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Impulse source localization with background noise in a reverberant environment by multiple sensors

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ABSTRACT

During vibration testing of space hardware, breaking sounds can be heard when cracks happen in the structure. However, much efforts are needed to determine the position of fault in a complex large-scale model such as a satellite with background noise in reverberant environment. This paper focuses on implementing non-contact localization of the sound sources which are attributed to structure failures. In order to achieve this goal, a system for fault detection by sound source localization with limited number of sensors under the moderate reverberant environment is established based on the TDOA technology. The low-quefrency filtering technique is adopted as a dereverberation pre-processing approach to alleviate the reverberation effects due to the echoes from the reflections of the laboratory. In order to reject the interference sound sources, two criteria, the geometrical criterion and the cyclical check criterion, are introduced. A series of experiments are conducted to verify the performance of this system. The results show that this system can localize the crack sources and hitting sources accurately in a short time.

Keywords: Impulse source localization, reverberant environment, TDOA

1. INTRODUCTION

During vibration testing with the input signal of sine-sweeping or random amplitude, breaking sounds can be heard when cracks happen in the structure. However, it is often a difficult task to localize the structure failure position in a complex large-scale model with background noise in reverberant environment. Consequently, it often requires much effort for engineers to check over the whole structure in order to determine the position of fault.

The current work aims at implementing non-contact localization of the sound sources which are attributed to structure failures. In order to achieve fault detection by sound source localization with limited number of sensors, the TDOA technique is adopted (1-3). Comparing with another classic sound source localization technique the beamforming method, the microphones required for implementing TDOA is much less. At least three omni-direction microphones are capable to obtain the location of a point source in 2-D case, which is often the case when the fault positions are considered on the surfaces of the space hardware. However, in the current study, one more microphone is employed for each surface to enhance the performance of the localization system against the interference sources and background noise.

By using the TDOA method, an impulse source can be localized accurately with four microphones for one surface of the test item in a short time in the free-field condition. However, when the environment is reverberant as in most application cases, the echoes due to the wall reflections will significantly degrade the performance of the TDOA method. In order to overcome this problem, the homomorphic filtering technique is adopted to reduce the reverberation effect. This technique eliminates the reverberation contributions by decomposing the original measurement signals into the

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minimum-phase components and the all-pass components, and by applying low pass filtering in the quefrency domain on the minimum-phase components. This technique can effectively reduce the reverberation effects in the measurement signals and therefore make the TDOA method more robust in reverberant environments.

Following the methodology, a prototype of 16-channel structural failure localization system is established and tested on a satellite model. The experiments show that this system can localize the crack sources and hitting sources quickly and accurately.

2. Methodology

As a classic technique for sound source localization, the TDOA technique has been applied in many fields for its simple principle, less sensors and less computation amount comparing with other techniques. Many researches have been done on TDOA in recent years (4-7). The TDOA technique is utilized in the localization system established in the current work so as to reduce the complexity of the microphone array and to reduce the measurement channels and therefore the economic cost.

Consider the acoustic wave S_k(t) emitted from a point source S_k. The received acoustic signal by a microphone j is expressed as:

$$x_j(t) = \left(4\pi\left|\mathbf{r}_{jk}\right|\right)^{-1} \cdot s_k\left(t + \left|\mathbf{r}_{jk}\right|/c\right) + n_j(t) \quad j = 1, \cdots, M \tag{1}$$

where x_j(t) is the received signal by microphone j, $\mathbf{r}_{jk}$ is the vector from the location of the jth microphone to the source S_k, n_j(t) is background noise which is considered as incoherent with each other in each measurement channel, and c is the speed of sound in air.

For a microphone pair consisting the ith microphone and the jth microphone, the time difference of arrival (TDOA) from the source S_k to the two microphones is defined by the following equation:

$$\delta_{ij} = \left(\left|\mathbf{r}_{jk}\right| - \left|\mathbf{r}_{ik}\right|\right)/c \tag{2}$$

It can be seen from this equation that if the TDOA δ_{ij} is found out, the sound source S_k is then constrained on one branch of a hyperboloid determined by Equation (2) in the 3-D situation. A convenient way to obtain δ_{ij} is to evaluate the cross-correlation function (CCF) of the measurement signals of the microphone pair, which is given by:

$$R_{ij}(\tau) = \lim_{T\to\infty} \frac{1}{T} \int_{t=0}^{T} x_i(t) \cdot x_j(t+\tau) \tag{3}$$

Equation (3) is the definition of CCF for continuous signals. However, the signals are processed digitally by computers and therefore sequences are used instead of continuous functions. Consequently, the following equation take the place of Equation (3) in real applications.

$$R_{ij}[\tau] = \frac{1}{N_{sample}} \sum_{n=1}^{N_{sample}} x_i[n] \cdot x_j[n+\tau] \tag{4}$$

where N_{sample} is the number of total sampling points of the signals and τ is the possible time shift.

Equation (2) represents one branch of a hyperboloid with the focal points at the positions of the ith and the jth microphones. In order to determine the location of a point sound source in a 3-D field, at least four microphones that are not in the same plane are needed so that three microphone pairs can be formed. The hyperboloids obtained in each local coordinate system are then transformed into the global coordinate system by applying coordinate transformation. The position of the sound source is obtained at the intersection of the three corresponding hyperboloids.

In most real situations, the acoustic signal captured by a microphone contains not only the direct path signal from the sound source but also multiple echoes reflected by the surfaces of the reverberant environment as well as the incoherent noise. These echoes can be considered as a series of delayed and attenuated versions of the direct path signal. In such situations, the measurement signal can be expressed as the convolution of the source signal and the room impulse response function which describes how the echoes are delayed and attenuated.

The contribution of the reverberation effect can be reduced using the cepstral dereverberation method. To apply this technique, first, each measurement signal is transformed into its cepstrum (8).

Then, the cepstrum is decomposed into two parts, the minimum-phase part and the all-pass part. The former part is further low-pass filtered to suppress the contribution of the reverberation effects. Afterwards, the filtered minimum-phase part and the all-pass part are transformed into the frequency domain and multiplyed with each other to obtain the processed spectrum. And finally, a processed measurment signal is obtained by applying the inverse Fourier's transformation. This technique is adopted in the current work as a pre-processing method of the measurement signals. To get more details one can refer to Reference (9-11).

3. Some Consideration of the Structural Failure Localization System

3.1 Optimization of microphone arrangement

The TDOA method determines the position of a sound source according to the intersection of multiple hyperboloids which depend on the time difference of arrival between two microphones in each microphone pair. In order to localize a point source in a 3-D field, at least three microphone pairs are needed, that is, at least four microphones which are not in the same plane should be used.

However, in the current study, the sound sources of interest are all assumed on the surfaces of the test item. Typically, the profile of a satellite model can be simplified as a cube. For each side of the cube, the sound sources to be localized are within a known area, that is, the corresponding surface of the model. In such situation, two microphone pairs are enough to obtain two hyperboloids. The intersection line of the two hyperboloids meets the surface at the sound source location. Hence, at least three microphones on a plane parallel to the corresponding surface of the space hardware should be utilized.

In many real applications, however, there are often reverberation effects as well as some interference sound sources such as the sound of air conditioning system, the voices of workers. These interferences may lead to spurious peaks in the cross correlation functions of the microphone pairs and consequently lead to wrong intersection of the hyperboloids. In order to enhance the robustness of the localization system, another microphone is added in the sub-array for each surfaces of the test item. Therefore, six microphone pairs are formed and six hyperboloids are obtained using four microphones in each sub-array. In order to localize the structure failure sound sources on the four sides of a test item, a whole microphone array containing 16 microphones with four sub-arrays are established. The four microphones of each sub-array are placed near the four corners of the surface to capture the acoustic signals on each surface.

3.2 Rejection of the interference sound sources

In order to reject the interference sound sources and enhance the localization robustness, two criteria, the geometrical criterion and cyclical check, are introduced into the current localization system.

3.2.1 Geometrical criterion

In order to reject the undesired sound sources, a geometrical criterion is introduced. This criterion constrains the range of the cross correlation function in which the main peak position is considered as the TDOA of the sound source of interest. According to the geometrical relation between each microphone and the corresponding surface of the test item, the maximum possible TDOA for a sound source on this surface is the maximum one of the four corners (illustrated in Fig. 1). By comparing the four TDOAs on the four corners, the maximum possible TDOA $\delta_{ij,\max}$ is determined. Therefore, for the cross correlation function of the microphone pair containing Mic i and Mic j, only the highest peak within the range $-\left|\delta_{ij,\max}\right| \leq \delta_{ij} \leq \left|\delta_{ij,\max}\right|$ is considered.

By introducing this criterion, the peaks in the cross-correlation function formed by the undesired sound sources which are out of the surface of the satellite model will not be considered. Therefore, the interference sound sources are naturally rejected.

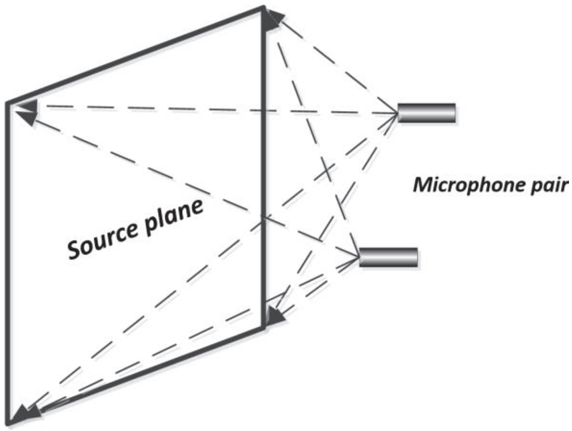

Figure 1 – Illustration of the geometrical criterion

3.2.2 Cyclical check

The TDOA method finds the source position by looking for the intersection point of the hyperboloids. Sometimes a pure quantity of measurement of one microphone will lead to a wrong TDOA and therefore a wrong hyperboloid. In this situation, the system will fail to localize the true source position even if the other two hyperboloids precisely correspond to the TDOAs of the microphone pairs.

In order to determine whether an intersection point is the true source position or not, a cyclical check procedure is introduced to check the time delay relations and to reject the wrong peaks in the cross-correlation functions of the measurement signals. The judgment is based on the following relation among the time delays of three microphone pairs that are formed by three microphones. When measuring the acoustic signal emitted from one single source s with two microphones Mic i and Mic j, the time delay of the two measurement signals are given by:

$$\delta_{ij}=\left(\left|\mathbf{r}_{jk}\right|-\left|\mathbf{r}_{ik}\right|\right)\Big/c=\tau_{kj}-\tau_{ki} \tag{5}$$

where $\tau_{kj}=\left|\mathbf{r}_{jk}\right|/c$ and $\tau_{ki}=\left|\mathbf{r}_{ik}\right|/c$ are the propagation time from the source S_k to Mic i and Mic j respectively.

Therefore, for any point source S_k and three microphones Mic i, Mic j and Mic n, the following relation exists:

$$\begin{aligned}\delta_{ij}&=\tau_{kj}-\tau_{ki}\\&=\left(\tau_{kj}-\tau_{kn}\right)-\left(\tau_{ki}-\tau_{kn}\right)\\&=\delta_{nj}-\delta_{ni}\end{aligned} \tag{6}$$

There are four microphones in each sub-array. Therefore, four three-microphone groups are formed. Those groups do not make Eq. (6) exist will not be used for localization.

4. Experimental validation

In order to validate the 16-Channel structure failure localization system, a series of experiments have been conducted.

4.1 Set-up of the experiment

The test set up is shown in Fig. 2. A series of cases are tested to localize impulse sources on the four vertical surfaces of a satellite model. These surfaces are numbered in such a way: Surface 1 is the front surface in Fig. 2, Surface 2 is the right surface facing the wall, Surface 3 is the back surface and Surface 4 is the left surface. Four microphone sub-arrays with four microphones in each one are set surrounding the model. Mic 1~4 facing Surface 1, Mic 5~8 facing Surface 2, Mic 9~12 facing Surface 3 and Mic 13~16 facing Surface 4. The distance from each surface of the model to the corresponding microphone array is 0.55 meter. The coordinate system is placed with the origin at the center of the first sub-array containing Mic 1~4. The x-axis lies on the horizontal plane pointing to the right when facing the model. The y-axis is placed horizontally pointing to the model, and the zaxis lies vertically pointing upwards to the ceiling. A series of cases are designed to verify the performance of the system.

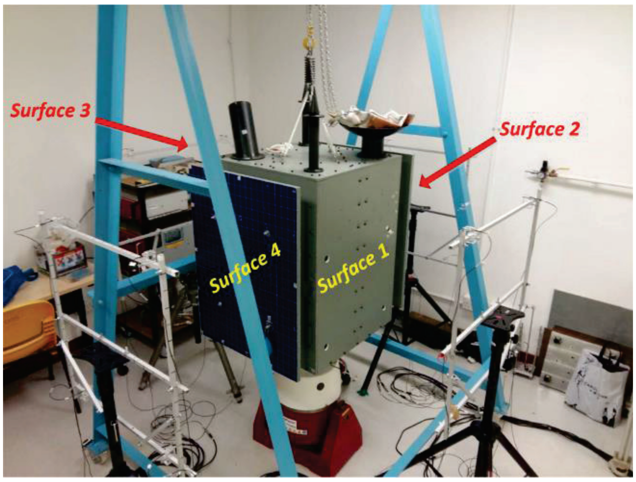

Figure 2 – The test set up

4.2 Cases and results

4.2.1 Case 1: Two sources on different surfaces

Two sound sources are designed in Case 1 to verify the performance of this system in localizing multiple sources in different surfaces. The first source S_1 on Surface 1 is generated by hitting the surface with a small hammer twice and S_2 is a small loudspeaker placed on Surface 4 playing a series of transient signals. The coordinates of the microphones as well as the sound sources are listed in Table 1.

Table 1 – The coordinates of the microphones and the sources in Case 1

Mic 1	Mic 2	Mic 3	Mic 4	S_1
(-0.25, 0, 0.33)	(0.25, 0, 0.33)	(0.25, 0, -0.33)	(-0.25, 0, -0.33)	(-0.1, 0.5, -0.25)
Mic 13	Mic 14	Mic 15	Mic 16	S_2
(-0.92, 1.05, 0.33)	(-0.92, 0.55, 0.33)	(-0.92, 0.55, -0.33)	(-0.92, 1.05, -0.33)	(-0.42, 0.6, -0.15)

The measurement signals of Mic 1 and Mic 16 are shown in Fig. 3 (a) and (b) respectively. In Fig. 3, the low-frequency noise has been eliminated by a high-pass filter. The two measurement signals both contain the signals from both sources. The five transient signals which occur periodically every one second are emitted from S_2 on Surface 4 while the other two impulses are the hitting sound of S_1 on Surface 1. Although S_1 is on Surface 1 and Mic 16 is facing Surface 4, the amplitude of the signal from S_1 is still larger than that from S_2 in the measurement signal of Mic 16 according to Fig. 3(b).

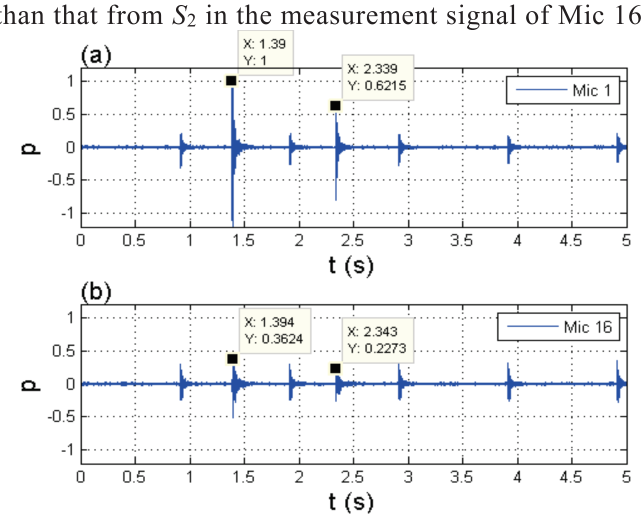

Figure 3 – The measurement signals of: (a) Mic 1, (b) Mic 16

By default, the system extracts the largest peak which occurs at around 0.92s for localization. This peak is the first transient signal emitted from S_2. The localization results are shown in Fig. 4. S_2 is localized at (-0.41, 0.62, -0.17) by Mic 14, Mic 15 and Mic 16. There is no localization result on any other surface of the test item. S_1 is not found because the extracted time block only contains the signal from S_2.

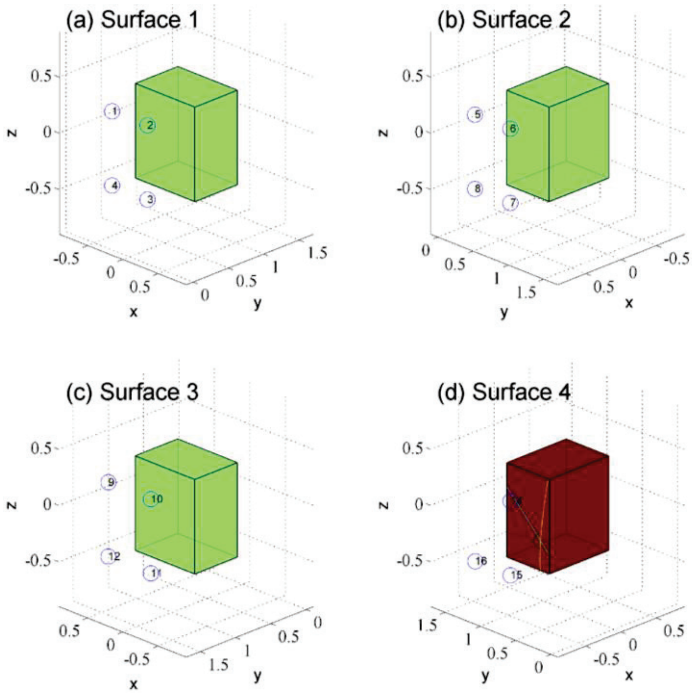

Figure 4 – The localization result for S_2

In order to localize S_1, the time-domain signal block at around 1.5s with the duration of 0.3s is extracted. The localization result is shown in Fig. 5. In this configuration, S_1 on Surface 1 is localized at (-0.07, 0.5, -0.25) and no miscarriage of justice is made on other surfaces.. The localization errors for S_1 and S_2 are both 0.03m.

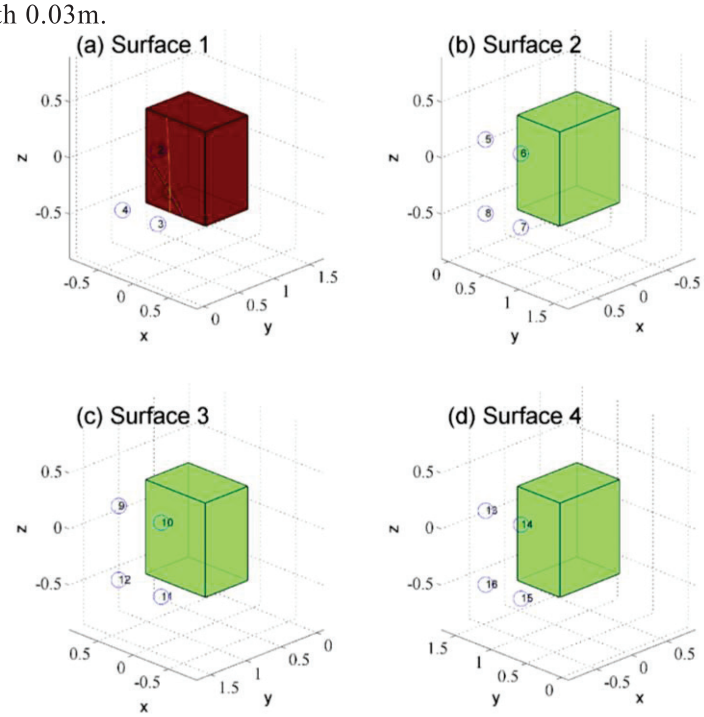

Figure 5 – The localization result for S_1

4.2.2 Case 2: Two sources on the same surface

Two sound sources are both placed on Surface 1 in Case 2 to verify the capability of this system in finding multiple sources on the same surface. S_1 is generated on Surface 1 by hitting the surface with a hammer and S_2 is a loudspeaker placed on the same surface playing a series of transient sounds with the interval of one second. The coordinates of S_1 and S_2 are (0.15, 0.5, 0.2) and (-0.22, 0.5, -0.2) respectively.

By default, the sound source that is localized first is S_1 which corresponds to the highest peak at around 2.12s. The localization result is shown in Fig. 6. S_1 is found at (0.16, 0.5, 0.16) by the three-microphone group of Mic 1, Mic 3 and Mic 4.

In order to localize S_2, the signal segment extraction is reset. The center time is chosen at 2.8s and the time duration is set of 0.3s. The localization results are shown in Fig. 7. In this situation, S_2 is localized by Mic 1 Mic 2 and Mic 4 at (-0.21, 0.49, -0.17). No miscarriage of justice is made on other surfaces. The localization errors for S_1 and S_2 are 0.0412m and 0.0332m respectively.

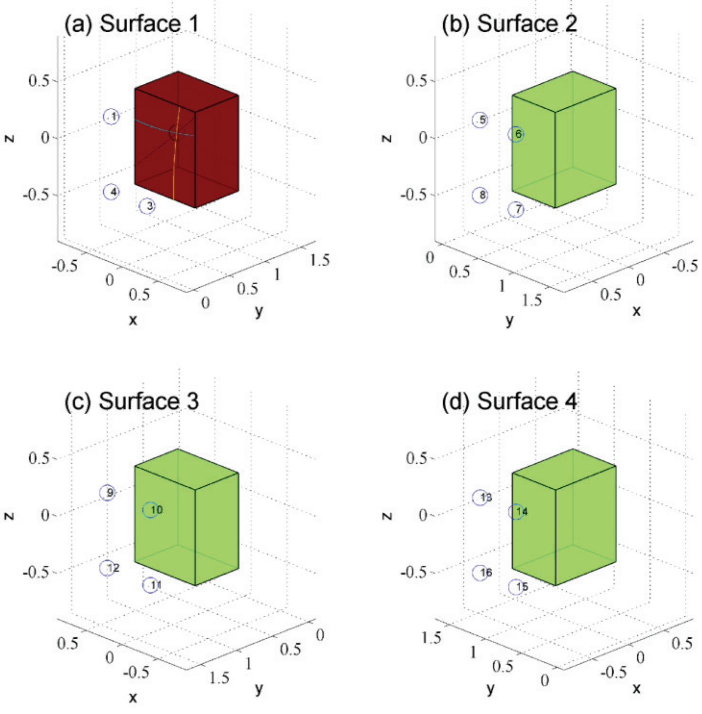

Figure 6 – The localization result for S_1

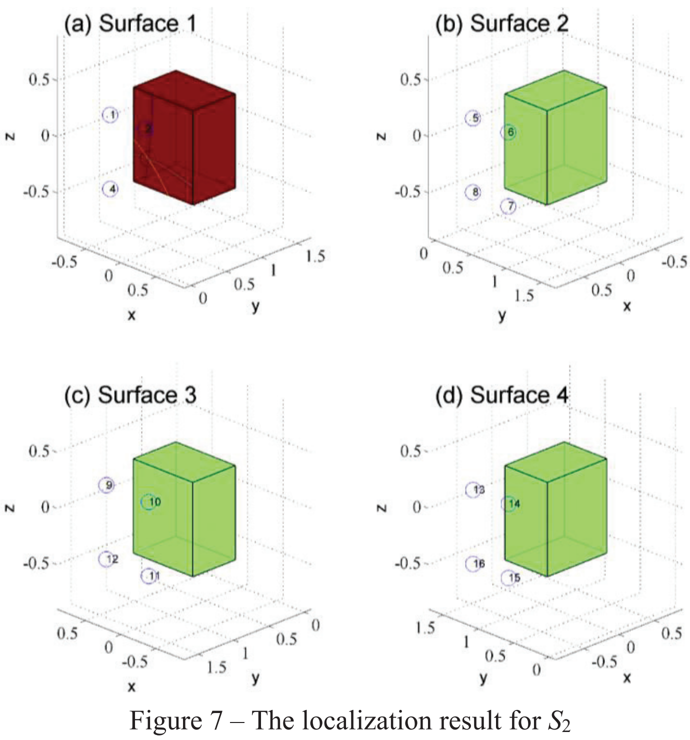

Figure 7 – The localization result for S_2

5. CONCLUSIONS

This paper aims at establishing a non-contact structure failure diagnosis system by localizing the impulse sound sources corresponding to the potential structural failures. The CCF method of TDOA technique is adopted. In order to alleviate the reverberation effect due to the echoes from the reflections, the low-quefrency filtering technique is utilized as a dereverberation pre-processing approach. By involving this technique, the robustness of the localization system is improved to a great extent.

A 16-channel structure failure localization system is established thereafter. In order to reject the interference sound sources, two criteria, the geometrical criterion and the cyclical check criterion, are introduced. A series of experiments are conducted to verify the performance of this system. The results show that the system can localize multiple sources on different surfaces, and even multiple sources on the same surface of the test item accurately (the maximum localization error is about 4cm) in a short time (less than two minutes in this experiment). The localization results are shown both in the image format and as the coordinates to help users to find the locations of the structure failures easily.

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