COMPARISON OF DIFFERENT MICROPHONE ARRAYS FOR SIGNAL PROCESSING OF AIR-ULTRASONIC SIGNALS IN NONDESTRUCTIVE TESTING

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ABSTRACT

Acoustic cameras consist of an array of microphones with well-defined positions, which receive individual sound signals. They provide real-time acoustic images after reconstruction mostly based on beamforming algorithms. In the work presented here, we transfer this technique to the ultrasonic frequency range. In the first step we examine different microphone configurations for localization of air ultrasonic sound sources. Afterwards we use a one-microphone-method to build a synthetic microphone array and reconstruct the source from the ultrasonic signals received by one transducer. With the help of this method, it is possible to evaluate the recorded signals with different microphone arrangements, by just performing one area scan. The spatial resolution and dynamic range of the array are investigated as a function of array-source distance and microphone arrangement. Furthermore, the reconstruction results are compared with simulations.

KEYWORDS: Beamforming, Microphone Arrays, Ultrasonic Sources, Signal Processing

1. INTRODUCTION

Ultrasonic phased arrays have been widely employed to detect defects in components for NDT applications. These active arrays, which typically operate in a frequency range between 1 MHz and 20 MHz, have to be in contact with the component surface, and the same elements send and receive the signals. A wide spread reconstruction technique in ultrasonic testing is the synthetic aperture focusing technique (SAFT) [1], which can either be applied to array data or to data recorded with a single transducer which was scanned within a synthetic aperture. Recently, passive microphone arrays have been used for acoustic source localization and considerable improvement has been achieved in this field [2]. These systems are adapted to the audible range with an upper frequency limit of 20 kHz or slightly more. These systems are able to provide real-time acoustic images based on beamforming algorithms, where the received signals at each microphone are delayed and summed to interfere constructively [3]. These conventional beamforming systems cannot be used for most of the NDT applications because of their frequency limitation. To overcome this limitation, the applicability of conventional beamforming in ultrasonic range will be investigated in this paper. A comparison between different microphone arrangements will be demonstrated and their performance will be examined with point source simulations and real measurements.

2. ARRAY SELECTION

A variety of microphone configurations have been investigated in the literature [4] to report that circular and spiral configurations have more benefits than linear or rectangular grid arrangements. In order to select microphone configurations, we simulated the beam pattern of different arrangements in the far field. A beam pattern provides information about directivity, beam width and side lobe level of receiving arrays [5]. To illustrate these properties, we first calculated a sample beam pattern of a ring array. A small inter element spacing and diameter are required when localizing high frequency sources and with respect to this, we chose a ring array with 32 omnidirectional microphones and 20 cm diameter. Figure 1.a depicts the calculated cut plane beam pattern at 75 kHz. The main beam located at 0° is called main lobe. The beam width determines the spatial resolution of the array to resolve two neighboring sources and is defined as the width of the main lobe at -3 dB. Beside the main lobe, some local maximums can be seen which are called side lobes. The level difference between the highest side lobe and the main lobe is...
known as maximum side lobe level (MSL). These two parameters have a significant role in the design of a microphone array arrangement.

![Figure 1(a)](image1.png) Far field calculation of beam pattern of a ring array with 32 microphones and 20 cm diameter at 75 kHz; (b) ring array configuration

To select the optimum sunflower arrangements, the sunflower spiral formula [6] which produces variety of arrangements with a fixed number of elements depending on a parameter $V$ was investigated. The formula in polar coordinates $r$ and $\theta$ is given by

$$ F(i) = (r(i), \theta(i)) = (\sqrt{i}, 2\pi \frac{\sqrt{V} - 1}{2}) $$

Where $i$ is the number of the $i^{th}$ microphone and $V$ is the corresponding parameter to each unique configuration. We studied the parameter $V$ from 1 to 10 and simulated the beam pattern of each array to compare their MSL. To find the best array for a wide frequency range, the simulations were done over a frequency band from 60 kHz to 100 kHz. Figure 2 shows the result of 273 beam pattern calculations and support the fact that $V = 1$, $V = 3.2$, $V = 8.2$ and $V = 9$ outperform over the mentioned frequency range. They have all a MSL more than 10 dB in common which is a good compromise for side lobe suppression at high frequency ranges. Although $V = 1$ and $V = 9$ are more advantageous, but because of their dense inter element spacing, it is very difficult to build their arrangement in practice. However, our one-microphone scan method which is explained in chapter 3 makes it possible to try this arrangement for real measurements. The $V$ arrangements are shown in figure 2 and they will be compared with the ring array in the next section.

![Figure 2](image2.png) MSL comparison of different configurations (from $V = 1$ to $V = 10$) at three different signal frequencies. The MSL value of $V = 3.2$ and $V = 8.2$ is shown with black arrows. $V = 1$ and $V = 9$ have same configuration.
3. SIMULATIONS

In order to evaluate the performance of selected arrays for real measurements, first we defined a 75 kHz spherical wave with 10000 samples (sampled at 20 MS/s in 0.5 milliseconds). The time signals of received by a microphone array at 10 cm and 18 cm distance from the source were calculated. These data were reconstructed using the delay and sum beamforming algorithm. The measurement distances were chosen to investigate the applicability of beamforming in localization of ultrasonic sources which are located slightly or much less than the diameter of the array. The simulation results are illustrated in figure 4 and it can be seen that the sources appear smaller when the array is located closer to the source. The results are displayed with a -10 dB dynamic range (MSL). It is also to be noticed that the ring array has better local resolution compared to the spiral configurations, but a higher side lobe level in the region of interest (area around the main source). The linear array shows extreme side lobe level in the direction perpendicular to its length axis. It can be seen that there is very low side lobes in other directions which is the advantage of linear arrays. The line array has a very small curved structure which cannot be seen in figure 3 but, it is more obvious in the reconstruction plane when the source is 10 cm far from the array. To measure the resolution of the arrays, we calculated the beam width with the simulations and an experimental formula. For the calculation of the beam width with simulations, we plotted the result at -3dB and measured the width of the source with a MATLAB tool. The spatial resolution as a rule of thumb using delay and sum beamforming is given by [7]

\[ B = \frac{425d}{Df} \]

Where \( d \) is the measurement distance, \( D \) is the diameter of the array and \( f \) is the signal frequency. The width of the main lobe depends also on array configuration which is not taken into consideration in the formula. Table 1 shows the calculated beam width from the formula and simulations. It is to be seen that the ring array has slightly smaller beam width and the spiral arrays \( V = 3.2 \) and \( V = 8.2 \) have almost same beam width. Surprisingly, the beam width of the \( V = 9 \) array at 10 cm is slightly bigger than its beam width at 18 cm measurement distance which again shows its unique properties. The beam width also defines the minimum distance where two in-phase neighboring sources are called to be resolved which is the measure of resolution of an array at a specific frequency.

<table>
<thead>
<tr>
<th>Array</th>
<th>( V = 3.2 )</th>
<th>( V = 8.2 )</th>
<th>( V = 9 )</th>
<th>Formula</th>
</tr>
</thead>
<tbody>
<tr>
<td>Beam width at 10 cm</td>
<td>1.8 mm</td>
<td>2.4 mm</td>
<td>3 mm</td>
<td>8.8 mm</td>
</tr>
<tr>
<td>Beam width at 18 cm</td>
<td>2.4 mm</td>
<td>4.2 mm</td>
<td>4.2 mm</td>
<td>8 mm</td>
</tr>
</tbody>
</table>

Table 1 Beam width comparison at -3dB calculated from simulations and the formula
4. EXPERIMENTS

Each microphone array possesses specific properties which can be extracted from its beam pattern. To find a suitable array for ultrasonic application we calculated the beam pattern of different arrangements (see chapter 2) and compared them with respect to their resolution and dynamic range. We chose a frequency range between 60 kHz and 100 kHz, which is above the typical limit of acoustic cameras but low compared to the MHz range of ultrasonic testing. However, it is within the common range of air-ultrasonic systems. To confirm the simulation results, we performed measurements with a ring and spiral configurations with 32 microphones and 20 cm diameter. Figure 5.c illustrates the measurement setup where the arrays were built synthetically by scanning one microphone. An ultrasonic transducer with 6 cm aperture and center frequency of 75 kHz (15% bandwidth) was placed behind a screen with a hole of 1 cm diameter to send the signals. The transducer was excited with a 75 kHz sinus burst signal with 3 cycles and 220 V sending voltage. A microphone with a frequency response of 6.5 Hz-140 kHz (±2 dB) and a diameter of 3 mm was moved with a manipulator in x and y directions. A quadratic area with a width of 21 cm * 22 cm was scanned with a step width of 1 mm. The data were sampled with 20 MS/s in 0.5 milliseconds. Afterwards, it was possible to extract the time signals recorded at each measurement point with a MATLAB code. The sample time signals captured at the positions of the ring array with 32 elements and 20 cm diameter and the average spectrum calculated with a Fast Fourier Transformation algorithm are depicted below. It can be seen that there are time shifts between the received signals in different microphones which is a requirement for reconstruction by delay and sum beamforming. Furthermore, it is shown that the captured signals have frequency components ranging from 60 to 100 kHz (two peaks at 75 kHz and 85 kHz) and have to be considered as broadband signals. Therefore, we decided to employ delay and sum beamforming in time domain which is a good choice for localization of broadband signals.
The reconstructed ultrasonic signals with our one-microphone-method using the ring and spiral configurations at 10 cm and 18 cm distance from the source are shown below in figure 6. The results are displayed with a -10 dB dynamic range for comparison with the simulations. As predicted by the simulations, the ring configuration provides better local resolution than the spiral configurations, but it is more affected by the side-lobe artefacts. This is to be noticed that the measurement distance has an influence on appearance of artefacts and the reconstruction plane consist of more side lobes when the measurement distance is much less than the diameter of the array. We did not consider any noise level in the simulated spherical waves, therefore, there are much less side-lobe artefacts in the reconstruction plane in figure 4 compared to those in figure 6. The spiral configurations fail to find the true source location at 10 cm distance which shows the advantage of ring array (symmetric configuration) in short distances measurements. Another important factor for the reconstruction with spiral configurations is the exact position of elements. Due to the step with of 1 mm which was set to the manipulator, we could not reach the exact position of elements in the array, which is for sure a reason for appearance of artefacts. The exact position of element do not have big influence on the results reconstructed with the ring array. The beam widths of each array calculated from the simulations are relatively narrower than those calculated from the real measurements and this is due to the fact that the simulated signals are narrowband, but in the measurements, we recorded broadband signals (sinus burst) which consist of more frequency components (60 -100 kHz).

Figure 6 reconstructed sources from the real data recorded from synthetic array: 1st col. Ring; 2nd col. V = 8.2; 3rd col. V = 3.2 and 4th col. V = 9. Measurement distance: 1st row 10 cm; 2nd row 18 cm.
Table 2 beam width comparison at -3 dB calculated from the real measurements

<table>
<thead>
<tr>
<th></th>
<th>Ring array</th>
<th>V = 8.2</th>
<th>V = 3.2</th>
<th>V = 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>Beam Width at 10 cm</td>
<td>2.75 mm</td>
<td>-</td>
<td>-</td>
<td>3.5 mm</td>
</tr>
<tr>
<td>beam width at 18 cm</td>
<td>4.3 mm</td>
<td>9 mm</td>
<td>7.5 mm</td>
<td>7 mm</td>
</tr>
</tbody>
</table>

3. CONCLUSION

In this paper, we presented the development of a one-microphone-method to build a passive synthetic microphone array for ultrasonic and NDT applications. Different microphone array configurations were simulated and compared with respect to their spatial resolution and dynamic range. Furthermore, the reconstruction results were compared with beamforming and point source simulations in order to test the functionality of the chosen arrangements. The spatial resolution was investigated with experimental formula, simulations and real measurements. It was demonstrated that beamforming in the ultrasonic frequency range is possible and that the local resolution improves compared to the acoustic frequency range as expected by theory. In the future studies, the Rayleigh Limit can be investigated where two sources are very close to each other. It is also proposed to apply a proper weighting factor to improve the reconstruction results or some other beamforming methods such as Functional or CLEAN-SC can be implemented to suppress the side lobes.

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REFERENCES