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Adaptive filter and noise cancellation*

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This article will mainly based on the principle of adaptive filter and its application technology and make a concrete analysis and research, mainly discusses the FIR filter and adaptive noise canceller two types of filter, and starting from the basic principle of two filters, adopt different LMS algorithm MATLAB simulation experiment of the system. Nowadays, Because of the global stability of the FIR system and can guarantee the system of linear phase characteristic, through the software to design FIR filter, the structure of the software can automatically generate filter. Through the adaptive filter can estimate the noise in the original input, thus in the system output signal. In the application of adaptive filter is very broad, can the purpose of this paper is from the FIR filter and adaptive noise cancellation in found in adaptive filter, the different response of the filter selection according to the required performance need.

Keywords: LMS; Adaptive; FIR Filter.

1. Introduction

Adaptive filtering is developed on the basis of Wiener filter and Kalman filter, linear filter[1] a filtering method, the design of fixed filter relies on a priori statistical knowledge of the signal and noise, and adaptive filter is not required or only very little statistical a priori knowledge about the signal noise, which is through the self-regulating process to gradually adapt or tracking so as to continuously changing environment non-stationary random signal, and ultimately achieve the optimal filtering performance. Adaptive filter in the communication and signal processing applications, can provide a very effective to solve the received signal due to noise and affected the reliability of the received signal, thereby causing the error rate is rising. At present, the adaptive filtering technique has been successful in the application in the field of sonar, communication, radar, earthquake and biomedical engineering, etc..

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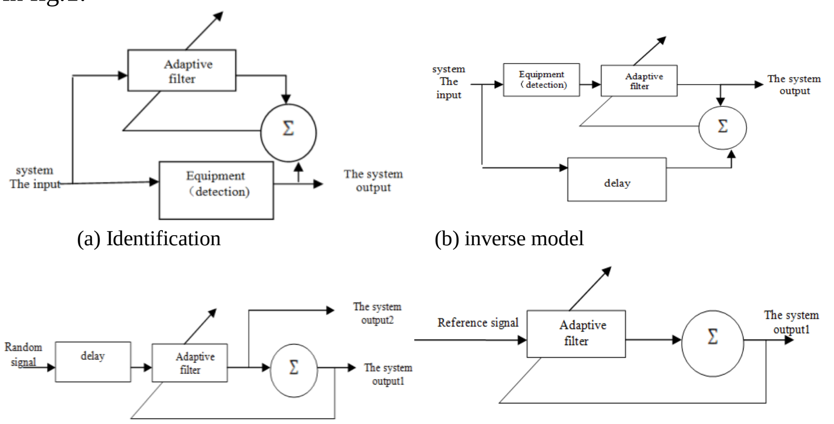
2. Adaptive Filter Theory Analysis and Calculation

2.1 *FIR Filter*

Adaptive filter has the ability to run well in unknown environment and the ability to track time-varying input statistics [1, 2], which makes the adaptive filter has become a powerful means of signal processing and automatic control applications. Applications of these characteristics have their own characteristics, also have a common feature: using the input vector, the expected response and to calculate the estimation error and the error in turn to control a set of tunable filter coefficients. The essential difference between the various applications of adaptive filters is that they are different in the way of obtaining the desired response [3, 4, 5]. Four basic types of adaptive filtering applications are shown in fig.1:



(a) Identification (b) inverse model

(c) Prediction (d) interference elimination

Fig. 1. Four basic types of adaptive filtering applications

FIR filter, it refers to the unit pulse response is limited to the length of the filter. Through the adaptive adjustment of the FIR filter, make it’s constantly modified its system function, make it good approximation and the unknown system parameters, eventually making error minimum, so as to achieve the purpose of system identification. In particular, the prominent characteristic of FIR filter is its unit sample response $h(n)$ is a finite length n point long sequence. The output $y(n)$ filter can be expressed as the input sequence $x(n)$ and unit sample response $h(n)$ of the linear convolution.

$$y(n)=\sum_{K=0}^{N-1}x(k)h(n-k)=x(n)*h(n) \tag{1}$$

The system function is:

$$H(Z)=\sum_{n=0}^{N-1}h(n)z^{-n}=h(0)+h(1)z^{-1}+...+h(N-1)z^{-(N-1)} \quad (2)$$

It can be seen that only exists in the origin of the filter poles, which makes FIR system has the global stability. The outstanding characteristic of FIR is linear phase characteristics of the filter can guarantee system, using MATLAB software to design FIR filter, the software will automatically generate the best filter structure.

2.2 *Adaptive Noise Canceller*

Adaptive noise cancellation technology [6, 7, 8] is widrow and puts forward a method of separation of the signal and noise, it can get a signal s n_0 from additive noise. It was the purpose of received signal to remove noise and to improve the signal-to-noise ratio. Its principle as shown in fig.2, it has two inputs, raw input $s+n_0$ and n_1 reference input. Except in the original input signal s n_0 also contains additive noise. s And n_0 is irrelevant, s n_1 and the reference input is irrelevant, but n_0 and n_1 is coherent. Fig.2 of the adaptive filter, adaptive filter for n_0 get associated with n_0 in the original input noise n_0^*, to offset the original interference noise n_0. Here, the system output y at the same time as the error e is used to automatically adjust the parameters of the adaptive filter, In this way, the prior knowledge of the signal and the noise does not need to know a lot, and the noise in the original input can be estimated by the adaptive filter, and then the signal is obtained in the system output.

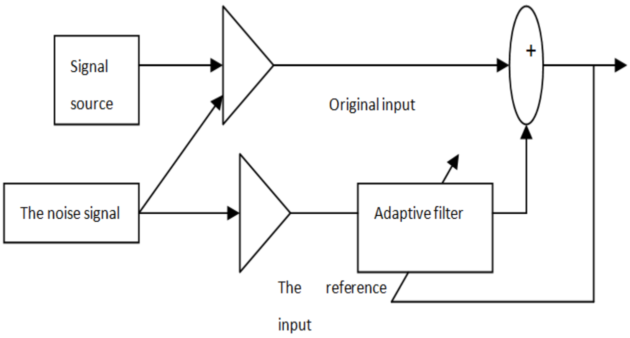


Fig. 2. Adaptive noise cancellation system

3. Conclusion

3.1 *Adaptive Matlab Modeling*

Adaptive modeling is divided into forward modeling and inverse modeling to forward modeling has been widely used in adaptive control system, coherence estimation and geophysical science and the design of digital filters. Reverse modeling is often used in adaptive control, speech analysis, channel equalization, digital filter, convolution, etc. Fig.1 based on the principle of adaptive filter identification fig.2, into the next phase of adaptive modeling, the modeling process is usually divided into three steps: select the model structure and order; Estimate the parameters of the model; Verify performance whether meet the requirements of the model, if does not meet the requirements, back to the first step to redesign. Based on the LMS algorithm for identification of FIR adaptive filter system, fig.3 results on MATLAB.

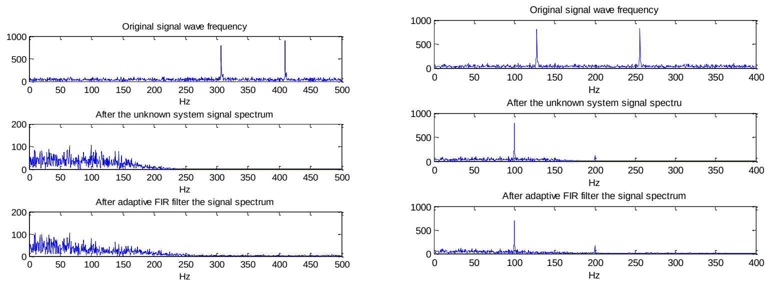


Fig. 3. FIR adaptive filter system

Adaptive FIR filter can well simulate the unknown system, the effect after it with the original signal processing is very close. Thus, as long as through the parameters of the adaptive FIR filter, get the unknown system function, and function of the unknown system can be the same hardware reconfiguration. This property has been widely used in engineering.

3.2 *MATLAB Offset of Adaptive Filter*

MATLAB by designing a weighted adaptive noise cancellation, for the additive white gaussian noise channel interference sinusoidal signal filtering, and eventually the following results were obtained as shown in fig.4: in the end by the adaptive LMS algorithm to adjust the weight coefficient of the linear combiner, and reference in the main channel and noise cancellation, the output error signal is the source of expected sine signal.

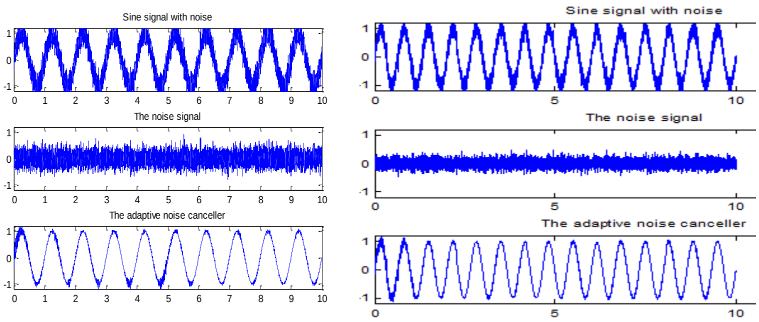


Fig. 4. Interference sinusoidal signal filtering

Adaptive filtering algorithm and parameter of digital filter can be adjusted by the adaptive filter, the content of this article first introduces the concept of adaptive filter principle and its application technology, based on the principle of adaptive filter, made of the two types of filter, and starting from the basic principle of two filters, adopt different LMS algorithm MATLAB simulation experiment of the system. And we introduce their application in daily life. It shows from the adaptive filtering has the very good situation, with the rapid development of science and technology, it involves in our daily life field more and more widely.

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