

# TJES

ISSN: 1813-162X

Tikrit Journal of Engineering Sciences

available online at: <http://www.tj-es.com>



## Suppression of the Narrow Band Jamming Signal in the Receiver of Spread Spectrum Communication System

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(Received 14 August 2012, Accepted 12 November 2012, Available online 18 September 2013)

### ABSTRACT

This paper describes the design of adaptive finite impulse response (FIR) filter with least mean square (LMS) algorithm by which we can reduce the narrow band jamming impacting the performance spread spectrum communication system (SSS). Simulation results shows that the jamming is rejected and the output signals free from error.

**Keywords:** LMS, Adaptive filter, FIR and MATLAB

أزالة اشارات التشويش ضيقة النطاق في المستقبل لمنظومات الطيف المنتشر

الخلاصة

البحث خاص بتصميم مرشح متكيف ذو استجابة نبضيه محدودة باستخدام خوارزمية نوع متوسط التربيع الأصغر. ذلك المرشح يقوم بتقليل التشويش ذو الطيف الترددي الضيق في منظومات اتصالات الطيف المنتشر. المحاكاة بينت ان التشويش المذكور تم حذفه نهائيا و الإشارة الخارجة اصبحت بدون خطأ.

### Introduction

Adaptive filters are very versatile device which is being researched continuously in various fields. It play very important role in the field of device noise cancellation, identification, noise reduction and equalization. There two most frequently applied algorithms for noise cancellation are least mean squares(LMS) and recursive least squares(RLS) algorithms. The main advantage of LMS is that it requires simple computation. FIR and IIR filters are used to satisfy some desired specifications. Our goal was to determine the coefficients of digital filter that meet the desired specifications. There are many digital signals processing applications in which the filter coefficients cannot be specified a priori. For example ,let as consider a high speed modem that is designed to transmit data over telephone channels. Such a modem employs a filter called a channel equalizer to compensate

the channel distortion[1].The modem must effectively transmit data through communication channels that have different frequency response characteristics and hence result in different distortion effects. The only way in which this is possible is if the channel equalizer has adjustable coefficients that can be optimized to minimize some measure of the distortion, on the basis of measurements performed on the characteristics of the channel. Such a filter with adjustable parameters is called an adaptive filter. The Pseudo noise(PN) signal is one of the spread spectrum signals with a special code has a low level of spectral density. The received additive narrow band jamming(NBJ) caused error of the PN signal detection. The adaptive filter with least mean square(LMS) algorithm is used to reduce the effect of NBJ.

Different researchers have worked on adaptive filters .In <sup>[2]</sup> Mbachu C.B. is worked on the use of

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the adaptive filter to reduce by about 13 dB power line interference electrocardiogram(ECG) recorders in medical fields. In [3] Pawan Nagle and Asst.Prof. Sumit Sharma are worked on reduction the noise impacting the performance of digital communication system using adaptive filter. The simulation results shows that the error has been reduced considerably adaptive filter compared with measurement results without adaptive filter. In [4] Yuka Morishita, Yusuka Tsuda, Toshiro Fujii and Tetsuya Shimamura are worked on the LMS adaptive equalizer using threshold in impulse noise environments ,where the variance of the received signal is calculated and use as the threshold for the LMS algorithm. In<sup>[5]</sup> Boza Krstajic, Zdravko Uskokovic are worked on adaptive channel equalizer with new LMS algorithm. In <sup>[6]</sup> Tan Zhong-ji, Shi Yan-li, Cui Ning are worked on the LMS adaptive filter used to noise cancelation when desired signal is sinusoidal and the noise sinusoidal or not sinusoidal.

### Direct form of adaptive FIR filter

Both FIR and IIR filters have been considered for adaptive filtering ,the FIR filter is by far the most practical and widely used. The reason for this preference is quite simple. The FIR filter has only adjustable zeros, hence it is free of stability problems associated with IIR filters that have adjustable poles as well as zeros[1].The direct form and lattice form structures(adaptive filter)are used for FIR filters. The direct form FIR filter structure with adjustable coefficients  $h(0),h(1),h(2),h(3) , \dots, h(N-1)$  is illustrated in Fig.1[1,2].

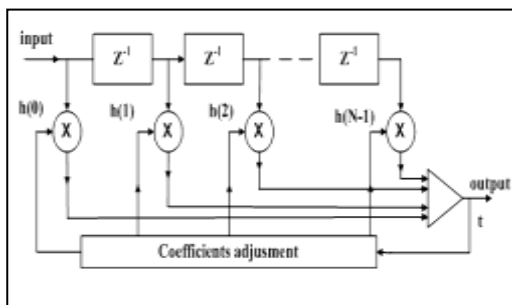


Fig.1.Direct form of FIR filter

### Least mean square (LMS) algorithm for coefficients adjustment

Suppose we have an FIR with adjustable coefficients  $\{h(k)\}$ ,  $0 \leq k \leq N-1$ . Let  $\{x(n)\}$  denote the input sequence to the filter and let the corresponding output by  $\{y(t)\}$ , where

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

Where  $n = 0, \dots, M$ .

Suppose that we also have a desired sequence  $\{d(n)\}$  with which we have compare the FIR filter output. Then we can form the error sequence  $\{e(n)\}$  by taking the difference between  $d(n)$  and  $y(n)$  [1,5,6] . That is,

$$e(n) = d(n) - y(n) \quad (2)$$

Where  $n = 0 \dots M$ .

Then coefficients of the FIR filter will be selected to minimize the sum of squared errors[2]. Thus we have

$$\begin{aligned} \varepsilon &= \sum_{n=0}^M e^2(n) = \sum_{n=0}^M [d(n) - \\ &\sum_{k=0}^{N-1} h(k)x(n-k)]^2 \\ &= \sum_{n=0}^M d(n)^2 - 2 \sum_{k=0}^{N-1} h(k)r(k)_{dx} + \\ &\sum_{k=0}^{N-1} \sum_{l=0}^{N-1} h(k)h(l)r(k-l)_{xx} \end{aligned} \quad (3)$$

Where,

$$r(k)_{dx} = \sum_{n=0}^M d(n)x(n-k), \quad 0 \leq k \leq N-1$$

And

$$r(k)_{xx} = \sum_{n=0}^M x(n)x(n+k), \quad 0 \leq k \leq N-1$$

we call  $\{r(k)_{dx}\}$  the cross correlation between the desired output sequence  $\{d(n)\}$  and the input sequence  $\{x(n)\}$ , and  $\{r(k)_{xx}\}$  is the autocorrelation sequence of  $\{x(n)\}$ .

The sum of squared error  $\varepsilon$  is a quadratic function of the FIR filter coefficients. Consequently, the minimization of  $\varepsilon$  with respect to the filter coefficients  $\{h(k)\}$  results in a set of

linear equations[2]. By differentiating  $\varepsilon$  with respect to each of the filter coefficients, we obtain

$$\frac{d\varepsilon}{dh(m)} = 0 \quad (4)$$

$$0 \leq m \leq N - 1$$

And hence

$$\sum_{k=0}^{N-1} h(k)r(k-m)_{xx} = r(m)_{dx} \quad (5)$$

$$0 \leq m \leq N-1$$

This is the set of linear equations that yield the optimum filter coefficients. To solve the set of linear equations directly, we must first compute the autocorrelation sequence  $\{r_{xx}(k)\}$  of the input signal and cross correlation sequence  $\{r_{dx}(k)\}$  between the desired sequence  $\{d(n)\}$  and the input sequence  $\{x(n)\}$ .

The LMS algorithm provides an alternative computational method for determining the optimum filter coefficients  $\{h(k)\}$  without explicitly computing the correlation sequences  $\{r_{xx}(k)\}$  and  $\{r_{dx}(k)\}$ . The algorithm is basically a recursive gradient (steepest descent) method that finds the minimum of  $\varepsilon$  and thus yields the set of optimum filter coefficients[1].

We begin with any arbitrary choice for the initial values of  $\{h(k)\}$ , say  $\{h_0(k)\}$ . For example, we may begin with  $h_0(k)=0$ ,

$0 \leq k \leq N-1$ . Then after each new input sample  $\{x(n)\}$  enters the adaptive FIR filter, we compute the corresponding output, say  $\{y(n)\}$ , form error signal  $e(n)=d(n)-y(n)$ , and update the filter coefficients according to the equation :

$$h(k)_n = h(k)_{n-1} + \Delta \cdot e(n) \cdot x(n-k) \quad (6)$$

Where  $0 \leq k \leq N-1$ ,  $n=0,1,\dots$ ,  $\Delta$  is called the step size parameter,  $x(n-k)$  is the sample of the input signal located at the  $k_{th}$  tap of the filter at time  $n$ , and  $e(n) \cdot x(n-k)$  is an approximation (estimate) of the negative of the gradient for the  $k_{th}$  filter coefficient. This is the LMS recursive

algorithm for adjusting the filter coefficients adaptively so as to minimize the sum of squared errors. The step size parameter  $\Delta$  controls the rate of convergence of the algorithm to the optimum solution<sup>[1]</sup>. A large value of  $\Delta$  leads to large step size adjustment and thus to rapid convergence, while a small value of  $\Delta$  results in slow convergence. However, if  $\Delta$  is made too large the algorithm becomes unstable. To ensure stability,  $\Delta$  must be chosen to be in the range

$$0 \leq \Delta \leq \frac{1}{10NP_x} \quad (7)$$

Where  $N$  is the length of the adaptive FIR filter and  $P_x$  is the power of the input signal, which can be approximated by

$$P_x = \frac{1}{1+M} \sum_{n=0}^M x(n)^2 = \frac{r(0)_{xx}}{M+1} \quad (8)$$

### Suppression of narrowband interference in a spread spectrum communication system

Let us assume that we have a signal sequence  $\{x(n)\}$  that consists of a desired spread spectrum [pseudo noise (PN)] signal sequence, say  $\{w(n)\}$ , corrupted by an additive narrowband interference sequence  $\{s(n)\}$ . The sequences  $w(n)$  and  $s(n)$  are uncorrelated. This problem arises in digital communications and in signal detection, where the desired signal sequence  $\{w(n)\}$  is a spread spectrum signal, while the narrowband interference represents a signal from another user of the frequency band or some intentional interference from a jammer who is trying to disrupt the communication or detection system[1,3,6]. For simplicity of analysis we use sinusoidal interference sequence  $\{s(n)\}$  as a jamming signal[7].

$$s(n) = A \sin(\omega_0 n) \quad (9)$$

Where  $0 \leq \omega_0 \leq 2\pi$

The suppression of jamming  $s(n)$  shown in Fig. 2.

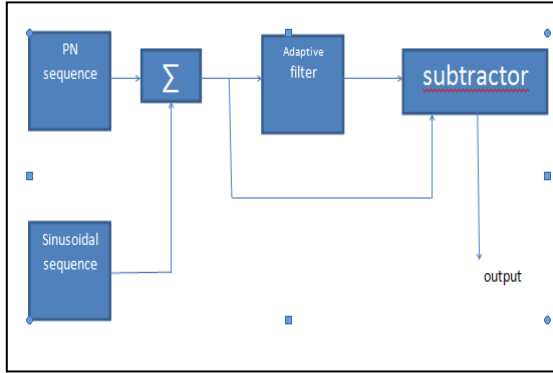


Fig.2. Block diagram of Jamming suppression.

**Simulation of the jamming suppression using MATLAB**

We make the program by MATLAB. By this program we generate the spread spectrum signal (pseudo noise signal) and narrow band jamming signal (for simplicity we used pure sinusoidal signal as the jamming). This program also consist the processing of the jamming rejection. Know we discuss the results as follow:

a). The generated sequence (input signal) as follow:  
 1 1 1 -1 -1 -1 -1 1 1 -1 1 1 1 -1 1  
 -1 -1 1 1 -1 -1 -1 1 -1 1 -1 1 1 -1 -1 1 -1 1  
 1 1 1 -1 -1 -1 1 1 1 -1 1 1 -1 1 1 -1 -1 1 -1 1  
 1 -1 -1 1 -1 -1 -1 -1 1 -1 -1 1 1 -1 -1 1 -1 1  
 1 -1 -1 -1 1 -1 -1 -1 1 1 -1 -1 1 1 -1 1 -1 1 -1  
 -1 -1 -1 -1 -1 -1 1 1 1 1 1 -1 1 1 1 1 1 -1 -1 1 1  
 1 1 -1 1 -1 1 1 1 -1 -1 -1 -1 1 1 1 -1 1 -1 -1  
 1 1 -1 -1 -1 1 -1 1 -1 1 1 -1 -1 -1 -1 1 -1 1 1 1  
 1 -1 -1 -1 1 1 1 -1 1 1 -1 1 1 -1 -1 1 -1 -1 1 -1  
 -1 1 1 -1 -1 -1 -1 1 -1 -1 1 1 1 -1 -1 1 1 -1 1 -1  
 -1 -1 -1 1 -1 -1 -1 1 1 -1 -1 1 1 -1 -1 1 1 -1

This sequence is shown in Fig.(3).

Fig.(4) shows the input signal plus jamming ,jamming, FFT of input signal and jamming. This jamming signal is very strong compared with pseudo noise(PN) signal shown in Fig.(3).

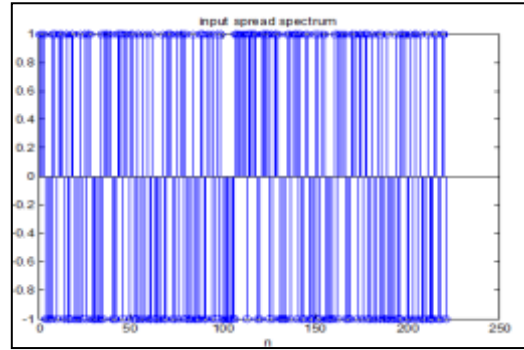


Fig.3. Input spread spectrum signal.

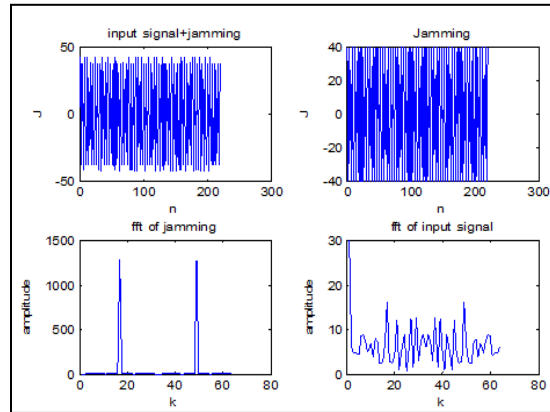


Fig.4. Input signal plus jamming jamming, FFT of input signal and jamming.

The adaptive filter automatically tuned at the frequency of jamming. This is a good characteristic of this filter because the frequency of jamming may be changed and the adaptive filter returned at new frequency. At the output of adaptive filter is only jamming. Subtractor in Fig.(2) subtract the jamming from the summation of PN signal and jamming signal. The output is spread spectrum signal (PN sequence) as follow :

1 1 1 -1 -1 -1 -1 1 1 -1 1 1 1 -1 1 -1 -1 1 1 -1 -1 -  
 1 1 -1 1 -1 1 1 -1 -1 -1 -1 1 -1 1 1 1 1 -1 -1 -1  
 1 1 1 -1 1 1 -1 1 1 -1 -1 1 -1 -1 1 -1 -1 1 -1 -  
 -1 -1 -1 1 -1 -1 1 1 1 -1 -1 1 -1 1 1 -1 -1 -1 1 -  
 -1 -1 1 1 -1 -1 1 1 -1 1 -1 1 -1 -1 -1 -1 -1 -1  
 -1 1 1 1 1 1 1 -1 1 1 1 1 1 -1 -1 1 1 1 1 -1 1 -1  
 1 1 -1 -1 -1 -1 1 1 -1 1 1 1 -1 1 -1 -1 1 1 -1 -1  
 1 -1 1 -1 1 1 -1 -1 -1 -1 1 -1 1 1 1 1 -1 -1 1  
 1 1 -1 1 1 -1 1 1 -1 -1 1 -1 -1 1 -1 -1 1 -1 -1  
 -1 1 1 -1 -1 1 1 1 -1 1 -1 1 1 -1 -1 -1 -1 1 -1  
 -1 1 1 1 -1 -1 1 1 -1

This sequence is shown in Fig.(5).

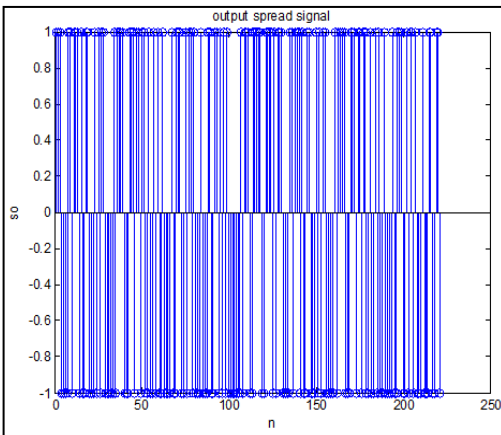


Fig.5.Output PN signal

We see from last sequence and Fig.(5) that the output and input signals are identical. This mean that the operation of adaptive filter is correct. Adaptive FIR is very stable and automatically tuned to new frequency jamming. The step size of the LMS adaptive filter in this paper we change continuously to find the good result for jamming suppression. Fig.(6) shows cross correlation between the output spread spectrum signal and the input spread spectrum signal for two threshold.

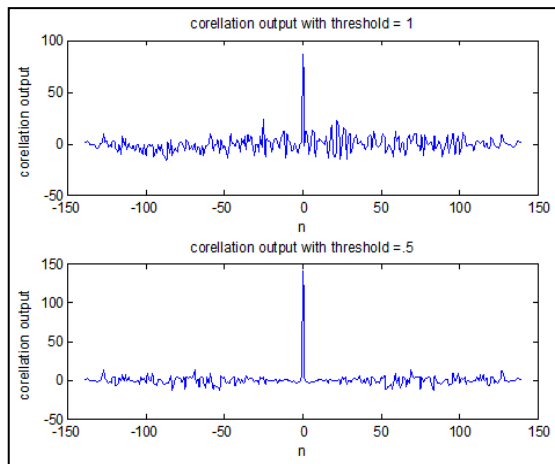


Fig.6.Correlation of output spread spectrum with two threshold.

First threshold equal '1' volt, this mean that threshold equal the pulse amplitude of input and output spread spectrum signals. Second threshold equal 0.5 volt, i.e. equal 50% of pulse amplitude of last signals. The peak of the

correlation output higher than maximum side lobe by about 11dB when the threshold is 0.5volt, and higher than maximum side lobe by about 4dBwhen the threshold is 1volt.

## Conclusions

This paper has studied the theory of adaptive filter based on LMS algorithm to suppress the narrow band jamming in the spread spectrum system. From the result of simulation by MATLAB we see that the output of PN sequence with very small error. The cross correlation between output and input spread spectrum PN sequence has a large one peak and small side lobs, i.e. FIR adaptive filter is effectively cancel the strong narrow band jamming.

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