Digital Adaptive Echo-Canceller for Room Acoustics Improvement

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***Abstract*—This paper presents a method to cancel the echoes generated by reflections in a room. The starting point is the full-band adaptive system identification method, where the unknown system is the room, providing several reflections of the sound. A sub-band filtering method is proposed and all the aspects regarding the filter banks, the structure, the number of sub-bands and the order of the required filters are analyzed. The performance is studied with respect to the provided error and the echo return loss enhancement. The asymmetric structure sub-band filtering offers a better performance than the full-band implementation. Increasing the number of sub-bands will also enhance the performance of the system.**

***Index Terms*—adaptive system identification, echo-canceller, reverberation, room acoustics, sub-band filtering**

I. INTRODUCTION

A closed space induces reflections of the acoustic signal generated inside, and together with other phenomena have influence on the auditory feeling. The combined effect of multiple sound reflections within a room is called natural reverberation. It starts with the production of a sound in a room. Then, the acoustic wave meets the walls, ceiling and other surfaces, where the energy is absorbed and reflected [1].

The acoustical properties of a room are accurately described by the impulse response. Figure 1 presents the impulse response of a concert hall. If a direct path exists between the source and the listener, the listener will hear the *direct sound* first, followed by reflections of the nearby surfaces, the later being called *early reflections*. After some tens of milliseconds, the number of reflected waves becomes very large, with a decreased evolution in time, characterized by a dense collection of echoes moving in all directions, their intensity being independent from the location in the room. These are *late reflections*. If the room is used for conferences, and there are many reflections due to the walls, the speaker message is no longer understandable, and an

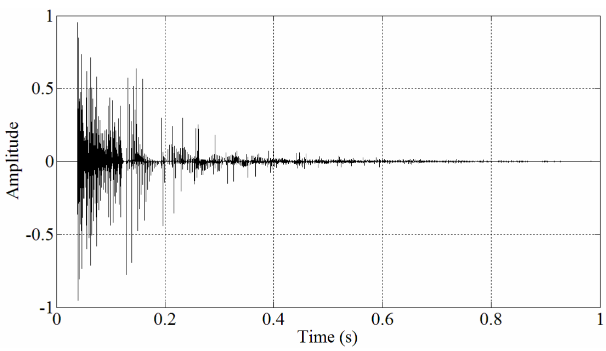

Figure 1. Impulse response of a concert hall.

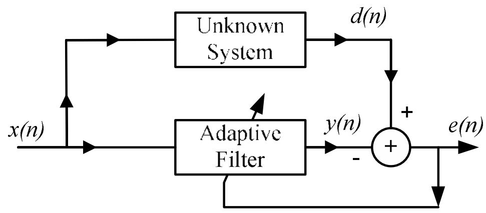

Figure. 2. Full-band system echo canceller.

echo-canceller is required. Their theoretical basis is in the field of adaptive filtering. Despite the fact that adaptive echo cancellation was conceived by the mid-1960s, practical implementation had to wait for the very large scale integration (VLSI) systems.

The second section of the paper presents the full-band adaptive algorithm for echo cancellation. Because its implementation results in a prohibitively complex hardware, a better solution is splitting the signal into several frequency bands and providing an echo canceller for each band. In Section III, the sub-band filtering system is presented and a detailed analysis is provided with respect to the filter bank and structure. Section IV is devoted to the conclusions.

II. ADAPTIVE FULL-BAND ECHO-CANCELLER

Fig. 2 depicts the full-band system identification application for echo-cancelling. The commonly employed algorithms are of Least Mean Square (LMS) type and their behavior is well understood. The classical LMS computes the tap weights as follows:

$$\begin{aligned} w(n+1) &= w(n) + 2\mu \cdot e(n) \cdot x(n) \\ e(n) &= d(n) - y(n) \end{aligned} \tag{1}$$

where *w(n)* is the tap weight, *x(n)* is the input signal, *y(n)* is the output signal, *d(n)* the desired signal, *e(n)* the error signal and μ the step size.

The normalized LMS (NLMS) algorithm sets an upper limit on the step-size μ, resulting in a more stable convergence and steady state coefficient values. Thus, the filter update for the NLMS algorithm is:

$$w(n+1) = w(n) + \beta \cdot e(n) \cdot \frac{x(n)}{\left\| x(n) \right\|^2} \tag{2}$$

where β is kept between *0* and *2*.

The performance of the LMS algorithms for echo cancellation is measured in terms of output error and of echo-return loss enhancement (ERLE). The last measure is defined as the ratio of the input echo signal power over the power of the residual echo signal:

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$$ERLE = 10\log_{10}\frac{E(x^2(n))}{E(e^2(n))} \quad (3)$$

TABLE I. ADAPTIVE FILTER ALGORITHM EVALUATION.

Algorithm	LMS	NLMS	RLS
Convergence time	Very slow	Slow	Fast
Stability	Very stable	Stable	Very instable
Complexity	Very simple	Simple	High
Implementation	Very simple	Simple	Difficult

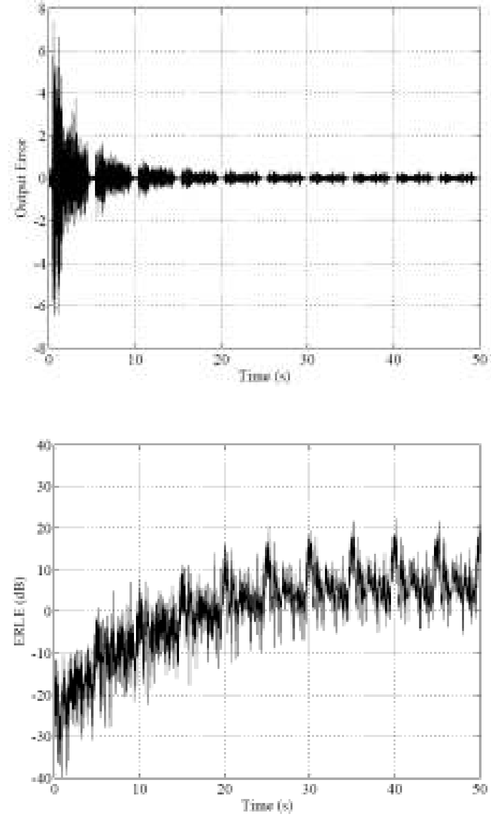
It is a smoothed measure of the amount (in dB) that the echo has been attenuated.

Different versions of the LMS algorithms were simulated and tested in Matlab. The input signals were generated as echoes added to a sound. The audio signal had the sampling frequency 22 kHz and lasted for about 50 seconds. The conclusions are presented in Table I. The best choice with respect to the stability and convergence speed was found to be the NLMS algorithm; the corresponding figure of merits is presented in Figure 3. The direct use of the NLMS algorithm is unsuitable for real-time applications because very high order finite impulse response (FIR) models are required for a good accuracy; to get an error of about 2.5 E-4, an order of 2048 NLMS is required. The audio signals have colored spectra and convergence is slow due to ill-conditioned covariance matrices.

III. ADAPTIVE SUB-BAND ECHO-CANCELLER

A better approach based on adaptive sub-band filtering has been considered for echo cancellation. This echo-canceller is depicted in Figure 4. Using the analysis filter banks *P(z),* the original signal is decomposed by subdividing its spectra into smaller intervals ($x_0(n)$, $x_1(n)$, …). Then echo-cancelling is performed in each sub-band by a set of independent adaptive filters ($h_0(n)$, $h_1(n)$,…). The outputs of these filters are subsequently combined using a synthesis filter bank *Q(z)* to reconstruct the full-band output.

In the analysis block, the basic element is the two-channel analysis sub-band filter. It decomposes the input into a high-frequency sub-band and a low-frequency sub-band, each with half the bandwidth and half the sample rate of the input. Therefore, in the analysis block, the signal is filtered by a pair of high-pass and low-pass FIR filters, followed by a down sampling by 2.



a)

b)

Figure 3. Performance of a full-band NLMS echo-canceller of order 2048; a) error; b) ERLE.

The filter banks introduce signal degradations, so an analysis regarding the influence of the filter characteristics upon the global performance was performed. Aliasing caused by the filter banks provides a distortion to the sub-band adaptive system, which forms a lower limit for the minimum mean squared error. In this work, the Kaiser, Blackman and Hanning FIR windows were taken into consideration. Simulations were provided for 2 and 4 sub-band echo canceller, for similar conditions: same unknown system, NLMS filter order, step size, but different types of filters. The best window was found to be the Kaiser's (Figure 5). Increasing the filter order decreases the error, but not in a linear relationship. The required order for an error 4.7E-4, step size 0.1, NLMS order 512 was found to be 64.

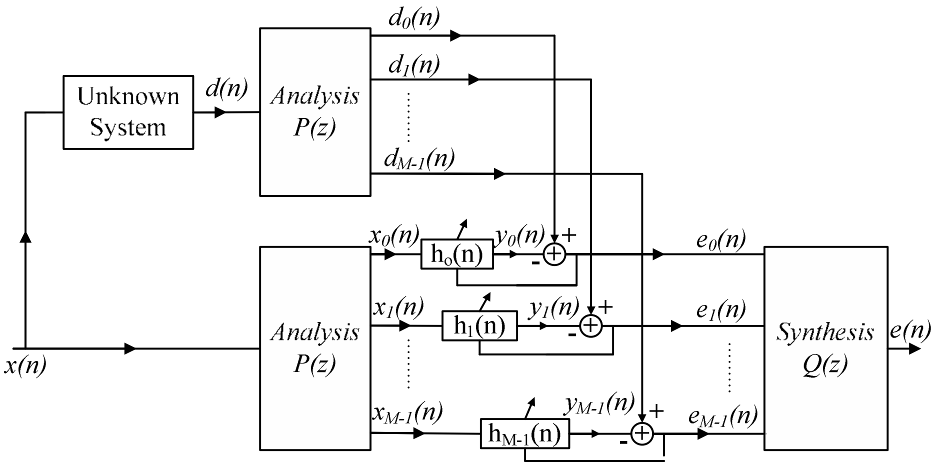


Figure 4. Sub-band system identification.

If the number of sub-bands is larger than 2, the structure of the sub-band filtering system may be either symmetric – the width of the sub-bands is equal – or asymmetric – the lower band being split into two parts, whereas the higher band remains unchanged (see the structure of a 4 sub-band analysis block in Figure 6). To choose the best structure from the two mentioned ones, simulations were performed for a 4 and 8 sub-band filtering with a Kaiser window. The best structure was found to be the asymmetric structure (Figure 7). This conclusion was expected because the acoustic sound is more colored at low frequencies and flatter at high frequencies. The adaptive algorithms are known to behave well for flatter bands, so dividing the spectrum into sub-bands at lower frequencies will enhance the system performance.

The next aspect to be analyzed is the influence of the number of sub-bands. Examining the simulations from Figure 8, one can see that the performance is enhanced with an increased number of sub-bands, maintaining the complexity in the same range. Not only the value of ERLE is higher with the number of sub-bands (around 5 dB for full-band and 20 dB for 8 sub-bands), but also the convergence speed is larger (the steady state is reached after 2500 samples for the full-band and 1000 samples for the 8 sub-band system).

The computational efficiency is achieved at the expense of some loss in the performance, especially in the asymptotic residual error. This is because achieving zero residual error requires an infinite tap size from the sub-band filters and sub-band models. Since we always use FIR sub-band filters and sub-band models, residual errors are unavoidable. This implies that in the design of a sub-band identification system, there is a tradeoff between asymptotic residual error and computational cost. This design becomes quite non-trivial if the convergence rate is also considered.

IV. CONCLUSIONS

A detailed analysis on sub-band identification for acoustic echo cancelling was performed. First, the classical full-band echo canceller was simulated for different LMS algorithms. The best choice, from stability and convergence point of view, was found to be assured by the NLMS algorithm; for a good accuracy, a high order of the adaptive filter is required.

Audio signals have colored spectra, and, it is well known that the performance of the LMS algorithms is suboptimal in this case, especially when extremely long FIR filters are being adapted. Thus, an alternative solution based on sub-band techniques was presented. Simulations of the system identification application were made, for signal splitting into two, four and eight sub-bands. The designs were made in Matlab-Simulink environment and the echo signal was simulated with an artificial reverberator. The order of the NLMS filter was chosen to be from $N = 64$ to $N = 4$, whereas the convergence step size was set to 0.1. The designs prove the improvement of the performance of the application, as the number of sub-bands increased. As filter bank type, the Kaiser was best suited and the asymmetric structure was found to be better than the symmetric one.

Further work will involve the implementation of this system into a hardware device and the use of the method for generating reverberation by modeling the acoustics of a cathedral.

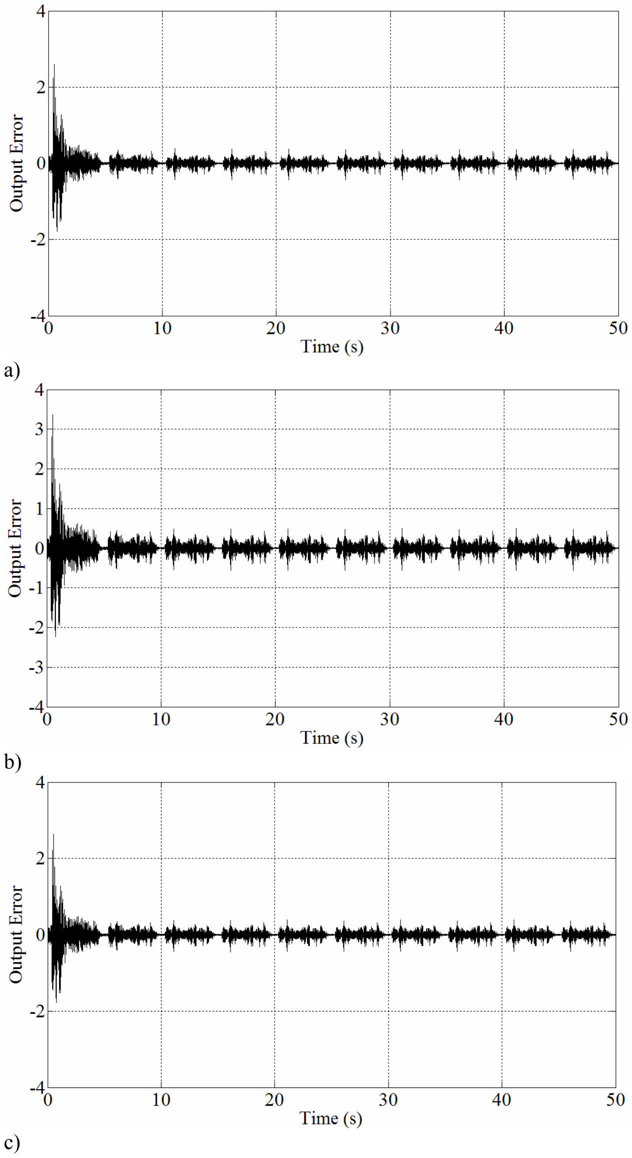


Figure 5. Output error for a 2 sub-band echo canceller with filter order=4, NLMS order=512 for filters of type: a) Kaiser; b) Blackman; c) Hann.

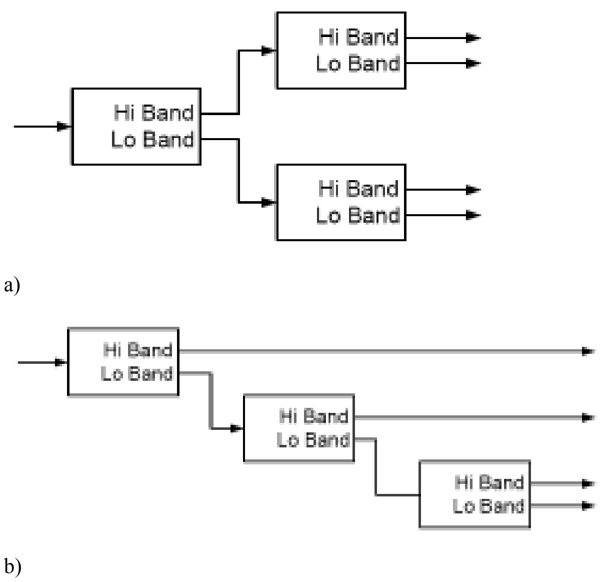
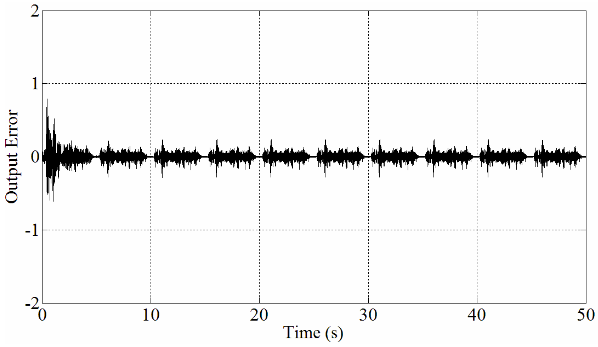


Figure 6. Four sub-band filter implementation a) symmetric structure; b) asymmetric structure.



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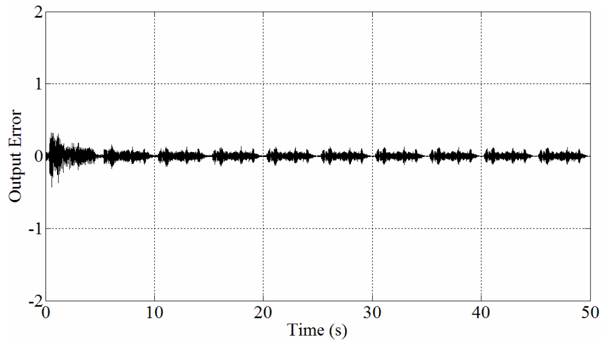

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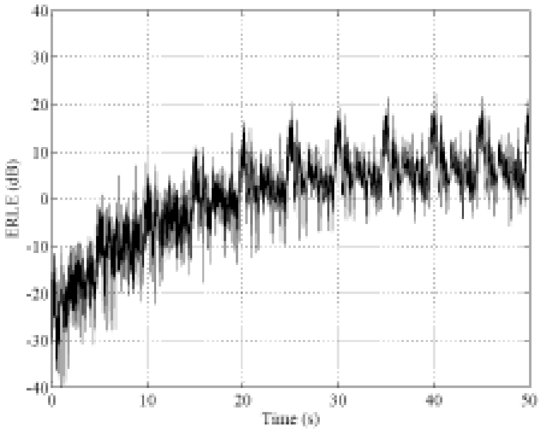
Figure 7. Output error for a 4 sub-band system with Kaiser filter order=4, NLMS order=256 having a symmetric structure; b) asymmetric structure.



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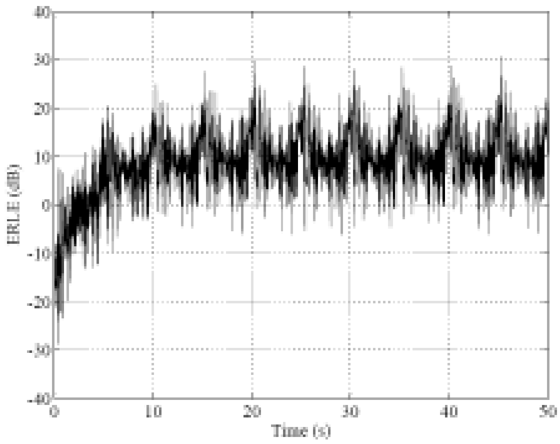

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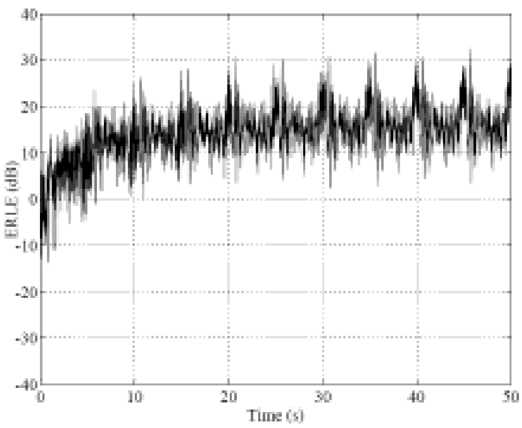

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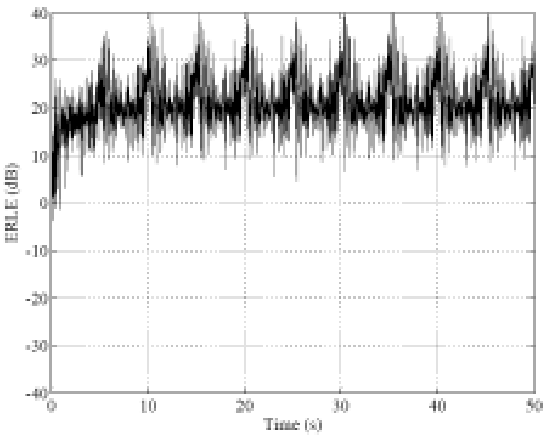

d)

Figure 8. ERLE for an echo canceller a) full-band - NLMS order = 1024; b) 2 sub-bands - Kaiser filter order = 4, NLMS order = 512; c) 4 sub-bands - Kaiser filter order =4, NLMS order = 256, asymmetric structure; d) 8 sub-bands - Kaiser filter order =4, NLMS order = 128, asymmetric structure.

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