Development of Noise Source Detection System using Array Microphone in Power Plant Equipment

배열형 음향센서를 이용한 발전설비 소음원 탐지시스템 개발

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Abstract

In this study, it has been initiated to investigate the specific abnormal vibration signal that has been captured in the power equipment. Array Microphone can be used in order to detect the direction and the position of the noise source. It is possible to track the abnormal mechanical noise in the power plant by utilizing the program and the microphone array system developed from this research. Array microphone system can be operated as a constant monitoring system

Keywords: Microphone Array, Source Detection, Acoustics, Noise, Vibration

I. INTRODUCTION

In this study, it has been initiated to investigate the specific abnormal vibration signal that has been captured in the power equipment. If there is an abnormal state of power equipment, abnormal noise happens to occur. But under noisy environment it is hard to know the location of noise source. The noise source has own frequency according to abnormal state. It is important to detect the noise source according to frequency in order to know the cause and the location of the problem [1].

It can be detectable for the direction and the position of the noise source by the array method using a large number of microphones. The use of the array microphone is indispensable for any signal source direction and the measurement of spatial position information. It is typically the human sensory organs, such as the eyes, the ears. Spatial information is measured by using the time delay and the amplitude difference generated by the path difference from the radiation source. Therefore, if one of the sensory organs is damaged, it becomes impossible to obtain a spatial position and directional information. It is possible to consider the application of the array microphone techniques for detecting the spatial position of the noise source. This technique is the one such as radar to detect enemy aircraft by using electromagnetic waves. It has the same principle as sonar by using sound waves to be used in submarines. As the cost of a microphone and signal collection equipment becomes lower, industrial use of these microphone arrays becomes increasing [1].

In this study, in order to verify the availability and usefulness of a sound source detection using these microphone arrays, we have performed a basic experiment in combustion facilities and anechoic chamber and developed a program that is able to track the location of the noise source in the power plant. It is possible to track the abnormal mechanical noise in the power plant by utilizing the program and the microphone array system developed from this research. Array microphone system can be operated as a constant monitoring system.

II. DETECTION OF NOISE SOURCE

It is possible to consider the application of the array microphone techniques for detecting the spatial position of the noise source. The principle of the array microphone can be explained from Fig. 1.

As shown in Fig. 2, if the sound source is located far, it can be assumed to be a plane wave. It can define the angle of incidence of the sound wave. The sound wave is incident on the microphone array at a constant angle as follows.

Here the angle between the wave and the microphone array is determined as the incident angle. Therefore, the sound pressure signal that is measured for each microphone can be readily understood to have a constant time delay. Since these delay times are determined according to the distance and the incident angle between the microphone and the wave. If it is possible to measure the delay time accurately, it is possible to estimate the future incidence angle. The distance between noise source and microphone can be calculated from the relationship between the wave transmission velocity and the angle of incidence. The direction and distance of noise source can be calculated without rotating the microphone array by using a virtually rotation effect with the delay time. And this method is called as Delay & Sum. Fig. 1 and 2 represent these Delay & Sum technique [2].

In order to consider detail mathematically, the difference between the sound wave transmission paths between the microphone can be shown in Fig. 4 [3].

If we assume to be constant spacing between the microphones, the first microphone signal with the reference microphone signal, the relationship between the second microphone signal is as follows.

$$y_1(t) = y_0(t-\tau),\ y_2(t) = y_0(t-2\tau) \tag{1}$$

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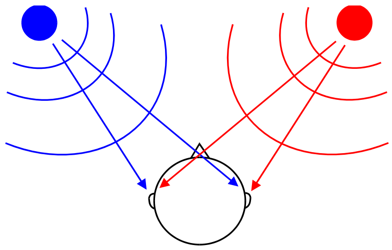


Fig. 1. Amplitude difference according to the path between sound generator and receiver

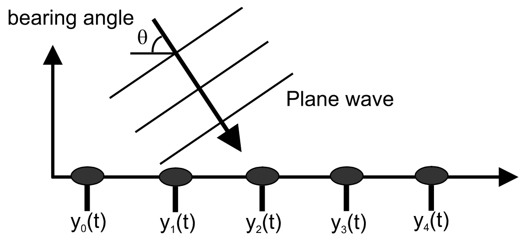


Fig. 2. The geometric relationship between the incident angle and the microphone in the plane wave model

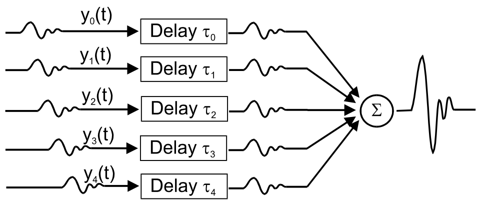


Fig. 3. Delay & Sum technique and calculation of output

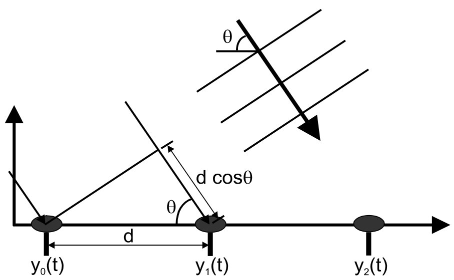


Fig. 4. The relationship between the transmission path of the plane wave and the distance between the microphones

Here, τ is defined the time delay between the microphone signals and can be calculated as $\tau = \frac{d\cos\theta}{c}$ from the acoustic velocity and path distance. Accordingly, by superimposing a signal to a time delay corresponding to the incident angle to be estimated from the measurement signals of the microphones, by comparing the power, we can find the angle of incidence. Considering in this frequency range for each frequency, first the signal of the microphone is the reference microphone signal is first, it will have the following relationship [4].

$$y_1(t) = y_0(t)e^{-i2\pi f_0 \tau} = y_0(t)e^{-i2\pi \frac{f_0 \cos\theta}{c}} \quad (1)$$

In addition, the relationship between the input signals of the other microphones can be determined as well.

$$y_n(t) = y_0(t)e^{-i2\pi n d_0 \cos\theta} \quad (2)$$

Eventually it is possible to calculate the power by adding

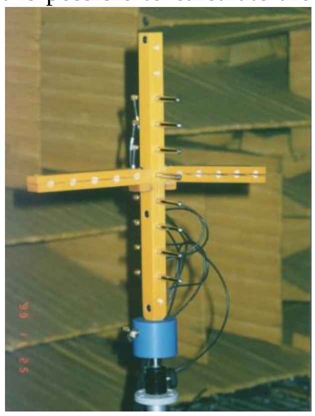


Fig. 5. Cross microphone antenna

Table. 1. The relationship between the transmission path of the plane wave and the distance between the microphones

Part	Function	Quantity
ICP Microphone	Convert to Voltage	16
ICP Amplifier	Amplify Signal	2
AD Board	Convert to Digital	1
PC	Signal Analyzer	1

the corrected time delay of the estimated angle for the input signals of all microphones. If it matches the angle at which the estimated angle is actually incident is as power is able to estimate the incident angle so greatest.

$$z(t) = \sum_{n=0}^{N-1} y_n(t)e^{-i2\pi n d_0 \cos\theta} = N y_0(t) \quad (3)$$

It is possible to know that the Fourier transform of the space transform of N spatial data to the frequency of the space. It calculates the power through the above process and becomes possible to estimate the position of the sound source. It is important to utilizing a finite number of measurement points on the space, since discrete Fourier transform can be seen the leakage problem and the resolution is determined from the space on the measurement [5][6].

It is confirmed the leakage in a different angle from the spatial resolution of the near 0 degree. It is possible to improve resolutions by reducing side lobe so as to increase the number of microphones.

Beamforming technique is well known as basis for the position detection of the noise source. MUSIC (MUltiple SIgnal Classification) method has been conducted to improve the resolution by using the microphone of the same number. However, when considering the general results robustness, a beam-forming technique is the most common [7][8].

The estimation of only the angle of incidence is converted into a spatial position estimation which is a method of using assuming a wave model as a point sound source radiation model. In many industrial applications using a microphone array, near-field measurements are mostly used with sphere wave model.

III. SYSTEM CONFIGURATION

A. System Configuration

This system consists of AD board, ICP amplifier, ICP microphone and PC. The acoustic signal from ICP microphone amplified at ICP amplifier and then is converted digital signal using AD board. The specification of system is as follows in Table 1. Microphone antenna is shaped as Fig. 5.

B. description of the program and use

The sound source position tracking system using the microphone array is described above. It calculates the position of the noise source from sound pressure measured by the microphone. The program is created based on Matlab. The inputs and output variables of the microphone array by mapping the beam-forming power of the sound source plane, would be to express the position of the noise source.

The first consider the frequency range of the noise source of interest, and calculates the wavelength with respect to this. Wavelength from the relationship between the frequency and the speed of sound, is easily determined in the following equation.

$$\lambda = \frac{c}{f} \tag{1}$$

Spacing between the microphones must be smaller than half of the wavelength calculated by considering the aliasing in space. Therefore, in consideration of the maximum frequency of interest, the distance between the microphones should be determined. We arranged a microphone to the + shaped at this interval. In this case, in order to determine the spatial position of each microphone, the program user can enter the height from the bottom surface of the center of the array. For the direction parallel to the floor surface in the x-axis, and the vertical direction to set the coordinate system in the y-axis, the center position of the array is (0, h). Spatial coordinates of each microphone turn is determined on the basis of the coordinate system is determined. This is an input, to measure the sound pressure. Area for detecting the position of the sound source, which can be a three-dimensional space, in the general case, to detect the position of a sound source in a random-determined sound source plane, and thus to output. Accordingly, a plane parallel to the array surface defined by the sound source plane in the program, and is adapted to provide the distance between both surfaces as input. Also, it is to be given an area for detecting the position of the sound source in the sound source plane, but to input the size of the x-axis and y-axis directions. Also, it wants to enter the lattice size of the sound source plane to calculate the position of the sound source. Entering such an input variable in the program, the program, the distance between the lattices of the microphones and the sound source surface is calculated, it becomes possible to calculate the power from the measured sound pressure by utilizing this. After entering the configuration variables of the array and the sound source plane and read the information of the sound pressure measured from the microphone. Sound pressure of the microphone, frequency and the reference microphone to attempt to calculate - generally No. 1 microphone - relative values of other microphone of the sound pressure signal, ie reads the transfer function. Such input data, as described above, calculates the power at each lattice point of the sound source plane, by two-dimensional mapping. This will be output. In this figure, the output and thus to determine to a large point, that is the power at which appears

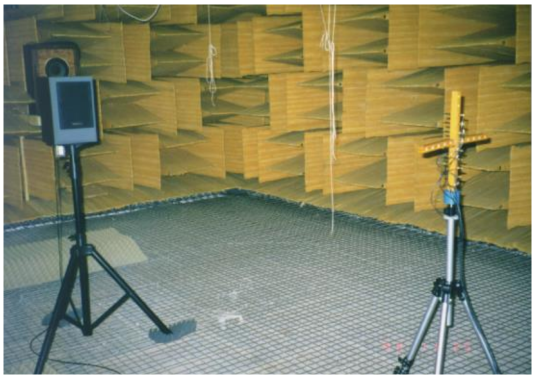


Fig. 6. Microphone array and speakers in anechoic chamber

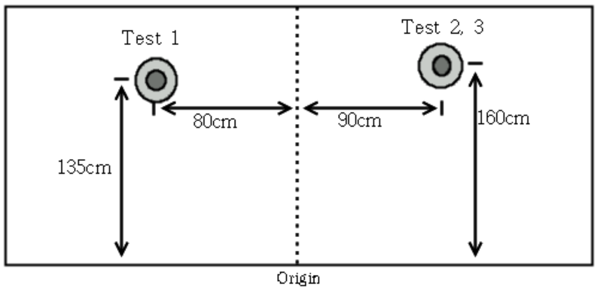


Fig. 7. Position of the noise source

largely sound source position. Such series of operations are run by a program created with such attachment 1 Matlab, the user, to execute the program, it may be input variables and sound pressure data. If continuous research future takes place, directly processed microphone measures sound pressure signal in addition to the development of a transfer device the gap adjustment can be automatically carried out between the microphones for measuring a desired frequency range, it can be expected improved program than outputs the result. Moreover, by complementing the GUI, it is possible to achieve a user experience.

IV. RESULT

Experiments were carried out divided into two major. First, it conducted experiments using a speaker in an anechoic chamber following, experiments were conducted in a real environment and a similar power generation facilities. At this time, the mic used were 8, respectively by measuring the sound pressure signal of the vertical and horizontal axes, and consequently uses the cross microphone antennas. Microphone arrays used were as follows photo. In addition, sound pressure signal measured from the microphone via the signal conditioner, which is inputted to the collection and processing unit of the multi-channel signal. Then, sound pressure signal that is collected in this way is to calculate the power to estimate the position of the noise source through a computation by a computer, and outputs the result.

A. Position investigation experiment of the noise source in anechoic chamber

Spacing between the microphone and the 3.5 cm in consideration of 3 kHz is a frequency component of the noise,

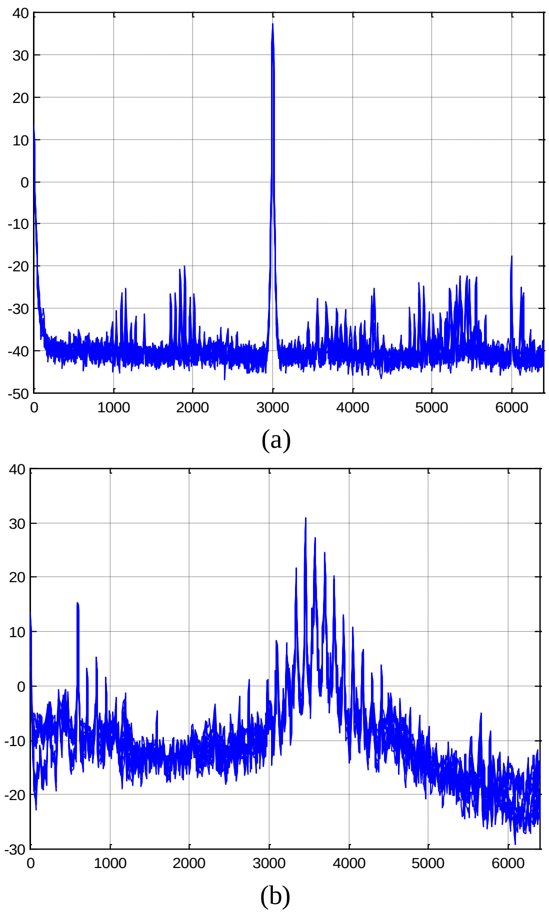


Fig. 8. Frequency characteristics of the noise source. (a) The spectrum of pure tone 3 kHz. (b) The spectrum of abnormal noise

was placed as shown in the photograph. In the case of the anechoic chamber experiments, the height of the center of the array is 120 cm from the floor, the distance between the array surface and the sound source plane was 150 cm. As the sound source, we have been using the omni-directional speaker, if the experiment 1 as shown in Fig. 6, the position of the speaker, is placed from the center of the array the remaining 90 cm, a height 135 cm, 3 kHz Tan It was emits one sound. In the case of Experiment 2, the right 80 cm speaker, and use a single tone of 3 kHz was placed in the height of 165 cm, in the case of Experiment 3, the position of the speaker, place in the same manner as in Experiment 1 Recording Hugh more than you were spinning. Fig. 7 shows the device was microphone array and the speaker of the anechoic chamber.

In this case, the frequency characteristics of the sound pressure signal which is measured from the reference microphone is a pure tone 3 kHz. The position of the pure tone and the abnormal noise of actual mechanical equipment is shown in Fig.8.

The results of Test 1,2 analyzed by emitting a single tone and abnormal noise are the same as Fig. 9 and Fig. 10. When compared with the sound source position in Fig. 8, anechoic chamber condition can be predicted for the position of the sound source very well.

From this result, the position investigation of the sound source using the microphone array can be confirmed that it is accurately performed in an anechoic chamber.

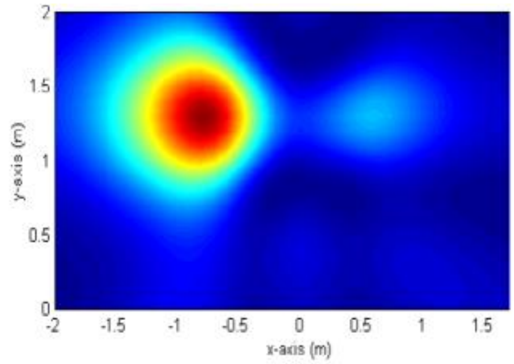


Fig. 9. Measurement results of the test 2

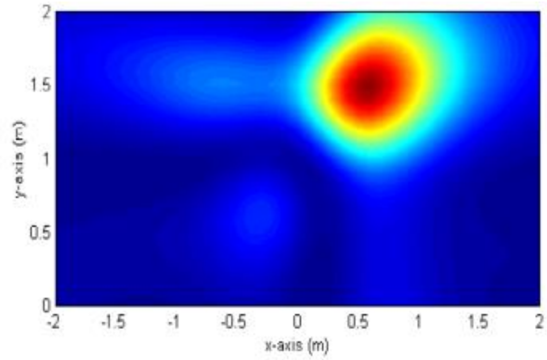


Fig. 10. Measurement results of the test 2

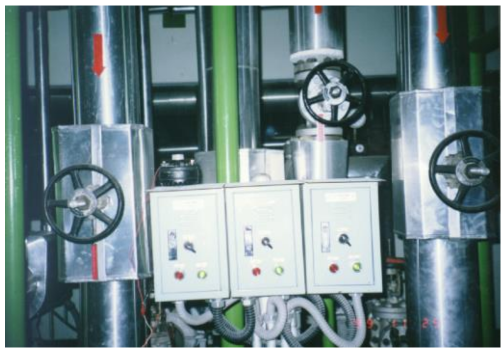


Fig. 11. Experiment using a speaker in the reverberation condition

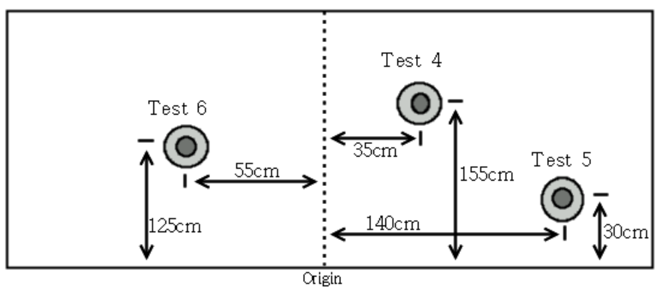
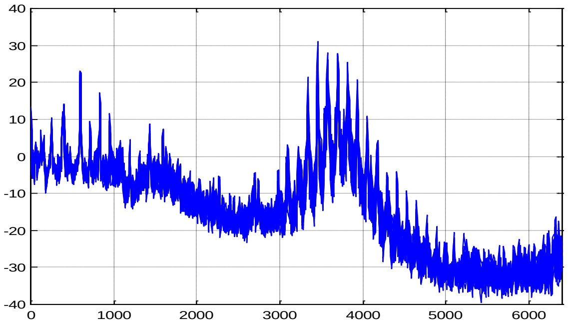


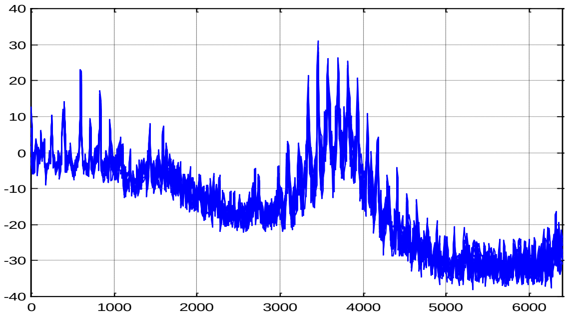
Fig. 12. Position of the noise source

B. position investigation experiment of the noise source in reverberation conditions

The following were performed position detection experiment of the noise source in the plant which is similar to the actual situation reflected waves there are many. Experiments, but the use of combustion equipment, the experimental apparatus, the same microphone array and a signal conditioner,



(a)



(b)

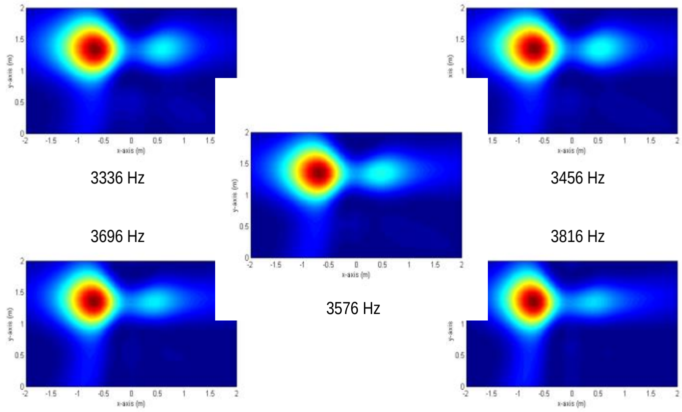
Fig. 13. Frequency characteristics from noise in power plant. (a) the frequency characteristics of the sound pressure signal which is measured in Test 4. (b) the frequency characteristic of the sound pressure signal measured in Test 5

and use the signal acquisition device, and the sound source, and using a horn driver emit unusual mechanical noise. Experimental systems and configurations are as in Fig. 12. Sound source, photographs have conducted experiments to be positioned in three positions, respectively, it shows an example. The center position of the microphone array, weave height of 135 cm from the floor, the distance between the measurement surface and the sound source plane was 300 cm. As shown in Fig. 13, in Experiment 4, there is the position of the sound source is from the center of the microphone array 35 cm to the right, to the height 155 cm, in Experiment 5, right 140 cm, a height 30 cm, in Experiment 6 , the remaining 55 cm, it was placed in height 125 cm. For this case the experiment 6, the sound source is placed in the pipe back in position behind the 130 cm from the sound source plane. Then, the hue or that has been recorded and is emitted, after collecting a sound by using a microphone array, and interprets it to track the location of the sound source.

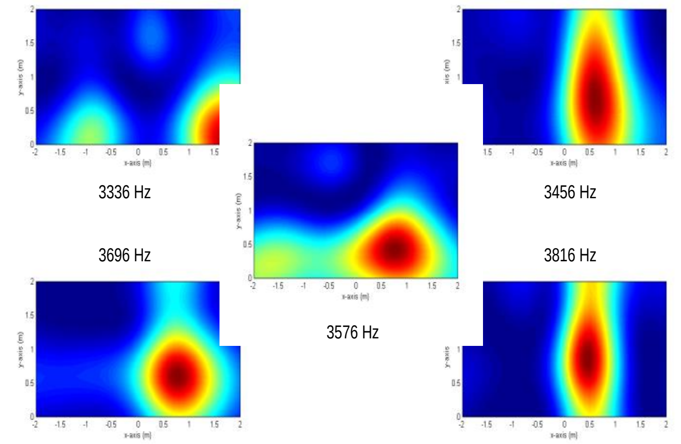
The following figure shows the frequency spectrum of the sound pressure signal measured by the reference microphone. We were able to Hugh of frequency characteristics to confirm a well-observed or larger.

The position of the noise source in test 4 and 5 was obtained as 16 and 17. Frequency shown in the Fig. 13 has the peak required to investigate the noise source.

It can be seen that the position investigation of the sound source using the microphone array can be confirmed from the experimental results. However, tracking capability of the noise source position as compared to the results of test carried out in an anechoic chamber is observed relatively to decrease due to



(a)



(b)

Fig. 14. Noise source analysis results of the experiments. (a) the frequency characteristics of the sound pressure signal which is measured in Test 4. (b) the frequency characteristic of the sound pressure signal measured in Test 5

reflected waves. Impact actual tracking capability the location of the noise source in situations acting greatly that it is possible to know the relatively poor, if the influence of the reflected wave is large, it is to infer the fact that errors are present it can.

For experiments 6, in situations where direct sound pipe is present before it is not transmitted of the noise source, to give the results shown in Fig. 14. Thus, as the device noise source to the back tubing, no sound directly, if the reflected sound is large, a plurality of reflected sounds is incident from a plurality of directions, in such a case, an accurate noise sources connecting can know the fact that the position investigation of difficulty. Therefore, it is necessary to carefully consider the influence of the reflected wave in the actual experiment.

Despite these reflected waves effects throughout the experiment, it was possible to confirm the availability of the microphone array in order to ascertain the position of the noise source.

V. CONCLUSION

Array Microphone System can detect an abnormal indication generated in complex mechanical equipment in order to find out the exact location and cause. We conducted experiments which consist of anechoic conditions and reverberant conditions in order to apply array microphone to power plant equipment. We also improved accuracy of the position investigation by considering the power plant environment. The exact size and phase of the array microphone was measured.

- The prediction of the position of the sound source is very well in the conditions of no anechoic chamber.

- Sound source position prediction in reverberation condition can be found by tracking relatively maximum peak frequency in comparison with the results of experiments conducted in anechoic chamber.

- In situations where sound is transmitted directly, the position investigation of the precise noise source is difficult due to incident plurality of reflected sounds. At actual application, it must be analyzed in consideration of the reflected wave.

The position investigation using array microphone in power plant needs to consider the reflected wave with a plurality of locations. It is possible to track the abnormal mechanical noise in the power plant by utilizing the program and the microphone array system developed from this research. Array microphone system can be operated as a constant monitoring system.

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