Adaptive Noise Cancellation Based on NLMS Algorithm

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Abstract

The main goal of this paper is to present an adaptive filter system using NLMS(Normalized Least mean square) adaptive algorithm for noise cancellation. The proposed algorithm has less computational complexity and better convergence property than the former algorithms like spectral subtraction algorithm, etc. We use TIMIT criterion voice and Noisex-92 for the experiment. The experimental result shows the feasibility of our algorithm for filtering noise from voice effectively.

1. Introduction

Noise Cancellation is a variation of optimal filtering that involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise [1]. Adaptive gradient algorithm includes Least Mean Square (LMS) algorithm and various improved LMS-type algorithms [2]. In 1960, Whidrow presented LMS algorithm which could be widely used in filed of automatic control, radar and signal processing.

The initial convergence speed, steady state misadjustment and tracking performance are three important technology indicators that determine the superiority and inferiority of the adaptive algorithm. Since the disturbing noise exists inevitably in the input signal, LMS algorithm will generate adjective noise. The louder disturbing noise $v(n)$is, the louder misadjustment noise will cause. Reducing μ, the step-size factor, can reduce the misadjustment noise generated by adaptive algorithm, and improve the convergence accuracy [3]. But with the reduction of step-size μ, the convergence speed and tracking performance will be reduced too. So a fixed step-size LMS algorithm has contradictory requirement in convergence rate and steady-state error. In order to solve this problem, various step-size LMS algorithms have been researched to achieve both fast convergence and small steady-state error. The variable-step-size LMS(VSS-LMS) algorithm is using some value provided in the adaptive processing as a standard measure and adjusts the step-size. Some simple ways are using the error signal or input signal, trying to make a function relationship between step-size and them.

Section 2 presents the theory and structure of NLMS filter and section 3 explain the experimental results. In section 4, we suggest conclusions and future work.

2. Adaptive Filter

2.1 LMS Algorithm

The Last mean square algorithm is a widely used algorithm for adaptive filtering. It have been extensively analyzed in the literature, and a large number of results on its steady state misadjustment and its tracking performance has been obtained. Figure 1 is the basic block diagram of LMS adaptive filter[4].

The algorithm is described by the following equations (1),(2),(3):

$$y(n) = W^T(n)X(n) \tag{1}$$

$$e(n) = d(n) + v_0 - y(n) \tag{2}$$

$$W(n+1) = W(n) + 2\mu e(n)X(n) \tag{3}$$

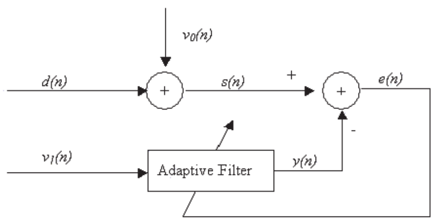

Fig.1:Adaptive filter diagram

Equation(1) calculates the output $y(n)$ of adaptive filter, $X(n)$is the input vector. Equation(2) is the error, $d(n)$ is the desired output. Equation(3) shows the weight iteration, $W(n)$ is the weight coefficient vector of adaptive filter in time n, ,μ is the step-size parameter which controls the stability and convergence speed of the LMS algorithm. The LMS algorithm is convergent in the mean square if and only if u satisfies the condition: $0 < \mu < 1/\lambda\max$, where $\lambda\max$ is the maximum eigenvalue of the autocorrelation matrix R_{XX} of the input sequence $X(n)$.

2.2 An Updated NLMS Algorithm

The main drawback of the LMS algorithm is sensitive to scaling of its input$X()$.This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm[5].The Normalized least mean squares filter(NLMS) is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input the NLMS algorithm can be summarized as:

$$W(n+1) = W(n) + \frac{\mu}{\epsilon + X^T(n)X(n)} e(n)X(n) \tag{4}$$

$$\mu(n) = \frac{\mu_0}{\epsilon + \| x(n) \|^2} \tag{5}$$

The equation (4) has been improved by equation (3). Equation (5) is the step-size iteration equation. In these equations: ϵ is a constant, usually $\epsilon \approx 0.0001$ for preventing the undersize of $\| x(n) \|^2$ from causing the oversize of step-size μ,resulting in the divergence.[2].The only difference with respect to the LMS algorithm is in the coefficient updating equation.

3. Experimental Results

In this section we evaluate the performance of NLMS algorithms and compare it with a Fractional Fourier Transform filter[6]. The LMS adaptive filter uses the reference signal and the desired signal, to automatically match the filter response. As it converges to the correct filter model, the filtered noise is subtracted and the error signal should contain only the original signal. The desired signal is composed of colored noise and an audio signal from a wave file.

We use the noise sampling from Noisex-92 database[7], testing with the TIMIT standard voice database, sampling f_s=16kHz, use

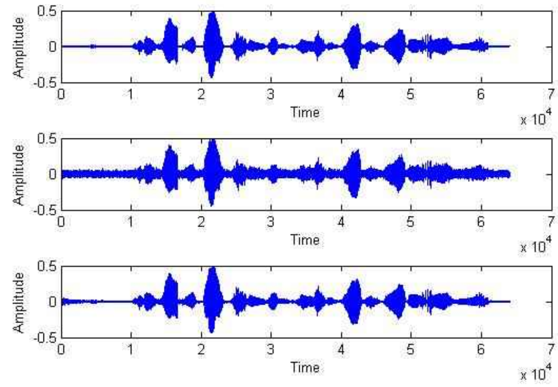

Fig.2:Processing with white noise signal

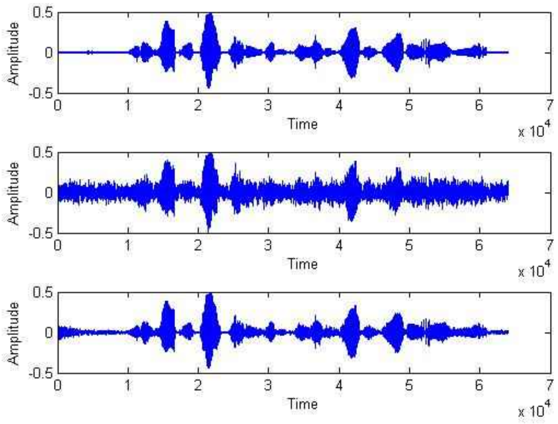

Fig.3:Processing with babble noise signal

the sampling FAKS0_SA1.WAV.We separately mix the original signal with different kinds of noise: white(Fig.2), pink, F16 aircraft, factory and babble (Fig.3). In these noises, we use white, pink and F16 aircraft are stationary noises, factory and babble are non-stationary noises respectively. From top to bottom in figure 2 and 3 are original signal, polluted signal and processing signal with NLMS filter in order.

We use signal to noise ratio(SNR) equation(6) for analysing the noise cancellation performance.

$$SNR = 10\lg\left(\frac{\sum_{t=1}^{N} signal^2(t)}{\sum_{t=1}^{N} noise^2(t)}\right) \qquad (6)$$

Table 1 Result using FRET filter

	white	pink	F16	factory	babble
SNR_{in}/dB	12.139	9.019	4.412	5.599	4.287
SNR_{out}/dB	13.246	10.191	7.360	7.927	7.203

Table 2 Result using NLMS filter

	white	pink	F16	factory	babble
SNR_{in}/dB	12.139	9.019	4.412	5.599	4.287
SNR_{out}/dB	21.815	18.550	16.068	17.011	15.674

Table.1 is the result using FRFT filter and Table.2 is NLMS filter result. From table.1 and table.2 we can see the NLMS algorithm has a much better performance, expressly in processing the non-stationary noise.

4. Conclusions

In this paper, we analyzed a NLMS algorithm and tested the noise cancellation program using this algorithm. The result shows the NLMS algorithm has faster convergence speed, better tracking performance, and adjusts the step-size mori effectively than other adaptive algorithms. This algorithm is also a good robust adaptive filter with non-stationary noise. In future work, we aim to optimize filter performance as finding updated various step-size algorithms

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