Noise Cancellation in Gamelan Signal by Using Least Mean Square Based Adaptive Filter

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***Abstract*— Gamelan is one of Indonesian traditional music instrument that has been worldwide. Noise reduction of identical instrument is a key challenge for instrument recognition, music processing and instrument analysis. Many theoretical analysis and experiments have been carried out to show that the optimal filtering technique can reduce the level of noise that is present in the instrument signal. In this paper, we conducted a study for noise removal on gamelan instruments using least-mean-square (LMS). Using the original signal mixed with noise, the result that enlarging the rate of convergence, filter order, and iteration can improve the LMS function in noise removal in the instrument gamelan. The performance of the designed adaptive filter is evaluated based on the mean square error by varying the additive white Gaussian noise levels. We found that the performance of Least Mean Square is satisfactory and is viable to be applied in gamelan signal.**

Keywords-gamelan; noise cancellation; adaptive filter

I. INTRODUCTION

Since we live in a natural environment where noise is inevitable and ubiquitous, speech signals can seldom be recorded in pure form and are generally contaminated by acoustic background noise. As a result, the speech signals have to be cleaned up with digital signal processing tools before they are stored, transmitted, or played out [1]. Noise reduction algorithms and systems for speech enhancement have received considerable interest in the past, primarily because the reduced speech intelligibility under noisy conditions is one of the major complaints in hearing impaired subjects. Recent years, noise reduction has been in great demand for an increasing number of audio applications, such as automatic speech recognition systems and cellular telephone [2].

Denoising can be achieved in many differet ways, such as adaptive filtering, temporal filtering, spatial temporal filtering, etc. Generally speaking, all of these algorithms can be classified into two categories: single-channel technique and multi-channel technique according to the number of sensors they needed. Compared to the single-channel technique, the multi-channel technique is substantially superior in reducing noise and enhancing speech, due to its spatial filtering capability of suppressing the interfering signals arriving from arbitrary direction other than the specified direction [3]. A multi-channel pose-filter is first presented by Zelinski with assumption of zero cross-correlation between noise and signals on different microphones. The linearly constrained adaptive beamformer, first presented by Frost, keeps the signals arriving from the desired look-direction distortionless while suppressing the signals from other directions by minimizing the output power of the beamformer. A small-scale subtractive beamformer based noise reduction algorithm has been proposed in [4]. Its superiority lies in the fact that no adaptive signal processing is adopted and high performance in reducing sudden noise. And its weakness lies in the assumption that only localized noise exist in the environment.

Gamelan which is a traditional music instrument that is recognized as a UNESCO cultural heritage should be preserved continuously. Not only by playing it but also need to do research on gamelan. Research on gamelan mostly done from the art or musicality while gamelan research in terms of technology is still rarely done.

Instrument signal is known as a non-stationery signal, and fixed digital filter cannot be successfully applied in signal processing because the complete range of input condition may not be known exactly and during the normal operation of the filter has been change the design criteria. Thus, we use adaptive filter, it can modify their response to improve their performance during operation periodically without any invention from the user. Adaptive filter operates satisfactorily in an unknown environment and can track time variations of input statistics. Recently, adaptive filtering has become effective and popular methods for processing and analysis of the ECG signal [5]. It is well known that adaptive filters with least mean square (LMS) algorithm show good

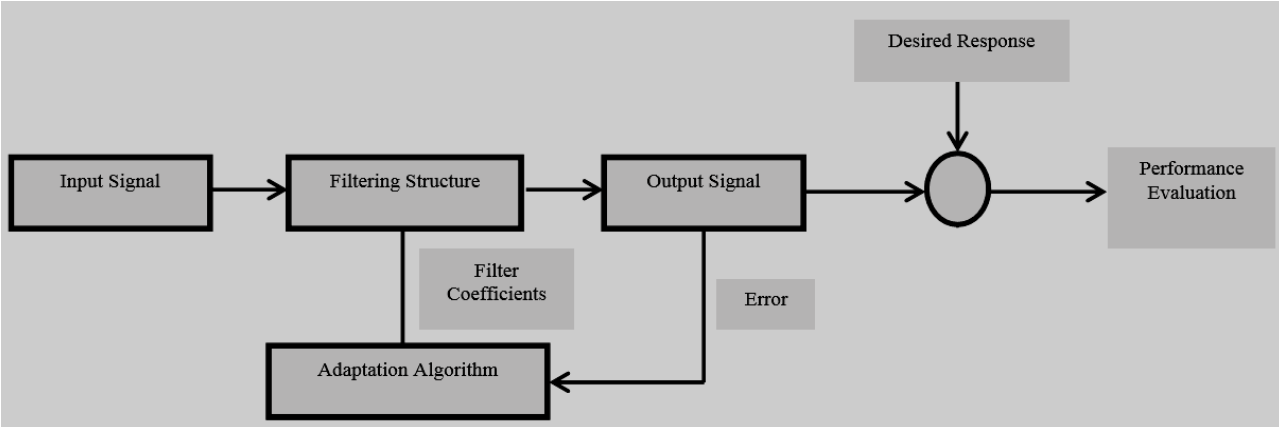

Figure 1. Design system of adaptive filter.

performance for processing and analysis of signal which are non stationary [6].

When a gamelan ensemble is played, the fastest tempo is 300 msecond, but some of them, two or more sarons, are played simultaneously within the same notations. Separating two or more signals is a very important to find signal features [7], but separating two or more identical music instruments is an extraordinary thing. The time delay among sarons sounds are very short, it is about less than 10 msecond so we enhance their methods to identify how many sarons are played. Basically two identical instruments have two identical frequencies too [7].

In gamelan ensemble, we have some grammatical features and signal features. Grammatical features are customs how to play music like tempo, notations, and signal feature is signal characteristic like timbre, amplitude. During the music is played, both features are very often changed by a conductor. The tempo changes faster after the music plays several notations, and back to slower. In additions, how to play an instrument is varied by another method likewise a music notation is played double, etc [7]. Denoising to identify some identical instruments played simultaneously using adaptive filter is very important to be applied in a gamelan ensemble to for instrument analysis.

II. FUNDAMENTAL THEORY

A. Gamelan

Gamelan is one of the traditional music instruments, it has monophonic characteristic and consist of 15 different percussion groups, i.e. saron, bonang, kempul, gong, celempung, gender, gambang and etc., [7].

A gamelan consists of several instruments that played together like an orchestra. Unlike an orchestra, in the gamelan there is no person who acts as a conductor. Instead, one of gamelan instruments is played to direct the other instruments of the gamelan [8].

The Balungan group is constructed by three instruments namely Demung, Saron and Peking [9]. Each Balungan instrument has one octave, so yhe other octave is played by another instrument. Demung has the lowest octave, the range frequency is about 200-500 Hz, and Peking has the highest octave, it is about 1000-2000 Hz[10].

Gamelan music instrument has its own search in play, like drum how to use it by being hit by hand directly without using tools, then musical instrument in other gamelan like peking, saron and demung played by beaten with tool made of wood.

B. Noise

Noise is the best characterized on the basis of its time and frequency dynamics. Spectrally white noise is per definition noise with a noise spectral density that is independent of frequency. In most systems noise has two important general properties: it has a Gaussian probability distribution and its statistical properties are stationary [8].

C. Adaptive Filter

The Adaptive Filter is a computing device that model the relationship between two signals by way of real time adventure.

Adaptive filters are usually associated with the broader topic of statistical signal processing. The operation of signal filtering by definition implies extracting something desired from a signal containing both desired and undesired components.

III. DESIGN SYSTEM

Noise cancellation in gamelan signal system is shown in Fig 1.

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A. *Signal Input*

Signal input is the original saron signal. The different types of noise signal are generated by using MATLAB®. The noise signal is then added with real saron signal.

The noise reduction problem considered in this paper is to recover a saron signal of interest (SOI) *x(n)* from the observation signal *y(n)* which is corrupted by the noise *v(n)*.

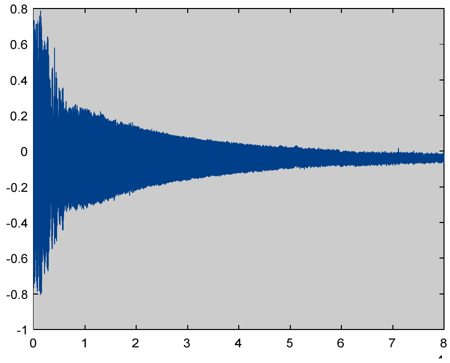

Figure 2. Saron signal

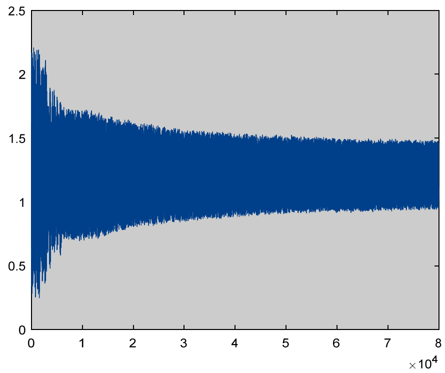

Figure 3. Saron signal+noise

B. *Filtering Structure*

This module forms the output of the filter using measurements of the input signal. The filtering structure can be linear or non-linear which is fixed by the designer, and its parameters are adjusted by the adaptive algorithm.

However, the basic block diagram for understanding the overall adaptive filtering process is depicted in Fig. 3

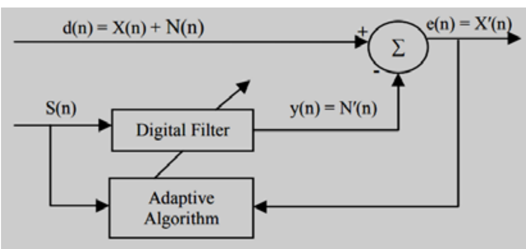

Figure 4. Principle of adaptive filter

The block diagram indicates that, if the value of N(n) is known, then after subtracting this from the mixed signal d(n), the original signal X(n) is obtained. But it is difficult due to the harmonics of noise signal. For this reason an estimated noise signal N′(n) is calculated through some filters and measureable noise source S(n). If N′(n) is more close to N(n), then the estimated desired signal is X′(n) more close to the original signal X(n). Mathematically the output is given by

$$e = X + N - y \tag{1}$$

The power or energy of this signal is computed by squaring it

$$e^2 = X^2 + (N - y)^2 + 2X(N - y) \tag{2}$$

Taking expectations of both sides results

$$E(e)^2 = E(X)^2 + E(N - y)^2 + 2EX(N - y) \tag{3}$$

$$E(e)^2 = E(X)^2 + E(N - y)^2 \tag{4}$$

Adapting the filter to minimize the error energy will not affect the signal energy. Therefore the minimum error energy is

$$E(e)^2{}_{min} = E(X)^2 + E(N - y)^2 \tag{5}$$

$E(e - X)^2$ is also minimized since, (e – X) = (n - y). Therefore, minimizing the total output energy is the same as minimizing the noise energy. The LMS algorithm produces the least mean square of the error signal by changing the filter tap weight, whose coefficient updating equation is

$$W_{k=1} = W_k + 2\mu e_k X_k \tag{6}$$

Where, μ is an appropriate step size to be chosen as $0 < \mu < 0.2$ for the convergence.

C. *Adaptive Filter*

The adaptive algorithm uses the value of the criterion of performance, the measurements of the input and desired signals so as to modify the parameters of the filter to improve its performance. Adaptive digital filters are currently widely used for signal processing applications. This digital filter is widely used to obtain the desired signal spectral characteristics, eliminating unwanted signals (such as noise or interference signals). The adaptive digital filter structure is shown in Figure 1. The adaptive definition refers to the ability of the system to change the weights or filter

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coefficients to adapt with the immediate environmental conditions.

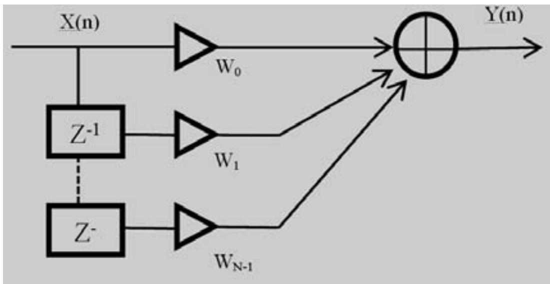

Figure 5. Digital filter structure

IV. RESULT AND DISCUSSION

Experiment performed several identical instrument. Data used to perform experiments is audio recording of saron instrument which played on the same time and the same notation.

In this section, we apply a LMS adaptive algorithm to noise reduction in saron signal to achieve a good performance the speech signals must be segmented.

The signal used is a saron signal with sampling number is 10000 and an amplitude of 1 mV. White noise is generated using MATLAB®. Then, this noise is added to the original gamelan signal to get the desired mixed signal. Thus, noise can be removed by using adaptive filters based on the LMS algorithm.

The audio signal noise reduction is shown in Fig. The noise canceled signal has a time domain waveform that is nearly equal to original. In this figure we have plotted the dissimilarity between the signals waveform of the clean audio signal and the noisy signal and then the noise canceled signal. It is noted that since the iteration of the adaptive filter is 1000.

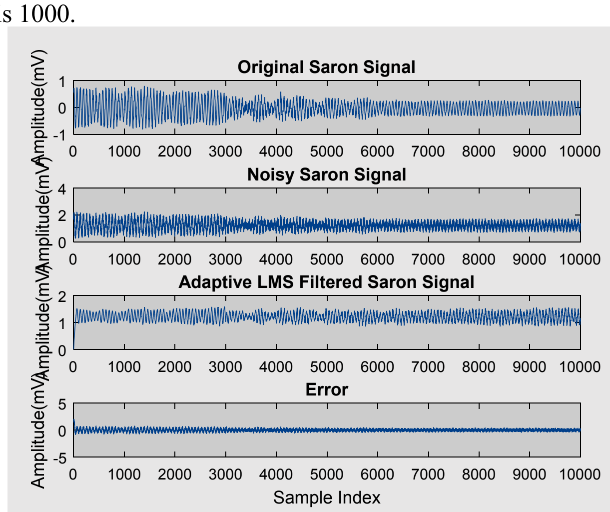

Figure 6. Graphical representation of LMS filtering signal for μ=0.0005 after removing White Gaussian Noise

From Figure shows that if the amplitude of the reconstructed signal increases, there will be a high distortion and vice versa.

To visually observe the denoising performance of adaptive LMS filter we use three parameters MSE.

According to the Table, it can be seen that the larger the larger the weight filter the smaller the MSE. And of course, the MSE on the signal that there is noise will be bigger.

To understand the performance of removing noise from saron signal, we represent signal fidelity measure. The MSE of mixed signal is shown graphically in Fig 7. and tabular form in Table I.

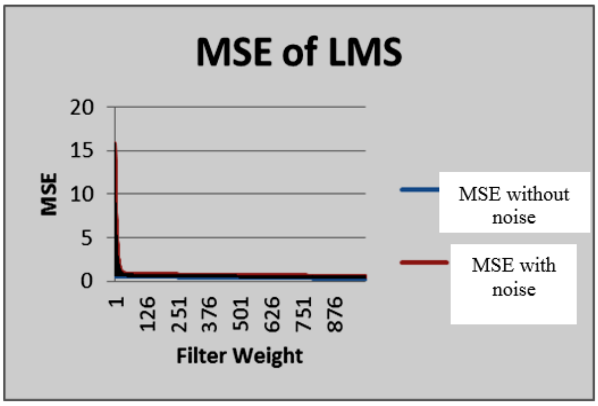

Figure 7. Graphical representation of LMS filtering signal for μ=0.0005 after removing White Gaussian Noise

The Mean Squared Error (MSE) was used to measure the performance.

TABLE I. COMPARISON OF MSE

Filter Weight	*MSE with noise*	*MSE without noise*
1000	0.44602427	0.81620051
2000	0.363645587	0.263178127
3000	0.127256284	0.20667388
4000	0.048984785	0.131716921

Obtained signal energy and noise energy values are still very volatile at the time of increasing the value of filter length, this proves that the increase in filter length does not affect the energy value of the signal either noise or noise signal.

V. CONCLUSION AND FURTHER WORKS

In this paper, we have presented a noise reduction method for gamelan signals by applying adaptive linear filtering technique. The noise reduction problem has been formulated as a filtering problem which is efficiently solved by using the LMS method. Simulation results indicate that the proposed method can improve the performance the quality of noisy gamelan signal. Through computer simulations, we have demonstrated that the proposed method is quite effective in noise reduction, especially in the case of stationary white Gaussian noise.

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REFERENCES

[1] J. Chen, J. Benesty, and Y. (Arden) Huang, "On the optimal linear filtering techniques for noise reduction," *Speech Commun.*, vol. 49, no. 4, pp. 305–316, 2007.

[2] Y. Liu, Y. Liu, M. Xiao, and Y. Tie, "A Noise Reduction Method Based on LMS Adaptive Filter of Audio Signals," pp. 1001–1008, 2013.

[3] B. J and S. K. U, "Super directive microphone arrays. In: Microphone arrays signal processing techniques and applications," *Springer, Berlin, 19-38*, 2001.

[4] M. Akagi and T. Kago, "Noise reduction using a small-scale microphone array in multi noise source environment," in *ICASSP*, 2002, pp. 909–912.

[5] R. A. Shaik, "Baseline wander and Power line interference elimination from Cardiac Signals using Error Nonlinearity LMS Algorithm," no. December, pp. 217–220, 2010.

[6] H. T. Nagle, "A Comparison of the Noise Sensitivity of Nine QRS Detection Algorithms," vol. 37, no. January, 1990.

[7] T. Kitahara, M. Goto, K. Komatani, T. Ogata, and H. G. Okuno, "Instrument Identification in Polyphonic Music : Feature Weighting to Minimize Influence of Sound Overlaps," vol. 2007, 2007.

[8] A. I. Technology, "HIGH PERFORMANCE GAMELAN ANALYZER USING," vol. 71, no. 1, 2015.

[9] S. The and F. Music, "VWB - Verlag für Wissenschaft und Bildung Bärenreiter Schott Music GmbH & Co . KG Florian Noetzel GmbH Verlag Review by : Rüdiger Schumacher," 2016.

[10] C. Description, "332 : 525 – Optimum Signal Processing – Spring 2011," 2011.