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Separation of Mixed Audio Signals Using Independent Component Analysis

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***Abstract*– This paper presents a simple method that dealt with independent component analysis and Finite Impulse Response (FIR) filter in separating convolutive mixtures. The original source was retrieved from the set of filtered versions of each mixed signals using independent component analysis method as well as filtering mixture of voices (audio) recorded in a noisy environment.**

***Index Terms*– Independent Component Analysis, Adaptive Filtering, Convolutive Mixture and Audio Signal**

I. INTRODUCTION

DIGITAL Signal Processing (DSP) is concerned with the theoretical and practical aspects of representing information bearing signals in digital form and with using computers or special purpose digital hardware either to extract information or to transform the signals in useful ways [1]. At achieving this, digital filters remain as the backbone for digital signal processing. The available types of digital filters are: Infinite Impulse Response (IIR) filters; and Finite Impulse Response (FIR) filters. An improvement on these resulted into the use of the digital filters at making adaptive filters where Finite Impulse Response filters remains the most used in this application because of its stability and short length of convergence [2].

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 than the IIR filters and their convergence is achieved faster than that of the IIR filters [3]. Some of the adaptive filter performs its task using correlation principle mainly cross correlation.

It is this method of adaptive filter filtering using cross correlation method to achieve signal separation coupled with using Least Mean Square adaptive algorithm that is employed in this paper to separate to mixed digital audio signals.

II. PROPOSED ICA MODEL

The proposed ICA model is shown in Fig. 1 and the optimization criterion is in general taken in the least squares family in order to work with linear operations [4]. By applying the adaptive filter coefficients, the general LMS is capable of removing noise or obtaining a desired signal [5].

Where β_i and h (Fig. 1) are respectively the Adaptive filter weight and the Adaptive algorithm.

Let S be time t indexed, *k*-dimensional independent signals from the linearly mixed observable variables. The ICA model is written as:

$$\hat{S}_o(t) = S(t) + N(t) - \hat{N}(t) \quad (1)$$

$$U(t) = A\hat{S}_o(t) \quad (2)$$

$$X(t) = \hat{S}_o(t) + N(t) - \hat{V}(t) \quad (3)$$

$$Y(t) = BX(t) \quad (4)$$

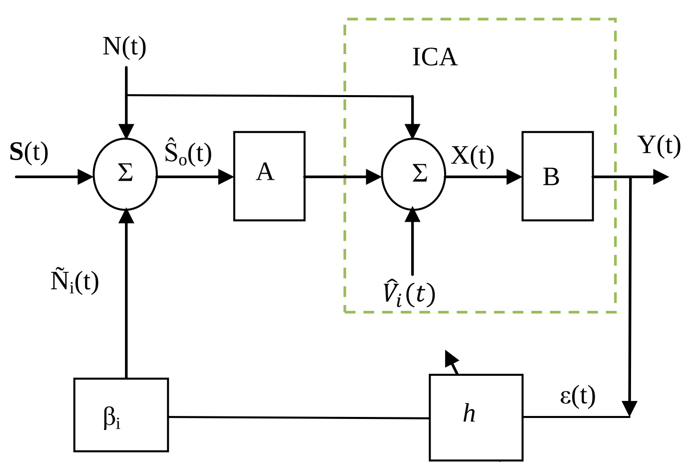

Fig. 1: Adaptive ICA process model

where $\hat{S}_o(t)$, $S(t)$, and $N(t)$, are respectively the optimum independent mixed observed signals, originally mixed unobserved signals, and N(t) is the background noise from rotating fans, working air conditions or machine. A and B are mixing and separating matrices. The optimum noises $\hat{N}(t)$, $\hat{V}(t)$ are defined as:

$$\hat{N}(t) = \sum_{i=0}^{L} \beta_i(t) N_{th}(t-i) \tag{5}$$

$$\hat{V}(t) = \sum_{i=0}^{L} \beta_i(t) N_{th}(t-i) = \hat{N}(t) \tag{6}$$

where N_{th} is the acceptable noise threshold, and *L* is the filter order. The proceeding coefficient of the filter can be estimated from the present coefficient and other thresholds:

$$\beta_i(t+1) = \beta_i(t) + 2\eta S(t) N_{th}(t-i) \tag{7}$$

where □ is the convergence constant. It should be noted that the filter order, *L*, might not necessarily be of the same order (or dimension) as that of the independent variables.

The LMS adaptive filter adapts the filter coefficients to achieve desired signal ensuring convergence; that is, minimizing error $\varepsilon(t)$ at each time index:

$$\varepsilon(t) = Y(t) - BX(t) \tag{8}$$

Convergence is slow coming; hence a local minimum is sought leading to establishing threshold values. We noted the difference in change of the filter coefficients as a measure of establishing rate of convergence, specifically

$$\xi_\beta(t) = \sum_{j=0}^{L} \left| \beta_t(j) - \beta_{t-1}(j) \right| \tag{9}$$

If $\xi_\beta(t)$ is large, filter converges to small order. We adjust the filter order to using $\xi_\beta(t) \le \alpha_{th}$ for quick convergence, where α_{th} local minimum threshold. The adaptation gain $G(n)$ is introduced for coefficient updating recursion for the period of the signal measurement:

$$G(n) = \left| \sum_{i=0}^{n} \frac{\hat{S}_o(i)}{S(i) + N(i)} \right| \tag{10}$$

where *n* is the period of the mixed signal for our signals of interest.

Detection of voice is achieved by comparing the extraneous noise V(t) (separate from background noise) introduced into the recording. At each time index, the intensity of V(t) is measured relative to the noise threshold level N_{th}. The noise is considerably removed if

$$\|V_i(t)\| \le N_{th}(t-1), \tag{11}$$

otherwise, the noise is still present, noted and removed. Where $\|V_i(t)\|$ approximates closely to that of $N_{th}(t-1)$, additional filtering is applied ensuring that the noise is filtered to bearable minimum [6].

III. SIMULATION AND RESULTS

To validate the algorithm, it was experimented with real audio signals recorded in a lecture room. The two audio signals used are myvoice and noise both of them are in .wav file. ICA algorithm was written for the separation of the two signals where A1 and A2 are the first and second audio signal respectively. The code *[y,e] = filter(halms,x,A2)* shows that filtration was achieved using a filter object *halms* and the filter object depend on *adaptfilt.lms* algorithm with filter length of 22 and step size of 0.0353.

With the code, the maximum step size for the filter was obtained using mumaxlms, while the maximum mean-square lms step size was obtained using mumaxmselms. However, a step size of 0.01 was used to improve the accuracy of convergence to match characteristics of the unknown to the time taken for it to adapt. The filter that performs the correlation of the second audio signal, A2 with the mixed audio signal was designed. The correlation compares the second signal with the mixed signal (Fig. 4 and Fig. 5), after which the difference between them is obtained and the difference is called the error, e while the output is given as y. The error is fed back into the adaptive filter and the iteration process continues until the error value becomes 0, showing that the adaptation has been done successfully and the filter has successfully adapt. Fig. 2 and Fig. 3 show the magnitude/ phase and impulse response of the adaptive filter used.

IV. ANALYSIS OF RESULTS

From the simulation result above, it could be observed that after the removal of the noise signal, the desired input signal and the desired output signal are close to each other while 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.

V. CONCLUSION

This paper has demonstrated that convolutive ICA can be regarded not only as a mathematical generalization of an instantaneous model, but also as a more powerful tool for accurately separating the mixtures of signals and the effectiveness of an adaptive filter in filtering the noise mixture from an audio signal.

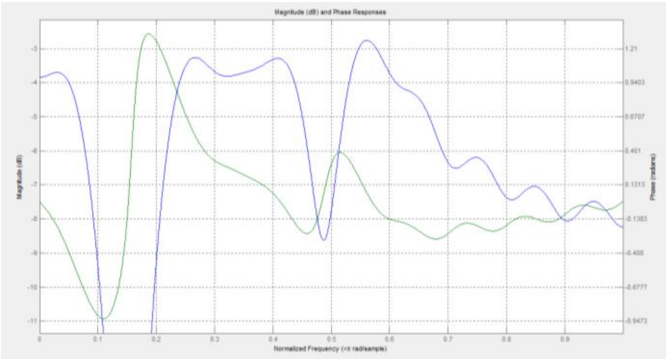

Figure 2: Magnitude and phase response of the adaptive filter used

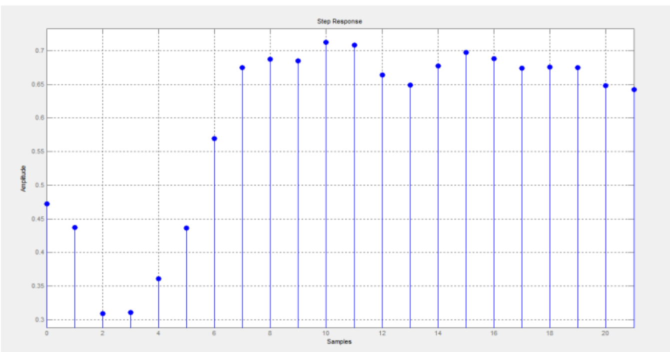

Figure 3: Impulse response of the Adaptive filter used

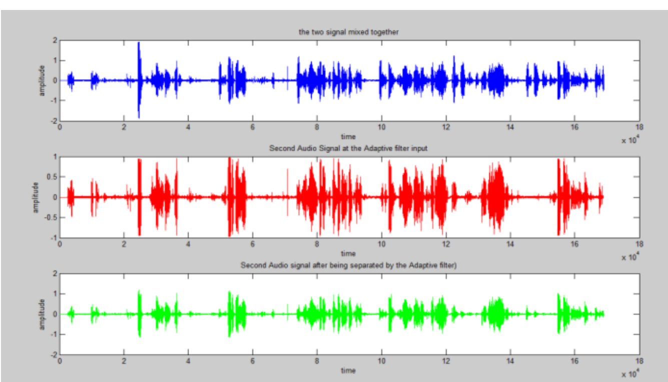

Figure 4: simulation result showing separation of signal A1

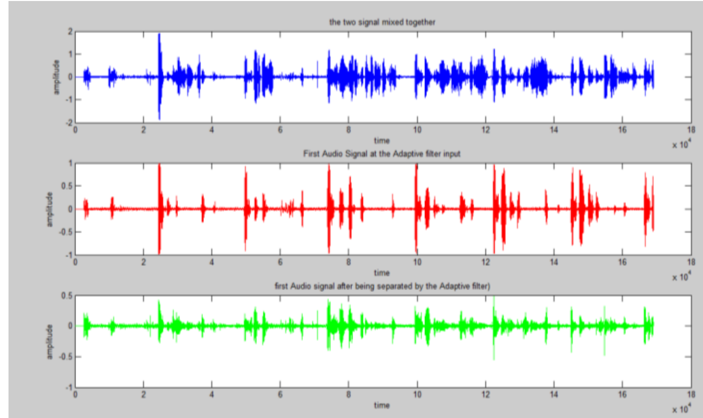

Figure 5: Simulation result for separation of signal A2

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