2 ANC system (SS1). Attenuation measured for a single tone at the SM1 microphone using: (a) the FXLMS algorithm and (b) the AP algorithm for N=2.

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