Rotating equipment fault noise location based on adaptive beamforming algorithm

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Abstract. Rotating machinery often produces continuous impact during operation due to the change of load and speed, which shows the characteristics of unsteady state and time-varying. Its working state cannot be comprehensively judged by a single vibration state parameter. Therefore, this paper proposes to use acoustic sensors to collect the fault noise signal of rotating machinery, and use the whole column of sensors to detect the fault noise signal. Based on the microphone array, this paper studies the adaptive beamforming algorithm (MVDR) to locate the fault source of rotating machinery in space. The effect of fault source location is verified by simulation and equipment measurement experiments. The acoustic sensor does not contact with the equipment, which will not damage the generator set, but also provide more effective information for fault source location and fault diagnosis and analysis.

1. Introduction
Array signal processing technology was applied in the 1930s[1]. It was first applied in the field of radar detection and achieved remarkable results. After decades of development, array signal processing technology has formed a relatively perfect system. Array signal processing technology has become the basis of modern signal processing technology[2-3]. Therefore, it is of great significance to locate mechanical equipment faults through array signals[4-5]. For example, Hou et al, used holographic array to measure azimuth acoustic signal, directly obtained the amplitude distribution of two-dimensional sound field sound pressure at the holographic measurement position, and reconstructed the two-dimensional sound field information of azimuth sound source position by using NAH technology[6-7]. To sum up, with the development of positioning technology, the purpose and significance of this paper is to use microphone array to study and improve the sound source enhancement algorithm based on adaptive beamforming, so that the improved algorithm has stronger noise suppression ability[8-9].

2. Sound source localization based on microphone array

2.1. Sound source and microphone array model
The sound source localization should consider the far-field and near-field model according to the nature of the sound wave. The distance between the sensor and the sound source and the wavelength of the main sound source affect the selection of the model. The far-field model and the near-field model represent two different geometric calculation models.
When the sound source signal is a near-field model, the sound propagates outward in the form of spherical waves. If the sound source is a far-field source model, then the sound source can be regarded as a plane wave.

Compared with the near-field model, the far-field model can be regarded as a kind of near-field model, which is less difficult to calculate. The difference caused by the amplitude attenuation in the sound propagation process can be ignored. In general, when the sound source is close to the microphone array, the incident angle of the signal received by each microphone array element is different, and a more accurate sound position is required. At this time, the near-field model is used. In most cases, the far-field model is used.

The choice of microphone array is also an important factor that affects the localizing accuracy. The topological structure of the array is one-dimensional, two-dimensional and three-dimensional, with common wire arrays, circular arrays, and spherical arrays. Compared with the linear array and the ball array, the circular array has a higher localizing accuracy than the linear array and less calculation than the ball array. In this paper, the circular array is selected as the microphone array for sound source localization.

2.2. Sound source localization algorithm

Sound source localization algorithm is an important part of real-time fault localization. This paper uses beamforming algorithm and MUSIC localization algorithm.

The basic principle of the beamforming algorithm after beamforming the signal received by the microphone, the output data is represented by $y(t)$:

$$y(t) = w^H x(t) = w^H \alpha(\theta, \phi)s(t) + w^H u(t)$$

(1)

The MUSIC localization algorithm is a high-resolution azimuth spectrum estimation algorithm, which overcomes the defects of beamforming and is superior to beamforming algorithm in localization accuracy. But it cannot overcome two or more sound sources that are closer together in the same direction. In recent years, the various high-resolution azimuth estimation methods that have been continuously improved theoretically overcome the limitations of the Rayleigh criterion.

2.3. Simulation experiment results and analysis

Comparing the performance of the adaptive beamforming algorithm and the MUSIC algorithm, this section compares and analyzes the differences between the two methods through simulation. First of all, this paper simulates a linear array and simulates the incident angle of a spatial sound source. Set the incident angle of the sound source to $10^\circ$, $30^\circ$, $60^\circ$, and the microphone array to be a linear array with 10 elements. The simulated signal-to-noise ratio of the array is 10, and the ratio between the radius and the wavelength is 0.5. The obtained angle and amplitude comparison chart is shown in the figure 1 below:

Fig 1. Localization algorithm performance comparison chart
It can be seen from the figure that both algorithms can locate the angle of incidence, but compared to the beamforming algorithm, the MUSIC algorithm has a higher resolution. But its array error has a great influence on its localizing accuracy, and the sound source and noise are required to be statistically stable. The ability to deal with interference sounds such as reflections and noises in a complex environment is weak, and high sound source signal energy is required for long-distance sound source localizing. The beamforming accuracy is not as high as that of the MUSIC localization algorithm, but the direction of arrival can be enhanced to suppress noise in other directions.

In addition, the localizing effect is also affected by the number of array microphones and the size of the array aperture. This paper uses a circular array and uses an adaptive beamforming localization algorithm. As the number of element microphones of the circular array increases, the effective time delay of the entire system will increase, and the localizing accuracy will also increase, but the amount of calculation will also increase. The number of microphones in the array can be reasonably and effectively selected within the range allowed by the hardware. When the sound source wavelength is set to 0.17m, the localizing effects of the three arrays with the number of array elements of 16, 32, and 64 are shown in Figure 2 below:

![Fig 2. The localizing effect diagram of the different number of array elements](image1)

In addition, the parameter of the array aperture that affects the localizing accuracy is also the parameter. We simulated the actual situation and set the number of array microphones to 16 to explore the influence of the array aperture on the localizing accuracy of the azimuth angle. In this paper, we have selected three different array apertures of 1.36, 2.72, and 5.44 to observe the overall changes. The effect diagram is shown in Figure 3 below:

![Figure 3. Simulation renderings of array aperture and localizing accuracy](image2)
Since white noise is added to the simulated sound source, it can be clearly seen that there are side lobes in the localizing simulation, but it does not affect the localizing accuracy. The three-dimensional simulation is shown in Figure 4.

Fig 4. Capon localizing simulation result graph

It can be seen from the three-dimensional localizing simulation diagram that the peak of the sound source direction is significantly higher than other directions, and the localizing effect is better. In order to compare and select the appropriate algorithm, we compared the localizing effect of the MUSIC localization algorithm and the beamforming algorithm in the case of different positions of the spatial sound source in the simulation situation. The comparison error table is shown in Table 1:

| Calibrated incident angle | MUSIC algorithm measurement results | MUSIC algorithm measurement error | Beamforming measurement results | Beamforming measurement result measurement error |
|---------------------------|-----------------------------------|----------------------------------|--------------------------------|-------------------------------------------------|
| (20.21,13.65)            | (22,15)                           | (1.79,1.35)                      | (21,13)                        | (0.21,0.65)                                     |
| (35.5,22.8)              | (37,20)                           | (1.5,2.8)                        | (35,24)                        | (0.5,1.2)                                       |
| (48.2,36.2)              | (48,37)                           | (0.2,0.8)                        | (49,35)                        | (0.8,1.2)                                       |
| (61.8,50.6)              | (60,52)                           | (1.8,1.4)                        | (61,50)                        | (0.8,0.6)                                       |
| (76.2,60.4)              | (75,61)                           | (1.2,0.6)                        | (76,60)                        | (0.2,0.4)                                       |
| (82.5,76.58)             | (82,76)                           | (0.56,0.58)                      | (83,76)                        | (0.44,0.58)                                     |

3. Conclusion

In this paper, according to analysed and compared the difference of near-field sound and far-field sound, narrow-band and broadband signals. Comparing to the advantages and disadvantages of MUSIC positioning algorithm and beam formation algorithm, selecting adaptive beam formation algorithm as the positioning algorithm, and proposing improvement Comparing the positioning results of the improved algorithm and the traditional beam formation algorithm through simulation experiments, it is proved that the improved algorithm is better in accuracy.

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