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ADAPTIVE FILTER ALGORITHMS FOR PINK NOISE CANCELLATION IN MUSICAL INSTRUMENTS SIGNAL

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ABSTRACT:
Almost all the signals are get corrupted by the noise signals when they are propagated via transmission medium , whether the medium is guided or non guided. As the information signals are interfered by the noise , the noise may be any type White, pink or impulse will distorted the desired signal and become difficult to detect or decode the signals. So it is very important to recover the desired signals by removing or minimizing the added noise. As we know that the characteristics of signals varies very fast and handling of the varying signals characteristics is uphill task. Most of the times for handling the noise FIR filter are used. But the problems with the FIR filter is that their convergence rate is very low for such high varying characteristics noise signals. Hence we have go for other method which can handle noise in better way oblivious choice is adaptive filters. Least Mean Square (LMS) and Averaging Algorithm (AA) can be used for noise cancellation. In the present work we are implementing LMS and AA algorithm for noise cancellation in the signals of the musical instruments. For that we have done experiments with pink noise. The implementation is done on Matlab software.

Keywords: Adaptive filter, Noise cancellation, Adaptive Averaging algorithm(AA), Least mean square (LMS), Pink noise .

INTRODUCTION

Cancellation of pink noise from the signals of musical instruments is the objective of this work. For that adaptive averaging algorithm and Least mean square algorithm are used. Many work have been intended for handling the guassian noise. As we know that simple FIR filters are not having convergence rate high so that they can handle fast varying characteristics of noise signals as well as of the desired signal. For obtaining the optimized results we have opted the adaptive filters, as we know that in adaptive filter we can vary the weight of the parameters using different adaptive algorithm to get the desired result. Every algorithm have some advantages and disadvantages. Some have high convergence rate but adds more complexity. Some have less complexity but convergence rate is low. Some of the filters have stability problems. So we have to select the filters according to the applications and as per requirement. We have to deal with the trade off between convergence rate, complexity, stability and others. Hence for noise cancellation purpose adaptive filters plays significant role [1]. Here in the present work we are dealing the pink noise with two algorithms Least Mean Square (LMS) and Averaging Algorithm (AA) . In the present work we are trying to get the desired output by maintaining robustness, fast convergence and less computational complexity. Earlier for noise cancellation from speech signlal is having priority but noise cancellation from musical instruments signals is of equal importance. Dewasthale, Mugdhaa and Kharadkar, R.D. work work on Least Mean Square (LMS) and Normalized-Least Mean Square (NLMS) algorithms, which are very popular and frequently used algorithms for noise cancellation in speech[2]. Sa'adah, Mamba'us; Wulandari, Diah Puspito; Suprapto, Yoyon Kusnendar work on Gamelan instrument. Gamelan instrument is Indonesian traditional music instrument. They conducted a study for noise removal on gamelan instruments using least-mean-square (LMS). They mixed the white noise in the original signal and the result show enlarging the rate of convergence [3].

ADAPTIVE NOISE CANCELLATION

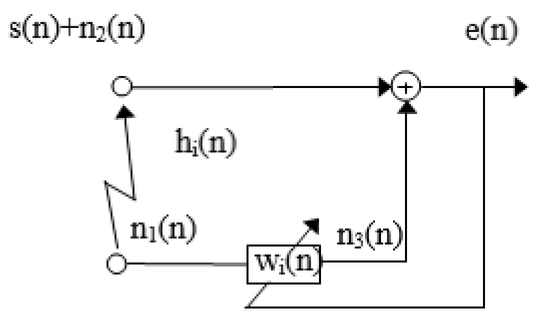
In the Fig. 1 FIR digital filter is shown in which it is used for adaptive noise cancellation. In the figure the source input is information signal s(n) and noise $n_2(n)$. In that reference signal is noise $n_1(n)$. The noises $n_1(n)$ and $n_2(n)$ are correlated and noise path impulse response is $h_i(n)$. The impact of the noise in the input signal is reduce by the system by correlating the two noise signals. It is same as to minimize the mean square error which is denoted by $E[e^2(n)]$ where,

$$e(n) = s(n) + n_2(n) - n_3(n)$$

by assuming that s(n) is not correlated with $n_2(n)$ and $n_1(n)$. Then we have,

$$E[e^2(n)] = E[s^2(n)] + E[n_2(n) - n_3(n)]^2$$

We can say that when the difference between $n_2(n)$ and $n_3(n)$ is minimize, $E[e^2(n)]$ is also minimize. $E[e^2(n)]$ will be minimum when $n_3(n) \approx n_2(n)$, it means that impulse response of the noise path and impulse response of the adaptive filter is almost same. This minimization of $E[e_2(n)]$ will be obtained by altering the taps of filter $w_i(n)$. NLMS and RLS algorithms are mostly used.



MUSICAL INSTRUMENTS:

For the generating the signals, musical instrument are having strings, air cavity or key boards. So signal obtained depends mainly on the kind of musical instruments. The musical instruments are categorized as,

String Musical- Instrument whose tones are produced by vibrating chords made up of material like copper or plastic. Every vibrating chords has its fundamental frequency producing complex tones. Brass Musical instrument- whose tones are depends on blowing air like woodwind. Woodwind musical instrument- consists of an open cylindrical tube at both ends. Percussion musical instrument example piano, most of the tones produces non harmonic tones. In the present work we are using instruments flute, guiter and table.

NOISE:

Noise is present everywhere and it is unwanted signals which interferes with the information signals and creates problems in the recovery of desired signals at the receiver end. The type of noise depends on the source. Source may internal or external. In the communication system we have to deal mostly Guassian white noise, Pink noise and impulse noise .

Gaussian noise is present in all the frequency spectrum. It is invented by Carl Friedrich Gauss. Its probability density function (PDF) follows the normal distribution or Gaussian distribution hence name Guassian Noise. As it has continuous distribution so also called white noise.

Pink Noise having power spectral density 1/f. Since power decrease as frequency increase sound of pink noise is softer than white noise.

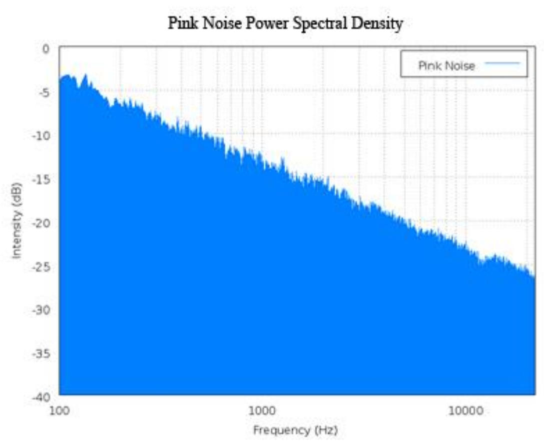


Fig . 2 Pink Noise

In pink noise each octave has equal amount of noise energy. The name is pink because of appearance of visible is having pink colour.

Impulse noise is a instantaneous noise like a spike having sharp sound like clicks and pop.

ADAPTIVE FILTERS:

For LMS algorithm;

Noise estimation:

By assuming autocorrelation matrix ΓM and cross correlation vector Yd are known, then coefficients can be computed iteratively as,

$hM(n+1) = hM(n)+1/2\Delta(n)s(n)$, n=0,1,.......

to obtain the minimum of J(h). Where,

hM(n) – vector of coefficient at the nth iteration.

$\Delta(n)$ – step size at the nth iteration.

s(n) – direction for the nth iteration.

hM(0) is chosen arbitrarily.

For minimization of J(hM) with ΓM and Yd are known uses gradient vectors. Steepest Descent search method are used. In that the direction vector is S(n)= - g(n) where g(n) is the gradient vector at the nth iteration. The recursive algorithm is

$$hM(n+1) = hM(n)-1/2\Delta(n)g(n)$$

and by substituting for g(n) we get,

$$hM(n+1) = (I- \Delta(n)\, \Gamma M)\, hM(n)+\Delta(n)Yd$$

hM(n) converges to hopt as $n\rightarrow \infty$, the sequences of step size $\Delta(n)$ should be absolutely summable with $\Delta n \rightarrow 0$ as $n\rightarrow \infty$, also as $n\rightarrow \infty$, $g(n) \rightarrow 0$.

For Averaging algorithm(AA)

Noise estimation:

$$n3(n) = \sum_{i=0}^{N} wi\,(n)\, n1\,(n- i)$$

N – filter order

Error estimation:

$$e(n) = s(n) + n_2(n) – n_3(n)$$

Coefficients update:

$$\bar{W}_i(n)= 1/n \sum_{k=1}^{n} Wi(k)$$

$$\overline{n1e_1(n)} = 1\backslash n \sum_{k=1}^{n} n_1\,(k-i)\, e(k)$$

$$W_i(n+1) = \bar{W}_i(n)+ 1\backslash n^{\gamma}\, \overline{n1\, e_1(n)}$$

For $0\le i\le N$

And

1/2<□<1

EXPERIMENT SET UP:

To obtain result and waveforms for LMS and Averaging filter, they are implemented on the MATLAB software with Filter order = 32. For LMS algorithm step size taken is 0.008. In Averaging algorithm for current_value = (Average of previous 16 , current and next 16 sample) is evaluated. The instruments used are Flute, Guitar and Tabla. In first step the signals from these instruments are taken then they are mixed with the pink noise and afterwards noise is minimized from the mixed signals. The waveforms of instruments signals, noise signals, mixed signals and recovered signals are shown in the result. Also PSNR is measured for both the filters and for all the instruments.

RESULT:

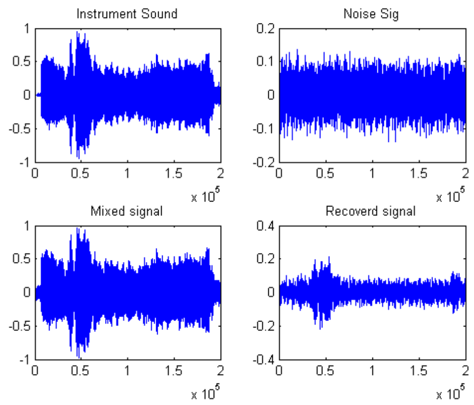


Fig. 3. Flute_Pink_Avg_11.49

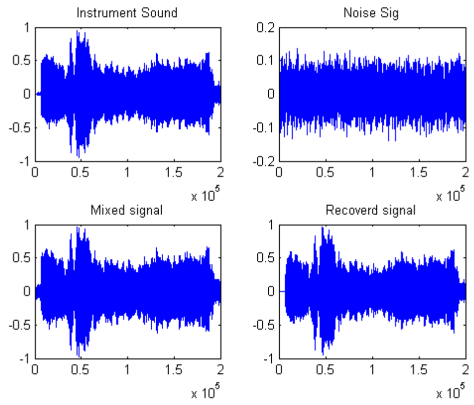


Fig. 4 Flute_Pink_LMS_28.53

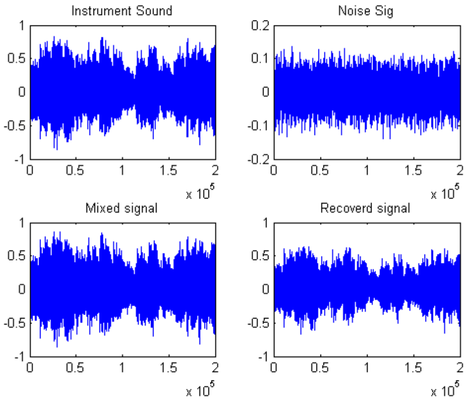


Fig. 5 Guitar_Pink_avg_20.89

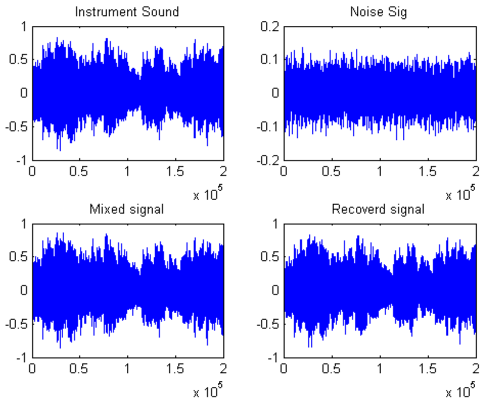


Fig. 6 Guitar_Pink_LMS_27.77

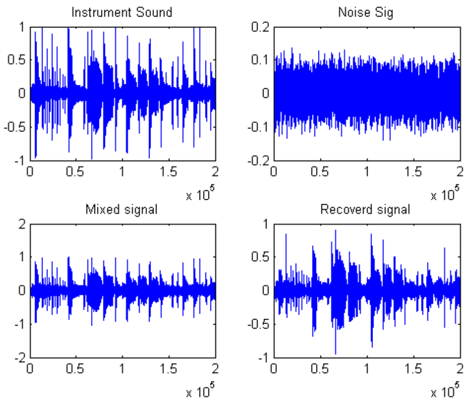


Fig. 7 Tabla_pink_avg_23.42

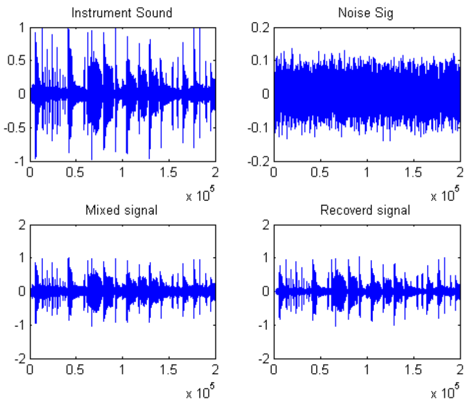


Fig. 8 Tabla_pink_LMS_29.01

Instrument	Noise	Filter	PSNR
Bansuri	Pink	Avg	11.49
		LMS	28.53
Guitar	Pink	Avg	20.89
		LMS	27.77
Tabla	Pink	Avg	23.42
		LMS	29.1

Table 1

CONCLUSIONS:

The pink noise in the musical instruments signals are reduced through the LMS and Averaging algorithm . The waveforms are shown in the result PSNR value is more in LMS as compared to averaging algorithm..

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