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# Alternative Solution for Non Uniform Sensitivity of Experimental Equipment Case Study: Ambient Noise Measurement of Underwater Acoustic Channel

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**ABSTRACT—*In this paper, will be presented the normalization process of some measuring instruments whose have different sensitivity. This step is performed as the initial preparation before starting experimental measurements in the field. The process of normalization was carried out on the device sound audio processing, and hydrophones as front end sensors. Normalization is conducted in frequency domain, and shows that the comparison of sensitivity was fluctuate at different frequencies. Overall, have obtained results that sensitivity ration among hydrophones are: H-123 has a sensitivity ratio of 0.85 compare to sensitivity of H-2, and has a sensitivity ratio of 1.5 compared to the H-ref. While H-2 has sensitivity ratio of 2.3 compare to H-ref. From these results, we hope to contribute in the research topics based on measurements. So that the problems in the sensitivity of measuring devices that are not uniform can be solved.***

**Keywords—** measurement system; normalization; time domain analysis; frequency domain analysis.

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## 1. INTRODUCTION

Ambient noise generally appears continually at a certain location. This can corrupt the signal emitted by underwater acoustic equipment. So, the detection of background noise is important to get the signal-to-Noise ratio (SNR) of acoustical based underwater instrumentation. Ambient noise determines the baseline sound scape, which in general is background noise that appears in a location at sea at certain time. Practically, the ambient noise is associated with self noise, like the flow of water around the sonar. At the sonar system processing, background noise, has a similarity in the time, locations, and dept, where the signal is to be detected [1] and [2].

Research of wind effects on ambient noise have been carried out [3]-[5]. Generally, the research is related to the effects of wind and ship movement on acoustic noise that appeared. The discussion is focused on the observation of noise level and its relationship with the wind speed, other effects of weather on noise, and other noise from sea traffic. In the measurement process, that involving some of measuring instruments, the equipments should have a same sensitivity. In real conditions, researchers are often faced with the condition of the equipment which has not the same sensitivity. To overcome these problems required a uniformity of sensitivity and a re-calibration of the instruments. So that the sensitivity difference of the measuring instruments can be anticipated.

Research on the calibration process at low frequency hydrophones has been presented [6]-[8]. Calibration has been carried out for the hydrophone which has a working frequency of 0 ~ 3 kHz. The calibration method is based on absolute method, by using one calibrated hydrophone as a reference. Generally, there is a deviation from the result with deviation of 0.5 dB at the frequency of 400 Hz.

Calibration method by utilize optical multi-layer hydrophone as second technique of calibration method has been proposed [9]. In this method, optical multi-layer is used as calibration reference for piezoelectric layer at hydrophone. This method is robust, but its sensitivity is not good.

Hydrophone calibration for shallow water environment with sedimentation at the bottom has been proposed [10] and [11]. Both papers are based on a simple finite element method to improve performance of water-based calibration method.

Overall, the proposed calibration methods have advantage and disadvantages. But, from all of papers above, there is not describes how to handle a condition if there is not any certified instrument as a reference.

This paper propose a simple method to solve a problem when measurement instrument show a different sensitivity, and in this case there is no a certified instrument as a calibration reference. The proposed method is very simple, with normalized all of sensor sensitivity from data received. In this case, sensors are hydrophones that will be used for underwater acoustic channel measurement experiment.

This paper is organized as follows. In Chapter 2, will overview about experiment set-up that will be carried out. In Chapter 3 will describe analysis in the time and frequency domain. The summary and conclusion will give in Chapter 4.

## 2. EXPERIMENTAL SET UP

### *2.1. Set Up of Input Channel Normalization*

The objective of normalization process is to obtain receiving characteristics of input channels of M-Audio equipment. It is to anticipate if there is any different sensitivity among input channels.

Normalization is carried out by using a Set Up as fig 1. Underwater speaker as a sound source was placed in the bottom of a box, with a radiation pattern directed to a surface. One hydrophone is placed in the surface position, with the sensor directed to bottom, face-to face with UW speaker. The hydrophone has the following specifications: frequency ranges of 1Hz ~ 100kHz, a sensitivity of −190 dB re: 1V/µP a(+4dB, 20Hz − 4kHz), and internal capacitance of 25nF. The capturing pattern of hydrophone is almost omni directional, having the strongest response for the incoming signal with an angle of $0^o$ to the perpendicular surface to hydrophones. The response will be reduced when the signal arrives with an angle ≥ $10^o$, and the attenuation up to 0.8 relative to the strongest signal ($0^o$).This setting is subjected to minimize a multipath effect that possible to be happened. Sound will propagate from bottom to surface with no any reflection.

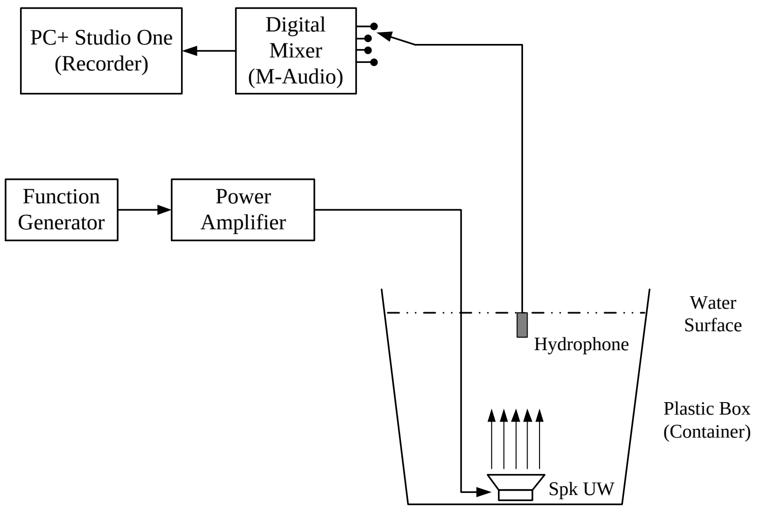


**Figure 1:** Set Up of M-Audio input channel normalization

Linear frequency modulation (LFM) signal with frequency sweep of 1~15 kHz, and the duration of 2 sec is used as tested signal. This signal is generated from a PC as a transmitter. The output is amplified by using a power amplifier with Aux-gain is set to 2.5 dB. The electrical signal is converted to acoustic signal by using underwater speaker. In the receiver part, hydrophone is works as a front line sensor, and converts the audio signal to be an electric signal. Gain of M-audio is set on 10 dB, and recorded at PC of receiver part. This off line processing is subjected for more detail and more flexible.

Measurement initialized by setting hydrophone at channel 1 of M-Audio, and continued testing with LFM signal. Testing signal is composed of three LFM signal with duration of 2 sec, and between successive signals with 0.2 sec. The results of recording is stored, and used for offline data processing. The next step is to replace the hydrophone output to Channel 2 of M-Audio. This step will continue with Channel 3 and Channel 4 of M-Audio.

### *2.2. Set Up of Hydrophone Normalization*

Normalization process aims to determine the characteristics of the reception of three hydrophones. This is to anticipate if there is a difference of sensitivity among the hydrophones. Equipment Set-Up is similar to normalization process of M-Audio reception, but in this case three hydrophones are used simultaneously. Hydrophone H-2 is positioned at Channel 1, hydrophone H-123 at Channel 2, and hydrophone H-ref at Channel 3. Gain power amplifier is same with the previous setting. Tested signal are generated 5 times for 5 data recording.

## 3. DATA PROCESSING AND MEASUREMENT RESULT

### *3.1. Sensitivity Ratio of M-Audio Channels*

The output signal from the hydrophones to Channel M-Audio is saved by using Studio One, with a value of 0 dB gain. Processing is done by using Matlab. The goal is to obtain an accurate result, and is expected to have good accuracy.

Processing begins with the signal recording division into three parts, each of which lasted 2 seconds, or the appropriate duration of generated LFM signal. The cutting process is done accurately by utilizing synchronization with Chirp signal. In this case the use of Chirp signal has an advantage in terms of ease in the synchronization.

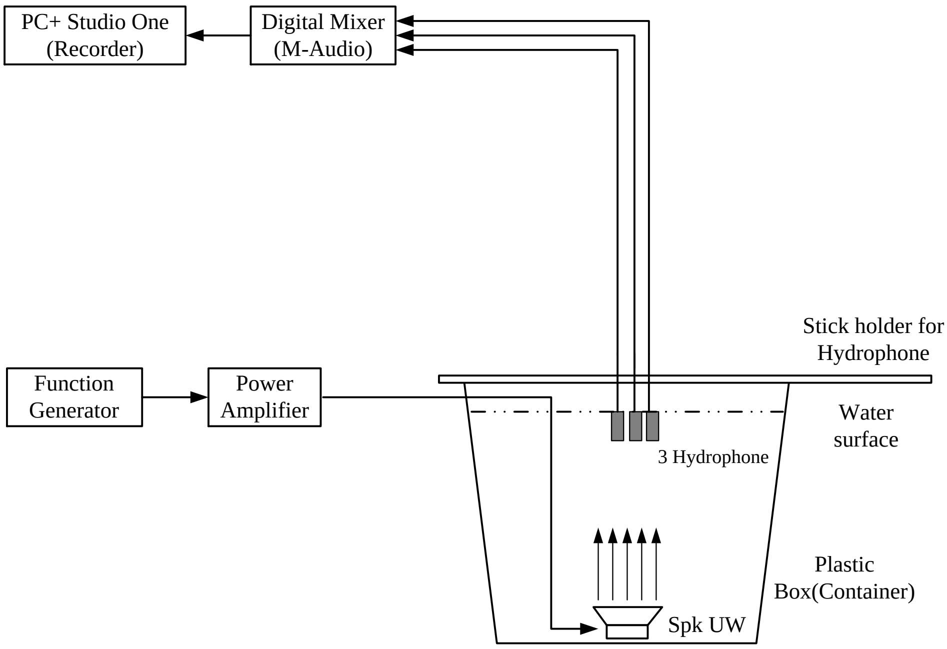


**Figure 2:** Set Up of hydrophone normalization

Signal in each frame is transformed into the frequency domain by utilizing the periodogram function, which gives an overview of the estimated power spectral density (PSD). If $x(t)$ is a discrete time signal $\{x(t);\ t = 0, \pm 1, \pm 2, \ldots\}$ is assumed to be a sequence of random variables. The covariance of $x(t)$ is defined as:

$$r(k) = E\{x(t)x^{*}(t\text{-}k)\}. \tag{1}$$

Where, $E\{\cdot\}$ denotes the expectation operator which averages over the ensemble of realization. The value of $x(t)$ is assumed to depend only on the lag between the two samples averaged. Variable $x^*(t\text{-}k)$ is complex conjugate of $x(t)$ with shifted version. The PSD is defined as discrete time Fourier transform (DTFT) of the covariance sequence

$$\phi(\omega) = \sum_{k=-\infty}^{\infty} r(k) e^{-j\omega k} \tag{2}$$

Periodogram output of all the frames averaged to represent one of the channel responses of M-Audio equipment. Results were compared between the channel alignment with each other, and outputs the resulting shape is like Figure 3.

Sensitivity Channel 1 and Channel 2 have similarities, so the reception ratio, in the frequency range of 1 ~ 12 kHz is about 1.2. But, at a frequency of 14 kHz Channel 2 tends to 2.3 times more sensitive than Channel 1. Channel 4 also has similar sensitivity with Channel 1. Channel 3 is more sensitive compare to the other channels, comparing with Channel is twice more sensitive at the frequency of 1 kHz, and almost 2.5 times more sensitive at higher frequency. It can be shown at top part of the chart. Channel 3 is also more sensitive than the channel 2, so that the ratio of Channel 2 to Channel 3 is about 0.5. Comparison of Channel 4 to Channel 3 also describes the same thing with the channel 2, which shows the sensitivity of 0.5 compared to Channel 3. Picture comparison sensitivity is a reference in the compensation transform and normalized the data input on a different channel in the M-Audio.

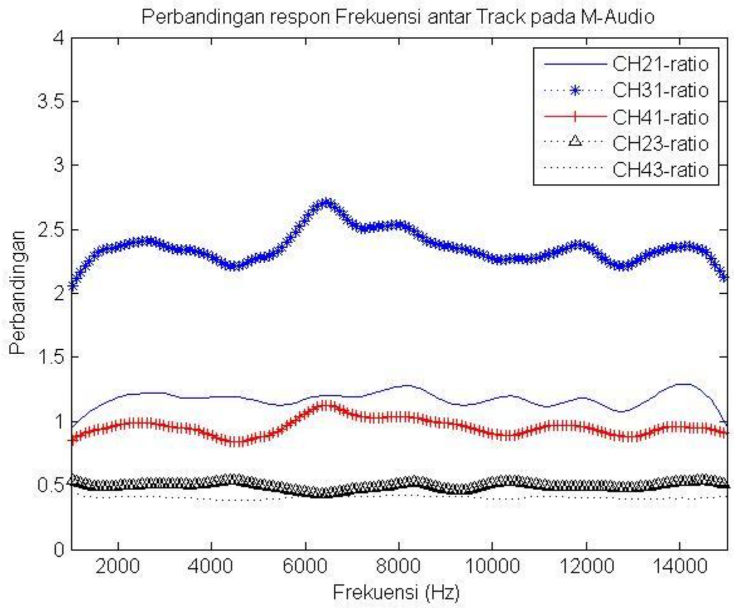


**Figure 3:** Sensitivity ratio among M-Audio channel

Processing of the data output from the hydrophones are normalized based on the channel that has been used. In this case the specified channel 1 as a reference, so that data is entered hydrophones H2 channel 1 is weighted 1, while data from the hydrophones H-123 entering into channel 2 is weighted 0.83, and hydrophones H-ref on Channel 3 is weighted 0.4. The signal outputs from the hydrophones are multiplied by weighting factor, and continued with the process of transformation to the frequency domain. The comparison results in the frequency domain provide a comparison of sensitivity hydrophones as a function of frequency as in Figure 4.

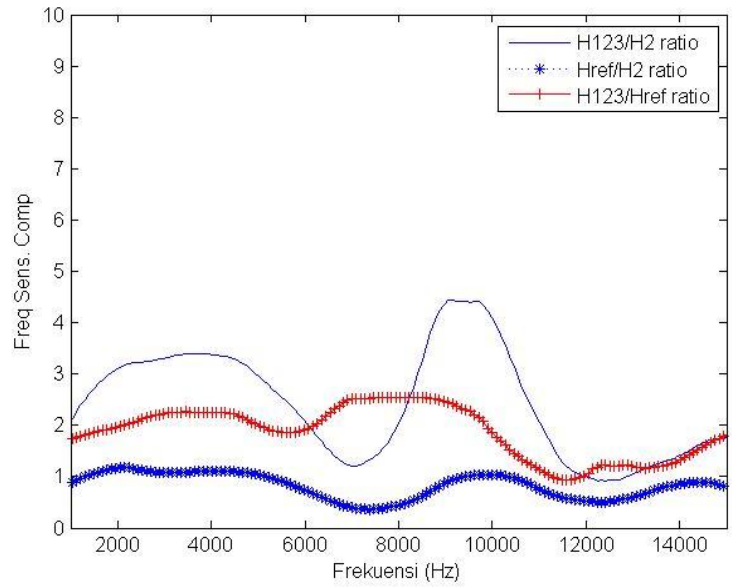


**Figure 4:** Sensitivity ratio among three hydrophones

Among hydrophones have different sensitivities, it is also influenced by the frequency of the input signal is observed. Hydrophone H-2 had similar sensitivity with hydrophone H-ref in the frequency range of 1 ~ 5 kHz, but its sensitivity is twice than hydrophone H-ref at frequency range of 6 ~ 9 kHz. At a frequency of 10 ~ 15 kHz, the both hydrophones have the same sensitivity. Hydrophone H-123 has a higher sensitivity than hydrophones H-2 and H-ref. Hydrophone H-123 is 2~3 time more sensitive compare H-2 at frequency range of 1~6 kHz. At the 6~8.5 kHz, sensitivity ratio is about 1.4, and At frequency range of 8.5 ~ 10.5 kHz the sensitive of H-123 is 3 time more sensitive compare to hydrophone H-2. At frequency range of 10.5 ~14 kHz the sensitivity of H-123 is 1.3 times compare with hydrophone H-2. The sensitivity of H-123 is twice than sensitivity of H-ref at frequency range of 1 ~ 10 kHz, and sensitivity of hH-123 is almost same with H-ref at frequency range of 11 ~ 14 kHz, with the sensitivity ratio of 1.2. This sensitivity ratio can be used as a reference in the analysis of measurement data which was obtained with hydrophones that has recorded with M-Audio.

### *3.2. Normalization of Ambient Noise Measurement*

The normalization output of hydrophone sensitivity is used for normalizing the measurement results. In this case study, discussion is concentrated on environmental noise measurements in Kenjeran bay, Surabaya. Output of the signal processing from measurement is simplified as a block diagram in Figure 5. Signal result of taking the measurements of all the hydrophones are used formed into a block frame with a predefined size corresponding to the length of FFT to be used. In this case set at 512 samples, and the output of the frame blocking converted to the frequency domain with FFT. The normalization has done by utilizing the results of the comparison sensitivity of hydrophones, as already obtained in the previous sub-chapter. Audio signals that have been normalized with IFFT were converted back into the time domain. The process of adjustment of the sample matrix form made for the recovery of audio signals.

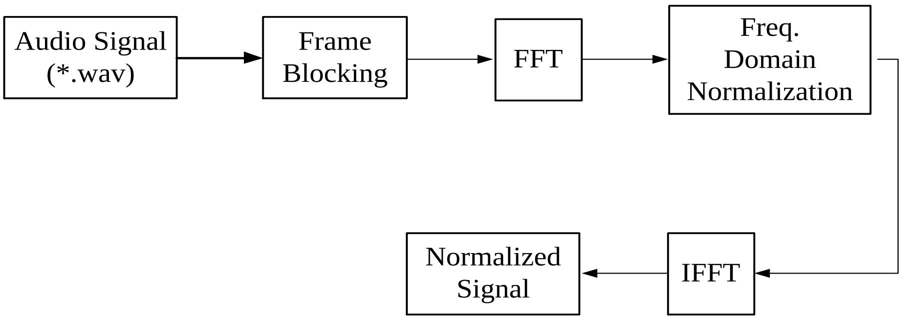


**Figure 5:** Block diagram of frequency domain normalization

Comparison of the results and the normalization of the original signal from the measurements by using hydrophones H_ref can be seen in Figure 6. By zoom the samples one seconds, it looked the change in signal level. In some samples signal amplification up to the levels up to the value of 0,005 volts, and in some other samples decreased to -0.005 volt. This illustrates that the signal level actually recorded by hydrophones H_ref amplitude distortion.

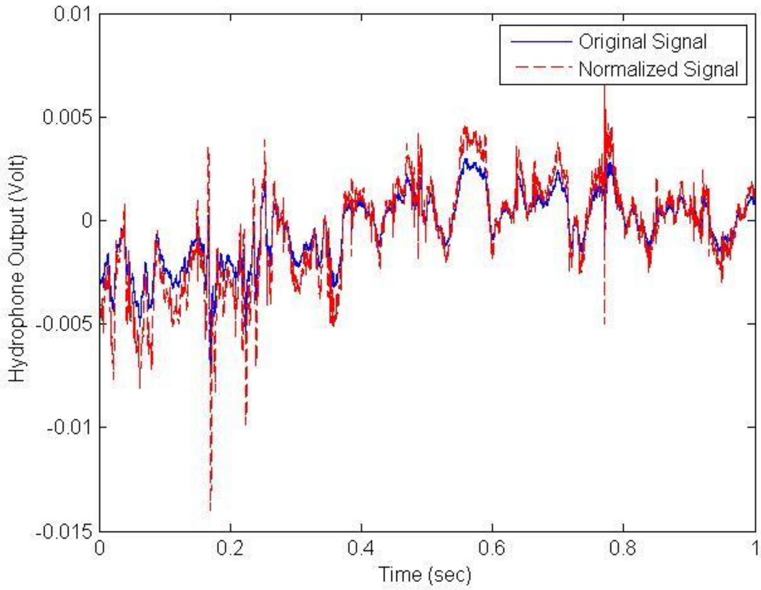


**Figure 6:** Time domain comparison of original and normalized signal.

Observations in the frequency domain illustrates that the process of normalization provide reinforcement signal power level of the original data. At frequencies less than 100 Hz, it was almost no change, but at frequencies above 100 Hz power changes began to occur. The increase in power levels occur about 5 ~ 10 dB of the original signal power levels that achieved by H_ref. At some certain frequency also attenuated, for example at 1250 Hz. Here, a decreasing in level until large enough to output is lower than -150 dB. But, overall are more going on the process of strengthening the level of data. The measurements result of ambient noise that has been in normalization has been compared with the results on the reference paper [4] and [5]. The result is not exactly the same, but in general has a similarity pattern.

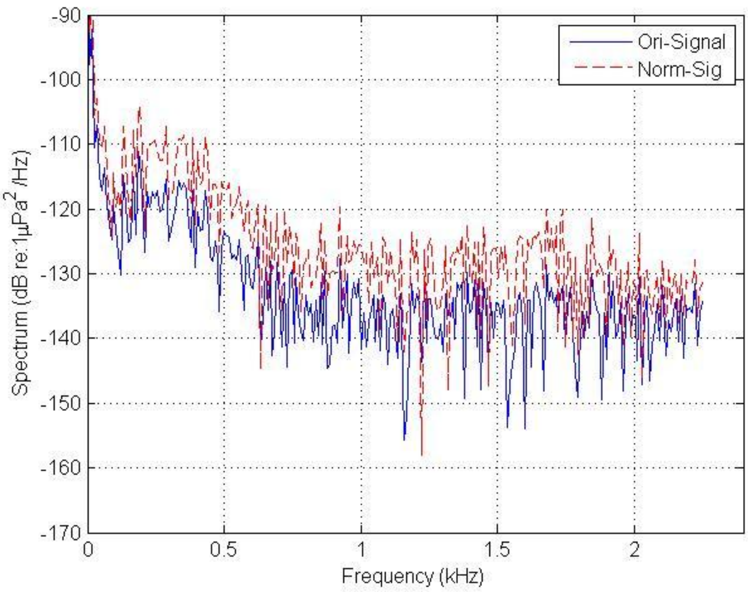

**Figure 7:** Frequency domain comparison of original and normalized signal

## 4. CONSLUSION

This paper presents a normalization process measuring instrument sensitivity of hydrophones. This step is done to anticipate the level of sensitivity that is not uniform among the hydrophones that are used for the measurement of underwater acoustic signals. In the frequency domain shows that the ratio of the sensitivity ratio of hydrophones are fluctuates. From the results of this study showed that between hydrophones H_123 picture with hydrophones H-2 have an average ratio sensitivity of 0.85. The sensitivity ratio between hydrophones H-123 with hydrophones H-ref is 1.5 and between H-2 and H-ref was 2.3. From this study, we expected to be used as a reference in the preparation of the measuring instrument prior to the implementation of an experimental measurement in the field, so that the condition of the sensitivity of the measuring instrument is not uniform can be anticipated.

## ACKNOWLEDGEMENT

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**The answer for reviewer, Major weakness and Suggestion:**

1. Failed to set up the benchmark in analysis and measurement of underwater acoustic signals.

*The Answer is described in Introduction.*

Ambient noise generally appears continually at a certain location. This can corrupt the signal emitted by underwater acoustic equipment. So, the detection of background noise is important to get the signal-to-Noise ratio (SNR) of acoustical based underwater instrumentation. Ambient noise determines the baseline sound scape, which in general is background noise that appears in a location at sea at certain time. Practically, the ambient noise is associated with self noise, like the flow of water around the sonar. At the sonar system processing, background noise, has a similarity in the time, locations, and dept, where the signal is to be detected [1] and [2].

Research of wind effects on ambient noise have been carried out [3]-[5]. Generally, the research is related to the effects of wind and ship movement on acoustic noise that appeared. The discussion is focused on the observation of noise level and its relationship with the wind speed, other effects of weather on noise, and other noise from sea traffic. In the measurement process, that involving some of measuring instruments, the equipments should have a same sensitivity. In real conditions, researchers are often faced with the condition of the equipment which has not the same sensitivity. To overcome these problems required a uniformity of sensitivity and a re-calibration of the instruments. So that the sensitivity difference of the measuring instruments can be anticipated.

2. Expecting to describe use of Matlab for analysis.

*The Answer is described in Sub Section 3.1.*

Signal in each frame is transformed into the frequency domain by utilizing the periodogram function, which gives an overview of the estimated power spectral density (PSD). If $x(t)$ is a discrete time signal $\{x(t); t = 0, \underline{+}1, \underline{+}2, \ldots\}$ is assumed to be a sequence of random variables. The covariance of $x(t)$ is defined as:

$$r(k) = E\{x(t)x^*(t-k)\}. \tag{xx}$$

Where, $E\{\cdot\}$ denotes the expectation operator which averages over the ensemble of realization. The value of $x(t)$ is assumed to depend only on the lag between the two samples averaged. Variable $x^*(t-k)$ is complex conjugate of $x(t)$ with shifted version. The PSD is defined as discrete time Fourier transform (DTFT) of the covariance sequence

$$\phi(\omega) = \sum_{k=-\infty}^{\infty} r(k) e^{-j\omega k} \tag{xx}$$

3. More focus is given on experimental setup rather than signal analysis.
*The Answer is described in Subsection 2.1.*

Underwater speaker as a sound source was placed in the bottom of a box, with a radiation pattern directed to a surface. One hydrophone is placed in the surface position, with the sensor directed to bottom, face-to face with UW speaker. The hydrophone has the following specifications: frequency ranges of 1Hz ~ 100kHz, a sensitivity of −190 dB re: 1V/μP a(+4dB, 20Hz − 4kHz), and internal capacitance of 25nF. The capturing pattern of hydrophone is almost omni directional, having the strongest response for the incoming signal with an angle of $0^o$ to the perpendicular surface to hydrophones. The response will be reduced when the signal arrives with an angle ≥ $10^o$, and the attenuation up to 0.8 relative to the strongest signal ($0^o$).

4. Comparison of the result with referred paper will be creditable.
*The Answer is described in Subsection 3.2.*

The measurements result of ambient noise that has been in normalization has been compared with the results on the reference paper [4] and [5]. The result is not exactly the same, but in general has a similarity pattern.