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# Separation of Digital Audio Signals using Least-Mean-Square (LMS) Adaptive Algorithm

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**ABSTRACT**

Adaptive filtering is one of the fundamental technologies in digital signal processing (DSP) in today's communication systems and it has been employed in a wide range of applications such as adaptive noise cancellation, adaptive equalization, and echo cancellation.Signal separation remains a task that has called for attention in digital signal processing and different techniques have been employed in order to achieve efficient and accurateresult. Implementation of adaptive filtering can separate wanted and interference signals so as to improve performance of communication systems. In the light of this, this paper usesa least-mean-square (LMS) adaptive algorithm for separation of audio signals.The simulated results show that the designed LMS based adaptive filtering techniqueconverge faster than conventional LMS adaptive filter.



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## 1. INTRODUCTION

Digital Signal Processing (DSP) is concerned with the theoretical and practical aspects of representing information bearing signals in digital form. Also, it involves using computers or special purpose digital hardware either to extract information or to transform the signals in useful forms [1]. To achieving this, digital filters remain as the backbone for digital signal processing. The existing types of digital filters are: Infinite Impulse Response (IIR) filters; and Finite Impulse Response (FIR) filters. An improvement on these digital filters result into the use adaptive filters where FIR filters remains the most used in this application [2, 3].

Adaptive filters are filters that can easily adjust their properties to suit the environment (conditions) under which they are used. The properties that are adjusted include: coefficients; step-size; and length.They can be implemented using either of the two available types of digital filters i.e. the Infinite Impulse Response (IIR) filter or the Finite Impulse Response (FIR). However, the FIR filter is preferred for the implementation of the adaptive filters because they are more stable and converge faster than the IIR filters [4, 5]. Some of the adaptive filter performs its task using correlation principle mainly cross correlation.

Adaptive filtering method using cross correlation method for signal separation coupled with the Least Mean Square adaptive algorithm is employed in this paper for the separation of digital audio signals.

## 2. DESIGN METHODOLOGY

The cross correlation method is used to separate two audio signals using adaptive filter. A signal buried in another signal can be estimated by cross correlating it with an adjustable template of the second signal. The template signal is adjusted by trial and error guided by the fore knowledge until the function is maximized; the template is then the estimate of the signal. Adaptive filter configuration employed in this work is shown in Figure 1. The filter iteratively alters its parameters so as to minimize the cost function of the adaptive algorithm of the differencebetween the desired output *d(n)* and its actual output *y(n)*. Adaptive algorithm adjusts thefilter coefficient included in the vector w (n) The adaptive filter aims to equate its output *y(n)* to the desired output *d(n)*. For each iteration, the error signal is given by:

$$e(n) = d(n) - y(n) \tag{1}$$

The error signal is fed back into the filter, where the filter characteristics are altered accordingly.

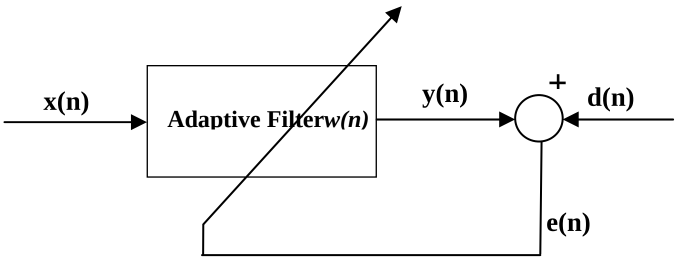


Figure 1. Adaptive filter configuration

### 2.1. The Least Mean Square (LMS) Algorithm

The LMS algorithm is widely used due to its computational simplicity. It is a form of adaptive filter known as stochastic gradient-based algorithms because it employs the gradient vector of the filter tap weights to converge on the optimal wiener solution [6, 7]. The iteration of the LMS algorithm leads to the update of the adaptive filter tap weights according to [8, 9]:

$$w(n+1) = w(n) + 2\mu\ e(n)\ x(n) \tag{2}$$

where $x(n) = [x(n)\ \ x(n-1)\ \ldots\ x(n-N+1)]^T$ is the input vector, the coefficients of the adaptive FIR filter tap weight vector at time n is

$$w(n) = [w_0(n)\ w_1(n)\ \ldots\ w_{N-1}(n)]^T$$

and μ is known as the step size parameter and is a small positive constant parameter. This step size parametercontrols the effect of the updating factor and determines both the stability andconvergenceof the adaptive filter becomes unstable and its output diverges.

The number and type of operations needed for the LMS algorithm is nearly the same as that of the FIR filter structure with fixed coefficient values, which is one of the reasons for the algorithm's popularity. It depends explicitly on the statistics of the input and desired response signals. In effect, the iterative nature of the LMS coefficient updates is a form of time-averaging that smoothed the errors in the instantaneous gradient calculations to obtain a more reasonable estimate of the true gradient.

## 3. IMPLEMENTATION OF THE LMS ALGORITHM

The LMS algorithm is used in designing adaptive filter and the MATLAB program is used for the simulation. The name given to the two audio signals used are; myvoice and noise. Both of them are .wav file because that is the only audio file format that MATLAB can take.

The following code is used for signal separation and A1 and A2 are the first and second audio signal respectively. The code *[y,e] = filter(halms,x,A2)* shows that filter ratingis achieved using a filter object *halms* and the filter object depend on *adaptfilt.lms* algorithmwith filter length of *22* and step size of *0.0353*, the frequency response of the filter used showing the amplitude and phase response of the filter is given by Figure 2 and Figure 3 shows the step response of the filter. The maximum step size for the filter is obtained using*mumaxlms*, while the maximum mean-square lms step size was obtained using *mumaxmselms.*

However, it is advisable to use the smaller step size i.e. 0.01 because it improves accuracy of convergence to match characteristics of the unknown to the time taken for it to adapt.

The next stage is the design of the filter that performs the correlation of the second audio signal, *A2* with the mixed audio signal, *x* as shown below:

*[y,e] = filter(halms,x,A2);*

The correlation compares the second signal with the mixed signal. After which the difference between them is obtained this difference is called the error, *e* while the output is given as *y*. The error is then fed back into the adaptive filter. The iteration process continues until the error value becomes 0. When the error value becomes zero, it means that adaptation has been done successfully, that is the filter has successfully adapted.

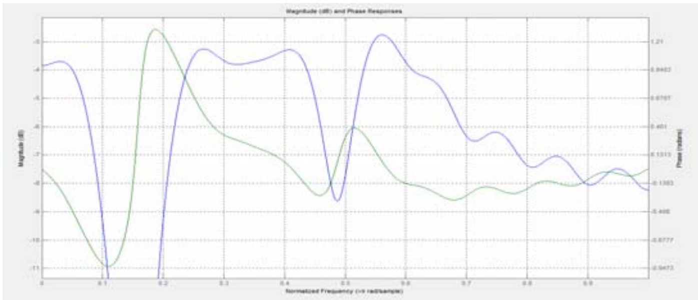

Figure 2. Magnitude and phase response of the adaptive filter used

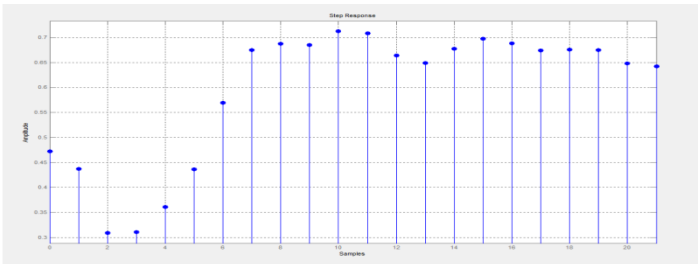

Figure 3. Impulse response of the Adaptive filter used

## 4. SIMULATION RESULT AND ANALYSIS

The LMS is used for updating the filter coefficients. The LMS algorithm is simulated using MATLAB. The filter length of *22* and step size *0.0353* is used. The MATLAB code written was run so that the betterperformance is achieved by varying the filter length and step size. Figure 4 and Figure 5 show the plot of separation of the first (desired) and the second (noise) audio signals. It is observed that after the removal of the noise signal, the desired input signal and the desired output signal are close to each other. Also, the adaptive filter output and the noise signal are close. This shows that in adaptive noise removal, the output of the filter is simply the noise signal while the correlation result is the desired output signal.

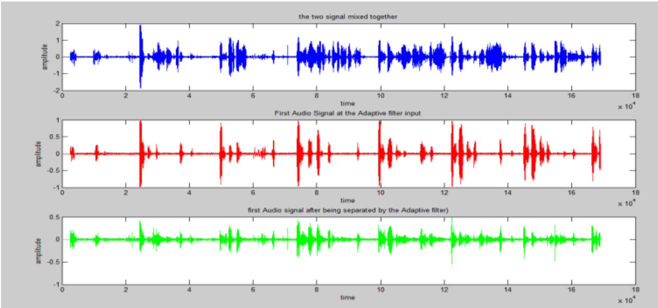


Figure 4. Separation of the First Audio Signal: the Desired Signal

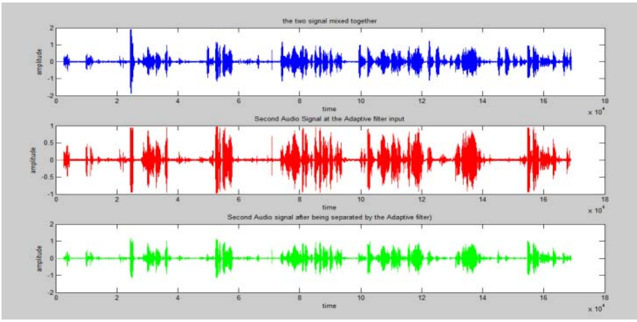


Figure 5. Separation of the Second Audio Signal: the Noise Signal

## 5. CONCLUSION

Adaptive filters are very important tools in Digital Signal Processing. The aspect of Mixed Audio Signal separation has been looked into in this paper. The LMS algorithm has been employed because of its simplicity. The implementation of this algorithm results in reliable adaptive noise removal from impaired audio signals.

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