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Malang, November 27, 2014

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Sekolah Tinggi Informatika dan Komputer Indonesia

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GREETINGS

Head of Committee IC-Itechs

For all delegation participants and invited guest, welcome to International Conference on Information Technology and Security (IC-Itechs) 2014 in Malang, Indonesia.

This conference is part of the framework of ICT development and security system that became one of the activities in STIKI and STTAR. this forum resulted in some references on the application of ICT. This activity is related to the movement of ICT development for Indonesia.

IC-Itechs aims to be a forum for communication between researchers, activists, system developers, industrial players and all communications ICT Indonesia and abroad.

The forum is expected to continue to be held continuously and periodically, so we hope this conference give real contribution and direct impact for ICT development.

Finally, we would like to say thanks for all participant and event organizer who involved in the held of the IC-Itechs 2014. We hope all participant and keynote speakers got benefit from this conference.

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LIST OF CONTENT

Active Noise Cancellation for Underwater Environment Using Raspberry Pi

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Abstract

Noise can interfere activities both on land and in the underwater environment. In this paper we propose a noise cancellation system using ANC (Active Noise Control) techniques to reduce acoustic noise signal using Raspberry Pi, to receive an amplitude and phase of the noise signal, calculate each sampled signal using normalised least means square (NLMS), and finally it will produce a replica of the transmitled noise by shifting the phase noise at 180°, through the speaker in real-time environment so that the signal noise can be reduced and decrease. The system is designed using Matlab and Simulink, the model is compiled for Raspberry Pi board as the hardware target. Raspberry Pi is a small computer so it make a system more efficient. The experimentation results shows that ANC can work in an optimal mode in delay=5 and filter length=64 and can reduce noise level until 0,894 dB in underwater environment and maximal reduction level in 2 Khz frequency. The system can influence in around the system and reduce the signal level about 1,97 dB in 10 cm depth, and 1,02 dB in 40 cm depth.

***Keyword** : Active Noise Control, Noise Cancellation, ANC, NLMS, Raspberry Pi*

1. INTRODUCTION

1.1. Active Noise Control

The activity in the aquatic environment cause noise or commonly referred to as noise, which can affect the ecosystems that exist under the sea. Acoustic noise generated by ships coming from one part of the ship that is the propeller that is easily recognizable because it is the largest producer of noise in a ship. The existence of propeller noise can be used to detect the presence of the ship [1]. Noise has always cause a variety of problems, so it takes a noise control techniques could be overcome by making the tool to remove noise in underwater environment. Active Noise Control (ANC) is a technology that is useful for removing unwanted noise sound. The principle works is to generate a magnitude sufficient frequency waves to neutralize the noise frequency wave signal from the sound source. The basic concept of active noise control (ANC) is a process of elimination or removal of the primary noise (unwanted noise) with anti-noise that makes noise residue and will be close to zero [2]. ANC effectiveness is highly dependent on the accuracy of the amplitude and phase of the anti noise. ANC basic concept is illustrated in figure 1 as follows:

(a) (b)

Figure 1. Concept of Noise Elimination, (a) Noise Wave and (b) Anti Noise Wave

In figure 1 above show that the wave of the primary noise and anti-noise have mutually opposite phase. The successful use of ANC based on the effectiveness of noise control compared to that done by passive noise control. Damping ANC gives meaning elimination of a number of noise in large numbers by using a device with small dimensions, especially in the low-frequency noise. Active noise control can result in very high noise reduction when the frequency is low frequency processing [2]. At low frequencies where only the best frequencies are allowed, the ANC provide great advantages to using passive techniques. The ANC system has two methods as follows [2]:

a. Feed Forward Control

Feed Forward Control is when the reference input signal is detected before busting noise propagated to the speaker. Illustration of Feed Forward Control method illustrated in figure 2 below.

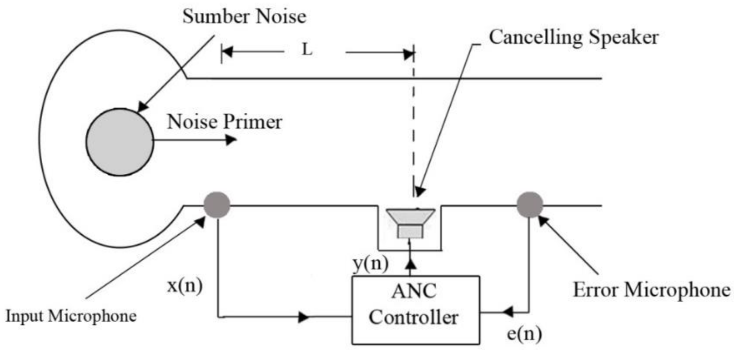


Figure 2. Feed Forward Control

b. Feedback Control

Feedback Control ie when processing noise remove noise without taking into account the excess of the reference noise [2]. Illustration of Feedback Control method illustrated by figure 3 below.

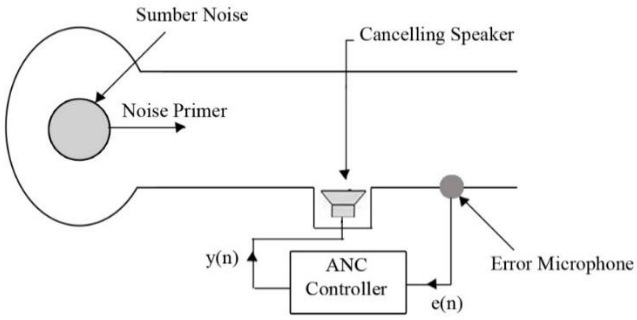


Figure 3. Feedback Control

In the ANC feedback control applications, most recently as the primary signal *e(n)* is not visible during the operation of the ANC since been reduced by the secondary noise. Thus, the basic principles of adaptive feedback ANC (AFANC) which provides estimates of the primary noise, and use it as a reference signal as the signal *x(n)* (feed forward control) in figure 3 for ANC filter feedback control. AFANC will make the reference signal itself is not as adaptive feed-forward which uses a microphone to pick up the reference signal. Feedback ANC system algorithms can be formulated as a predictor of adaptive as shown in figure 4. This system uses only one error microphone to resolve the problem of acoustic feedback. The core of this algorithm is to estimate the primary noise *d(n)* and use it as a reference input to the adaptive filter [3].

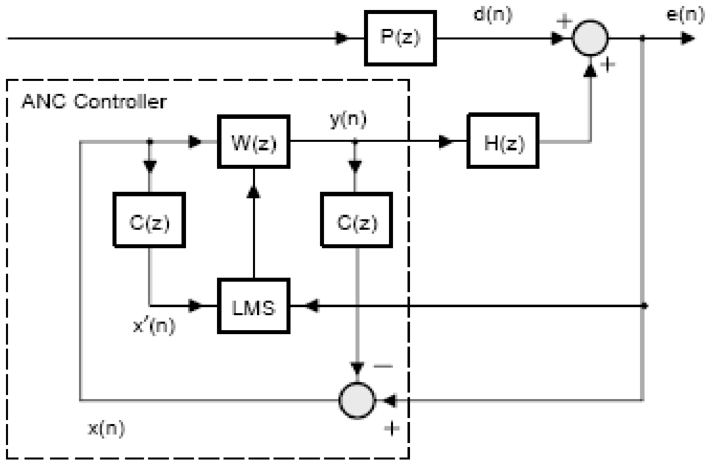


Figure 4. Diagram Block of Feedback ANC

From figure 4, we obtained:

$$D(z) = E(z) - H(z)\,Y(z) \tag{1}$$

Where *E(z)* and *Y(z)* exists if the transfer function *H(z)* of the secondary path is modelled by *C(z):*

$$D(z) \approx X(z) = E(z) - C(z)\,Y(z) \tag{2}$$

The error signal is indicated by:

$$E(z) = D(z) - W(z)\,H(z)\,X(z) \tag{3}$$

The error signal of the feedback ANC system will be 0 when:

$$W(z)\,H(z)\,X(z) = D(z) \tag{4}$$

This is possible if the primary noise *D(z)* and the periodic transfer function *W(z) H(z)* is equal to the equivalent delay for some period of the signal.

1.2. NLMS Adaptive Filter algorithm

Adaptive filter is a filter that has the characteristics can be adapted to a given input and generates the necessary filter coefficients to obtain the desired output [4]. Adaptive filter has two distinct parts, namely a digital filter with coefficients that can change and adaptive algorithms that serve to change the coefficients of the digital filter [5]. Block diagram of adaptive filters as noise reducer shown in figure 5. The purpose of the system is to obtain the most optimal an estimates signal (noise) *ne[k]*.

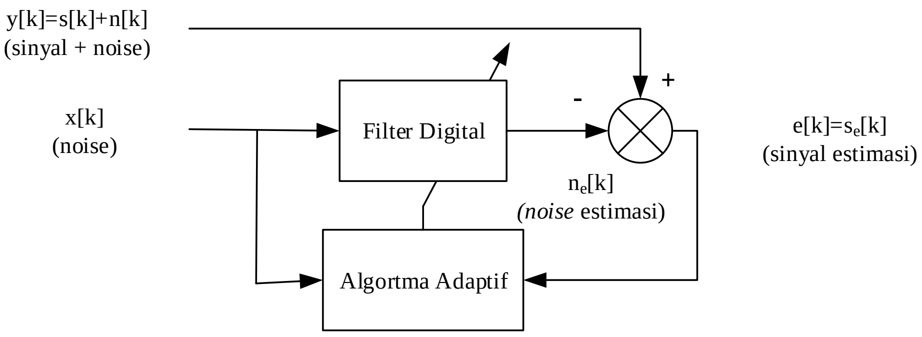


Figure 5. Adaptive Filter Diagram Blok

Figure 5 is one of the algorithms that can be implemented by an adaptive filter that is NLMS (Normalized Least Mean Square) which is the development of algorithms LMS (Least Mean Square). LMS algorithm is widely used for the application of adaptive filters as effective in terms of computational and storage. NLMS algorithm is very simple to implement and has an efficiency of the computing process. A previous study [5], states that this algorithm has an average reduction of -27.9 dB with variable step size that is stable for arbitrary signals. That makes this algorithm suitable for implementation in real-time systems. NLMS algorithm has been found in the block diagram in Simulink Signal Processing so no need to program the coding to simplify and shorten the implementation of the model.

2. RESEARCH METHOD

At this chapter, we will be explained the process of designing the system from making sound input through a microphone and then processed by the processing of acoustic signals to be retransmitted through the active speaker against the signal to be removed. At this stage of the development of every component the ANC diagram that has been planned in sequence.

a. Determination of the noise source

Sources of noise that would be muted are recorded propeller noise from underwater footage that has been determined by the adaptive filter. Recording voices are underwater noise, which is the result of recording in PCM with a sampling frequency of 44100 Hz. It is appropriate that the Nyquist criterion sampling frequency value must be worth twice the maximum frequency of the signal information that is for the audio frequency range between 20 Hz - 20 kHz. So, we need a minimum of 40000 Hz sampling frequency so that no aliasing frequency. In the figure 6 below, signal voltage value will be multiplied by a constant multiplication by -1 so product block of elements is the inverse of the signal.

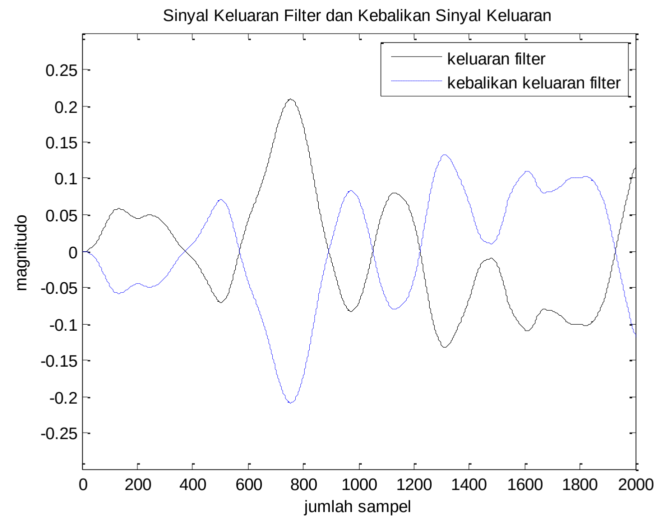


Figure 6. Example of Reverse Signal of Filter Output

b. Software Development

Software is developed according of the block diagram of feedback ANC system by using the NLMS adaptive filter method. It designed in Matlab as depicted in the figure 7, then it implemented on the Raspberry Pi.

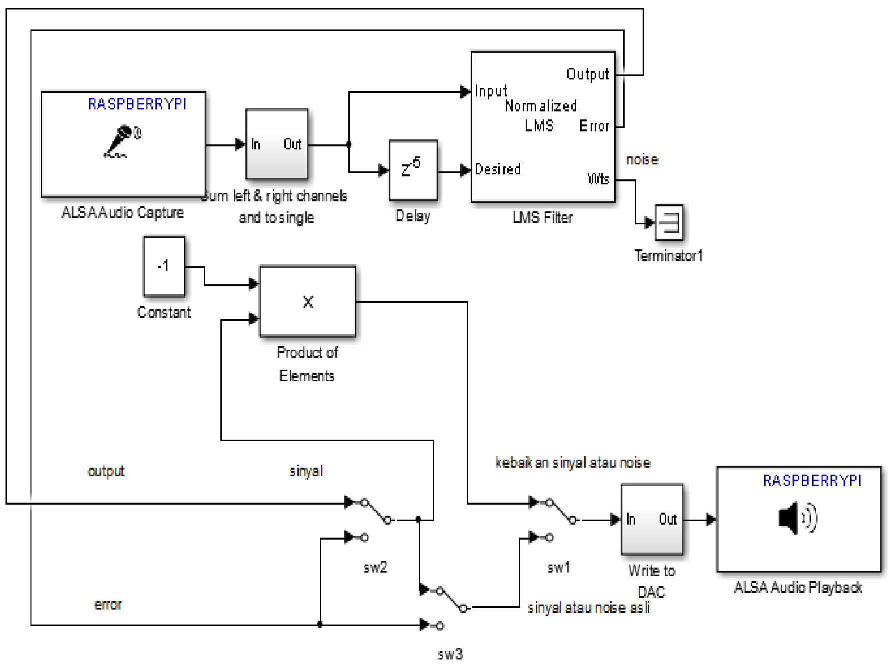


Figure 7. Noise Silencer System

c. Hardware Construction

At this stage it does of a prototype hardware construction according to the principles of the ANC and the feedback is in the barrier tube to avoid the interference of other sounds that would interfere with the performance of the system. Thus the system is assumed that there is only one noise to be reduced.

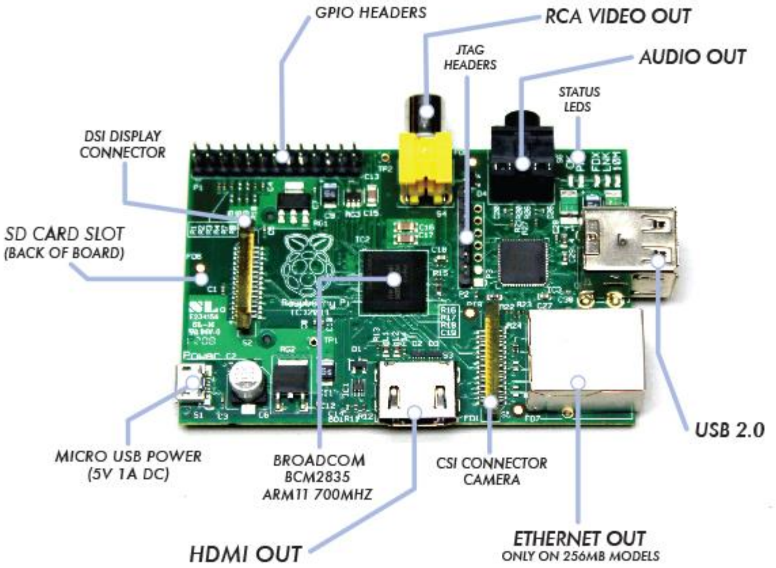
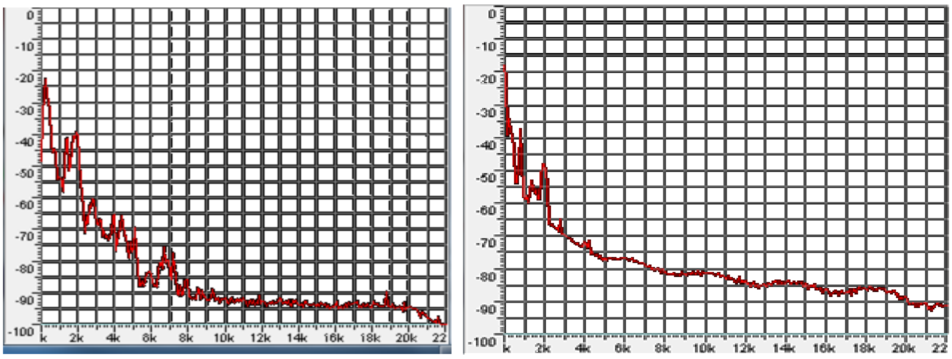


Figure 6. Raspberry Pi Overview

Raspberry Pi as depicted in the figure 6 above contains a program results from Matlab Simulink code generation, it has input from the microphone input and the output of the inverse noise signal issued by the anti-noise speakers.

3. RESULT AND DISCUSSION

In this chapter discussed about experiment and analytical results, based on the data of empirical result. In the figure 7 below, a comparison of the signal in the frequency domain in order to facilitate any frequency observations are successfully suppressed by the system. The frequency spectrum of the signal before muted shown by figure (a) and after a muted signal is shown by figure (b).



(a) (b)

Figure 7. Frequency Spectrum (a) Signal Before muted (b) Signals After muted

It can be seen that the signal component that wants to be processed are scattered throughout the audio frequency band. The signal has a frequency value at the maximum level of less than 1 kHz with a value of -24 dB, and then appeared frequency of 2 kHz with -40dB level value, then the frequency of 4 kHz worth -65dB and a low level value with the highest level of -90dB at a frequency above 8Khz. While in figure 7 (b) less than 1kHz frequency decreased with a maximum value of -35dB, frequency of 2 kHz down to the level of -49dB and -73dB worth 4kHz frequency but increased the level at frequencies above 8Khz with the highest level value of -80dB. From the experimental data it can be seen that the noise damping system works at low frequency band with the highest level of power reduction occurred at a frequency of less than 1kHz. But less effective at high frequencies above 8 kHz. This is due to the presence of other noise that affects during signal processing.

4. CONCLUSSION

From the measurement data under the water, it can be seen that the noise damping system works on a frequency of less than 2 kHz and capable of reducing signal at 1kHz-2kHz range of -53db be -85dB in the amount of 32dB. This is because the water pressure in the membrane that prevents vibration signal issued. From the test results and analysis of the system in the previous chapter, it can be notice that noise dampening system is capable to produce an anti-noise signal to the inverse of the estimated signal defined by the adaptive filter. The noise dampening system is able to work at a low frequency band with the highest level of power reduction at a frequency of less than 1kHz, but less effective at high frequencies above 8 kHz due to the influence of other noise in the system. The noise dampening system capable of working at a frequency of less than 2 kHz and capable of reducing signal at 1kHz-2kHz range of -53db be -85dB that is equal to 32dB.

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