Distant Sound Source Positioning by a 3D Octagonal Microphone Array Optimized by Improved Weighted Beamforming Algorithm

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ABSTRACT:
Audio source localization is one of the most important topics in different fields of signal processing e.g. entertainment, military and security applications. In this way, this article presents a novel approach for intercepting and localizing distant audio sources using a three-dimensional (3D) octagonal microphone array consisting of 23 microphones. The proposed system employs an efficient algorithm for analyzing the information obtained from the microphone array to accurately determine the spatial position of sound sources across a wide frequency range, from infrasound to ultrasonic frequencies. To validate the system efficiency, numerical simulations were conducted through 80 tests using different audio sources with frequencies ranging from 20 Hz to 1 MHz, located at spatial distances of 52 to 402 meters from the array. The results demonstrate the high accuracy of the proposed system in identifying the spatial position of sound sources. The proposed system's performance is optimized using an improved weighted beamforming algorithm, which is implemented in MATLAB software on a core i7, 64-bit, RAM-8GB computer system. The proposed approach has the potential to enhance the capabilities of military and security systems for detecting and localizing sound sources in complex environments.

KEYWORDS: Source Localization 3D Microphone Array; Weighted beamforming; Array Geometry; Sound Signal

1. INTRODUCTION

In recent years, the positioning of signal generation sources has emerged as a key research area across various domains, including global positioning systems, video conferences, radar and sonar-based interceptions, mobile phone positioning, robotics, and human-computer interaction. This research field also encompasses topics such as speaker tracking [1]-[5], sensor networks, and touch-based human-computer interactions. In this article, we propose a new method for identifying the position of sound sources using audio imaging. Audio imaging is a technique used to track audio signals and detect the location of their sources [6]. To better understand our proposed system, we first examine the fundamental concepts underlying this issue. Our method has the potential to enhance the accuracy of sound source localization, which is essential in a wide range of applications, including speech recognition, noise reduction, and audio surveillance.

1.1. Microphone array

A microphone array refers to a set of microphones arranged in a specific pattern and a geometric shape to capture sound from the surrounding environment. These arrays can be designed in two-dimensional (2D) or three-dimensional (3D) configurations, with various array types available, as illustrated in Fig. 1 [7]. Microphone arrays have gained widespread use in diverse applications such as speech recognition, audio surveillance, acoustic imaging, and noise reduction, among others. The design and configuration of microphone arrays play a crucial role in determining the accuracy and resolution of sound source localization, which is an essential aspect of many audio applications.

Paper type: Research paper
https://10.71822/mjtd.2025.1092446
Received: 17 February 2024; Revised: 28 April 2024; Accepted: 21 June 2024; Published: 1 March 2025
How to cite this paper: M. R. Tayuri, S. Karimi, **"Distant Sound Source Positioning by a 3D Octagonal Microphone Array Optimized by Improved Weighted Beamforming Algorithm"**, *Majlesi Journal of Telecommunication Devices*, Vol. 14, No. 1, pp. 1-17, 2025.

Several studies have shown that the accuracy and speed of sound source localization are directly and inversely related to the dimensions and number of microphones, respectively [8]. These findings suggest that increasing the size and reducing the number of microphones can enhance the accuracy and speed of sound source localization, respectively. However, the optimal design and configuration of microphone arrays depend on the specific application requirements and environmental factors.

1.2. Beamforming

Beamforming is a signal processing technique used in sensor arrays to transmit or receive a signal while also determining the direction of the source.

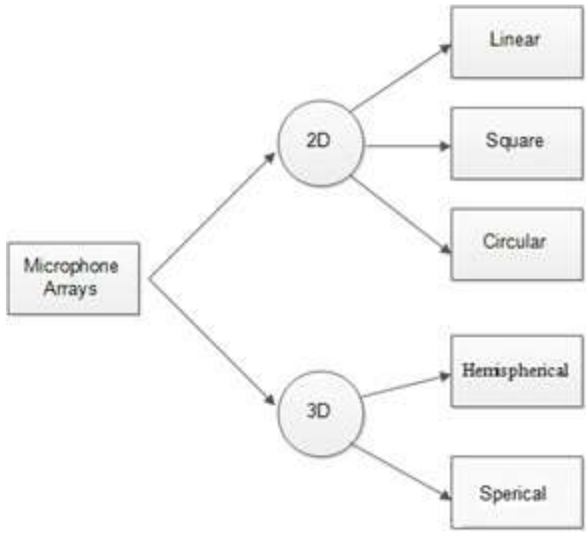


Fig. 1. The most common types of arrays based on appearance.

This technique is designed such that the desired signal enters the array microphones at different angles and creates constructive interference to determine the position of the array source [9]. Beamforming, also known as radiography, is a versatile technology that finds applications in various fields of science, such as seismology, radio astronomy, acoustics, radars, and sonars. In audio systems, beamforming is classified into two categories based on the type of microphone array and the working frequency of each microphone: active and passive audio imaging [10].

- Active audio imaging employs a specific wavelength to send a signal for imaging, instead of using the sound of environmental sources directly. In this approach, the sent audio signal collides with the elements in the surrounding environment and will be analyzed after returning [6]. These structures are widely used in various medical and military fields, such as ultrasound and subsurface radars. In these systems, the speed of sound is calculated as a compressed wave using equation (1), and the time interval between sending and receiving the signal, according to equation (2), is used to determine the distance of the desired object from the source [11]. Beamforming in audio systems is a powerful tool for enhancing the accuracy and resolution of sound source localization, which is essential in various audio applications, including speech recognition, noise reduction, and audio surveillance. The effectiveness of beamforming depends on factors such as the design and configuration of the microphone array, the characteristics of the sound source, and the environmental conditions.

$$c = 331.1 * \sqrt{1 + \frac{T}{273.15}} \tag{1}$$

$$l = \frac{c * t}{2} \tag{2}$$

where C is the speed of sound in m/s, T is the temperature in degrees Celsius, t is the time difference between sending and receiving the signal, and l is the distance between the source and the target.

- Passive audio imaging is a technique in which the audio imaging system does not emit any signals but rather is sensitive to specific frequencies and wavelengths received from the surrounding environment. By analyzing and calculating the intensity and angle of these received signals, the system determines the position of the object [6] and [12]. This approach is useful in scenarios where active imaging is not feasible or may interfere with other systems. Passive audio imaging finds applications in various fields, including speech recognition, audio surveillance, and acoustic imaging. The effectiveness of passive audio imaging depends on factors such as the design and configuration of the microphone array, the characteristics of the sound source, and the environmental conditions.

This article proposes a novel method for determining the spatial position of sound sources in the frequency range from Hertz to Mega Hertz with high accuracy, by optimizing the microphone array structure. To the best of our knowledge, this is the first study to present such a method, which has potential applications in various audio-related fields such as speech recognition, acoustic imaging, and audio surveillance. The proposed method enhances the accuracy and resolution of sound source localization by optimizing the design and configuration of the microphone array, which plays a crucial role in determining the performance of passive audio imaging. Some of the most important applications of the proposed system can be stated as follows:

1- Automotive and aircraft industries: to identify the weak points of sound insulation and engine and moving parts troubleshooting [13].

2- Military industries: passive positioning of moving targets [14], [15].

3- Music: the direction of tuning and tuning the instruments and their correct placement in the performance hall [16].

4- Police traffic control cameras: to identify cars that cause noise pollution in unauthorized places [17].

5- Construction: evaluation of the walls, doors and windows of buildings, to prevent the entry of noise pollution [18], [19].

6- Power industry: Specifying the insulators of high pressure distribution and transmission systems that do not have a complete and correct connection with the cables, and this improper connection causes sounds caused by electric discharge [20], [21].

7- Robotics: to detect the position of the speaker's voice in noisy environments and also to detect the distance of the speaker's voice [22].

8- Subsurface applications: This technology is used to investigate the position of marine organisms such as fish masses and whales and to detect the position of moving targets such as submarines [23], [24].

The paper proposes a novel array configuration for microphones, along with a new approach to constrained optimization for optimal noise suppression. This is achieved by replacing the multi-dimensional optimization process with an efficient two-dimensional search.

The paper is organized as follows: Section 2 introduces different types of microphone arrays. Section 3 evaluates various techniques used for positioning sound sources. Section 4 discusses the proposed method, and its results are illustrated in Section 5. A comparison of the proposed algorithm’s outcomes with other methods is provided in Section 6. Finally, the conclusion is presented in Section 7.

2. TYPES OF MICROPHONE ARRAY

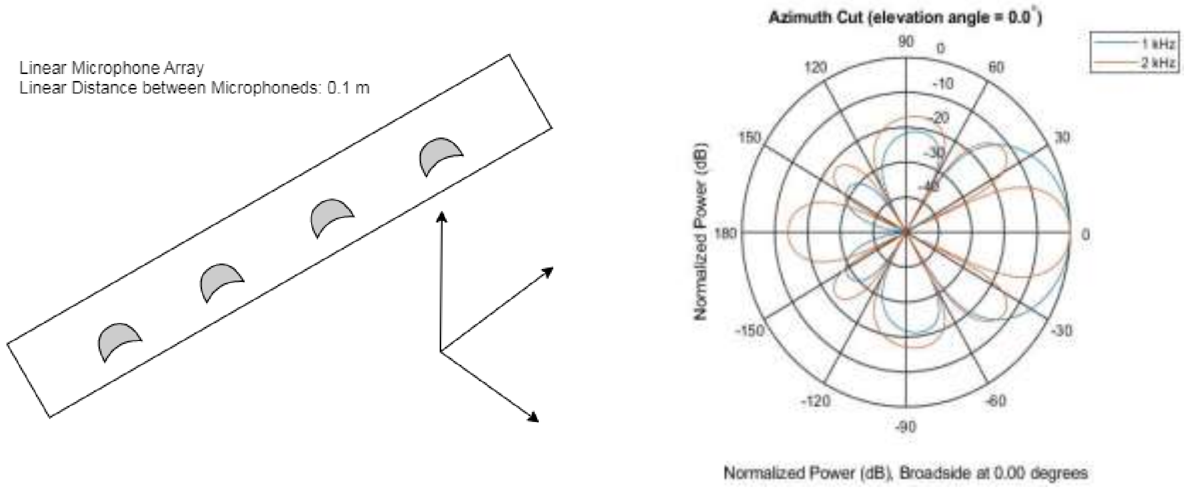
In this section, we will examine the types of microphone arrays and their different applications.

2.1. Linear Array

In a linear microphone array, the microphones are arranged in a row. The distance between the microphones is an important factor that affects the performance of the array and varies depending on the specific application. Two main approaches are commonly used to consider the distance between the microphones in linear arrays. The first approach is to consider the distance in terms of the physical separation between the microphones, which is typically expressed in meters or centimeters. The second approach is to consider the distance in terms of the phase difference between the signals received by the microphones, which is typically expressed in degrees or radians. The optimal distance between the microphones depends on several factors such as the frequency range of the signals, the desired accuracy of sound source localization, and the environmental conditions. They are arranged in two main models:

- Uniforms; in which the distance of all microphones from each other is the same.
- Non-uniform; in which the distance between the microphones increases as the distance from the center of the array increases.

Fig. 2 shows the radiation pattern of a uniform linear microphone array consisting of four microphones placed at an equal distance of 10 cm. The graph indicates that the number of loops obtained at frequencies of 1 and 2 kHz is small and has an asymmetric shape. Therefore, this type of two-dimensional array may not provide sufficient accuracy in audio imaging systems, particularly in the range of speech frequencies. However, this structure offers the simplest calculation method to determine the sound source direction [25]. The optimal design and configuration of a microphone array depend on various factors, including the characteristics of the sound source, the size of the array, and the desired accuracy of sound source localization. Linear microphone arrays find applications in diverse domains such as speech recognition, audio surveillance, beamforming, and acoustic imaging.

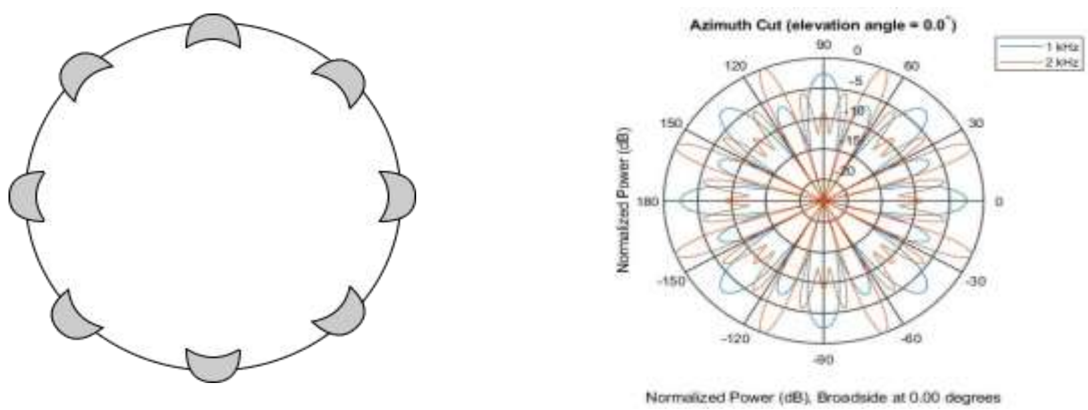


(a) Structure (b) Steering pattern

Fig. 2. Linear array.

2.2. Circular Array

The circular microphone array is a type of array in which the microphones are placed equidistantly on a circle with a certain radius [10]. This structure is widely used in various industries due to its inherent symmetry and the ability to capture sound from different angles. To increase the efficiency of this structure, microphones can be placed on several concentric circles. Fig. 3 shows an example of this type of array, and Fig. 3b depicts the radiation pattern of an eight-microphone circular array with a radius of 0.5 meters at frequencies of 1 and 2 kHz. As evident from the graph, the radiation pattern is perfectly symmetrical at various angles. Notably, the number of loops at a frequency of 2 kHz is higher, resulting in improved accuracy of the system. The optimal design and configuration of the circular array depend on various factors, such as the size of the array, the frequency range of the signals, and the environmental conditions. Circular microphone arrays find applications in diverse domains such as speech recognition, audio surveillance, beamforming, and acoustic imaging.



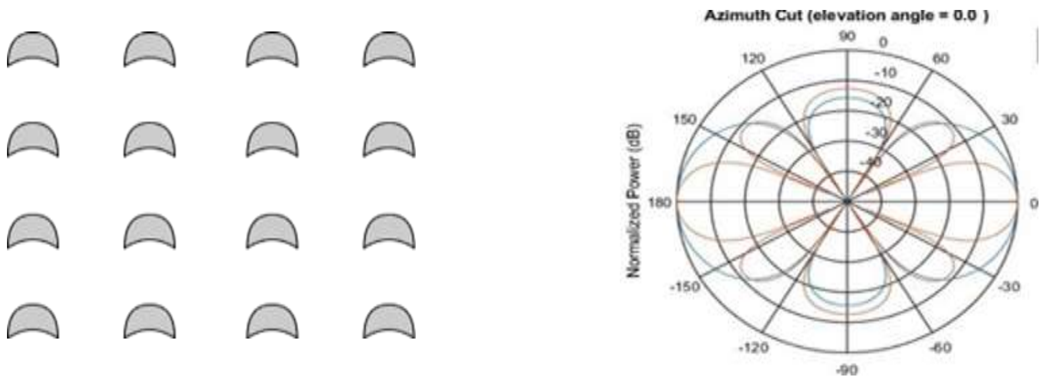
a) Structure b) Steering pattern

Fig. 3. Circular array.

2.3. Square array

The square microphone array is a type of array in which the microphones are arranged in a square structure with a certain distance between them. Fig. 4 shows an example of a square microphone array with 16 microphones arranged in four rows and four columns, each placed at a distance of 0.1 meters from each other. It is evident from the figure that the number of microphones used in the square structure is significantly higher than that used in the circular type.

However, the radiation pattern of the square structure is less accurate than that of the circular structure. Fig. 4 also depicts the radiation pattern of the square array at frequencies of 1 and 2 kHz.



a) Structure b) Steering pattern

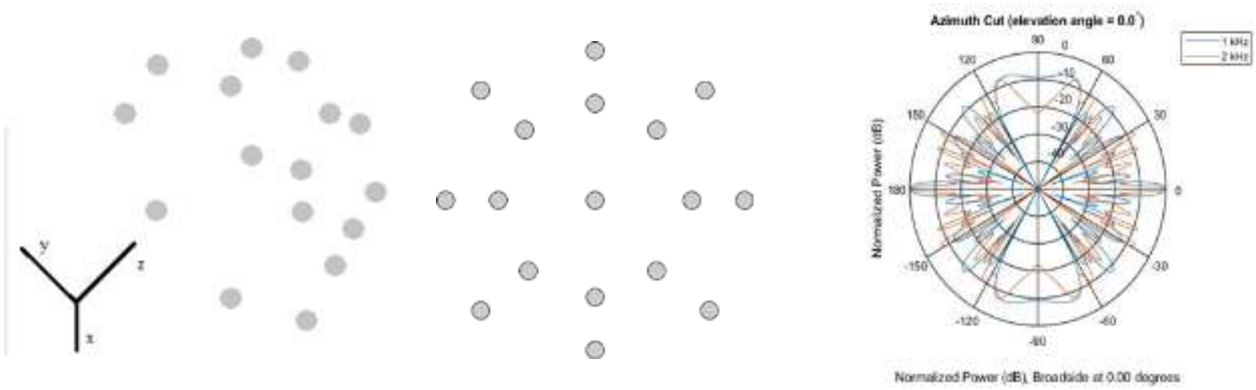
Fig. 4. Square array**.**

2.4. Hemispherical array

The hemispherical microphone array is a three-dimensional structure that consists of multiple circles with different radii placed together with a certain distance and radius. This structure forms a hemisphere and has diverse applications in various systems [26]. Fig. 5 presents an example of a hemispherical array with 17 microphones arranged in three circular layers with radii of 1, 1.5, and 2 meters, designed for implementation at frequencies of 1 and 2 kHz.

2.5. Spherical array

The spherical microphone array is a fully three-dimensional structure in which the microphones are arranged in a spherical shape. This structure allows for the capture of sound from all possible angles due to the shape of the microphones. The accuracy of this structure is very high, making it one of the most practical types of array [27]. Fig. 6 shows an example of a spherical array with 36 microphones placed on a sphere with a radius of 2 meters, illustrating the radiation pattern of the array.



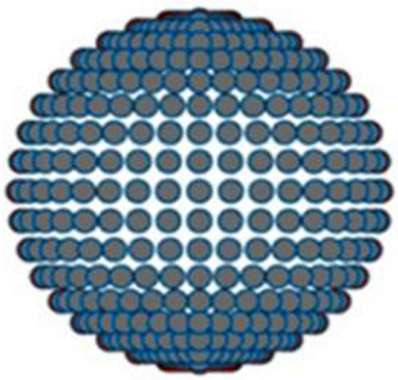
a) Structre b) View from the top c) Streering pattern

Fig. 5. Hemispherical array.

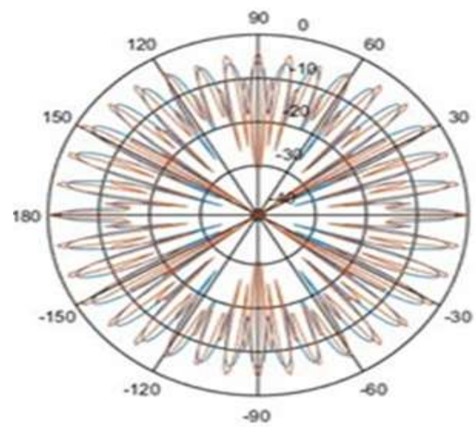
3. SOURCE POSITIONING

An audio camera is a microphone system that comprises a microphone array [28]. Each microphone in the array receives the sound produced by the source. Due to the geometric displacement between the microphones, the sound arrives at each microphone at different times, leading to a time delay between the signals recorded in the microphone array. This time difference of arrival (TDOA) is used to identify the source position [29].

To determine the source position, changes are made to the microphone's radiation pattern, a process called beamforming. Beamforming is a signal processing technique that enables the localization of the sound source. In general, sound source localization involves more complex calculations compared to the localization of radio or optical frequencies, but it can be achieved with cheaper equipment. Fig. 7 depicts a sound collection system that amplifies the target sound using the delay-and-sum method while suppressing environmental noise without introducing a delay. Fig. 8 shows the received sound patterns of each microphone based on their angle and location.



a) Structure



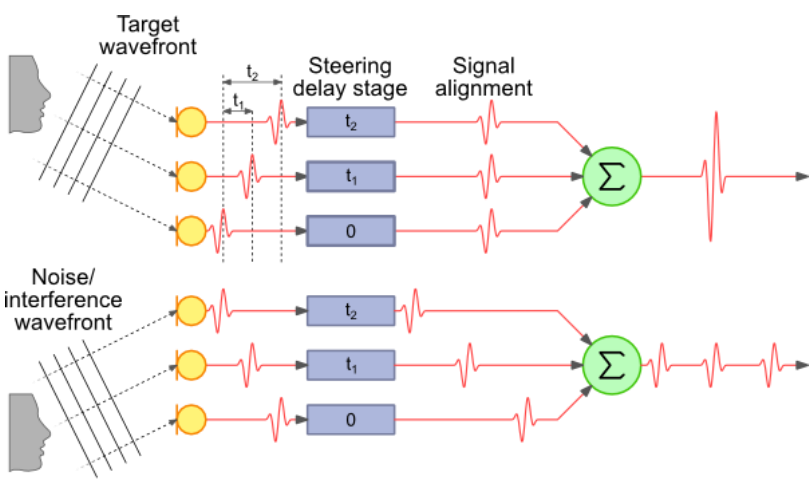
b) Steering pattern

Fig. 6. Spherical array.

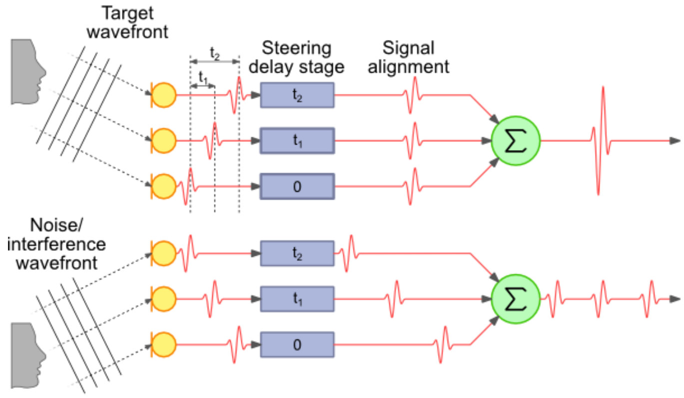
To determine the position of the audio source, we use a two-step algorithm as follows.

1) For broadband signals, many well-known algorithms, such as the Capon or MUSIC method, cannot be used because they use the phase difference between elements, which is only applicable for narrow-band signals. In the case of broadband signals, instead of phase information, the difference in signal arrival time between microphones can be used. In this research, we use the generalized cross-correlation algorithm using phase transformation (GCC-PHAT) to calculate the arrival time differences.

2) We calculate the position of the source by triangulation method. For this purpose, first, straight lines are drawn from the microphones along the directions of arrival. Then, the intersection of these lines is determined as the source location.



(a)



(b)

Fig. 7. The sum system of the received sounds; (a) Using delay; (b) Without delaying the sounds.

Assume that the microphones are located at two-dimensional coordinates (0,0) and (L, 0) and the unknown source is located at (X,Y).

Based on Fig. 9, the distance between two microphones can be obtained using equation (3) [28].

$$L = Y\tan\theta_1 + Y\tan\theta_2 \tag{3}$$

In this way, the values of Y and X are calculated using equations (4) and (5) [28]:

$$Y = L/\left(\tan\theta_1 + \tan\theta_2\right) \tag{4}$$

$$X = Y\tan\theta_1 \tag{5}$$

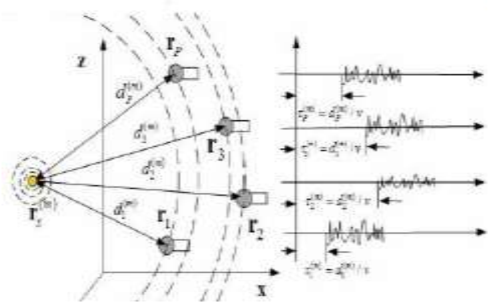


Fig. 8. Sound reception delay of each microphone based on position.

4. THE PROPOSED SYSTEM

This section focuses on the proposed system and the resulting sound source positioning accuracy achieved using the three-dimensional microphone array and its unique structure, which does not have any frequency restrictions. The accuracy of the proposed system is very high, and it provides a robust and reliable method for sound source localization in various domains.

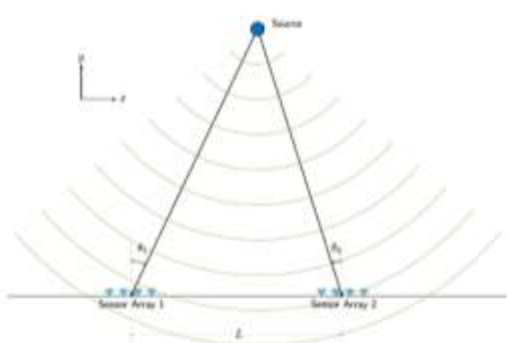


Fig. 9. Propagation of the sound source waves.

4.1. Proposed 3D array

This section presents a novel three-dimensional array structure and assesses its properties. Specifically, the proposed array's microphone placement was optimized via a genetic algorithm in three-dimensional space to enhance sound source localization accuracy at distances ranging from 52 to 402 meters for military and security applications. The optimization process aimed to augment spatial coverage across various angles while amplifying received sound.

4.2. Genetic Algorithm

This research employed the Genetic Algorithm (GA) to optimize the placement of microphones in an array for improved sound source localization. GA was selected for its ability to navigate a vast solution space and identify optimal solutions. GA was implemented to generate a range of candidate solutions for the microphone array's placement, and each solution was evaluated using an objective function that quantified the accuracy of sound source localization. The optimal microphone array configuration was then identified as the best solution. The findings indicate that the GA-optimized microphone array outperformed traditional methods, such as random placement and uniform spacing.

The pseudocode described in Algorithm 1 is presented to optimize the microphone placement in the schematic of our proposed microphone array.

Algorithm 1. Pseudocode to optimize the position of microphones in the proposed microphone array.

```
1. Define the objective function f to evaluate the quality of the microphone array.
2. Define the population size N, maximum number of generations G, and mutation rate pm.
3. Initialize the population of candidate solutions X = {x1, x2, ..., xN} with random microphone positions.
4. Evaluate the fitness of each candidate solution using the objective function: f(xi) for i = 1 to N.
5. Repeat for a set number of generations:
   a. Select the best individuals for reproduction based on fitness using tournament selection:
      i. Randomly select two individuals from the population.
      ii. Choose the individual with the higher fitness to be a parent.
      iii. Repeat steps i and ii to select a second parent.
      iv. Apply crossover operator to create new offspring.
   b. Apply crossover operator to create new offspring:
      i. Randomly select a crossover point k between 1 and the number of microphones M.
      ii. Create a new offspring by swapping microphone positions after the crossover point.
   c. Apply mutation operator to randomly modify offspring:
      i. Randomly select a microphone position to mutate.
      ii. Perturb the position by adding a normally distributed random value with mean 0 and standard deviation
σ = 1 multiplied by the mutation rate pm.
   d. Evaluate the fitness of new offspring using the objective function: f(xi) for i = 1 to N.
   e. Replace the worst individuals in the population with the new offspring:
      i. Select the two worst individuals in the population.
      ii. Replace the worst individual with the best offspring.
      iii. Replace the second-worst individual with the second-best offspring.
6. Return the best candidate solution found during the optimization process: argmax(f(xi)) for i = 1 to N.
```

Fig. 10 displays the outcome of the proposed three-dimensional array model's design. The array structure comprises two pyramids with identical sections, aligned in a mirror-like fashion.

4.3. Evaluation of the Sound Matrix

Following the design of the proposed array structure, we examined and analyzed sounds received from the microphones to identify the source's location.

The sound source's location is represented as (x1, y1, z1), the center of the array as (x2, y2, z2), and each microphone's location as (xi, yi, zi).

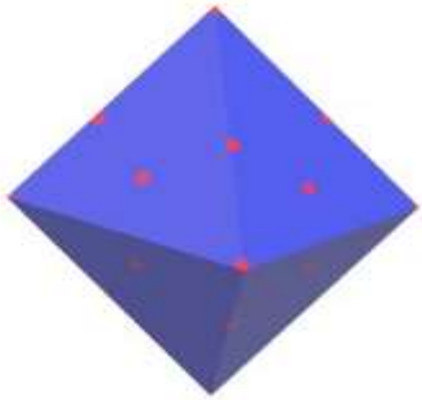


Fig. 10. Proposed 3D microphone array.

Equations (6) and (7) describe the pseudo-three-dimensional distance between the sound source's position and the microphones and the center of the microphone array in the two-dimensional x-z plane, respectively.

$$R_{1a} = \sqrt{((x,k_{a2}) - x_i)^2 + {y_i}^2 + ((z,k_{a1}) - z_i)^2} \tag{6}$$

$$R_{2a} = \sqrt{((x,k_{a2}) - x_2)^2 + {y_2}^2 + ((z,k_{a1}) - z_2)^2} \tag{7}$$

Where:

$k_{a1} \in \{1, 1+\gamma, 1+2\gamma, .., z_1\}$, $k_{a2} \in \{1, 1+\gamma, 1+2\gamma, .., x_1\}$

specify the amount of displacement in the z and x axes, respectively. In this way, by considering the difference of the above values, the sound intensity can be obtained based on the equation (8).

$$b_a = e^{-jw_n(k) f_d R_a / c} \sqrt{a^2 + b^2} \tag{8}$$

where $R_a = R_{1a} - R_{2a}$, c is the speed of sound, $w_n(k)$ is the wave equation based on the angular frequency, and f_d is the amplification factor of the system. In this way, using relation (9), the sound measurement matrix can be obtained in the specified scale.

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$$P(k_{a1}, k_{a2}) = | b_a \times R_a \times b_a^{'} | \tag{9}$$

Utilizing this relation, the P matrix represents the noise measurement matrix in the x-z plane, where each value in the matrix's rows and columns indicates the received sound power according to a specified scale. This matrix can be interpreted as a grayscale image pattern, where each layer represents the sound intensity in that part of the screen. The maximum value in the matrix corresponds to the point where the maximum sound occurred based on the specified scale. Subsequently, the three-dimensional equation (10) employs the obtained results to determine the exact coordinates of the target point with high precision.

$$R_{3a} = \sqrt{((x.k_{a2}) - x_i)^2 + ((y.k_{a3}) - y_i)^2 + ((z.k_{a1}) - z_i)^2)} \tag{10}$$

Where:

$k_{a1} \in \{1, 1+\gamma, 1+2\gamma, .., z_1\}$,

$k_{a2} \in \{1, 1+\gamma, 1+2\gamma, .., x_1\}$,

$k_{a3} \in \{1, 1+\gamma, 1+2\gamma, .., y_1\}$

Similarly, in equation (11), the three-dimensional distance of the microphone array center from the source location is obtained.

$$R_{4a} = \sqrt{((x.k_{a2})-x_2)^2+((y.k_{a3})-y_2)^2+((z.k_{a1})-z_2)^2)} \tag{11}$$

Based on this, similar to equation (9), the sound measurement matrix can be accurately obtained using relation (12).

$$P(k_{a1},k_{a2},k_{a3}) = | b_a \times R_{a2} \times b_a^{'} | \tag{12}$$

where $R_{a2}= R_{3a}- R_{4a}$. In this way, P is the sound measurement matrix in x, y and z plane, which determines the maximum sound received in this plane.

To enhance the usability of the proposed algorithm, a maximization filter is applied to the matrix. This filter identifies the maximum value in each row and column, sets the maximum value to 1, and zeroes out the remaining values. Consequently, the precise location of the target point can be determined using the binary matrix presented in equation (13).

$$ZO(\mathbf{W}) = (x_m, y_m, z_m) = \max_X\left(\max_Y\left(\max_Z(\mathbf{W})\right)\right) \tag{13}$$

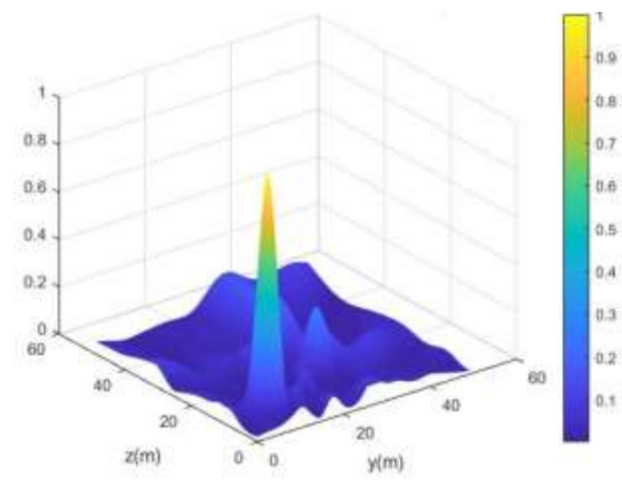
In which, W is the matrix resulting from the three-dimensional positioning of the source and the resulting matrix of zero and one, only its maximum dimension is equal to one and the rest of the elements are equal to zero.

5. RESULTS

In this section, we examine the results obtained from the implementation of the proposed algorithm and compare the results with similar works.

5.1. Preliminary 3D Array Results

In this section, we use the proposed 3D array to locate the sound source and we use the 3D space equations that were introduced in the previous section. To track distant sound sources for military and security applications, we first position the sound source on the three-dimensional coordinate plane at (10, 10, 10) meters from the center of the array, which is equivalent to a distance of 52 meters. To enhance tracking accuracy, we set the evaluation step to 0.01 meters. Subsequently, we evaluate the performance of the proposed system by varying the source's position and frequency and analyzing the results of the proposed algorithm's sound source localization. Figs. 11 and 12 illustrate some simulation results for varying the distance and frequency components of the source.



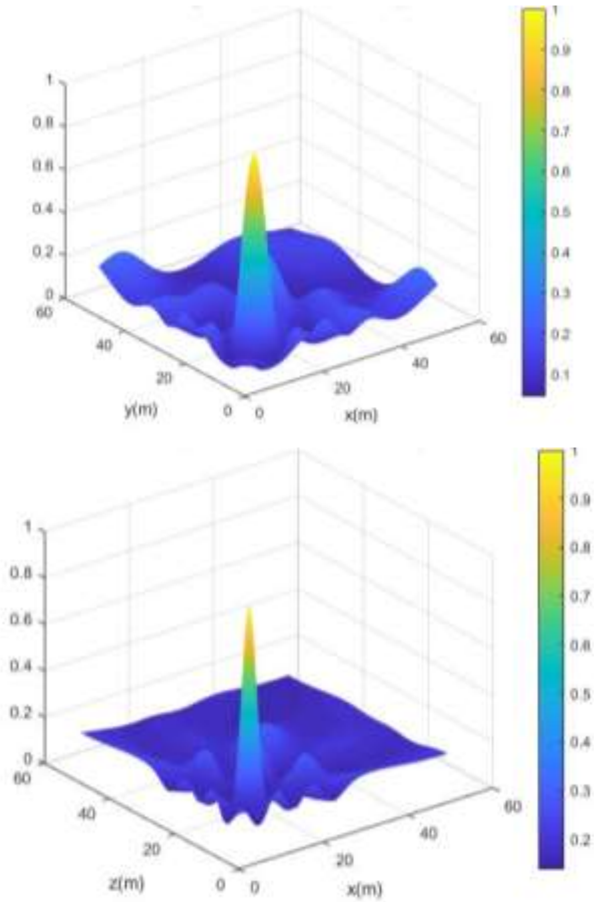


Fig. 11. 3D-localization of a 50 Hz sound source located at (10, 10, 10) using the proposed system.

Fig. 11 demonstrates the capability of the proposed system to determine the precise location of a sound source based on the received sound's intensity. However, some regions with strong noise could indicate potential errors in sound source detection. Another observation from the presented images is that as the distance between the source and the array increases, the localization error increases due to the impact of noise and the reduction in signal-to-noise ratio (SNR). The next section will examine the results of incorporating the proposed system, including the input signal amplifier equation, to evaluate the three-dimensional positioning of the sound source using the binary matrix.

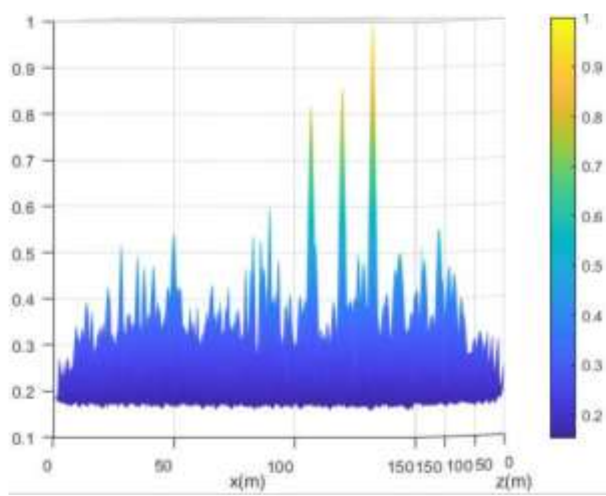


Fig. 12. Localization of a 1 kHz sound source located at (100, 120, 130) using the proposed system.

5.2. Proposed 3D Array System with Gain Coefficients

To evaluate the effectiveness of the amplification factor in enhancing the proposed system's performance, we conducted the following experiments. As previously mentioned, the distance of the sound source from the microphone array and the low frequency of the source signal are two significant factors that contribute to errors in sound source localization. In this section, we simulated these two factors in two experiments, depicted in Fig. 13. Specifically, we set the frequency of the sound source to 20 Hz and 100 Hz in the respective experiments and positioned it at (75, 70, 60) meters and (130, 120, 100) meters in the (x, y, z) space, respectively.

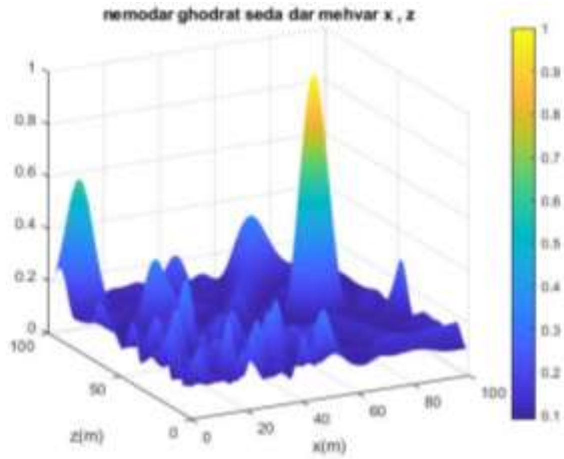


Fig. 13. Localization of a 20 Hz sound source located at (60, 75, 70) (in the x-z plane) using the amount of received sound power in the proposed 3D array with the amplification factor of the system.

The above results demonstrate that the proposed enhanced system can accurately locate a sound source with a frequency of 20 Hz at a distance where previous algorithms utilizing stronger frequencies experienced significant errors. Hence, it can be concluded that the proposed system is effective in identifying the sound source's position even at challenging distances and low frequencies.

5.3. Results of the Applying Zero and One Matrix

In addition to the amplifier system's contribution to enhancing sound source localization accuracy, this section presents the application of the zero and one filters on the system's output. These filters isolate the primary response, representing the sound source's location, while suppressing various visual disturbances and noises that may cause errors in determining the sound source's position.

The component matrix of the received sound power, obtained by analyzing equations (6-9), is based on the three-dimensional coordinate axis scale and a specific pitch, which indicates the received sound's power at each point with a specific pitch. The highest value in this matrix, located in a specific row and column, corresponds to the received sound power of the sound source. Therefore, this position is related to the source's x and y coordinates, which are represented in the first matrix. Using the position of the received sound power and the combination of the y position in this matrix with the highest value's position in the x and z plane, we can determine the sound source's precise location. This approach enables us to analyze the three-dimensional position of the surrounding space using only two matrices, which results in high accuracy and requires less time for analysis than traditional three-dimensional equations. We will elaborate on this approach in the next section.

In this experiment, we examine an example to illustrate the sound source's three-dimensional position. The sound source's position on the (x, y, z) coordinate axis is set at (72, 94, 30) for a 50 Hz source, and the range of 1 to 200 is monitored with a step of 0.1, as in previous experiments. Fig. 14 displays the sound source's position after applying the zero and one filter, which isolates the source's position while removing other noises and disturbances. Fig. 15 depicts the sound source and existing microphones on the array.

6. COMPARISON OF THE RESULTS

To evaluate the proposed algorithm's performance, we placed a 50 Hz sound source at the coordinates (30, 30, 30) and examined three modes: two-dimensional array, three-dimensional array, and three-dimensional array with the system amplification factor. The results of this comparison are presented in the figure, where the yellow area represents the system's output in detecting the sound source's position. As shown, the proposed algorithm significantly reduces the probability of error, although some residual local noises still affect the sound source's position. To address this issue and determine the sound source's precise location, we applied the proposed zero and one filter to the three output matrices of the proposed algorithm. Each matrix examines the sound source's position in two dimensions (x-y, x-z, and y-z matrices). This approach enabled us to identify the sound source's exact three-dimensional position.

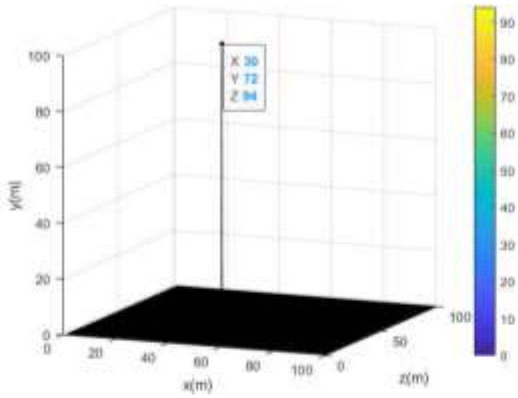


Fig. 14. Sound source position localization using the proposed method boosted by the zero and one matrix.

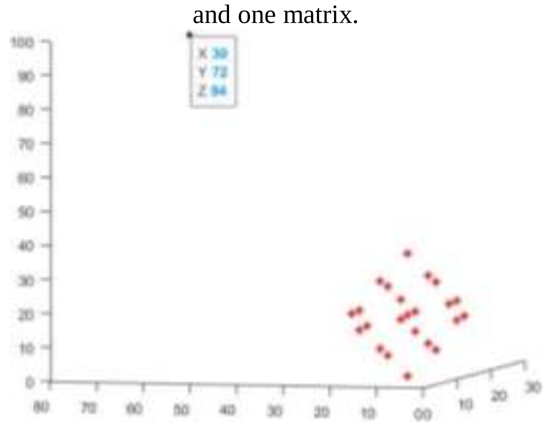
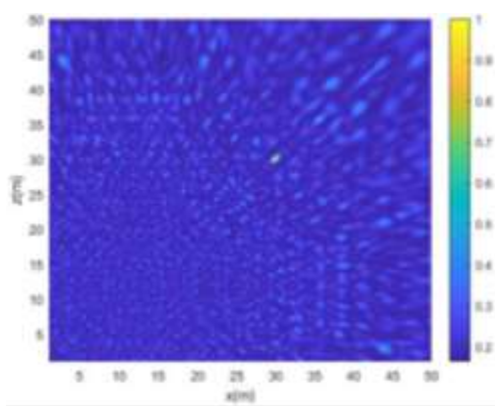
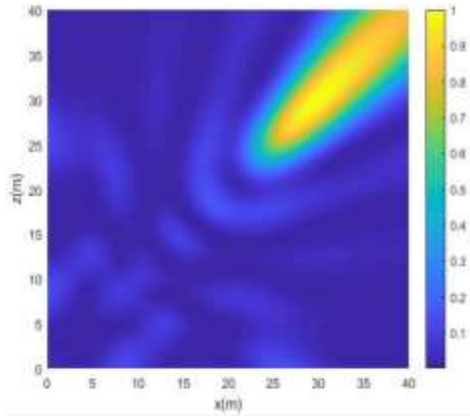


Fig. 15. Location of the sound source and the 3D microphone array.

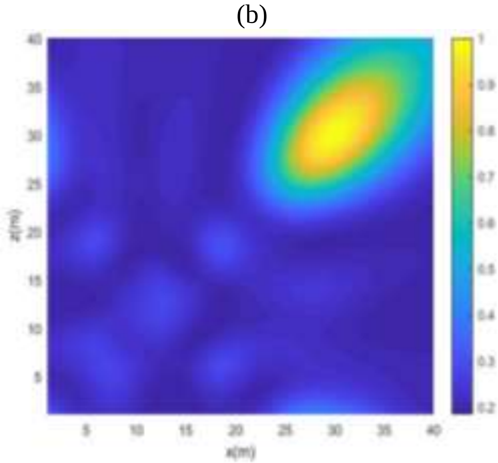
The comparison tables of the results were designed based on the Euclidean distance pattern to demonstrate that the proposed system, which can amplify the received signal, enhances the accuracy of sound source identification compared to similar systems. We performed 160 tests for both near and far distances. Specifically, we conducted 80 tests on the first system without considering the signal amplification and the other 80 tests on the amplified system. The experiments also encompassed signals with frequencies of 20, 50, 100, 1K, 5K, 10K, 100K, and 1M Hz. Tables (1) and (2) present the results of these tests.



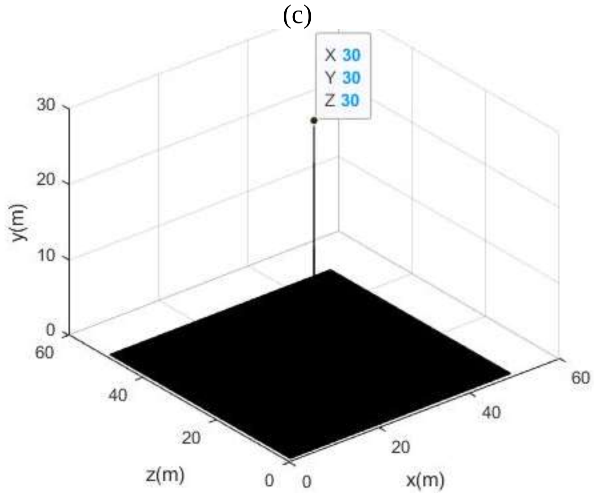
(a)



(b)



(c)



(d)

Fig. 16. Comparison of sound source positioning at 50 Hz using four structures (a) 2D array; (b) 3D array; (c) 3D array by applying the system amplification factor (proposed method 1); (d) 3D array by applying amplification factor and implementing zero and one filter on the output matrices (proposed method 2).

Table 1. Error percentage in locating the sound-source position using the delay and sum system without signal amplification.

		Sound source position $\begin{bmatrix} x \\ y \\ z \end{bmatrix}$ (in meters)									
		30 30 30	50 50 50	50 60 70	70 80 90	90 100 110	110 130 140	110 140 170	170 145 190	200 210 230	210 240 250
Sound source frequency (Hz)	20	37%	43%	47%	50%	57%	64%	68%	75%	80%	84%
	50	36%	41%	45%	48%	54%	61%	65%	72%	76%	80%
	100	33%	37%	41%	44%	50%	56%	60%	66%	70%	74%
	1K	26%	29%	34%	35%	41%	47%	50%	56%	59%	63%
	5K	23%	25%	29%	31%	36%	42%	45%	49%	53%	57%
	10K	20%	22%	26%	27%	31%	37%	40%	43%	47%	51%
	100K	17%	17%	24%	25%	28%	29%	32%	34%	38%	42%
	1M	10%	19%	22%	24%	26%	27%	27%	29%	30%	31%

7. CONCLUSION

In order to solve the distant sound source localization problem, this article explores various arraying methods and signal analysis techniques. By combining the advantages of each of these methods and providing an efficient microphone arrangement pattern, we propose a new method for accurately identifying the sound source's position based on the evaluation of received signals. To this end, we design an array structure comprising 23 microphones arranged in an octagonal pattern and two pyramids whose cross-sectional surfaces are stacked. Additionally, we introduce a delay and weighted summation system to improve signal analysis and increase the proposed system's positioning accuracy. By considering the wave equations, we present equations to strengthen the proposed system, which enhance the positioning accuracy for both clean and noisy signals and increase the speed of 3D processing calculations for sound source location. Finally, we apply a digital filter to the system's output, enabling us to determine the sound source's position with high accuracy.

Table 1. Error percentage in locating the sound-source position using the proposed delay and sum system with signal amplification and zero and one filter.

		Sound source position $\begin{bmatrix} x \\ y \\ z \end{bmatrix}$ (in meters)									
		30 30 30	50 50 50	50 60 70	70 80 90	90 100 110	110 130 140	110 140 170	170 145 190	200 210 230	210 240 250
Sound source frequency (Hz)	20	4%	7%	8%	9%	11%	11%	12%	13%	13%	15%
	50	3%	6%	6%	7%	9%	10%	10%	11%	12%	13%
	100	2%	4%	5%	6%	8%	9%	8%	9%	9%	12%
	1K	<1%	2%	4%	5%	7%	8%	6%	7%	8%	11%
	5K	<1%	<1%	2%	3%	5%	6%	5%	6%	6%	9%
	10K	0	<1%	1%	2%	4%	4%	4%	5%	5%	8%
	100K	0	0	<1%	1%	2%	3%	4%	5%	5%	7%
	1M	0	0	0	<1%	<1%	1%	3%	4%	4%	5%

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