A microphone position calibration method in a reverberant environment for a randomly distributed array

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ABSTRACT

A microphone position calibration method in a reverberant environment is investigated for randomly distributed large-size array in this paper. The microphone positions need to be known exactly in order to localize sound sources. Time delay estimation (TDE) is an important step in calibrating the microphone positions using a few calibrating sound sources for a randomly distributed array. However, TDE method degrades severely in a reverberant environment which can introduce serious position errors. The microphone position error analysis is made due to misleading estimation of time delay. In this paper, impulsive sound source as a calibrating source is investigated compared with chirp source and white noise. The direct sound from an impulsive sound is isolated to calculate exact time delay in a reverberant environment while early reflection and reverberation sound are cut off. The simulation experiments are carried out to demonstrate that accurate microphone positions can be obtained using the proposed method. The proposed method is also applied in a project to calibrate the microphone positions for a spiral-like ceiling-mounted array with 64 elements and aperture of 3.5m, which proves the effectiveness.

Keywords: Position calibration, reverberant, distributed, microphone array I-INCE Classification of Subjects Number(s): 74.7

1. INTRODUCTION

Acoustic source localization has been a hot topic using microphone array in the last decades and widely applied in various fields including multimedia communication, audio conference(1,2), medical imaging, and machinery trouble shooting and diagnosis(3), etc. Utilizing localization techniques, an acoustic camera has been invented to visualize sound field by overlapping the captured optical image and the calculated sound field image represented by color map(4). The visualization of sound field allows to identify main emitting sources, which plays an important role in noise reduction. Numerous methods have been studied to localize acoustic sources such as CBF, MUSIC, and MVDR(5,6), etc. In most cases, the microphone positions are assumed to be known exactly. Otherwise, the position errors of microphone can degrade the performance of localization severely, especially for MUSIC and MVDR methods.

It is not trivial to determine microphone positions in machine health monitoring and noise sources localization in a industrial plant where microphones are randomly distributed and mounted conformally on the ceiling in order to construct a large-size array. The position calibration method need to be employed to calculate the exact microphone positions. Several methods are proposed to solve the problem (7-10), which are dealt with in the free field. In this paper, we investigate the microphone position calibration method in a closed industrial plant which is reverberant. The time delay estimation method is utilized in the calibrating microphone positions in most cases(11), but it introduces serious errors in a reverberant environment. Three kinds of calibrating sound sources are investigated in this paper. we propose a method that impulsive sound sources are used as calibration sources and the direct sound is truncated to improve time delay estimation.

This paper is organized as follows. In Sect.2, we propose an mathematic model to calibrate the

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microphone positions and make the error analysis by the Monte-Carlo simulations . In Sect.3, the calibration of microphone position in a reverberant environment is dealt with and three kind of calibrating sources are discussed including white noise, chirp signal and impulsive sound. The direct sound of the impulsive source is employed to improve the calibration precision in a reverberant environment. Experimental results are presented in Sect.4 and the conclusion are drawn in Sect.5.

2. FORMULATION

The computation of microphone positions of array is carried out in two steps. Firstly, the difference of distance from the calibrating sound source to the reference microphone in known locations and to an pending microphone is computed by general cross correlation method. Secondly, the computed difference of distance is employed to determine the positions of microphone according to the Euclidean distance.

2.1 Mathematic model

Given a set of M microphones in unknown locations, N calibrating sound sources and one reference microphone in known locations, which is illustrated in Figure 1. Let $\boldsymbol{S}_i=[x_{s_i},y_{s_i},z_{s_i}]^T$, $\boldsymbol{m}_j=[x_{m_j},y_{m_j},z_{m_j}]^T$ and $\boldsymbol{m}_r=[x_{m_r},y_{m_r},z_{m_r}]^T$ be location coordinates of the i^{th} calibrating sound source, j^{th} microphone in unknown locations and the reference microphone, respectively. The distance between the source $\boldsymbol{S}_i$ and the microphone $\boldsymbol{m}_j$ is defined as $d_{S_im_j}$. The distance between the source $\boldsymbol{S}_i$ and the reference microphone $\boldsymbol{m}_r$ is defined as $d_{S_im_r}$. The difference of distance is written as:

$$d_{ij}=d_{S_im_j}-d_{S_im_r} \tag{1}$$

where

$$d_{S_im_j}=\sqrt{(x_{s_i}-x_{m_j})^2+(y_{s_i}-y_{m_j})^2+(z_{s_i}-z_{m_j})^2} \tag{2}$$

$$d_{S_im_r}=\sqrt{(x_{s_i}-x_{m_r})^2+(y_{s_i}-y_{m_r})^2+(z_{s_i}-z_{m_r})^2} \tag{3}$$

The time delay estimation is defined as τ_{ij} using cross correlation method. The difference of distance can also be rewritten as:

$$d_{ij}=c\cdot\pi_{ij} \tag{4}$$

where c is the velocity of sound. It is well known that at least three sources are required to determine a position in the three-dimensional space. The equations become over-determined if the number of calibrating sound sources is larger than three. Then the positions of microphone are computed with a maximum likelihood to increase the precision.

$$\boldsymbol{d}_j=c\cdot\boldsymbol{\tau}_j \tag{5}$$

where $\boldsymbol{\pi}_j=[\tau_{1,j},\tau_{2,j},\cdots,\tau_{i,j},\cdots,\tau_{M,j}]^T$,and $\boldsymbol{d}_j$ is denoted as

$$\boldsymbol{d}_j=\begin{pmatrix} d_{1j}\\ d_{2j}\\ \vdots\\ d_{ij}\\ \vdots\\ d_{Mj}\end{pmatrix}=\begin{pmatrix} d_{S_1m_j}-d_{S_1m_r}\\ d_{S_2m_j}-d_{S_2m_r}\\ \vdots\\ d_{S_im_j}-d_{S_im_r}\\ \vdots\\ d_{S_Mm_j}-d_{S_Mm_r}\end{pmatrix} \tag{6}$$

The equation(5) is nonlinear and over-determined and the solution can be solved by seeking a minimum of cost function J which is written as

$$\min J_j=\min\left\|\boldsymbol{d}_j-c\cdot\boldsymbol{\tau}_j\right\|=\min(\sum_{i=1}^{N}(d_{S_im_j}-d_{S_im_r}-c\cdot\pi_{ij})^2) \tag{7}$$

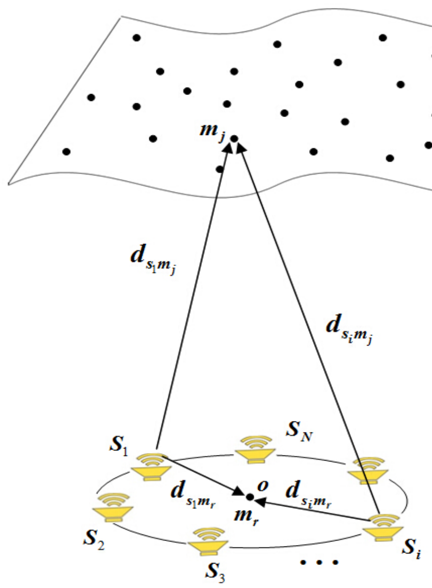


Figure 1 – Schematic diagram of calibrating microphone positions

2.2 Error analysis

The microphone location is calibrated according to the geometric relationship among the calibrating sound source, the reference microphone and the pending microphone, which is represented by equation(1) and equation(4). It can be seen that $\boldsymbol{m}_r = [x_{m_r}, y_{m_r}, z_{m_r}]^T$ and $\boldsymbol{S}_i = [x_{s_i}, y_{s_i}, z_{s_i}]^T$ are known in location, and $\boldsymbol{m}_j = [x_{m_j}, y_{m_j}, z_{m_j}]^T$ is to be determined. Therefore the time delay estimation value τ_{ij} calculated from the reference microphone $\boldsymbol{m}_r$ and $\boldsymbol{m}_j$ using cross correlation method is the only parameter to affect the precision in location. The location error is analyzed due to the error of time delay estimation as depicted below.

The error analysis can be made using simulation data.

$$\min \tilde{J}_j = \min \left\| \boldsymbol{d}_j - c\cdot(\boldsymbol{\tau}_j + \boldsymbol{\varepsilon}_{\tau_j}) \right\| = \min \sum_{i=1}^{N} (d_{S_i m_j} - d_{S_i m_r} - c\cdot(\pi_{ij} + \varepsilon_{\tau_{ij}}))^2 \tag{8}$$

Then the location error can be represented as

$$\varepsilon_p = \min \tilde{J}_j - \min J_j = \min \left\| \boldsymbol{d}_j - c\cdot(\boldsymbol{\tau}_j + \boldsymbol{\varepsilon}_{\tau_j}) \right\| - \min \left\| \boldsymbol{d}_j - c\cdot\boldsymbol{\tau}_j \right\| \tag{9}$$

Without loss of generality, all errors of time delay estimation are assumed to be identical and gaussian distribution. The Monte-Carlo simulations are carried out 2000 times at each time delay error and the result is shown in Figure 2. It can be seen that the time delay need to be estimated exactly to secure the microphone position with high precision. The time delay error is affected by the SNR and reverberation. In most cases, the ratio of the calibrating source signal and noise is large enough, therefore the reverberation is the main factor to degrade time delay estimation. The reverberation condition is discussed in Section 3 and an effective method is proposed to reduce reverberation effect on time delay estimation.

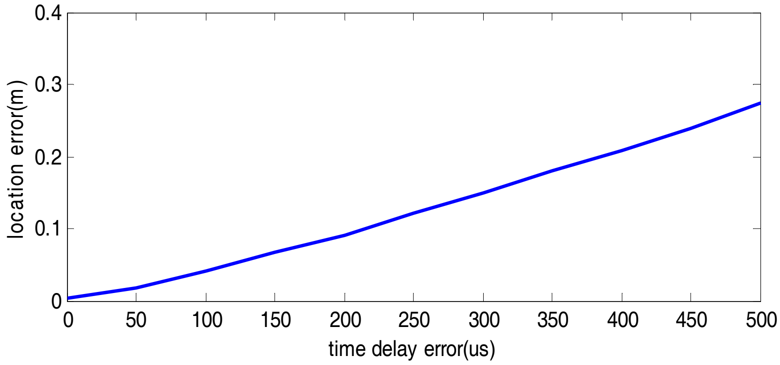


Figure 2 – Location error of calibrating microphone positions with time delay error

3. POSITION CALIBRATION IN A REVERBERANT ENVIRONMENT

In a reverberant environment, a microphone receives not only the direct sound from the source, but also the reflected waves from the walls and floor. The reflected waves is coherent with sound of source, which reduce the precision of time delay estimation and even lead to a mistake estimation. In many applications, white noise or chirp source is used as a calibrating sound source(9,10). In these scenarios, the direct sound, early reflection and reverberation are mixed up. Therefore the reflected wave disturbs severely the performance of time delay estimation. In this paper, we employ to use an impulsive source as the calibrating sound source and truncate the direct sound from the reflected waves. The direct sounds from the reference microphone and a pending microphone are utilized to calculate the time delay using general cross correlation method by padding zeros. The simulation is carried out in this section. The sample frequency is 48kHz and SNR is 20dB. There are eight microphones in a room with the size of [6m,6m,8m]. The coordinates of microphones are list in Table 1. The reverberation data is generated with RT=0.4s by reference(12).

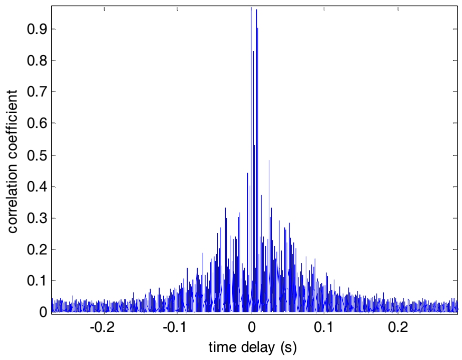


Figure 3 – correlation coefficient between the reference microphone and a pending microphone for white noise calibrating sources in a reverberant environment with RT_{60}=0.4s

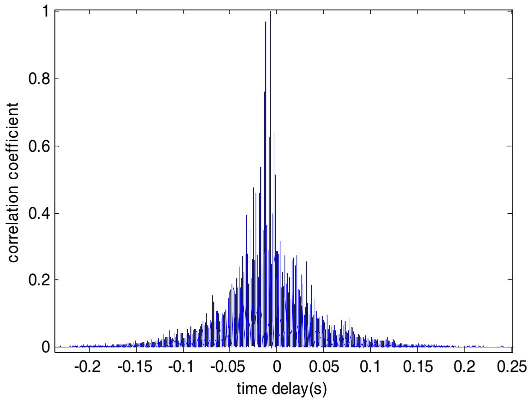


Figure 4 – correlation coefficient between the reference microphone and a pending microphone for chirp calibrating sources in a reverberant environment with RT_{60}=0.4s

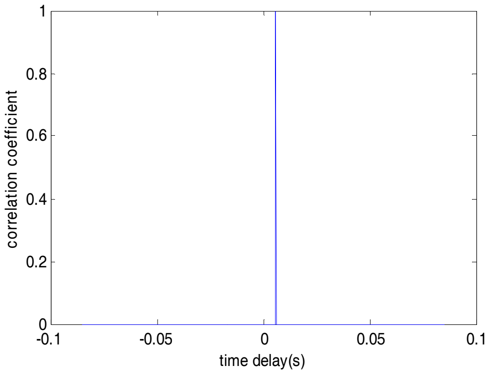


Figure 5 – correlation coefficient between the reference microphone and a pending microphone for impulsive calibrating sources in a reverberant environment with RT_{60}=0.4s

Figure3-Figure5 show the correlation coefficients between the reference microphone and a pending microphone for white noise, chirp source and impulsive source as calibrating sources in a reverberant environment with RT60=0.4s, respectively. It can be seen that the maximum of correlation coefficients for chirp source and white noise is difficult to find out, which indicates that there is a error or even a mistake estimation. In contrast, the maximum of correlation coefficients for impulsive source is quite distinct. Table 1 shows the time delay estimation error using three different schemes. As the Table 1 shows, time delay error for impulsive source is smaller than the others. Based on the Table 1 and Figure 2, the small location error can be concluded. The position calibration results are shown in Figure 6, which also indicates smaller errors using the proposed method.

Table 1 – time delay estimation error due to three calibrating sources

Microphone location (m)	White noise error (us)	Chirp source error (us)	Impulsive source error (us)
(6.0,5.5,5.0)	-59	-63	-2
(3.0,6.0,3.5)	9	-10	-5
(1.0,5.0,3.6)	1072	-18	6
(5.0,30,4.8)	-1996	-17	-3
(0.0,1.0,3.5)	-1	-27	-5
(3.0,5.0,5.5)	2	-18	-6
(6.0,2.0,6.0)	1632	1694	13
(4.5,3.0,6.5)	7025	2959	16

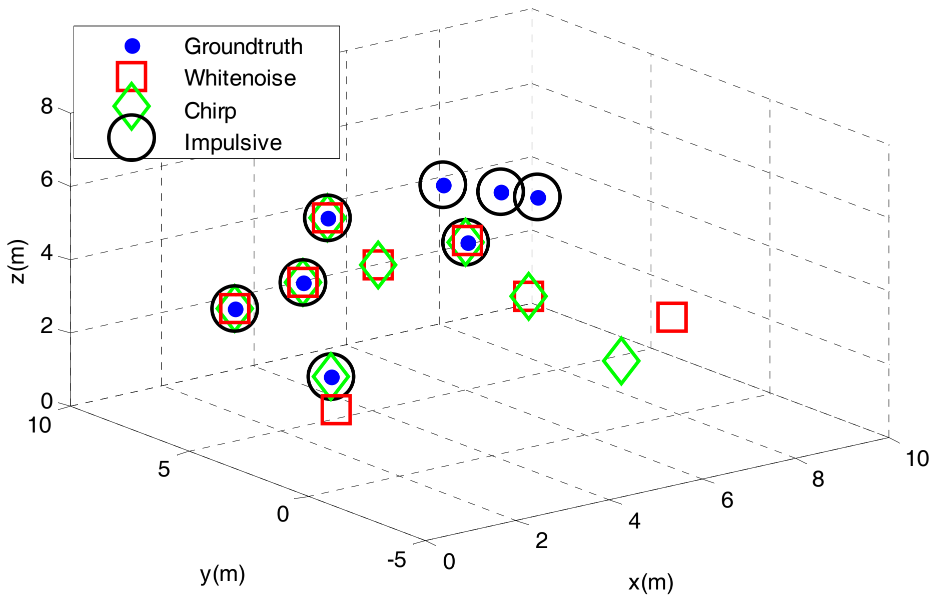


Figure 6 – Simulation results of microphone position calibration using white noise, chirp and impulsive source in a reverberant environment with RT_{60}=0.4s.

4. EXPERIMENTAL RESULTS

4.1 Experimental setup

In order to validate the proposed method, we carried out experiments of microphone position calibration in a reverberant room by constructing a large-size microphone array with aperture of 3.5m. As illustrated in Figure7, microphones are randomly mounted on the ceiling. The reverberation time is measured shown in Figure 8. The average reverberation time is above 0.5s from 100Hz to 8kHz. The impulsive calibrating sources are generated by a starting gun in four locations. The sampling frequency is 15kHz.

Figure 7 – Setup of randomly distributed microphone array mounted on the ceiling

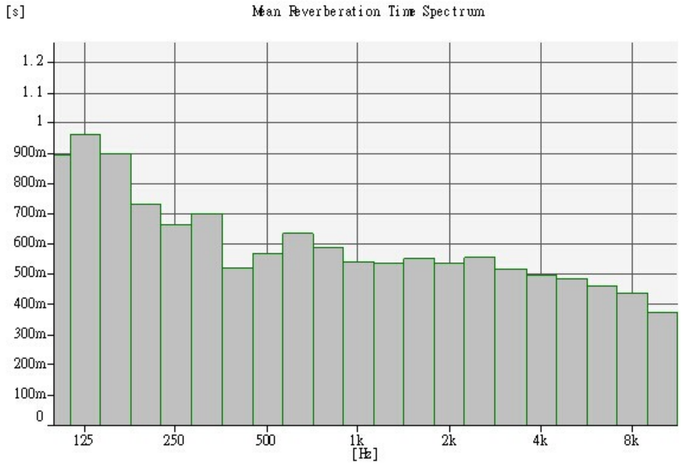
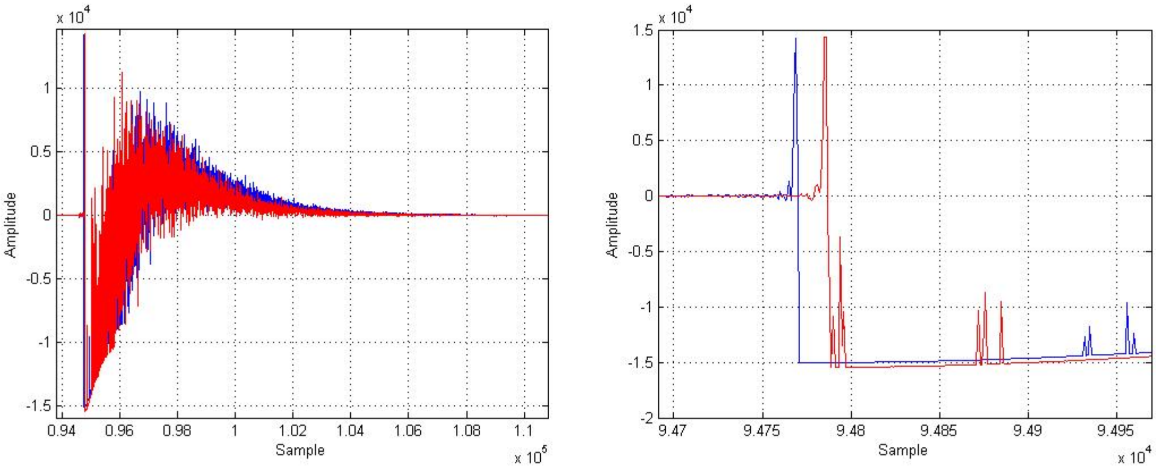


Figure 8 – Reverberation time with frequency band in the experiments.

Figure 9 shows two signals received by the reference microphone and a pending microphone. Figure 9(b) shows partial enlarged drawing of Figure 9(a). The blue line represents the signal received by reference microphone and red one represents the signal received by a pending microphone.



(a) microphone signal (b) partial enlarged drawing

Figure 9 – Microphone signals received by reference microphone(blue) and pending microphone(red)

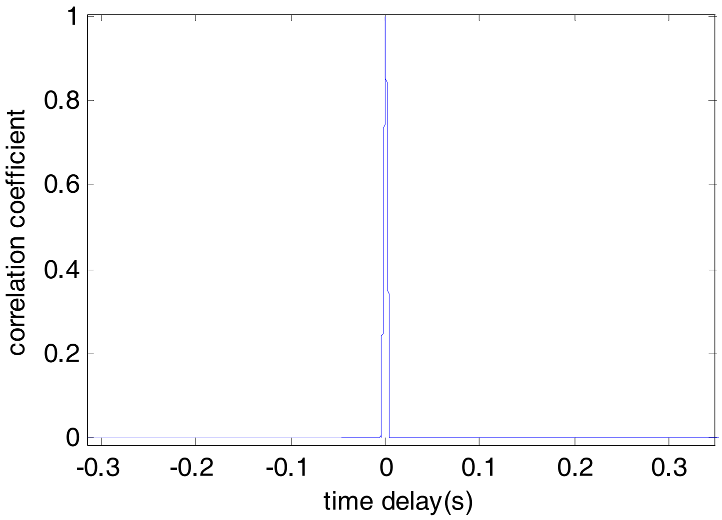


Figure 10 – correlation coefficient between the reference microphone and a pending microphone using the direct sound from impulsive calibrating sources in a reverberant office.

Figure 10 shows the correlation coefficient between the reference microphone and a pending microphone using the direct sound from impulsive calibrating sources in a reverberant room. Table 2 shows the calibrated results of microphone positions with ground truth plus error. The ground truth is obtained by measuring the projection of microphones on the ground using the plumb. The measurements are made several times to average in order to reduce the measurement error. It can be seen that the maximum error in Cartesian coordinates is less than 20mm.

Table 2 – The results of microphone position calibration

Microphone No.	（ground truth+error）（mm）			Microphone No.	（ground truth+error）（mm）		
	X	Y	Z		X	Y	Z
0	-78+2	455+2	0+0	32	-7+1	-600+11	0-6
1	-78-1	720-2	0+1	33	23+2	-860-5	0-4
2	-340-5	970-2	0+2	34	310+2	-1110-3	0+5
3	-855+2	940+2	0-3	35	840-10	-1080+0	0-5
4	-1130-2	915-1	0+0	36	111+5	-1060+6	0-2
5	-1380-1	900-3	0-8	37	1384+0	-1050-8	0+6
6	-1615+3	880+0	0-5	38	1648-3	-1030-2	0+0
7	-1863+0	620+1	0+3	39	1954+1	-745-6	0+7
8	-305+0	180+5	0+5	40	288+2	-315-2	0+2
9	-563+3	420+4	0-6	41	560+2	-595+0	0+1
10	-1103-2	410+2	0-6	42	1110+1	-530-2	0+3
11	-1343+3	130+2	0+5	43	1363+2	-255-4	0+0
12	-1593-2	-150+0	0+4	44	1604+14	-20-5	0-9
13	-1573+0	-410-2	0+3	45	1563+5	320-10	0+3
14	-1820-3	-675+0	0-8	46	1820-5	75-3	0+7
15	-1783+2	-925+5	0-3	47	1780+18	840-6	0+5
16	-555-3	-110+2	0+2	48	493+2	-20-4	0-4
17	-793+3	-110-2	0-2	49	785+4	0+0	0-4
18	-1073-4	-370+3	0+6	50	1053-3	265+0	0+3
19	-1008-2	-905-3	0-7	51	1015+0	900+8	0+3
20	-990-8	-1155+2	0+6	52	968-8	1050-7	0+6
21	-960-2	-1435-3	0+0	53	938-5	1310+2	0-9

22	-930-2	-1640+3	70-2	54	878+2	1460+9	70+2
23	-635+1	-1640+2	320+5	55	490-10	1460+9	410-3
24	-263+2	-350-2	0-3	56	243+4	230-13	0+3
25	-503-2	-620-5	0+7	57	475-3	500+0	0+8
26	-465-2	-1130-7	0+0	58	430+3	1020+10	0+5
27	-190-4	-1400+2	0+5	59	150+5	1235+5	0-2
28	65-2	-1640-5	65-4	60	-130-4	1450+15	70-12
29	370-2	-1640-7	70-3	61	-375+10	1450+15	80-4
30	530+6	-1640-7	250-5	62	-610-6	1450+13	2+2
31	820+6	-1640-7	260+6	63	-800-4	1450-14	6+3

4.2 Results of acoustic imaging system

In order to validate the calculated microphone positions further, the coordinates of all microphone are input into an acoustic imaging system (AIS) built in IACAS to localize multiple sound sources. As illustrated in Figure 11, the acoustic imaging system localizes accurately two sound sources generated by two small speakers.

Figure 11 – Result of localization of two point sources

Figure 12 shows the results of localization of a line sound source. The line sound source is generated by installing a speaker into one end of pipe, sealing the other end of the pipe and cutting a slot.

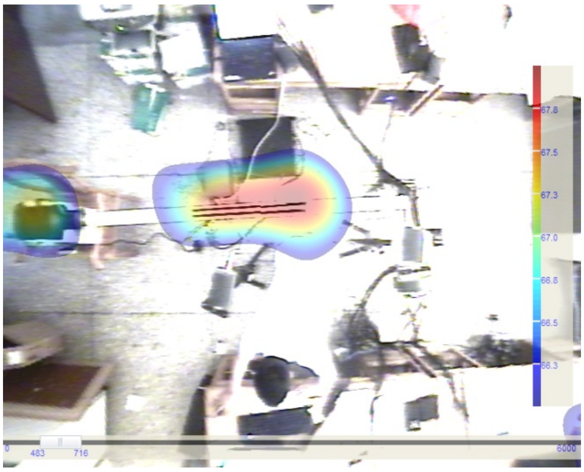


Figure 12 – Result of localization of the line sound source

The accurate localization of sound sources of the acoustic imaging system indicates that the coordinates of microphone have been calibrated with high precision, which proves the effectiveness of the proposed method.

5. CONCLUSIONS

In this paper, we propose a new effective solution for calibrating microphone positions in a reverberant environment. Three kinds of calibrating source are investigated and the impulsive sound source are proved to reduce the effect from reverberation by isolating the direct sound from early reflection and reverberation sound. The simulation experiments are carried out to demonstrate the results among impulsive sound, chirp sound and white noise, which indicates the accurate time delay can obtained by using the proposed method. The proposed method is validated using both Monte-Carlo simulations and a real-time experimental setup.

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