Applied Variable Step Size Algorithm to Dual-Adaptive Noise Canceller

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Abstract: - Adaptive filter is one of the important instruments for digital signal treatment. Basically, it would change with the alternations of system environment condition, which is to say, adaptive filter would make adaptation and adjustment in accordance with the estimated system and signal parameter. In recent years, adaptive LMS has gained wide attention and application for this advantage and its simplicity of design. However, it still has the shortcomings: this convergent speed is very slow, and there is no proper method for its selection of step size. Therefore it is very difficult to carry out this method. Many scholars work on improving the aforementioned shortcomings. This paper implemented alternative step size function arithmetic to realize the adaptive Noise Canceller (ANC). The adaptive noise canceller proposed in this paper is composed of two adaptive filters: the main filter (MF) and Subfilter (SF). SF is used to estimate the inputted signal-to-noise ratio (SNR), and select the step size parameter for the MF algorithm to operate, and finally reduce the noise of output after the constant adjustments. Through the experimental simulation of MATLAB and the comparison between the linearity step size function proposed by Ikeda and the non-linearity step size function algorithm raised by Ramadan, it is proved that adaptive step value function algorithm could reach the expectant result effectively.

Key-Words: - adaptive noise cancellation, dual-adaptive, LMS algorithm.

1 Introduction

Noise exists in our daily life. According to the research, if human stays in the noisy environment for a long time, he may suffer from hearing loss physiologically; people have paid more and more attention to this negative impact. So the reduce and clear up of noise has attracted people's attention for a long time. And some noise-proof research and method has been proposed in progression. Active noise control theory is first raised by Lueg [1] in 1936. This basic technical concept is to reproduce the sound wave of noise source and then use this man-made noise to produce destructive interferences for the original noise source. In the earlier state, for the practical technology can not meet the requirement, this active noise control is restricted to the theory research stage. In recent years, with the breakthrough and exaltation of digital signal processor and electronics, the faster, cheaper and more powerful DSP chip has been gradually developed and come into the market. Its performance has been gradually evoluted to the kernel of active noise controller. With the fast advancement of processor and estimation technology, the active noise cancellation system becomes practical.

In 1975, Widrow and Hoff [2] proposed to use the adaptive theory of least squares method to carry out active noise cancellation. Its biggest advantage is the simple calculation, but such advantage is coupled with a growing problem, which is the selection of step size, which could only be gotten after the constant attempt of mistakes. So some scholars endeavor to improve the selecting of step size for better effect. Ikeda [3] proposed a new technology which is to use the filter pack of MF and SF. It would substitute the estimated SNR into linearity step size function to adjust step size parameter in the process of clearing up the remnant noise of feedback, so that its convergent speed is faster, and this speed is especially quick in the mini-distorted signal, otherwise if the noise interference is larger, the effect of minimum mean-squared error response is bad. In the last few years Ramadan [4] used non-linearity step size function to replace aforementioned linearity mode. No matter what the size of noise interference is, it could reduce minimum mean-squared error to reach the purpose of interference elimination. We proposed the adaptive step size algorithm to further improve the effect when the system is in the stable status. It could overcome the following problems, the smaller step size required by small distorted signal, and the larger step size required by large disturbing signal to reach the better effect. We used experiment practical measurement and computer simulation simultaneously to evaluate the practicability of these relevant methods, and the result has shown the effect and feasibility of these innovations.

2 Adaptive noise cancellation theory

Standard adaptive noise canceller block diagram is shown in Fig. 1. In the current algorithm $s(k)$, $n_0(k)$ of execution time k have represented the signal source and noise source respectively, while $h_j(k)$ is the impulse response of noise transmission path from the noise source to main microphone. Main signal $X_P(k)$ and reference signal $X_R(k)$ is shown as below:

$$X_P(k) = s(k) + n_1(k) = s(k) + \sum_{j=0}^{N-1} h_j(k) \cdot n_0(k-j) \quad (1)$$

$$X_R(k) = n_0(k) \quad (2)$$

where N is the order number of filter. After the error signal $e(k)$ acquired by the arithmetic processing of adaptive filter and primary input signal is shown as bellow:

$$e(k) = s(k) + n(k)_1 - y(k) \quad (3)$$

$$y(k) = \sum_{j=0}^{N-1} w_j(k) \cdot n_0(k-j) \quad (4)$$

where $y(k)$ is the output of adaptive filter, while $w_j(k)$ is the parameter of j order adaptive filter.

If the travel path of noise $h_j(k)$ uses the normalized least mean square algorithm to estimate, while the renewal of $w_j(k)$ is through the following formulae:

$$w_j(k+1) = w_j(k) + \frac{\mu \cdot e(k) \cdot n_0(k-j)}{\sum_{i=0}^{N-1} n_0^2(k-i)} \quad (5)$$

where μ is step size and $0 \le j \le N-1$. The stability and rapidity of convergence of the algorithm is impacted by μ and reference signal, while from formulae (5) we could see that $w_j(k)$ and μ form a direct ratio relation. For μ is a fixed value, the global convergence speed is impacted by it, and response of the rapidity of convergence for the signal with faster variations is not so good, so we need the adaptive filter algorithm for rapid convergence, therefore we have chosen NLMS algorithm to improve the impact of input signal for convergence factor.

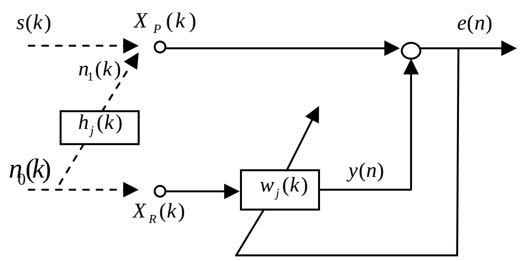


Fig. 1 Standard adaptive noise canceller block diagram

3 Structure of noise canceller

The adaptive noise canceller combined by MF and SF is shown in Fig. 2. SF is used to estimate the SNR of main input signal. And then use the estimated SNR to control step size of MF, and to raise the global noise cancellation performance of the system.

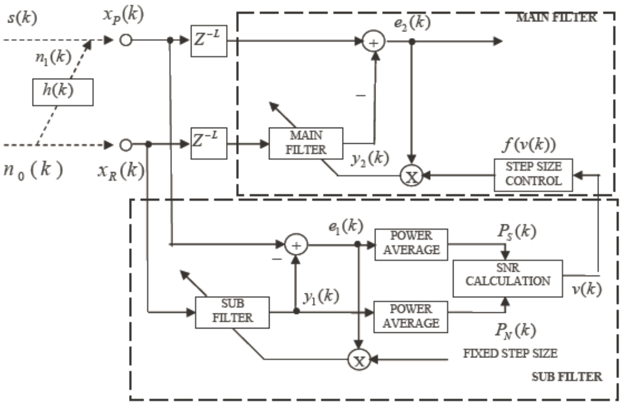


Fig. 2 Block diagram for the two adaptive filters using parallel arithmetic

3.1 Subfilter

Subfilter is a typical ANC model; its main function is to estimate the SNR of Subfilter. It uses NLMS algorithm to finish the renewal of filter parameter. For the fast convergence and fast trace of the noise transmission path variations, the step size of general μ_{sub} would use larger parameter. If the μ_{sub} has adopted a small value, although it could carry out the estimation more precisely, it is inapplicable for the rough estimate of SNR; it is the method used for the explanation of the MF. The estimate of SNR is shown in the following formulae which take the reference signal of MF as average power $P_N(k)$ and the error signal average power $P_S(k)$:

$$P_N(k) = \frac{1}{M}\sum_{j=0}^{M-1} y_1^2(k-j) \tag{6}$$

$$\begin{aligned} P_S(k) &= \frac{1}{M}\sum_{j=0}^{M-1}\left[X_P(k-j) - y_1(k-j)\right]^2 \\ &= \frac{1}{M}\sum_{j=0}^{M-1} e_1^2(k-j) \end{aligned} \tag{7}$$

where $y_1(k)$ and $X_P(k)$ are the output of Subfilter and main signal, M is the sampling numbers for the estimation of $P_N(k)$ and $P_S(k)$, and the SNR $V(k)$ calculated by $P_S(k)$ and $P_N(k)$ is shown as bellow:

$$V(k) = 10\log_{10}\{P_S(k)/P_N(k)\}(\text{dB}) \tag{8}$$

Use the calculated $V(k)$ value to control the step size parameter for MF.

The SNR estimated by Subfilter would have operation time delay, it depends on the sampling numbers of M value for estimating $P_N(k)$ and $P_S(k)$. To compensate for the operating delay time of Subfilter, we have set delay unit z^{-L} before MF. The time delay required for disposing voice distortion is general half of required M, so L is set to $M/2$.

3.2 Main filter

The primary function of MF is to use a precise step size to realize noise cancellation, and the step size is controlled by SNR $V(k)$. If $V(k)$ is the smaller value, step size would use the larger parameter for fast convergence, while the step size has to use the smaller value for the adaptive filter to output less signal distortion. The following equation expresses the function of $V(k)$ to control step size $\mu(k)$.

$$\mu(k) = \begin{cases} \mu_{\min} & V(k) > V_{\max} \\ f(V(k)) & V_{\max} \ge V(k) \ge V_{\min} \\ \mu_{\max} & V(k) < V_{\min} \end{cases} \tag{9}$$

where $\mu_{\max}$, $V_{\max}$, $\mu_{\min}$, $V_{\min}$ and $f(\cdot)$ are the maximal step size, SNR, minimal step size, SNR and a $V(k)$ function respectively. The large SNR value estimated by Subfilter would take a small step size, so that we learn that it is a descending function.

3.3 Application of different step size function

The primary impacts of step size are on system stability and rapidity of convergence, in algorithm, the proper step size would pick bigger μ value when the error is large to accelerate the rapidity of convergence, while when the system is in a steady state it would choose a smaller μ value for the error signal becomes smaller and smaller to

reach the optimal noise abatement effect. This paper selected three different step size functions, including: the linearity degressive step size function (10) proposed by Ikeda [3], its arithmetical complexity is low, while the corresponding step size selected can not effectively improve error signal in real time. The non-linearity degressive step size function (11) proposed by Ramadan [4], it could reduce the error signal of the algorithm, but it is still open to improvement, and the alternative step size function (12) proposed in this paper, it would provide a better effect.
Linearity step size function proposed by Ikeda:

$$f(V(k)) = \mu_{\max} \times \left[\frac{V_{\max} - V(k)}{V_{\max} - V_{\min}}\right] \tag{10}$$

Non-linearity step size function proposed by Ramadan:

$$f(V(k)) = \mu_{\max} \times \exp\left[\left(\frac{V(k) - V_{\min}}{V_{\max} - V_{\min}}\right)\ln\left(\frac{\mu_{\min}}{\mu_{\max}}\right)\right] \tag{11}$$

Alternative step size function:

$$f(V(k)) = \mu_{\max} \times \exp\left[\left(\frac{V(k) - V_{\min}}{V_{\max} - V_{\min}}\right)^2 \ln\left(\frac{\mu_{\min}}{\mu_{\max}}\right)^2\right] \tag{12}$$

Visually, when SNR is smaller, it represents larger noise, so we could use larger step size parameter to get the better convergent effect; when SNR is larger, the noise is smaller, so the system is stable, then we would use smaller step size parameter to maintain the system at the optimum condition. Figure 3 has shown the relation among linearity (10), non-linearity (11), alternative step size (12) and SNR. When SNR is at -50dB the noise is larger, comparing the linearity, non-linearity mode and alternative step size mode, we could find out that the faster rate of decline is caused by larger selected step size; when SNR is at -10dB, we could find out that the differences among linearity, non-linearity and alternative step size are larger, so alternative step size is more ideal, and it could reach the minimum mean squared error.

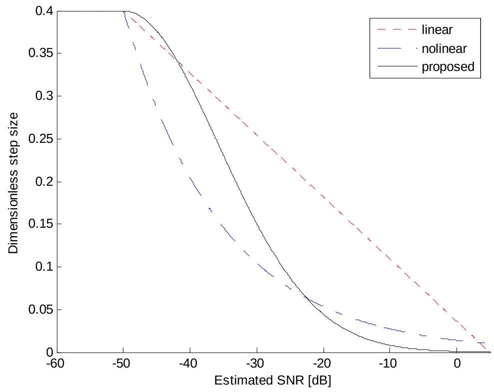


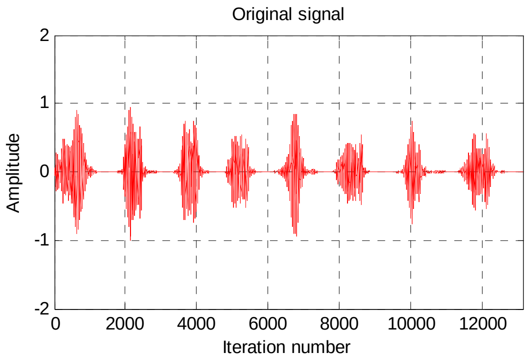
Fig. 3 The corresponding step size chart for the SNR of three different step size equations

4 Simulation results

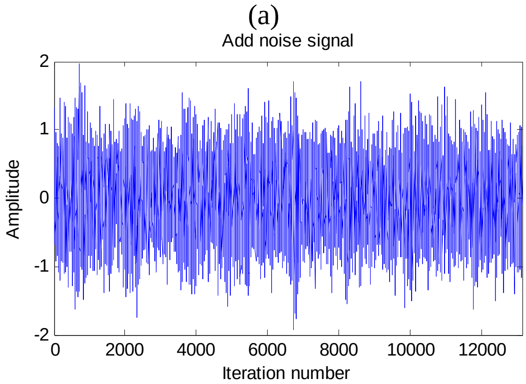
4.1 Simulation 1

Here the adaptive noise cancellation system of Figure 2 has been used, and it has compared the different step size mathematical expression with same sound wave signal under same condition. Noise constituent is to add the resound generated by the noise transmission path of noise source to the sound wave signal and the signal distortion caused by them. General sound wave contains the noise constituents without relativities, which is like the noise we heard in the recording that we recorded. The original signal sampling frequency used in this paper is 8KHz as is shown in Fig. 4(a), $n_0(k)$ is the white gaussian noise signal of zero mean, and its variance is 0.4, its combination with original signal is shown in figure(b).

The following parameter setting M = 128, L = 64, μ_{sub} =0.4, $V_{\min}$ =-50dB, $V_{\max}$ =5dB, $\mu_{\max}$ =0.4, $\mu_{\min}$ =0.01. Bringing the above data into the aforementioned three different step size functions (10)~(12), then we could got the minimum mean squared error (MSE) comparison diagram as is shown in Fig. 5. It is observed from the aforementioned diagram that we could get better result through the alternative step size function (12) proposed in the paper. Besides this, use the three different step size mentioned in Table 1 to figure out the comparison table of mean-square error and operation required time. In the analogy of alternative step size function, although the arithmetical complexity is higher, it has preferred minimum mean squared error when data - 54.632dB is at a steady status according to Table 1.



(a)



(b)

Fig. 4 (a) undisturbed original signal figure, (b) add noise signal figure

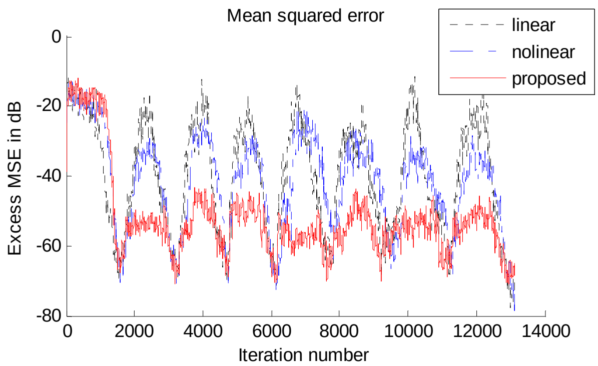


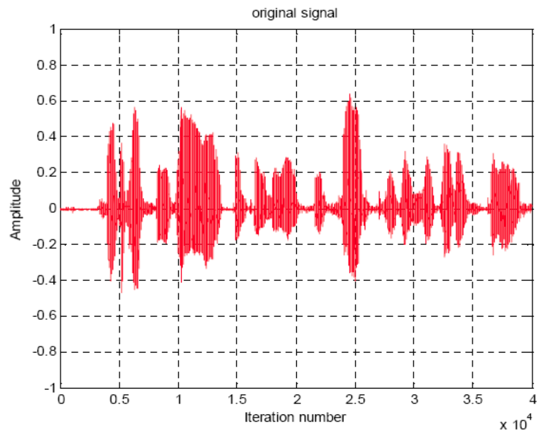
Fig. 5 the MSE of simulation one using three different step size methods

4.2 Simulation 2

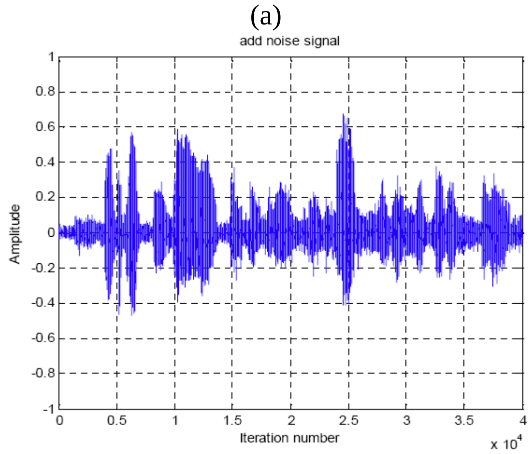
If the system environments setting remains the same, and put a 10KHz waveform sampling from the practical human speaking at the input end as is shown in Figure 6(a), while the noise is the passing sound of a actual train, after its mixture with the original signal, a 10KHz waveform sampled is shown in Figure 6(b). Use $V_{\max} = 30dB$ to carry out simulation and then observe its mean squared error variations as is shown in Fig. 7.

Table 1 The minimum mean squared error and operation time comparison table for simulation 1

Step size function	MSE(dB)	CPU time
Linear step size	-42.552 dB	49.887
Nonlinear step size	-45.954 dB (-7.99%)	49.959 (+0.14%)
Variable step size	-54.632 dB (-28.39%)	50.473 (+1.17%)



(a)



(b)

Fig. 6 (a) real signal figure, (b) added passing noise of a train signal figure

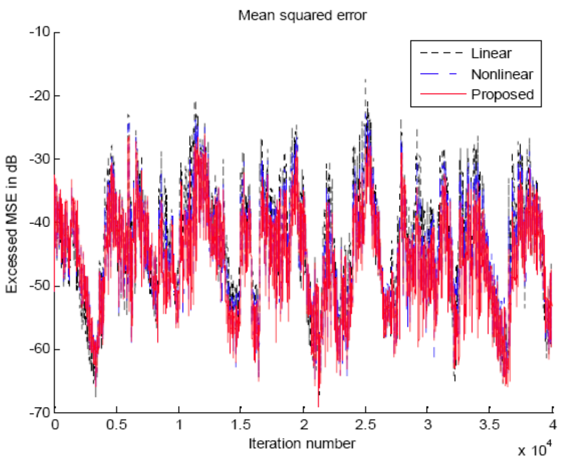


Fig. 7 the MSE of simulation two using three different step size methods

It is observed from this experiment which takes the real sound as the adaptive noise cancellation that although the minimum mean squared error of these three different algorithms, the linearity step size function proposed by Ikeda and the non-linearity step size function proposed by Ramadan and the alternative step size algorithm, is not as good as the result obtained in the aforementioned experiment, the alternative step size algorithm is still able to reduce the noise effectively. It is observed from Table 2 (comparison table of mean squared error and that operation time) that alternative step size approach is better than other step size algorithms, after the calculation of 40000 data; it could reduce the noise by -45.84dB.

Table 2 the minimum mean squared error and operation time comparison table for simulation 2

Step size function	MSE(dB)	CPU time
Linear step size	-43.096 dB	300.31
Nonlinear step size	-44.48 dB (-3.21%)	304.63 (+1.43%)
Variable step size	-45.84 dB (-6.36%)	305.49 (+1.72%)

5 Conclusion

The alternative step size algorithm proposed in this paper is to use the combination of two filters to realize noise cancellation, and this study has also compared it with the different adaptive algorithms simulating different step size. From the research direction, we could find out that the step size function is related to the rapidity of convergence as well as the size of final error. It is exactly for this reason that the adjustment of step size becomes more important. It is observed from the final simulation that although the arithmetical complexity of alternative step size is comparatively higher than other algorithms, it would produce a smaller error, proving that the step size algorithm is better.

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