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ADAPTIVE LINE ENHANCEMENT IN PRESENCE OF IMPULSIVE AND GAUSSIAN NOISES

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Abstract

In this paper, a proposed scheme for an adaptive line enhancement (ALE) in presence of impulsive and Gaussian noises is presented. The proposed scheme is composed of an adaptive line enhancer and a median filter. The object of this scheme is to obtain a best estimate of the desired signal which is combined with an additive impulsive and Gaussian noise at very low input signal to noise ratio. The median filter is used mainly to remove the impulsive noise from the input observation. The resulting signal is applied to the adaptive line enhancer to obtain a good estimate of the desired signal. The adaptive line enhancer is implemented by an adaptive transversal filter whose output is assumed as a good estimate of the desired signal based on the minimization of the mean square error criterion. The filter coefficients are updated according to the least mean square (LMS) adaptation algorithm. The performance of the proposed scheme is evaluated in presence of the impulsive and Gaussian noise. It is concluded that using the adaptive line enhancer and median filter exhibits a better performance than that of the stand alone adaptive line enhancer.

Keywords

Adaptive line enhancer, adaptive filter, adaptation algorithm, median filter, impulsive noise, Gaussian noise.

1. Introduction

The adaptive signal enhancement is applied successfully in signal estimation and spectrum analysis especially when the observation signals are deeply corrupted with

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noise or interference at a very low signal to noise ratio [1,4]. The smoothing techniques used in signal processing are generally based on the assumption that the noise is Gaussian. This is not usually justified and may not be a reasonable assumption for practical systems. Indeed, the environments commonly seen in wireless communications are primarily characterized by bursty and impulsive statistics, which are non-Gaussian [9]. The presence of impulsive noise generally degrades the performance of the adaptive line enhancer, and in extreme cases can lead to instability. This paper presents a scheme for adaptive line enhancement in presence of impulsive and Gaussian noises. The proposed scheme is based on a combined structure of the adaptive line enhancer and median filter. The median filter is used to cancel the impulsive noise and its output is applied to the adaptive line enhancer to obtain a good estimate of the desired signal based on the minimization the mean square error criterion.

The paper consists of six sections. Section 2 explains the principle of the adaptive line enhancer and its adaptation algorithm while section 3 presents the median filter. Section 4 presents the proposed scheme for adaptive line enhancement in presence of impulsive and Gaussian noises. Section 5 is concerned with simulation results. Section 6 presents conclusion of the paper.

2. The Adaptive Line Enhancement

The principle of the adaptive line enhancer (ALE) as depicted in Figure 1 is to estimate a band limited signal that is buried in white noise or wide band interference signal. The observed or received signal is delayed by integer values of the estimated fundamental period of the band-limited signal. The effect of the time delay is to decorrelate the band-limited signal and the broad band combined noise. The delayed observation is adaptively filtered and is subtracted from the undelayed observation to produce an error signal. The role of the adaptive filter is to obtain a good estimate of the desired signal based on the minimization of the mean square of the deviation error between the undelayed noisy observation and the filter output [1,2,7].

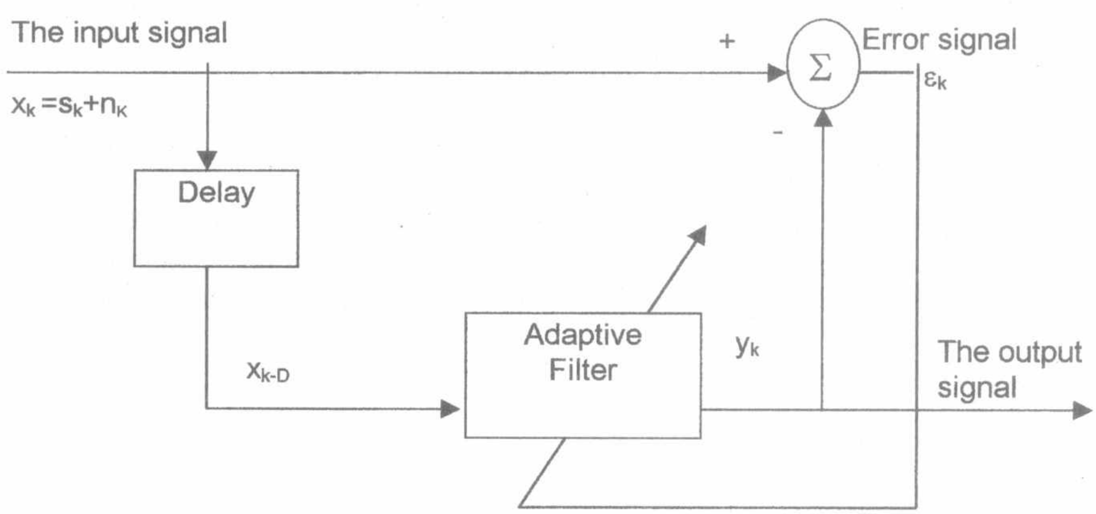

Fig. 1 The adaptive line enhancer configuration

The received signal, $\mathbf{x_k}$ can be written as:

$$\mathbf{x_k = s_k + n_k} \tag{1}$$

where $\mathbf{s_k}$ denotes the desired signal and $\mathbf{n_k}$ denotes the interference or noise signal. The input signal to noise ratio (ISNR) is defined by:

$$\mathrm{ISNR} = \frac{E[s_k^2]}{E[n_k^2]} \tag{2}$$

where $\mathbf{E[s_k^2]}$ and $\mathbf{E[n_k^2]}$ refer to the signal and noise powers respectively. The filter output is subtracted from the undelayed observation signal to generate an error signal as:

$$\varepsilon_k = \mathbf{x_k - y_k} \tag{3}$$

Substituting eq.(1) in eq.(3) yields:

$$\varepsilon_k = \mathbf{s_k - y_k + n_k} = \eta_k + \mathbf{n_k} \tag{4}$$

Where η_k is defined as the deviation error between the desired signal and its estimate; i.e.

$$\eta_k = \mathbf{s_k - y_k} \tag{5}$$

The mean square error criterion [1,3]. Is usually used in adaptive filtering. Assuming that the additive noise $\mathbf{n_k}$ is uncorrelated with the error signal η_k, the mean square error is given by

$$\mathbf{E[\varepsilon_k^2] = E[\eta_k^2] + E[n_k^2]} \tag{6}$$

The expectation of the error signal η_k can be expressed as:

$$\mathbf{E[\eta_k^2] = E[s_k^2] - 2E[s_k\, y_k] + E[y_k^2]} \tag{7}$$

It is apparent that $\mathbf{E[\eta_k^2]}$ is a function of the adaptive filter weights. The optimum filter weights are those that minimize $\mathbf{E[\eta_k^2]}$. In order to attain these coefficients, the least mean square (LMS) algorithm [1,2,7] is used. The LMS algorithm is given by

$$\omega_{k+1} = \omega_k + 2\,\mu\,\varepsilon_k\,\mathbf{x_k} \tag{8}$$

Where μ represents the step sizes that controls the adaptation speed and the steady state misadjustment of the adaptation algorithm and ω_k **is** the filter coefficient vector.

The output signal to noise ratio (OSNR) of the adaptive enhancer can be written as:

$$OSNR = \frac{E[s_k^2]}{MinE[\eta_k^2]} \tag{9}$$

The SNR improvement offered by the adaptive enhancer is given by

$$IMP = \frac{OSNR}{ISNR} \tag{10}$$

Substituting eqs. (2) and (9) in eq. (10) yields:

$$IMP = \frac{E[n_k^2]}{MinE[\eta_k^2]} \tag{11}$$

3. The Median Filtering

In most signal processing applications a linear smoothing is generally used to eliminate the noise-like components of a signal. For some signal processing, linear smoothers are not completely adequate due to the type of observation signals. For such cases, some type of nonlinear smoothing algorithm can be used [6]. The nonlinear smoothing technique that has probably found most common usage is the median filter [5]. Such filter is not linear and hence cannot be characterized in terms of simple transfer function. It is characterized by two key properties [4]:

i) Median filter completely removes short duration impulses from signal.

ii) Median filter passes edges without smearing.

Impulsive noises are infrequent bursts of high amplitude noise. They can occur from atmospheric disturbances and high power transients. In particular, the amplitude and arrival time are the main parameters of the impulsive noise [8]. The received signal x_K can be expressed in presence of impulsive and Gaussian noises as:

$$\mathbf{x_k = S_k + n_k + I_k} \tag{12}$$

Where $\mathbf{I_k}$ is the impulsive noise sample received at time k.

The role of the Median filter is to cancel the impulsive noise from a window that contains **M+1** samples of received signal in which one or more samples are corrupted with the impulsive noise. The selected window can be mathematically expressed as:

$$\mathbf{W_M = [\, x_{k-M} \; x_{k-M+1} \ldots\ldots \; x_{k-1} \; x_k \,]} \tag{13}$$

The median of **M+1** samples is obtained by algebraically ranking the samples from smallest to largest and then selecting the middle value. The median filter output is represented as the median value of the selected window samples [4,5,6]. This is designated as

4. The Proposed Scheme of The ALE

The proposed scheme depicted in Figure 2 depends mainly on the joint-operation of a median filter and an adaptive line enhancer. In the proposed model, Median filter is presented as a pre-Adaptive Line Enhancer (PALE) stage, the impulsive noise is cancelled by the median filter and the resulting signal is applied to the ALE to obtain a good estimate of the desired signal. In the proposed model, the median filter operates only when the estimated input signal power $\hat{\sigma}_x^2$ exceeds some selected threshold, and it is excluded as there is no impulsive noise to decrease the complexity computation. Moreover, the median filter impact negatively the system performance in case of the absence of the impulsive noise.

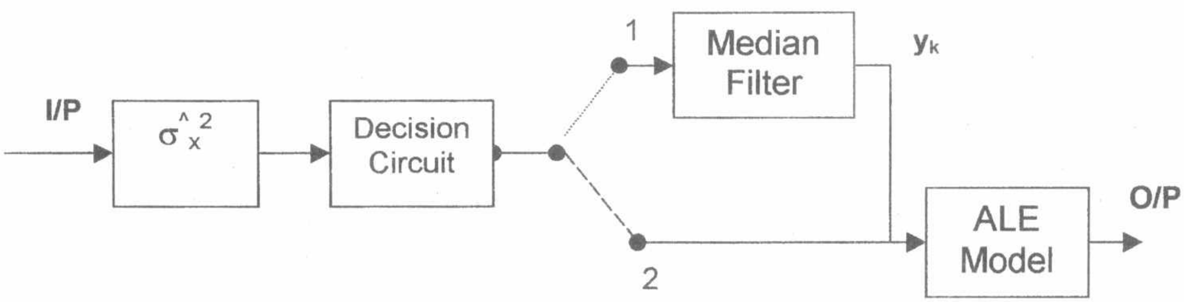

Figure 2. Combination of adaptive line enhancer and median filter.

5. Simulation Results

The adaptive line enhancer depicted in Figure 1 is used to enhance a signal that is corrupted with a broad band Gaussian noise, while the proposed system depicted in Figure 2 is used to enhance a signal that is corrupted with both a broad band Gaussian noise and an impulsive noise. The desired signal, s_k is modeled as: $s_k = A \sin(2\pi f T_s k)$. Where its frequency; f=1000 Hz. The sampling frequency is 10000Hz. A white Gaussian and impulsive noises are generated and added to the desired signal at different signal to noise ratios. The performances of stand alone ALE and the proposed schemes are evaluated in three cases.

5.1 Case 1 performance of ALE in presence of white Gaussian noise

Figure 3 illustrates the desired signal. Figure 4 gives the signal after adding a Gaussian noise at SNR=-7.46 dB Figure 5 gives the output signal of the ALE. The output SNR is 14.38 dB corresponding to an improvement factor of 21.84 dB.

5.2 Case 2 performance of ALE in presence of white Gaussian and impulsive noises

Figure 6 presents the signal with Gaussian and impulsive noise at SNR=-8.21dB. Figure 5 gives the output signal of the ALE. The output SNR is 1.24 dB corresponding to an improvement factor of 9.45dB.

5.3 Case 3 performance of the proposed scheme in presence of white Gaussian and impulsive noises

Combining of Median filter and adaptive line enhancer depicted in Figure 2 is used to enhance signal that is corrupted with Gaussian and impulsive noise as shown in Figure 6. Figure 8 presents the output signal of median filter. Figure 9 gives the final output signal of the proposed scheme. The output SNR is 15.7dB corresponding to an improvement factor of 23.91 dB. It is concluded from Figure 7 and Figure 8 that the median filter provides an improvement factor of 14.46 dB in the output signal to noise ratio.

By comparing the simulation results of the above cases, we note that the performance of the stand alone ALE in presence of Gaussian noise is good while its performance in presence of Gaussian and impulsive noises is bad. The proposed scheme exhibits high performance in presence of Gaussian and impulsive noises.

6. Conclusion

A proposed scheme, which consists of a median filter and an adaptive line enhancer, can be used to enhance signal that is corrupted with Gaussian and impulsive noises. It improves the output signal to noise ratio especially at very low input SNR. The presence of impulsive noise has degrading effect on the performance of the conventional adaptive line enhancer and can lead to instability. The stability of the proposed scheme and its performance in presence of Gaussian and impulsive noises are good. The effect of nonlinearty of the median filter will be studied intensively in the future work.

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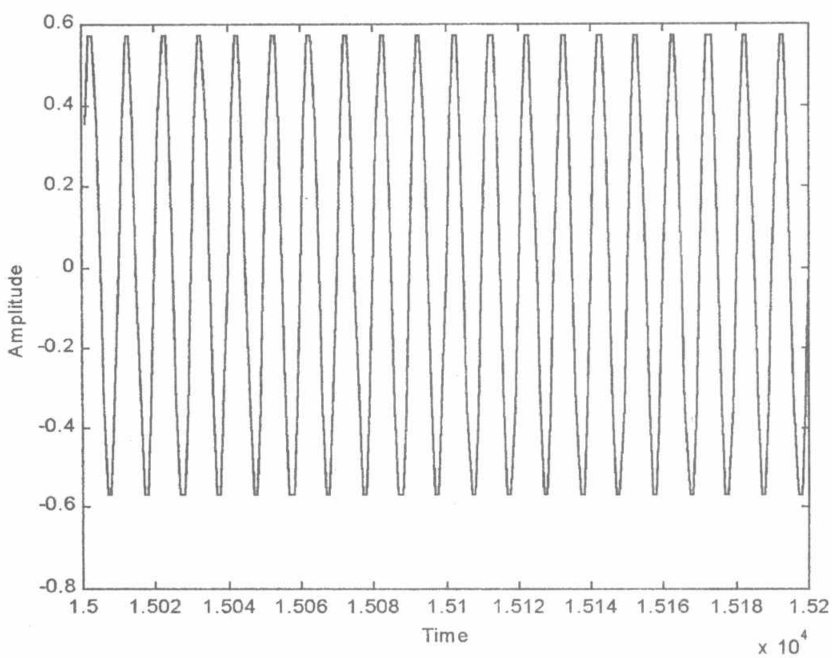

Fig. 3 The desired signal s_k (A=0.6)

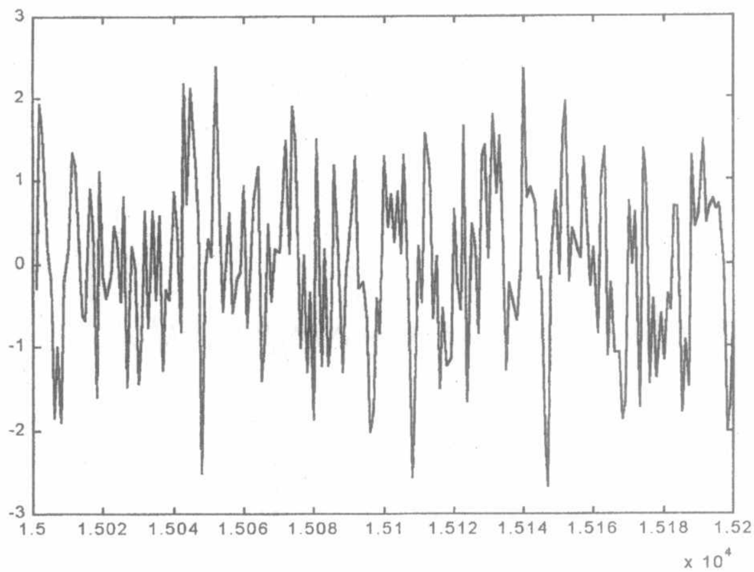

Fig. 4 The Gaussian noisy input signal x_k (ISNR=-7.46 dB)

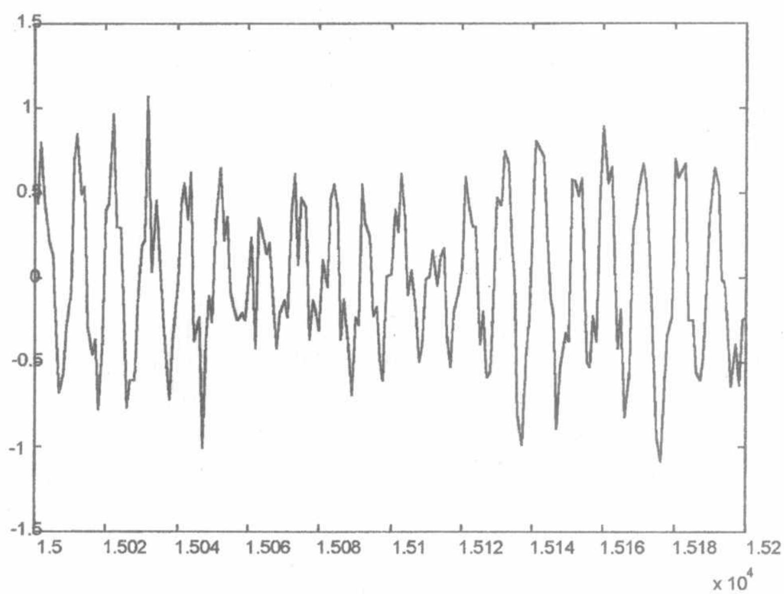

Fig. 5 The output signal y_k of stand alone ALE (N=30, μ=0.001412, SNRO=14.38 dB, IMP=21.8)

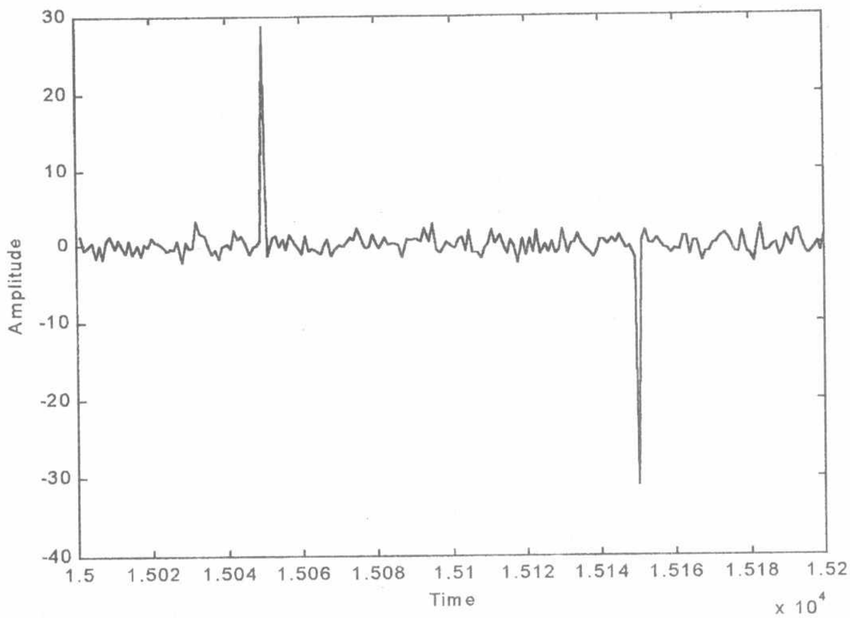

Fig. 6 The desired signal corrupted with Gaussian and Non-Gaussian noise x_k (SNRI=-8.21)

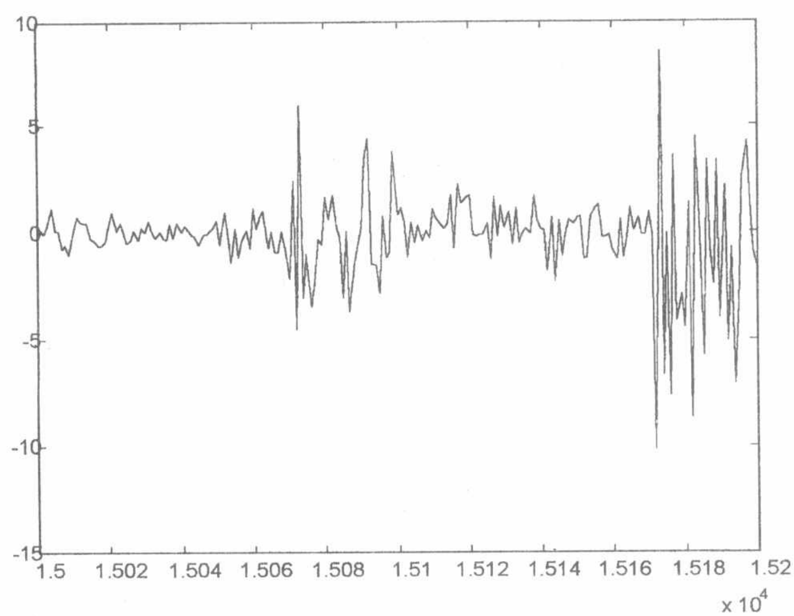

Fig. 7 The output signal y_k of stand alone ALE (N=30, μ=0 001412, SNR=1.24 dB, IMP=9.45)

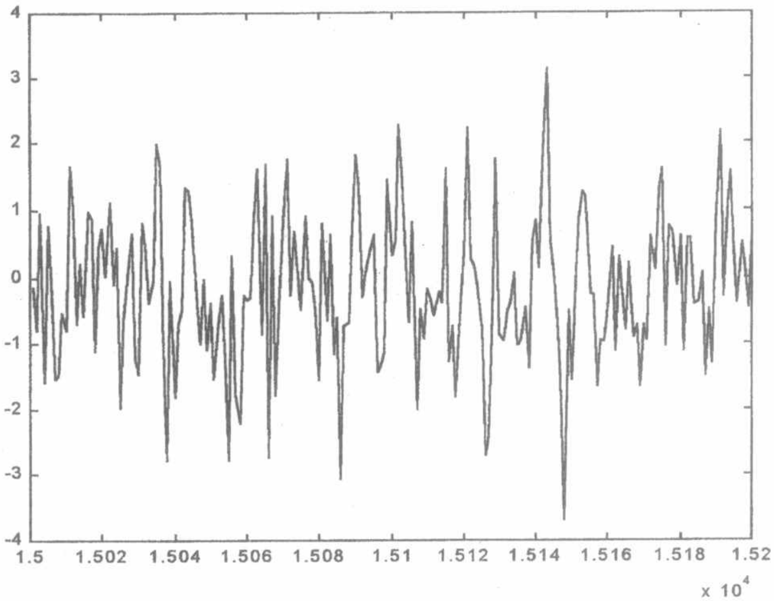

Fig. 8 The combined output of the median filter and the decision circuit (M=10)

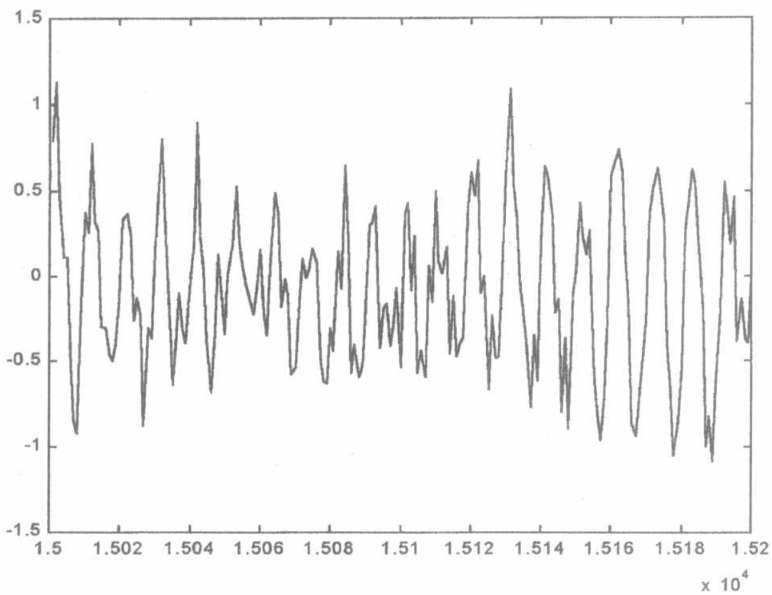

Fig. 9. The final output of the proposed scheme (N=30, μ=0 001412, SNRO=15.7 dB, IMP=23.91)