Analysis Of The Effect Filter Order Number On Noise Canceller System Using STM32F4

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***Abstract*— Communication plays an important role in society. This can be seen in the amount of information that is spread, such as information in the form of sound, images, and videos. But communicating is not always easy. Due to the large number of sounds emitted from different sources, other sounds will be disturbed. The interference caused by noise can distort the information signal, leading a sine wave to integrate a minor noise signal. As a result, the receiver cannot differentiate the actual information signal and the added noise. This study proposed a noise-reduction system by analysing the effect of the number of filter orders on the noise canceler system. The information signal will be processed using the STM32F4 noise canceller system, which will then be filtered using an adaptive filter with a Finite Impulse Response (FIR) structure and the Least Mean Square (LMS) algorithm. The test results show that the best SNR value is obtained at Order 40 of 5.3671 dB at a sound duration of 14 seconds, while the best PSNR value is obtained at Order 40 of 21.3557 at a sound duration of 9 seconds, and the higher the filter order value, the smaller the MSE value.**

Keywords*— *Noise Canceller, Filter, Order Number.

I. INTRODUCTION

Communication is essential in daily life. This is shown by the wide variety of information delivery systems that spread information in the form of sound, images, and videos. Audio communication is a popular mode of communication nowadays. This communication can be defined as the process of delivering information from one person or numerous people so that they can connect with others. Therefore, audio communication is the process by which a person can convey information and engage with others and their environment using auditory symbols (audio) in both spoken and non-verbal forms [1].

However, communication is not always easy. The amount of sound emitted from different sources makes other sounds disturbed, such as the received sound being mixed with other sounds so that it cannot be received. This is referred to as noise, an acoustic (sound), electrical, or electronic disturbance signal that is present in a system as interference that is not the desired signal [2]. Noise can disrupt the information signal. This causes a small noise signal in the sine wave. As a result, the receiver is unable to differentiate between the true information signal and the additional noise [3].

An adaptive filter is one method for reducing noise. An adaptive filter is a filter that alters the coefficients to adapt to changes in signal statistics to optimize the signal from distortion. The Least Mean Square (LMS) algorithm, which may be applied to the Finite Impulse Response (FIR) adaptive filter, is one of the most commonly used adaptive filter algorithms [4]. One of the techniques that can be used to change a digital filter's coefficients is the Least Mean Square (LMS) approach. The coefficient of the digital filter is modified using the Least Mean Square (LMS) technique.

This study proposes a system for noise reduction based on an adaptive digital filter and the STM32F4 microcontroller. STM32F4 is utilized as an implementation of a device that performs digital signal processing by examining the influence of the number of orders and filter repetition operations on the noise canceler system. Following the filtering process, the outcomes of the adaptive filter application on the STM32F4 device may be viewed through the computation of the Signal Noise Ratio (SNR) result.

II. METHOD

A. Noise

Noise is defined as undesired signals in a communication or information system. These noise signals may affect the quality of signal reception and reproduction of the transmitted signal. Noise can also limit the system's range at a certain transmit power, affect the sensitivity of the reception signal, and cause a reduction in system bandwidth [4]. Noise in the radio receiver might generate an obnoxious hissing sound in the speakers. While on television, noise can generate an unclean image that seemed like dots shaped like snow.

B. Noise Canceller

Noise canceller or Adaptive Noise Cancellation is an application of Active Noise Control (ANC). Noise canceller systems are often used for various needs, especially in the communication process because of the ability of this system to reduce noise. From 1957 to 1960, Howells and Applebaum conducted research on the implementation of a noise cancelling system for sidelobe cancelling antennas [5].

The noise canceler system can be applied using two methods: the single-channel method using the Wiener filter method, the Kalman filter, or the spectral subtraction method and the double-channel method. The double-channel method requires two inputs: one is an info signal mixed with noise, and the second is reference noise with a correlation with noise that interferes with the info signal at the first input. Figure 1 shows the noise canceller concept.

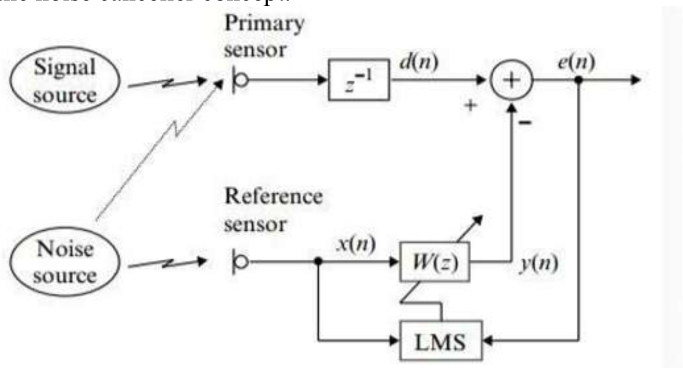

Figure 1. Basic Concept of Noise Canceller

C. Least Mean Square Algorithm

The LMS algorithm is a common and simplest method that can be used to solve signal processing issues such as noise, echo, and interference. This study uses the LMS (Least Mean Square) algorithm and Finite Impulse Response (FIR) Filter. LMS algorithms are often used in adaptive signal processing for a variety of applications.

The popularity of the LMS algorithm is also supported by the following reasons: easy and simple computation, no data repetition, and does not require gradient forecasting. The Least Mean Square (LMS) algorithm is cancelling that uses the Δ gradient operator in its adaptation process. The adaptation process of the tap-weight (filter coefficient weight) takes place recursively and starts with an initial value. Therefore, the results obtained will be better if the number of iterations is greater [6].

The final result that is expected from this iteration process is a value that converges to the solution of the Wiener filter method. The recursion process commonly used is steepest descent, which is shown in Equation (1):

$$(n+1) = w(n) + {}^{1}\!/_{2}\, m[-\Delta(J(n))] \quad (1)$$

To calculate the Δ gradient vector (J(n), the effective method is to substitute the correlation matrix R and the cross-correlation vector, as shown in Equation (2):

$$\Delta\big(J(n)\big) = -2p + 2Rw(n) \quad (2)$$

The simplest estimator for R and P is to use an estimation based on the input tap vector sample size {u(n)} and the desired response {d(n)}, as determined by Equation (3):

$$R(n) = d(n)x(n); p(n) = x(n)x^T(n)w(n) \quad (3)$$

Parameter H represents the value of the Hermitian matrix (conjugate complex). For the gradient vector value, it is obtained by substituting Equation (4):

$$\Delta\big(J(n)\big) = -2\big(x(n)x^T(n)w(n)\big) + 2\big(d(n)x(n)\big)w(n) \quad (4)$$

After obtaining the value of each parameter, the updated value of the tap-weight (weight of the filter coefficient) can be determined using Equation (5) as follows:

$$w(n+1) = w(n) + \mu(p(n) - R(n)w(n)) \quad (5)$$

Based on the overall formula, the conclusions of the LMS algorithm are as follows:

- Filter output: y(n) = w(n) x(n)
- Estimation error: e(n) = d(n) – y(n)
- Tap - weight adaptation: w(n+1) = w(n) + μ d(n)e(n)

This LMS algorithm does not require a complicated calculation process because it does not require the calculation of the correlation function or the calculation of the inverse matrix. The principles of this simple calculation can be easily applied in the form of a computer program. It is because of this simplicity that the LMS algorithm is often used in the adaptive filter's calculation.

D. STM32F4 Nucleo-F446RE

The STM32 Nucleo-64 board provides an affordable and flexible way for users to try out new concepts and build prototypes with the various combinations of performance and power consumption features provided by the STM32 microcontroller. For compatible boards, external SMPS reduces power consumption in Run mode significantly.

Arduino™ Uno V3 connectivity support and ST morpho headers allow for expansion of the functionality of the open STM32 Nucleo platform with a wide selection of custom shields. The STM32 Nucleo -64 board does not require a separate probe as it integrates the ST-LINK debugger/programmer. The STM32 Nucleo-64 board comes with a comprehensive STM32 software library as well as examples from the STM32Cube MCU package [7]. Figure 2 shows the shape of the STM32 used.

Figure 2. Mikrocontroller STM32 Nucleo

E. Adaptive filter

Adaptive filter is a filter that can adapt automatically based on a certain algorithm. The output of this adaptive filter is fed back and processed according to a predetermined algorithm to get the next filter coefficient so that the quality of the output matches (very close to) the expected [8].

Adaptive filters can be implemented with either FIR or IIR structures. FIR filters are typically realized non-recursively, whereas IIR filters are. The following is the differences between FIR and IIR structure:

1. Transverse filter structures, also known as tapped delay lines, are often used in adaptive filters with FIR architecture. This structure has an all zeros transfer function so that the adaptive filter with the FIR structure is always stable.
2. Adaptive filters with IIR structure usually use canaconic direct form because of their ease of realization and analysis. However, the adaptive filter with IIR structure tends to be unstable because the pole position may be outside the unit circle.

F. Mean Square Error (MSE)

MSE (Mean Square Error) is the average value of the output signal error [9]. The MSE value can be calculated using Equation (6):

$$MSE = \sum_{i=1}^{n} (S - S_e)^2 \qquad (6)$$

N is the signal length, S is the input signal, and Se is the output signal. The signal-to-noise ratio is the ratio of signal strength (valuable information) over noise strength (unwanted signal) [10]. SNR can be formulated in Equation (7):

$$SNR_{dB} = 10 \log_{10} \left(\frac{P_{signal}}{P_{noise}} \right) \qquad (7)$$

The peak signal-to-noise ratio (PSNR) is the ratio between the maximum signal value measured and the amount of noise that affects the signal [10]. PSNR is used to compare the quality of sound before and after processing. To determine the PSNR of a sound, the MSE value must first be known. The higher the PSNR value, the more comparable the sound quality of the processing result is to the original sound quality. PSNR can be determined using Equation (8), where MAX is the maximum value of the measured signal [11].

$$PSNR = 10 \log_{10} \frac{MAX^2}{MSE} \qquad (8)$$

G. Diagram Block System

Figure 3 shows a block diagram of an FIR adaptive filter. d(n) is an input audio signal mixed with noise, while x(n) is a noise signal. The x(n) signal is decomposed into a digital filter and an adaptive algorithm as a programming parameter of the algorithm. The signal that enters the digital filter will be filtered using an FIR filter. Next, the filter output will be a y(n) signal. The output signal from the FIR filter will be combined with the audio signal mixed with noise. Thus, there will be noise reduction in the d(n) signal [12].

Furthermore, the results of this signal will be entered into an adaptive algorithm which will adjust the coefficient where the filter parameters are set in such a way that it can optimize the signal from distortion (defects) to a minimum to minimize signal noise in the audio. Then the filter will be implemented using STM32F4.

Figure 4 explains that the sound signal mixed with noise is the input signal taken from the function generator by setting the required frequency as a test parameter and noise as a reference which is fed into the bias circuit. Reference noise is obtained from the audacity software from a laptop that is connected using an audio cable. The bias circuit is used as a voltage amplifier.

Meanwhile, the transistor requires a bias voltage to perform as a voltage amplifier. The voltage is amplified so that the resulting signal is only positive. The voltage will be adjusted by 1.5 V over the initial value.

An Analog to Digital Converter is an electrical device that converts analog signals (continuous signals) into digital signals. The ADC is attached to one of the STM32F4 device's pins, which serves to limit analog signal processing by digital systems and convert digital signals to analog [13][14]. ADC 1 serves as an amplifier in the adaptive circuit, whereas ADC 2 serves as a reference for the adaptive filter system.

The results of the ADC configuration will produce an audio signal in the form of a digital signal. In this process, the STM32F4 is configured using a program created on the Arduino software (IDE) and then processed and computed using an adaptive Finite Impulse Response (FIR) filter with the LMS algorithm. Furthermore, the filtered signal which is still a digital signal will be converted into an analog audio signal using a DAC. After the amplification process, the results of the analog signal can be seen using an oscilloscope.

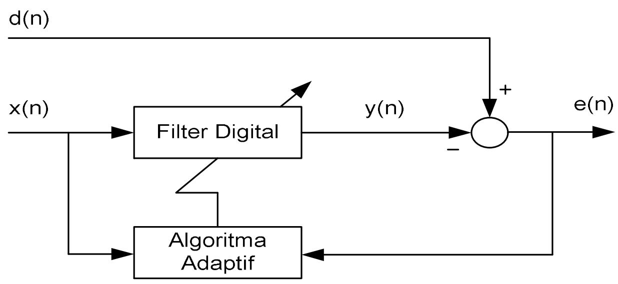

Figure 3. Filter Adaptif FIR Diagram Block

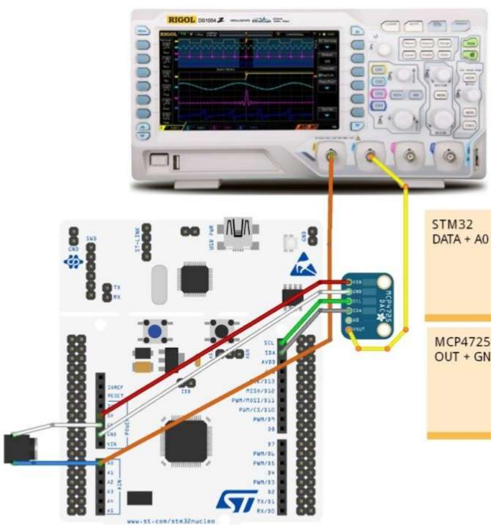

Figure 4. System Circuit Scheme

III. RESULTS AND DISCUSSION

In this section, the adaptive filter using the Least Mean Square algorithm tested using a two different types of sound samples with various order values and different signal durations.

The results of the adaptive filter test with a value of order 5 and a duration of 9 seconds can be seen in Figures 5 to 7.

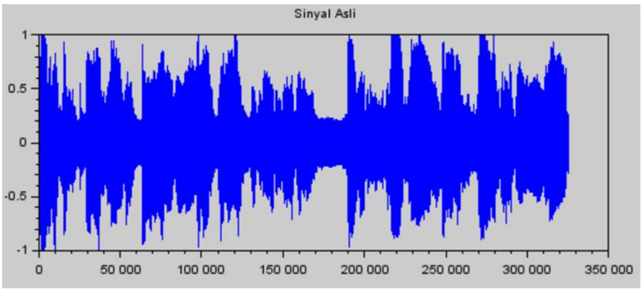

Figure 5. Original Signal Graph Output Results At 9 Seconds

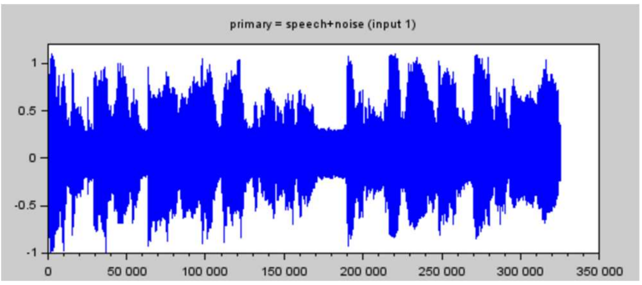

Figure 6. Combined Signal Graph Output Results at 9 Seconds

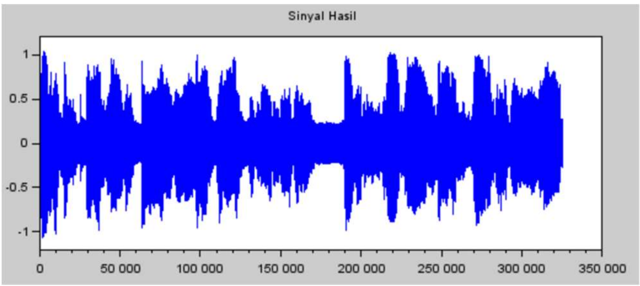

Figure 7. Final Graph Signal Result At 9 Seconds

Based on the signal image above, it can be seen that the adaptive method can keep the signal stable despite noise interference for 9 seconds. Figures 8 to 10 show the results of the adaptive filter test with an order of 5 and a duration of 14 seconds.

Same as the previous results, the use of order 5 in the adaptive filter system designed is able to maintain signal quality from the information signal sent to the information signal received.

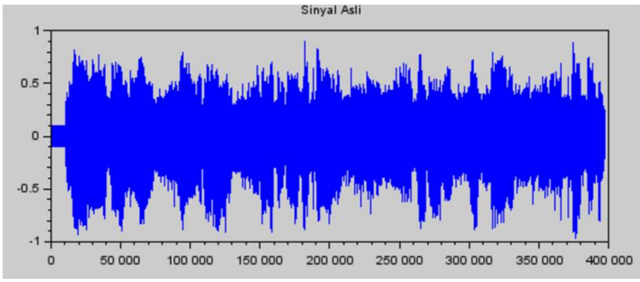

Figure 8. Original Signal Graph Output Results At 14 Seconds

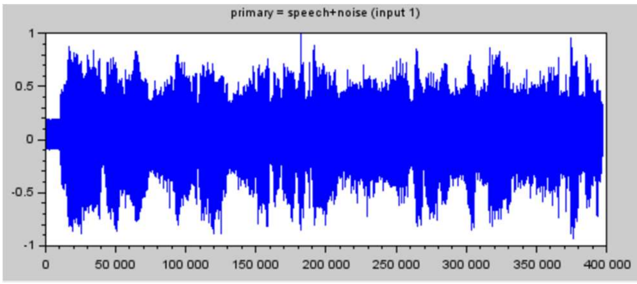

Figure 9. Combined Signal Graph Output Results at 14 Seconds

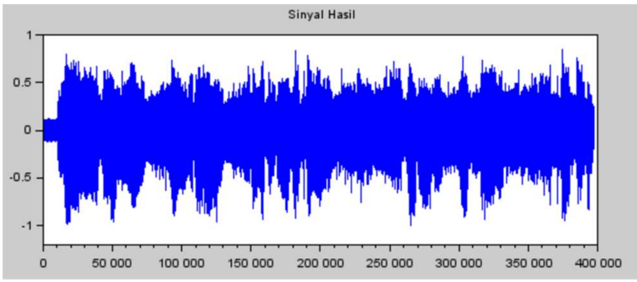

Figure 10. Final Graph Signal Result At 14 Seconds

To analyze the average value of the test results, the test results will be divided into three parts, namely the SNR, PSNR, and MSE values of the filter order.

The first test is the SNR of the voice signal. SNR is the filter's signal performance metric. SNR is a comparison of the value of the information signal and noise. The higher the SNR number, the better the filtered signal.

The second test is the PSNR of the voice signal. PSNR is another metric used to determine sound quality and filter performance. The higher the PSNR number, the better the filter.

The third test is the MSE test. MSE is another parameter used to define the noise canceler system's minimum MSE. The lower the MSE value, the higher the quality of the sound.

Table 1 shows the results of the SNR, PSNR, and MSE tests performed on female voice samples with two distinct durations of 9 seconds and 14 seconds using an order 5 value.

TABLE I
RESULTS OF SNR, PSNR AND MSE VALUES ON FILTER ORDER 5

Orde	Duration	Iteration	SNR	PSNR	MSE
5	9 second	10	2.744	145.241	0.01942
		50	27.868	146.867	0.00427
		100	27.923	146.952	0.00539
		150	29.138	147.939	0.00135
		200	32.778	169.666	0.00249
		300	34.464	169.793	0.01940
		400	35.559	169.936	0.00248
	14 second	10	27.598	145.259	0.00427
		50	27.918	146.889	0.00431
		100	29.324	163.822	0.00431
		150	3.184	16.43	0.00427
		200	32.898	164.663	0.00316
		300	34.476	164.092	0.00318
		400	37.071	165.016	0.00253

From the data above, the value of SNR and PSNR for time duration 9 seconds, the largest value is obtained at 400 iterations: 3.5559 dB with PSNR 16.9936 while the maximum MSE is found at 300 times iteration value of 0.01940. While in the sample duration of 14 seconds, the largest SNR, PSNR, and MSE values were obtained at the iteration value of 400 times, which are 3.7071 dB SNR, 16.5016 PSNR and 0.00253 MSE.

TABLE II
RESULTS OF SNR, PSNR AND MSE VALUES ON FILTER ORDER 40

Order	Duration	Iteration	SNR	PSNR	MSE
40	9 detik	10	46.363	189.583	0.00162
		50	47.034	191.164	0.01111
		100	49.095	190.322	0.02205
		150	4.947	201.159	0.02203
		200	53.444	200.796	0.00579
		300	52.775	213.551	0.02205
		400	53.022	213.557	0.02240
	14 detik	10	47.064	196.171	7 0.00336
		50	49.464	204.447	0.00575
		100	4.908	204.105	0.00721
		150	53.598	213819	0.00575
		200	53.265	213.794	0.00569
		300	53.671	221.707	0.00639
		400	53.596	221.449	0.00720

According to the data in Table II, the SNR value for a time span of 9 seconds is 5.3444 dB, the PSNR value at 400 iterations is 21.3557, and the highest MSE is 0.02240 at 400 iterations. While the duration sample 14 obtained the highest value at 300 iterations, which is 5.3671 dB, the PSNR value at the same iteration is 22.1707, and the maximum MSE is obtained at 400 iterations, which is 0.00720. Order 40 is the other order chosen since the outcomes are the best when compared to the other orders.

IV. CONCLUSION

The adaptive filter coefficient is affected by the effect of filter order on the performance results of the STM32F4 nucleo-board device. The influence of filter order was evaluated using three signal performance parameters: SNR, PSNR, and MSE. According to the data in the table, the higher the SNR and PSNR values acquired, the better the filtering process that occurs. At a sound duration of 14 seconds, the best SNR value of 5.3671 dB is attained at Order 40. The higher the order value, the higher the obtained SNR value. The PSNR value of 21.3557 for a sound duration of 9 seconds is the same.

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