A new variable step LMS algorithm and its application in blind satellite signals filtering

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Abstract. The variable step LMS algorithm can construct appropriate factor and adjust its filter parameters to achieve the optimal filter. But for the blind signals whose frequency bandwidth is different largely, the traditional variable step LMS algorithm can't meet the requirements. To design a new effective filtering algorithm, a correction function is introduced. It greatly improves the convergence speed of the algorithm, when the error is relatively large. The principle of the new algorithm is illustrated, and the simulation results show that the new algorithm is very effective. In the end, the new algorithm is applied in blind satellite signals frequency domain filtering, and good results are achieved.

Introduction

Developing satellite communications monitoring technology research, improving the level of monitoring of satellite communication technology and the ability of interference disposal are the effective ways to maintain normal order of satellite communications [1]. But the received signal affects by noise seriously and we may know nothing about the number of signals or the signals' center frequency, bandwidth and amplitude. It is very important to seek an effective way to deal with the satellite signal and obtain good waveform.

The variable step LMS algorithm can construct appropriate factor and adjust its filter parameters to achieve the optimal filter [2]. Adaptive filtering algorithm is an important link of the adaptive filter and there are several typical algorithms such as LMS adaptive filtering algorithm, RLS adaptive filtering algorithm, transform-domain adaptive filtering algorithm, affine projection algorithm, adaptive algorithm based on subband decomposition and conjugate gradient method, QR-decomposition-based least squares lattice adaptive filter algorithm[3-8]. LMS was put forward by Windrows and Hoff firstly and it is a typical algorithm of adaptive filter. It is simple in structure, small amount of calculation, stable in performance and easy to achieve, which is widely used in noise elimination, system identification, spectral enhancement and so on. Unfortunately, the contradiction between convergence speed and steady-state error restricts its application. In this paper, a new variable step LMS algorithm is put forward based on classic algorithm. It is applied in blind satellite signal frequency domain filtering and obtains good results.

The principle of LMS adaptive filter

Classic LMS adaptive filtering algorithm. The structure of adaptive filter is shown in Fig. 1.

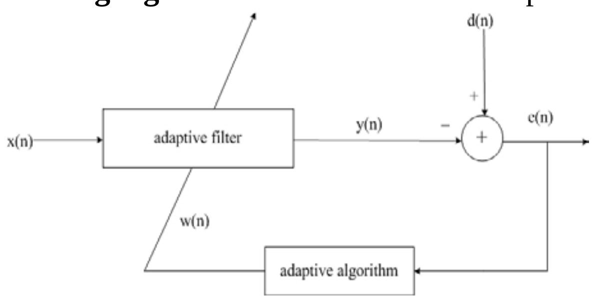

Fig. 1 Structure diagram of adaptive filter

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Where x(n) is the input of the adaptive filter, y(n) is the output of the adaptive filter, d (n)is the expected response, e (n) is the estimation error and w (n) is the weight coefficient.

And the basic formula is described as follow:

The output of the filter is given by Eq. 1

$$y(n) = W^T(n)X(n) = \sum_{i=1}^{M} \omega_i(n)x_i(n) \tag{1}$$

The error is calculated by Eq. 2.

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

The weight coefficient is updated by Eq. 3.

$$w(n+1) = w(n) + \mu e(n)x(n) \tag{3}$$

Where X(n) is the input vector, w(n) is the tap weight vector, M is the order of filter, μ is the step of the weight coefficient updating, and named as step length factor.

Variable step LMS adaptive filtering algorithm. Variable step LMS algorithm is a relatively simple and feasible algorithm and sigmoid variable step least mean square(SVSLMS)[9] put forward by Qin Jing-fan is a typical algorithm. Variable step function is defined as in Eq. 4.

$$\mu(n) = \beta\left(\frac{1}{1+e^{-c|e(n)|}} - 0.5\right) \tag{4}$$

Where c controls the Rising speed of the curve, β controls the scope of the dependent variable. In the initial stage, it has big step size and fast convergence speed because of the big value of e(n). And after the algorithm come into the steady state, step size becomes small with the value of e(n) becoming small gradually. To improve the performance of the algorithm, big step size is be applied to speed up convergence, when the error is big; and when the error is small, small step size is be used to obtain precision.

The algorithm in Literature [3] is improved on the basis of SVSLMS. Variable step function is defined as in Eq. 5.

$$\mu(n) = \beta\frac{1-e^{-c|e(n)|^3}}{1+e^{-c|e(n)|^3}} \tag{5}$$

The new variable step LMS adaptive filtering algorithm

The correction function. According to the analysis of the variable step adaptive filtering algorithm, the distortion of narrowband signal is mainly due to that the algorithm convergence speed is too slow in the rising edge and falling edge. A correction function is put forward according to the characteristic that the error is bigger in the Rising edge and Falling edge than others. And the algorithm is described as in Eq. 6.

$$y = 1 + N * \left(\frac{\arctan(a*(|e(n)|-b))}{\pi} + \frac{1}{2}\right) \tag{6}$$

Where y is correction factor, N>=0,a>=1,b>=0. N controls the scope of the dependent variable, a controls the rising speed of the curve, and b is determined by e(n).

The correction function curve is shown in Fig. 2.

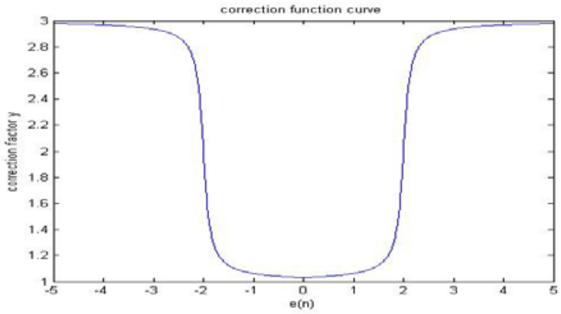

Fig. 2 The curve of correction function

The new filtering function. The new filtering function is concluded based on SVSLMS and correction function. The algorithm is described as in Eq. 7.

$$\mu(n) = \beta\left(\frac{1}{1+e^{-c|e(n)|}} - 0.5\right)\left(1 + N * \left(\frac{\arctan(a*(|e(n)|-b))}{\pi} + \frac{1}{2}\right)\right) \tag{7}$$

And the filter function curve is shown in Fig. 3.

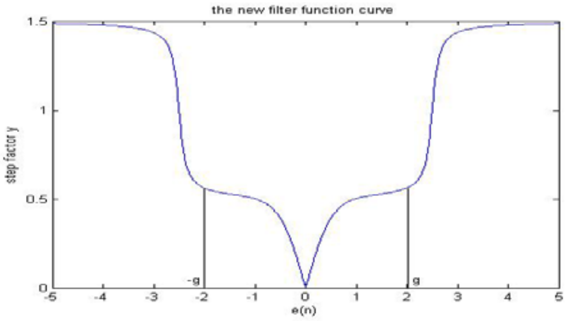

Fig. 3 The curve of new filter function

By the graph, the function maintains the nature of the SVSLMS when the error is less than the threshold g. But step size of the new function increases rapidly when the error is over the threshold g, which has a fast convergence speed. Generally, the value of parameter b and threshold g is very close because of the steep slope of the correction function and the Setting of threshold will be described in detail below.

The adaptive threshold. To filter the signal accurately, the setting of adaptive threshold is crucial. The new algorithm is based on the result of the SVSLMS. Assuming the separated noise obeys Gaussian distribution, we can realize that the location whose error value is more than two times the standard deviation is the location of rising edge and falling edge according to the nature of the Gaussian distribution(95.4% of the area within the scope of two standard deviation around the average). According to the above analysis, the algorithm of adaptive threshold is described as in Eq. 8.

$$g = u + 2\sigma \tag{8}$$

Where u is the mean and σ is the standard deviation of Gaussian distribution

Algorithm process. Specific algorithm process is described as follow:

①Preliminarily estimating noise based on SVSLMS；

②Automatically setting adaptive threshold by Eq. 8;

③ Blind satellite signal frequency domain filtering.

Performance analysis. The simulation signal is sine wave, whose amplitude is 4, frequency is 40hz and SNR is -5db. The algorithms of the literature[3], literature[9] and this paper are used to simulation signal filtering and the learning curve is shown in Fig. 4.

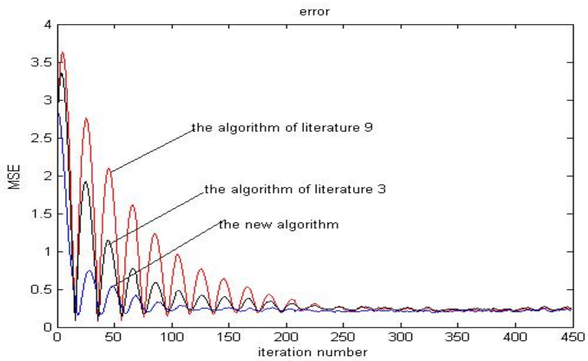

Fig. 4 The learning curve

From the graph, we can know that convergence speed of the new algorithm is significantly faster than the algorithm of the literature[3] and literature[9] while the steady state is the same as them.

Application in blind satellite signal filtering

The experimental data is a length of real-time acquisition satellite signal, whose sampling frequency is 50 Mhz, the number of sampling points is 10000. The satellite signal is shown as Fig. 5.

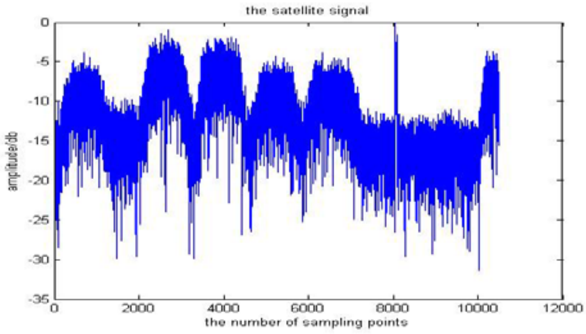

Fig. 5 The satellite signal

The algorithms of the literature[3], literature[9] and this paper are used to the satellite signal filtering . The wave after filtering is shown in Fig. 6 and Fig. 7.

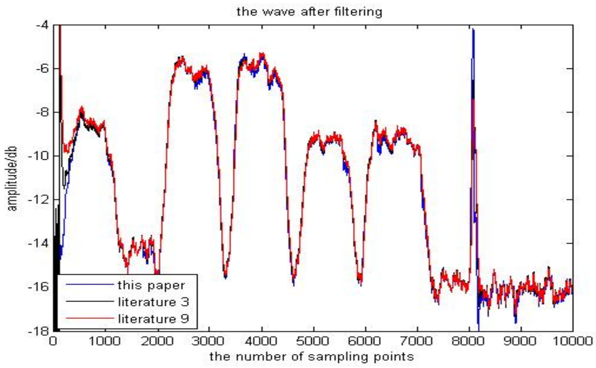

Fig. 6 The wave after filtering

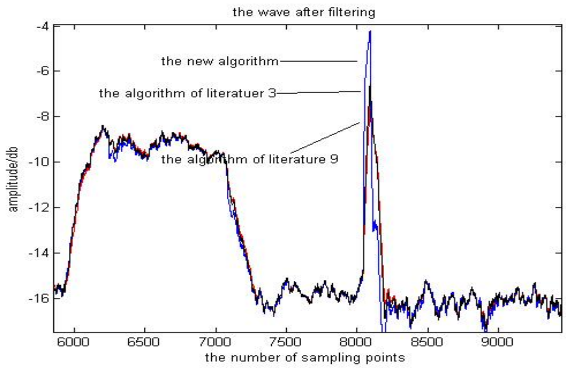

Fig. 7 a larger view of Narrowband

Through the comparison for the three algorithm filtering effect, we can visually find the new algorithm makes the amplitude increase nearly 1/4 and 3db bandwidth reduce approximately 1/2 about the narrowband signal. For the narrowband signal, the new algorithm restrains attenuation of amplitude and broadening of bandwidth while keeping the precision. So the new algorithm is very effective.

The error of three algorithms is shown in Fig. 8 and Fig. 9.

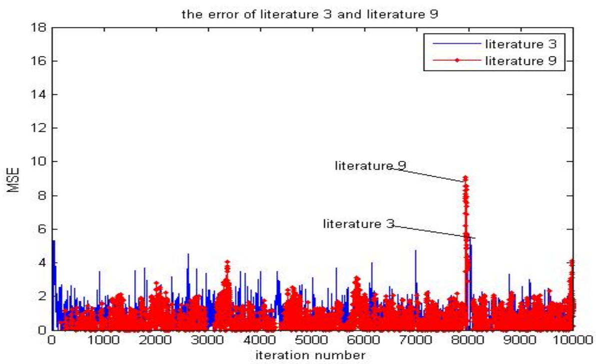

Fig.8 the error of the literature[3] and the literature[9]

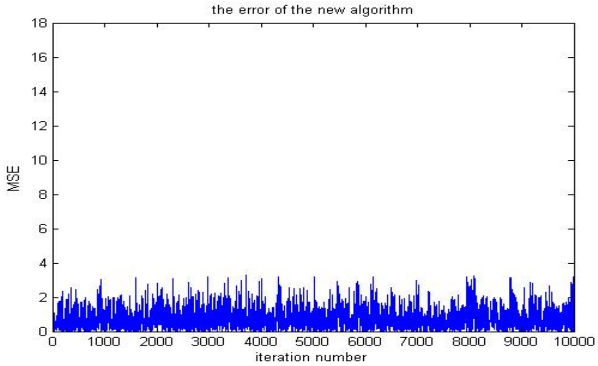

Fig. 9 the error of the new algorithm

From Fig. 8 and Fig. 9, we can visually find the error of rising edge and falling edge reduces evidently and especially in the location of narrowband (around 8000th point). It proves the validity of the new algorithm again.

Conclusion

To resolve the problem of blind satellite signals filtering, a new variable step LMS algorithm is put forward in this paper. It greatly improves the convergence speed of the algorithm by introducing a correction function, when the error is relatively big. And it can restrain attenuation of amplitude and broadening of bandwidth while keeping the precision of signals. The new algorithm is applied in the blind satellite signal frequency domain filtering and good results are achieved.

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