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Design and Research of Adaptive Filter Based on LabVIEW

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Abstract: LabVIEW is a figural virtual instrument program language, which has powerful digital signal processing function. As traditional tools such as MATLAB and Simulink are inconvenient to produce application systems. In order to solve the complex programming problem of adaptive filters, the Least Mean Square (LMS) and Recursive Least Square (RLS) algorithms based adaptive filters are designed based on adaptive filter toolkit of LabVIEW 2011. The parameters which can affect these two algorithms are carried on the analysis to filters' sensitivity. The simulation results show the function comprehensive characteristic of the adaptive filters with the good man- machine interface. It can be convenient for engineers to develop efficiently, So, the system developed in this paper has good practical engineering value. *Copyright © 2013 IFSA.*

Keywords: LabVIEW, Adaptive filter, LMS algorithm, RLS algorithm.

1. Introduction

Adaptive filter is an optimal filtering method, which is developed on the basis of the linear filters, such as the Wiener filter at the end of 1950's. This filter can achieve the best value through making the automatic adjustment parameters of adaptive filtering system. And in the design, little or no prior knowledge is required about the signal and noise [1]. Therefore, the adaptive filtering theory has the wide range of applications such as communication, remote sensing and telemetry, automatic control, radar and electronic jamming, pattern recognition, noise cancellation, channel equalization, echo cancellation, speech signal processing, biomedical and other aspects of access.

The core of the adaptive filter is the adaptive algorithms. At present, the realization of the adaptive filter is used by MATLAB, Simulink, DSP, FPGA and so on. Although this method can effectively realize the function of filtering, but the disadvantage is more complex programming and not friendly interface. At the same time, it is difficult to achieve the filter in practical engineering applications. Based on the mixed language programming of MATLAB and LabVIEW, the design of adaptive filter needs to use the MATLAB script node to invoke the complicated procedure. When the filter has higher accuracy requirements, the program will become quite complex, and it is not conducive to the rapid and effective design of adaptive filter [1].

Aiming at the shortcoming in the design of adaptive filter, This paper will use the toolkit of adaptive filter in LabVIEW 2011, which has realized the value of engineering application based on the least mean square(LMS) algorithm and recursive least square algorithm (RLS) of two kinds of adaptive filter. Its design is simpler in structure, and has good man-machine interface, which can be easily adjusted according to different requirements. It adapts to the actual signal analysis and processing for different occasions, and can bring very big convenient for the most optimal solution to the problem of research or engineering design.

2. Software of LabVIEW

LabVIEW is the abbreviation of Laboratory Virtual Instrument Engineering Workbench and is the graphical programming software which is launched by National Instruments Corp. LabVIEW is a graphical development environment which can produce the best code with compiler, and the operating speed equals to the compiled C or C++ program. It has the modular characteristics, which facilitates the reusability of the program. The knack of LabVIEW that it uses a graphical programming language to create the source in the block diagram. The diagram of program takes the place of the traditional code. So it is convenient and intuitive interface [2]. LabVIEW gives full play to the advantages of graphical language, which can greatly short of the virtual instrument development cycle, and eliminate the complex process of the virtual instrument programming [3].

3. The Principle and Algorithm of Adaptive Filter

3.1. The Principle of Adaptive Filter

The block diagram of adaptive digital filter is shown in Fig. 1. It is mainly composed of digital filter of adjustable parameters (also known as adaptive processor) and the adaptive filtering algorithm. The digital filter of adjustable parameters is FIR digital filter or IIR digital filter, and also is a lattice digital filter. The adaptive filter is divided into the linear adaptive filter and the nonlinear adaptive filter, where the nonlinear adaptive filter has better ability in signal processing, but the computation is complex, which is greatly limited in the practical engineering application. Therefore, the linear adaptive filter is more common in practical application. It has the advantages of simple structure and easy calculation algorithm.

In Fig. 1, x (n) represents the input of n moments, the output signal y(n) is generates by digital filter of adjustable parameters, so y (n) and the reference signal d (n) are compared to generate an error signal e (n). The error signal e (n) controls the impulse response of adaptive digital filter, which can be automatically controlled according to the e (n) value, and the input x (n+1) of the next moment (n+1) is suitable and in order to make the output y (n+1) is closer to d (n+l), until the error signal e (n) reaches to the minimum value, the y(n) is the best approximation of d (n), which enables the system to fully adapt to the two external added signals. The input signal x (n) and d(n) can be determined or random signal, and can also be a stationary random process or non - stationary random process [4].

The difference of adaptive filter and other filter is impulse response or filtering parameter which is changed with the external signal changes. The convergence time requires the period of automatic adjustment to achieve the best filtering effect. Adaptive filter has an algorithm according to the input, output, and parameter values, which modifies the filter parameters according to certain rules, so that it can effectively track the change of external signals. Therefore, an important aspect of the study on the adaptive filter theory is adaptive algorithm.

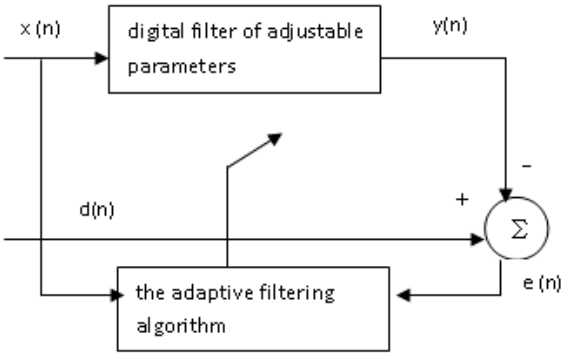


Fig. 1. The block diagram of adaptive digital filter.

3.2. The Algorithm of Adaptive Filter

The least mean square (LMS) and the recursive least square (RLS) are two basic algorithms of adaptive filter [4].

3.2.1. The Algorithm of the Least Mean Square (LMS)

In 1976 Widrow et al proposed the adaptive filtering algorithm of LMS, and due to the use of a special gradient estimation method, it avoids the disadvantages brought by general gradient estimation. Assuming the noise signal s (n) is zero, then the expected signal:

$$d(n) = h^T(n)x(n) \tag{1}$$

The estimation signal is described below.

$$y(n) = w^T(n)x(n) \tag{2}$$

Form (1) and (2), where x(n) is the input vector of the adaptive filter and $x(n) = \left[x(n), x(n-1), ..., x(n-L+1)\right]$; h(n) is the vector of the unknown system impact response and $h(n) = \left[h_0(n), h_1(n), ..., h_{L-1}(n)\right]^T$; $w(n)$ is the coefficient vector and $w(n) = \left[w_0(n), w_1(n), ..., w_{L-1}(n)\right]^T$;d(n) is the expected signal and $d(n) = d^{'}(n) + s(n)$. When the S (n) is zero, then $d^{'}(n) = d(n)$. Where e(n) is the error signal and

$$e(n) = d(n) - y(n) = d(n) - w^T(n)x(n) \quad (3)$$

The objective function is described below.

$$J(n) = E[e^2(n)] = E[(d(n) - w^T(n)x(n))^2] \quad (4)$$

For simplicity, usually in the definition of the objective function, the mean square error is replaced by the square error, which can prove that the error square is the unbiased estimate of mean square error. The objective function is also described below.

$$\hat{J} = e^2(n) \quad (5)$$

The objective function can be obtained by derivation.

$$\nabla\hat{J}(n) = \frac{\partial[e^2(n)]}{\partial[w(n)]} = -2x(n)e(n), \quad (6)$$

where it is available by the steepest descent method:

$$w(n+1) = w(n) + \mu[-\nabla\hat{J}(n)] = w(n) + 2\mu x(n)e(n) \quad (7)$$

Form (7) is the iterative formula of LMS algorithm, where n=1, 2, 3, … is time series; x (n) represents the input of n moments; e (n) is the output error signal and the output signal of the system; $w(n)$ is the weight vector of adaptive filter in n time.

3.2.2. The Algorithm of the Recursive Least Square (RLS)

Least square (LS) algorithm was as early as put forward by Gauss in 1795. But the computing amount of LS is big, therefore the recursive least square (RLS) algorithm is used in adaptive filtering, which is a recursive method for optimal solution of the algorithm, because RLS algorithm is better easily than LS and it is widely applied. In 1994, Sayed and Kailath established the relationship between Kalman filter and RLS algorithm, which proved that the RLS algorithm is actually a special case of the Kalman filter, and had the further understandings of the RLS algorithm. The large number of research results of Kalman filter can be applied to adaptive filtering, which played an important role in adaptive filtering technology [4].

RLS is a kind of least squares method of the most commonly fast algorithm. It is described below. Initialization is below:

$$P(0) = \delta^{-1}I, w(0) = 0 \quad (8)$$

On each sampling point of $n = 1, 2, ...$ for iteration:

$$\begin{aligned} k(n) &= \frac{\lambda^{-1}P(n-1)x(n)}{1+\lambda^{-1}x^T(n)P(n-1)x(n)} \\ e(n) &= d(n) - x^T(n)w(n-1) \\ w(n) &= w(n-1) + k(n)e(n) \\ P(n) &= \lambda^{-1}P(n-1) - \lambda^{-1}k(n)x^T(n)P(n-1) \end{aligned} \quad (9)$$

Form (9), during the initialization, δ is the small positive integer which is to ensure that the matrix is nonsingular and prevents the inverse of the matrix overflow.

4. The Design and Implementation of Adaptive Filter Based on LabVIEW

The toolkit of adaptive filter is provided in LabVIEW 2011, which has been designed by using various algorithms function VI and it can be very convenient to design the adaptive filter system which has practical engineering value. This method which is based on the graphical programming does not need complex programming, at the same time it can be more convenient and flexible to allow users to quickly develop adaptive filter. This paper gives the examples of adaptive filter which are based on the algorithms of LMS and RLS in LabVIEW 2011.

If the input simulation signal is set to the Sine wave signal which is superimposed by Gaussian white noise:

$$x(n) = A\sin(2\pi f_0 n) + v(n) \quad (10)$$

Form (10), where A is the amplitude of Sine wave signal, f_0 is frequency of the signal, $v(n)$ is Gaussian white noise with the mean value of zero, n is the length of the input signal, and σ is the amplitude of noise. Where the initial values are $A = 2.0$, $f_0 = 10\ Hz$, $\sigma = 0.1$.

The reference input signal is a sine wave signal, and it is described below.

$$x(n) = A\sin(2\pi f_0 n) \quad (11)$$

Form (11), where the initial values are $A = 2.0$, $f_0 = 10\ Hz$.

In addition to the RLS algorithm in the initial value, it is $x(0) = 0$.

The rest is similar with the LMS algorithm which $w(0) = 0$ is initial value of the weight vector. The parameters of the filter can be set on the front panel, such as the length, the iteration step and forgetting factor.

This paper gives the design of block diagram based on LabVIEW of graphical programming language. Through the input parameters are adjusted on front panel so that the filter output effect achieves

the best effect. Fig. 2 and Fig. 3 show respectively the adaptive filters' design and implementation of the above conditions based on the algorithms of LMS and RLS.

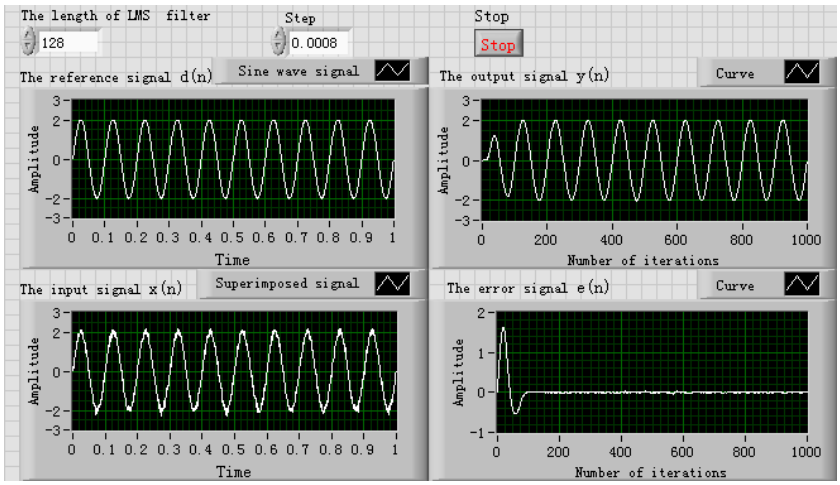
These experimental parameters are described below: the length of filter=128, step =0.0008, forgetting factor =0.995. From the simulation of Fig. 2 and Fig. 3 can show that the output effect of filter is more effective and the system's error is smaller.

5. The Parameters' Analysis of Algorithms of LMS and RLS

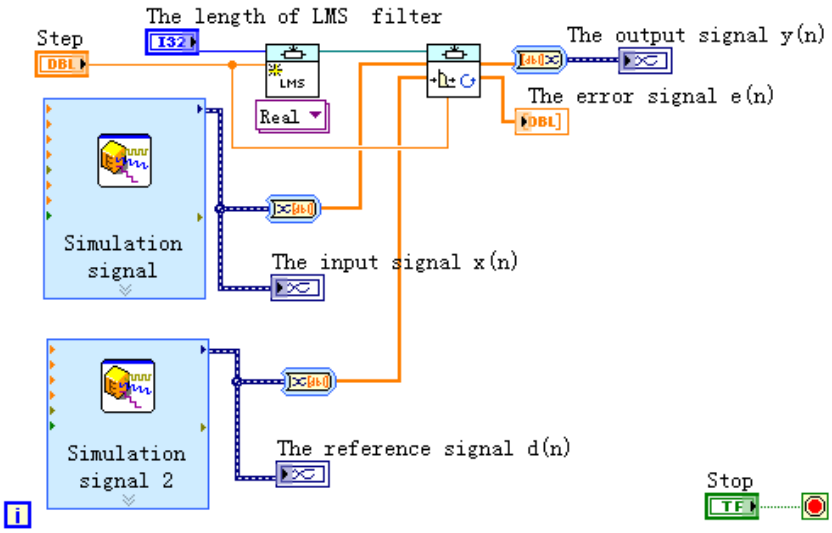
5.1. The Parameters' Analysis of LMS Algorithm

1) Where μ is the step factor. The theoretical range of μ is $0 < \mu < 1/\lambda_{\max}$ to keep the stability of the system. Where $\lambda_{\max}$ is the biggest feature value of correlation matrix R. When μ is smaller, the system is more stable and offset is smaller, so the adaptive time will be longer. So in considering the accuracy requirements, the user selects the appropriate step size factor to reduce the adaptive time [5].

2) In LMS algorithm, the fixed step size μ has the decisive impact on the performance of algorithm. It is used to describe the error adjustment according to the factor coefficients of digital filter. The convergence speed and the steady-state misadjustment size depend on the step size μ. For constant of μ, adaptive convergence speed and steady-state misadjustment size is a pair of contradiction. Getting a faster convergence speed can be increased the value of μ, but this will lead to increases of imbalance quantity.

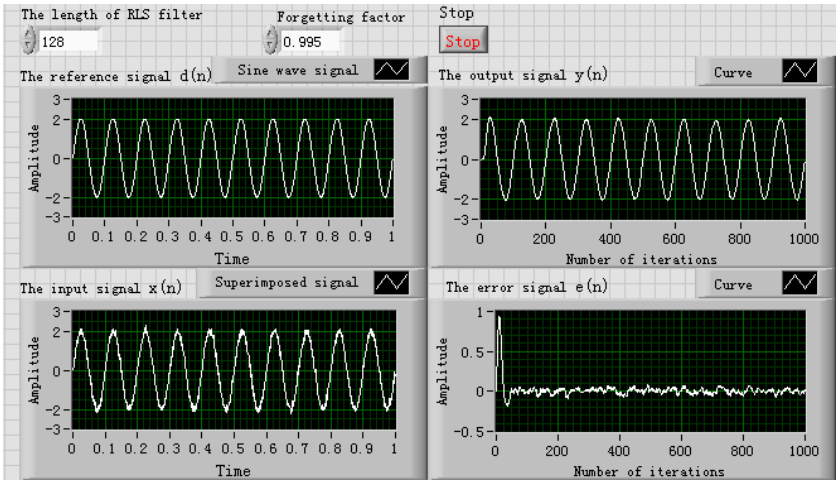


(a) The front panel of adaptive filter based on LMS algorithm.

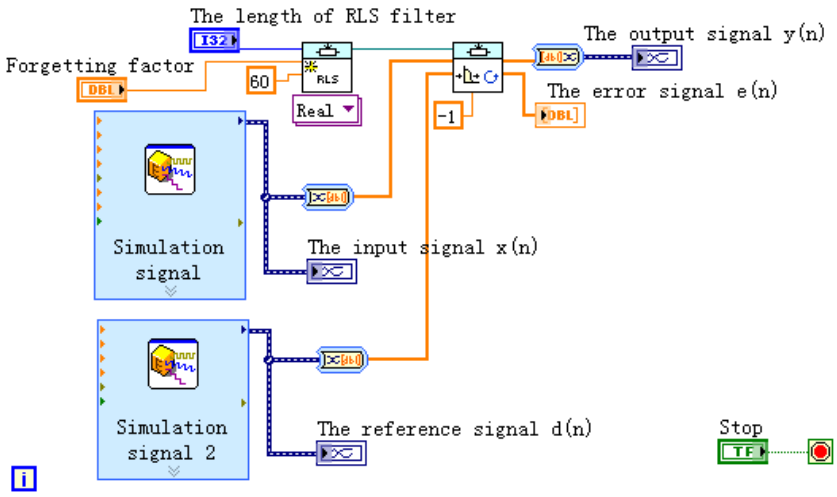


(b) The block diagram of adaptive filter based on LMS algorithm.

Fig. 2. The design and implementation of adaptive filter based on LMS algorithm.



(a) The front panel of adaptive filter based on RLS algorithm.



(b) The block diagram of adaptive filter based on RLS algorithm.

Fig. 3. The design and implementation of adaptive filter based on RLS algorithm.

If it wants to meet the requirements of the misalignment, the convergence process will be restricted. The convergence speed and steady-state misadjustment's error are two indicators which can not have both. The previous research on the performance of LMS algorithm has been made a lot of work, but it is still an important research topic.

5.2. The Parameters' Analysis of RLS Algorithm

1) RLS is an iterative algorithm for each step in the adaptive iteration requires. It is based on the least square criterion as the basis, the least squares criterion LS (Least Square) is to make a range of error square, i.e. $\sum_{i=k}^{k+m-1} e^2(n)$ is the minimum. The basic idea of least squares algorithm is to make the weighted least square error signal by adjusting the coefficients of filter.

2) The forgetting factor is smaller, then the time-varying parameters of the tracking ability are stronger, but also more sensitive to noise. In the design of filter, selecting the appropriate value of forgetting factor is very important. So some documents require that the value is $0.95 \leq$ forgetting factor ≤ 0.995 [6, 7].

3) The aspect of initial convergence speed, RLS algorithm is faster than LMS algorithm, and execution algorithm is more stability. But the computation of data is larger and more complex, at the same time the amount of storage required is larger, and timely effect is poor. If using the computer to store the mass data processing can bring greater convenience.

6. Conclusion

1) In this paper, it has achieved the design of adaptive filter based on LMS and RLS algorithm by using the toolkit of adaptive filter in LabVIEW 2011. Comparing to MATLAB, DSP, FPGA and other

software, the operation of adaptive filter in LabVIEW 2011 has good man-machine interface, and saves a lot of time for the engineering and technical personnel.

2) The simulation results show that the effect of filtering is good and strong practicability. In the design of LMS and RLS adaptive filter, this paper uses the signals which are superimposed by sinusoidal signal and Gaussian white noise. It can also choose other types of noise, and has good filtering effect.

3) But in the actual engineering application, the test environment is different so that the actual signal filtering is also different. The engineering and technical personnel choose the filter according to the actual situation of the parameters to achieve the filtering requirements. At the same time comparing with the traditional analysis method, it has obvious advantages in accuracy and operability etc.

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