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ELECTRET CONDENSER MICROPHONE AS SENSOR IN ARTERIAL PULSE RECORDING DEVICE

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ABSTRACT: One of the human pulses could be detected on an arterial radial that represents the human health condition. Several sensors could capture the pulse signal. However, it still needed a sensor with a small dimension and lightweight. One of the sensors that comply with those specifications is an electret condenser microphone (ECM). The ECM was designed to pick up the signal from the human radial arterial pulse. The ECM pick up system was equipped with an instrumentation amplifier and a signal processing system. The characteristic of the ECM was analyzed using a sixth-order polynomial equation. The equation was used to reconstruct the signal, which flattened the frequency response at the range of 0.5-4.5 Hz. The signal processing consists of an FFT, peak detection and a sixth-order anti-polynomial computational unit which is implemented in a computer. The minimum error was found at 1.5 Hz, which is closed to the designated frequency of the arterial pulse normal frequency.

Keywords: Sensor, ECM, an Arterial pulse signal, Frequency response

1. INTRODUCTION

Pulses present in several parts of the body. One of them is located in the radial artery. The pulses can represent the health condition of the human [1]. The pulse signal is a mechanical signal that has a frequency range of 0.5-4.5 Hz. A sensor is required to capture the mechanical signal and then used in arterial pulse recording. One kind of sensor can be realized by an Electret Condenser Microphone (ECM), which is well-known for its stability [2]. The ECM has a relatively flat frequency response within the audio frequency of 50-20 kHz. However, the frequency of the pulse around 0.5-4.5 Hz is far away from the microphone response range. Lately, a number of researches on the ECM showed the increase of accuracy of its mechanical signal capturing [3]. The increase opens possibilities of utilizing them as a sensor on the arterial pulse recording device.

In our previous study, the ECM has been used to detect the pulse in the human radial arterial area. Figure 1 shows the results of the recording of the human arterial pulse signals.

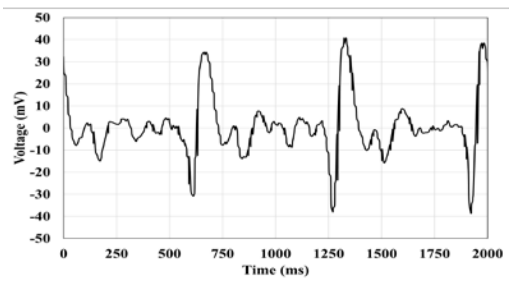

Fig. 1 Human Arterial Pulse Signal acquired using the ECM

The measurement produces a pulse signal recording, which is expectedly similar to that of Shusaka Nomura [4, 5]. However, the signal has not been verified related to the character of the ECM. The character of the microphone should be characterized at the range frequency of 0.5-4.5 Hz.

Some researchers have worked on the pulse signal measurement models using the ECM. One of them employed a soundproof sponge that is placed in a cylinder tube as an intermediate medium to capture the pulse signals [6, 7]. The model has advantages in terms of low cost and simple. However, the tube has a large dimension because it still requires air as a medium.

Another technique of the pulse signal measurement using was conducted by means of a Micro Electro Mechanical System Electret Condenser Microphone (MEMS-ECM). The MEMS-ECM is a type of microphone that has a small dimension. The technique using an integrated circuit has a flatter frequency response than a differentiation circuit compensation. However, the frequency response is still about -35 *dB* [4] and -40 *dB* [8], so it shows that the output signal still low compared to the input.

The application of the microphone as a sensor requires a predetermined frequency response obtained from the measurement. The measurement must be performed at the extremely low-frequency region. The very low- frequency response from 0.5 to 20 Hz can be reliably measured by removing the plastic cover and dust filter of ECM [3]. However, this method decreases the signal to noise ratio due to the removal of the filter and cover.

This paper proposes a new method of arterial pulse measurement using ECM. The system is intended to condition and process the signal and obtaining a flat closed to 0 *dB* frequency response in the range of 0.5-4.5 Hz.

2. METHODOLOGY

2.1 ECM Testing System

2.1.1 Electret Condenser Microphone

Electret Condenser Microphone (ECM) is a type of microphone that converts audio to an electric signal. The audio pickup section of the ECM has a structure of a condenser consisting of a diaphragm and a back plate opposite. The ECM that used in this study was KUC2123 from HOSIDEN with 9.4 mm of diameter, 6.5 mm of height and 0.8 g of mass. The operation of this microphone is in the frequency range of 50-20k Hz [9]. ECM has a fabrication frequency response of 0 *dB* in the frequency range 50 Hz-20 kHz as shown in Fig. 2

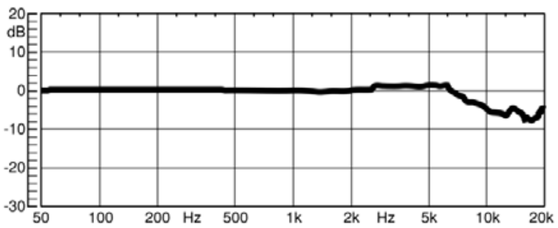

Fig. 2 Frequency response of commercial ECM [9]

However, the ECM frequency response in the range of 0.5-4.5 Hz for the pulse reading system is unknown.

2.1.2 Testing setup

The setup for testing ECM frequency response is designed, as shown in Fig. 3. The system consists of a signal generator, a low- frequency amplifier, a low-frequency loudspeaker as a source of mechanical vibration, instrumentation amplifier, and an oscilloscope.

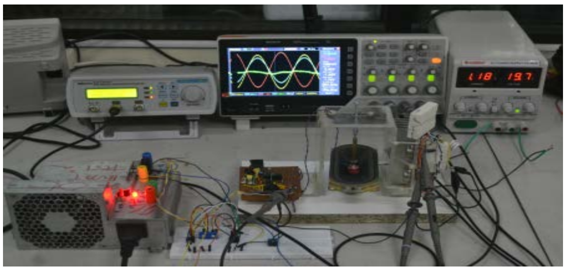

Fig. 3 The setup of the ECM testing

The signal from the signal generator was amplified by a low-frequency amplifier that vibrates the low-frequency loudspeaker. The vibration of the loudspeaker is felt by the ECM that covered by rubber housing through soft silicone rubber [10]. Soft silicone rubber functions as a wave medium between low-frequency loudspeakers and ECM.

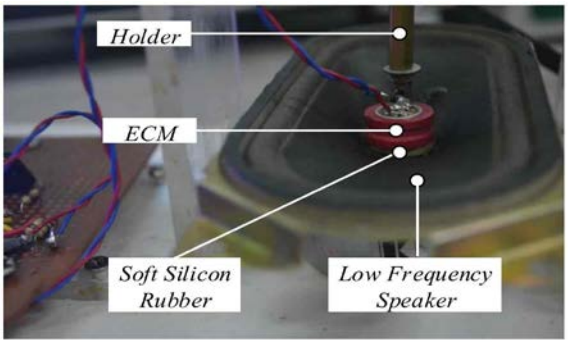

Fig. 4 The placement of soft silicon rubber in ECM testing [10].

Figure 4 shows the placement of the soft silicone rubber as the medium between the low-frequency loudspeakers and the ECM. The holder serves as a support to keep the ECM in place.

2.1.3 Data acquisition system

The ECM output signal at low frequency was quite small, so we designed an instrumentation amplifier to amplify the signal. Figure 5 shows the schematic wiring of signal instrumentation amplifier that consists of 2 OP Amp. The function of this circuit is to eliminate the low noise frequency and reducing noise interference from the power supply.

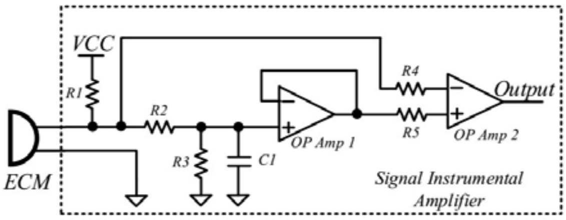

Fig. 5 Schematic of the instrumentation amplifier

We select a high input impedance Op-Amp which has a typical slew rate to match the high impedance output of the ECM to obtain an accurate signal response. The selected OP Amp was TL 062 from Texas Instruments with specification as follows, typical supply current: 200 µA (pre-amplifier), High Input Impedance: JFET-Input Stage and high slew rate: 3.5 V/µs Typical. Some passive components required such as R1 is 50 KΩ, R2 is 1 KΩ, R3 is 200 KΩ, R4 is 500 Ω, R5 is 500 Ω, and C1 is 1 µF.

The amplified output signal was connected to a computer for digital signal processing, as shown in Fig. 6.

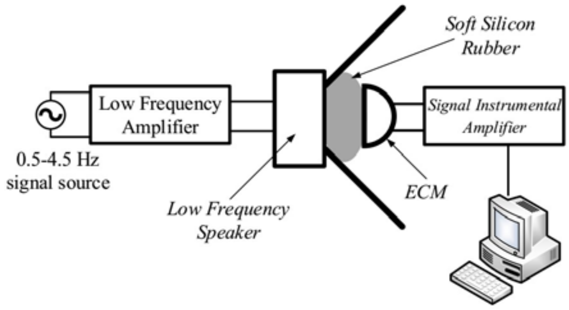

Fig. 6 Experimental setups of ECM test

Table 1 shows the test condition of the experimental setup. The test condition consists of specifications of silicone rubber, laptop specification, ECM specification, and digital storage oscilloscope.

Table 1 Test condition of the experimental setup

Items	Specification
Signal generator	K-Moon DDS Signal Generator/Counter (Dual Channel)
Computer specifications	Intel Core i5 CPU 2.30 GHz, 4 GB RAM
ECM	sensitivity: −45 *dB*, impedance: 1.0 KΩ PSU: 4.5 VDC Current : 0.8 mA
Oscilloscope	Digital Storage Oscilloscope Hantek MSO5074F 4 Channel 70 MHz, 1 GSa/s

2.2 Measurement of the ECM Signal

In Fig. 3, the output signal of the ECM was measured by means of an oscilloscope. The amplitude at different frequencies is displayed in Fig. 7. The characteristic of the ECM is critical for the design of signal processing.

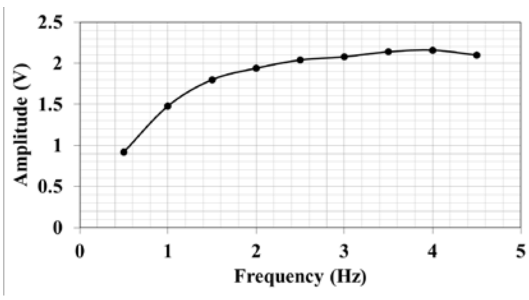

Fig. 7 The ECM characteristic at 0.5-4.5 Hz frequency range

Based on Fig. 7, the characteristic of ECM can be interpreted in the polynomial equation. The amplitude of the ECM's response increase and tend to flat in higher frequencies. The smallest amplitude of the ECM characteristic is 0.92 *Vpp* (peak to peak), and the highest amplitude is 2.16 *Vpp*.

2.3 Signal Reconstruction

Based on Fig. 7, we approached the character of the ECM into a polynomial equation. The equation consists of a sixth-order polynomial equation as given by Eq. (1).

$$y = 0.0093x^6 - 0.3226x^5 + 3.0306x^4 + 2.9419x^3 - 192.06x^2 + 1081.7x + 24.444 \tag{1}$$

From Eq. (1), we reconstruct the signal by using an anti-polynomial equation. The signal reconstruction was aimed to obtain consistency of frequency response at the range frequency of 0.5-4.5 Hz. Another goal of the anti-polynomial equation is to convert the signal to a digital mode so that it is suitable for digital processing. The anti-polynomial equation is formulated in Eq. (2) as follow.

$$y = 0.0001x^6 - 0.0041x^5 + 0.0576x^4 - 0.04274x^3 + 1.7692x^2 - 3.9478x + 4.9001 \tag{2}$$

Based on Eq. 2, we construct the anti polynomial graph presented in Fig. 8.

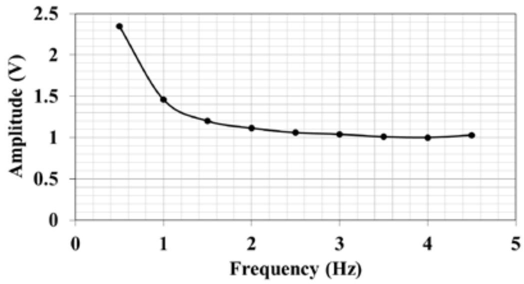

Fig. 8 Anti polynomial graph at 0.5-4.5 Hz frequency range

Based on Eq. 1 and Eq. 2, we design the signal processing system. Linear filtering and pattern matching are techniques for determining the presence of specific waveforms in a signal [11]. The FFT is a numerical computation which converts the time domain periodic signals into frequency domains data [1]. The FFT is performed for application purpose in signal processing and data manipulation embedded in a Digital Signal Processing (DSP) or a microcontroller [11]. On the other hand, the IFFT is the inverse processing of the FFT into the time domain. The implementation of the IFFT on a DSP or microcontroller is carried out using convolution and a correlation [12], respectively. Linear filtering is used to pass 0.5-4.5 Hz of frequencies and block others. The linear filter is the same as an FFT that the algorithms are often able to speed up their computation [11]. The FFT function for linear filtering, x*(k-i)* is the input sequence to the filter and *h(i)* is the unit pulse response of the filter. Then, for matching, *x (k + i)* is the input signal and the pattern to be found in the signal that is represented by *h(i)*.

$$y(k) = \sum_{i=0}^{M-1} x(k-i) * h(i) \tag{3}$$

$$y(k) = \sum_{i=0}^{M-1} x(k+i) * h(i) \tag{4}$$

Figure 9 describes the steps for the implementation of Eq. (3) and Eq. (4).

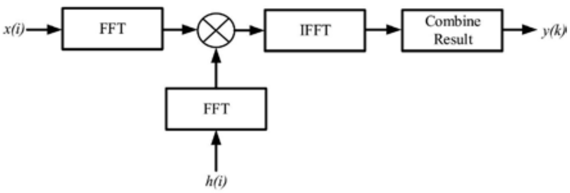

Fig. 9 Block diagram of frequency domain processing in DSP [11].

This processing system has high efficiency because it has advantages, i.e. reducing signal processing computing, increasing signal processing speed and reducing power consumption on DSP. The FFT must be multiplied by the sixth order anti polynomial equation to get a flat response and converted to the time domain using IFFT. The system of signal processing is shown in Fig. 10.

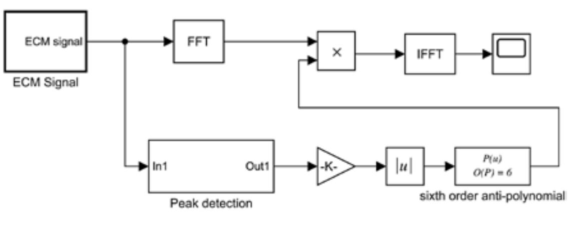

Fig. 10 Signal processing system

The reconstructed signal of the system is shown and compared with the signal before processing in the result and discussion below.

3. RESULT AND DISCUSSION

The amplitude of ECM before and after processing are tested and analyzed. The results show that the proposed system has a good response, as given in Table 2. From Table 2, we can define that the amplitude before processing is decreased when the frequency is reduced. On the other hand, the amplitude after processing is relatively constant at a range of 2 *Volt*.

Table 2 The amplitude of ECM before and after processing

Frequency (Hz)	Amplitude before processing (V)	Amplitude after processing (V)
0.5	0.92	2.000
1	1.48	2.057
1.5	1.8	2.080
2	1.94	2.093
2.5	2.04	2.100
3	2.08	2.102
3.5	2.14	2.108
4	2.16	2.117
4.5	2.1	2.102

From Table 2, we can calculate the gain value of ECM amplitude. The Voltage gain is determining in *Av*, where *Vo* is the output voltage, and *Vi* is the input voltage. We can obtain the *Av* by using the upper of the amplitude of ECM as *Vi*. In this case, the upper of the amplitude is 2.16 *Volt*. The voltage gain *(Av*) was determined from the value of *Vo* and *Vi* by using the Eq. (5) below.

$$Av = \frac{Vo}{Vi} \tag{5}$$

Based on Eq. (5), the gain of the system can be shown in Fig. 11. The gain of the system after processing is relatively flat in the range of 1. It showed that the outputs of the system were similar to the input.

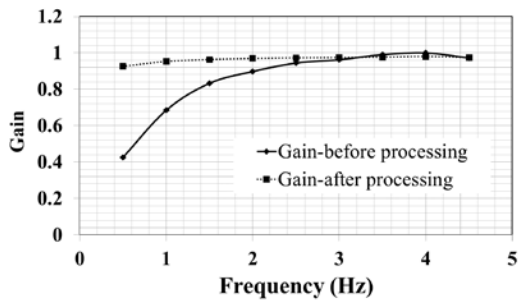

Fig. 11 The gain of before and after processing

The magnitude of the ECM frequency response is in decibel (*dB*) or ECM (*dB*) can be determined by using Eq. 6, where *Vo* is the output voltage, and *Vi* is the input voltage.

$$ECM\,(dB) = 20\log_{10}\frac{Vo}{Vi} \tag{6}$$

The frequency response of ECM (*dB*) at the frequency range of 0.5-4.5 Hz is shown in Fig. 12. The frequency response before processing was decreased. However, after processing the frequency response of ECM (*dB*) is relatively flat at the range around 0 *dB*. It showed that the output voltage was similar to the input voltage.

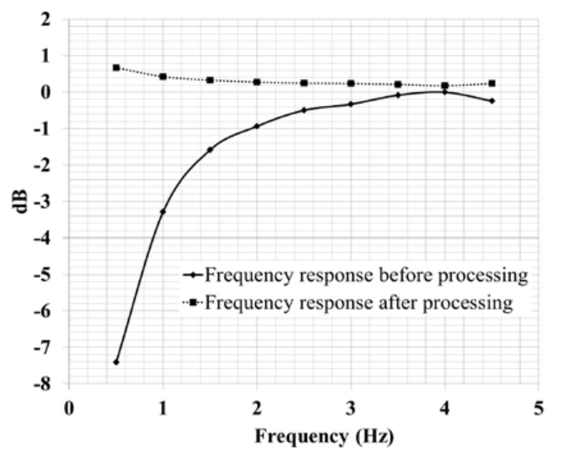

Fig. 12 The frequency response of before and after processing

Based on Eq. (6), the output voltage after processing is shown in Fig. 13. The output voltage before processing is compared to output voltage after processing. In this figure, the output voltage before processing is not flat. However, the output voltage after processing is relatively flat in the range of 2.1 Volt. It shows that the output voltage closed to the input voltage.

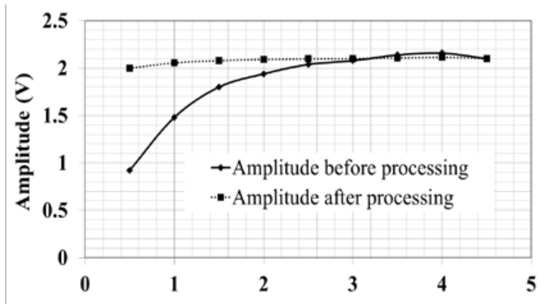

Fig. 13 The output voltage before and after processing.

Based on Fig. 13, the output voltage and input voltage are compared to get an absolute error value. The error value is determined in Eq. (7) as follows:

$$Error = \left|\frac{V_{after} - V_{average(after)}}{V_{average(after)}}\right| x100\% \tag{7}$$

Figure 14 shows the absolute error value from the experiment. The error value from the system is about 0.21-4.05%. The error is caused by the input characteristic of the ECM and its sixth-order polynomial representation. Furthermore, it also affected by the signal manipulation process which is the multiplication of the sixth-order anti-polynomial and the FFT output. The sixth order polynomial representation is the approach of the original signal. The error can be reduced by applying a higher-order polynomial. In previous research, Yang Mei applied an eighth-order polynomial implemented in DSP TMS320C67x using 'C67x library functions generates over 60% and up to 70.2% of total TI library function cycles [13]. So it can be concluded that sixth-order polynomial orders are capable of being implemented in the DSP processor.

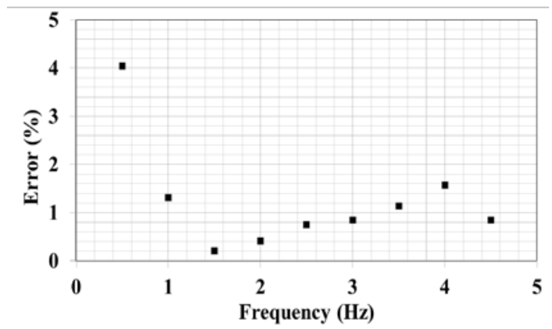

Fig. 14 The error of signal processing

This new method of arterial pulse measurement using the ECM covered by rubber housing [9] through soft silicone rubber obtaining a flat closed to 0 *dB* frequency response in the range of 0.5-4.5 Hz. This method can flatten the response frequency that closed to 0 *dB*.

4. CONCLUSION

In this study, the ECM was used as a sensor to record the signal of the human radial arterial pulse. A signal reconstruction was required to improve the output amplitude and to obtain consistency of frequency response at the range frequency of 0.5-4.5 Hz. The frequency response of the ECM is flat at 0 *dB* with the error value ranging from 0.21-4.05%

This work points out that the ECM will provide a better choice for arterial pulse sensor because of several advantages, such as simple, cheap, compact, portable and easy to use.

5. ACKNOWLEDGMENTS

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