CZECH TECHNICAL UNIVERSITY IN PRAGUE

Faculty of Electrical Engineering

Department of Measurement

Bachelor Thesis:

Application for Acoustic Signal Localization

Aplikace pro lokalizaci akustického signálu

Project of: Ma Yuchen
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BACHELOR'S THESIS ASSIGNMENT

I. Personal and study details

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Study program: **Electrical Engineering and Computer Science**

II. Bachelor's thesis details

Bachelor's thesis title in English:

Application for Acoustic Signal Localization

Bachelor's thesis title in Czech:

Aplikace pro lokalizaci akustického signálu

Guidelines:

Design an application for calculating a position of an acoustic event that will be detected by a defined number of units where each unit contains two microphones. The input data for created application will be the GPS coordinates (Latitude, Longitude) of the units, the speed of the sound, and the position of an intended event. The part of the work will be a simple GUI showing the result.

Bibliography / sources:

[1] R. C. Maher, "Modeling and Signal Processing of Acoustic Gunshot Recordings," 2006 IEEE 12th Digital Signal Processing Workshop & 4th IEEE Signal Processing Education Workshop, Teton National Park, WY, USA, 2006, pp. 257-261, doi: 10.1109/DSPWS.2006.265386.

[2] S. Astapov, J. Ehala, J. Berdnikova and J. -S. Preden, "Gunshot acoustic component localization with distributed circular microphone arrays," 2015 IEEE International Conference on Digital Signal Processing (DSP), Singapore, 2015, pp. 1186-1190, doi: 10.1109/ICDSP.2015.7252067.

Name and workplace of bachelor's thesis supervisor:

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Date of bachelor's thesis assignment: **06.02.2023** Deadline for bachelor thesis submission: __________

Assignment valid until:
by the end of summer semester 2023/2024

Ing. Jakub Svatoš, Ph.D.
Supervisor's signature

Head of department's signature

prof. Mgr. Petr Páta, Ph.D.
Dean's signature

III. Assignment receipt

The student acknowledges that the bachelor's thesis is an individual work. The student must produce his thesis without the assistance of others, with the exception of provided consultations. Within the bachelor's thesis, the author must state the names of consultants and include a list of references.

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Date of assignment receipt

Student's signature

DECLARATION

"I hereby declare that this bachelor's thesis is the product of my own independent work and that I have clearly stated all information sources used in the thesis according to Methodological Instruction No. 1/2009 – "On maintaining ethical principles when working on a university final project, CTU". In Prague.

Date Signature

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ACKNOWLEDGEMENTS

I would like to express my sincerest gratitude to my supervisor, Ing. Jakub Svatoš, Ph.D. for his continuous guidance and help all through the entire thesis period. This work is a product of his hard work, patience, and vision as much as it was mine.

Abstract

This bachelor's thesis focuses on developing a simulation detection procedure within a sound source localization system, leveraging time-variable data. The primary goal is to generate accurate position coordinates by filtering out invalid and erroneous data, ultimately pinpointing the precise location of the sound source. The program employs a sophisticated algorithm that not only identifies valid data but also detects and eliminates time-related discrepancies. Additionally, the research investigates the impact of external environmental factors, sensor configurations, timestamp accuracy, and other variables on localization errors. Through this comprehensive approach, the thesis aims to enhance the reliability and precision of sound source localization systems in diverse real-world scenarios.

Keywords

Sound source localization, Angle of arrival, Timestamp, Localization error

Abstrakt

Tato bakalářská práce se zaměřuje na vývoj apmilace pro simulaci lokalizace zdroje zvuku, využívající časově proměnná data. Primárním cílem je generovat přesné souřadnice polohy odfiltrováním neplatných a chybných dat av konečném důsledku určit přesnou polohu zdroje zvuku. Program využívá sofistikovaný algoritmus, který nejen identifikuje platná data, ale také detekuje a odstraňuje časové nesrovnalosti. Kromě toho se práce zabývá dopadem vnějších faktorů prostředí, konfigurace senzorů, přesností časové značky. Prostřednictvím tohoto komplexního přístupu se práce zaměřuje na zvýšení spolehlivosti a přesnosti systémů lokalizace zdrojů zvuku v různých scénářích reálného světa.

Klíčová slova

Lokalizace zvukového zdroje, Úhel příchodu, Časové razítko, Chyba lokalizace zvuku

LIST OF ACRONYMS

TDOA -- Time Delay of Arrival
AR -- Augmented Reality
AoA -- Angle of Arrival
PTP -- Precise Time Protocol
GPS -- Global Positioning System
GUI -- Graphical User Interface

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1. Introduction

In recent years, the pressing concern for public safety has underscored the critical importance of accurate and efficient sound source localization systems. The rise in societal challenges, particularly incidents of gun violence, has emphasized the need for advanced technologies to swiftly and precisely locate the origin of sounds in various environments. Sound source localization plays a pivotal role in security and surveillance applications, aiding law enforcement and emergency response teams in rapid and effective decision-making.

This bachelor's thesis is dedicated to addressing the challenges associated with sound source localization by developing a simulation detection procedure based on time-variable data. The primary goal is to enhance the precision of sound source localization, a factor crucial in responding to societal issues like gun violence. Traditional methods often face limitations in accurately determining the source of gunfire amidst complex urban environments, where echoes, obstructions, and ambient noise can obscure crucial information.

By improving the accuracy of position coordinates and mitigating the impact of external factors on localization errors, the research endeavors to provide valuable insights and solutions for enhancing public safety measures. This work aligns with the broader societal objective of leveraging technology to create safer environments, especially in the face of increasing concerns related to gun violence and other security threats. As such, this thesis not only contributes to the field of sound source localization but also holds implications for the broader societal context of public safety and security.

Sound source localization technology involves the use of multiple microphones to measure sound signals at different positions in the environment. The algorithm processes the measured sound signals to obtain the direction of arrival of the sound source point relative to the microphone, including azimuth angle, pitch angle, and distance. While more sophisticated positioning techniques exist today, the fundamental principle of phase enhancement remains the basis for modern sound source localization systems.

The historical perspective of sound source localization traces back to the First World War when complex acoustic defenses were invented. One notable example is the Télésitemètre Perrin, designed by French physicist Jean Baptiste Perrin. This early model utilized a receiver with dozens of small horns arranged in a hexagonal honeycomb nest, connected to a central horn. Today, modern sound source localization systems build on the principle of phase enhancement but have evolved to incorporate signal processing algorithms, eliminating the need for physical movement.[1]

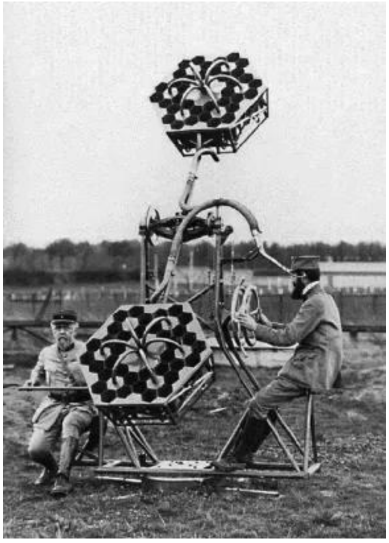

Fig.1 Picture of an early model of the Perrin telemetry, from unknown sources [1]

In military applications, sound source localization technology proves valuable for measuring artillery positions, locating hidden snipers, and determining ammunition test gun landing points. As stealth technology advances, passive sound source detection becomes a significant advantage. Industries also leverage this technology for detecting ships and vehicles, identifying noise sources in machines, and monitoring the condition of mechanical systems. Moreover, sound source localization methods find applications in acoustic design, non-contact vibration measurement, and virtual reality audio systems.

In the pursuit of real-time and accurate sound source positioning, this research acknowledges the common trade-offs between real-time processing and accuracy. The accuracy of positioning is crucial for decision-making and action. The paper's focus is to analyze the theoretical basis of an acoustic detection system proposed by my supervisor. It involves dissecting localization algorithms based on a two-microphone setup within a detection system, implementing the algorithm using MATLAB code, and creating a user-friendly test template with the MATLAB Application Designer. This template aims to facilitate future evaluations of detection system effectiveness. The research also delves into understanding how uncertainties, such as distance, speed of sound, and environmental factors, affect the accuracy of sound source localization, providing insights for optimizing performance in diverse scenarios and test conditions.

2. Theoretical Principles

Basic principles of acoustic positioning

Acoustic positioning entails the utilization of an array of microphones strategically positioned in accordance with a defined geometric arrangement to ascertain the spatial coordinates of a sound source. For a pair of microphones situated at distinct spatial locations, a discernible discrepancy in the distance from the sound source is evident unless they are precisely aligned along the midline. This phenomenon is depicted in the illustrative figure below.

It is noteworthy that a spatial disparity ΔL exists between the sound source and the two microphones, where ΔL is the product of the speed of sound C and the time delay τ, represented as $\Delta L = C \cdot \tau$. This temporal lag, τ, is inherently present due to the distinct distances traveled by the sound wave to reach each of the two microphones. In an ideal scenario, the signals received by microphones i and j adhere to the relationship $Si = Sj(t - \tau)$, encapsulating the temporal discrepancy associated with the sound wave propagation. [3]

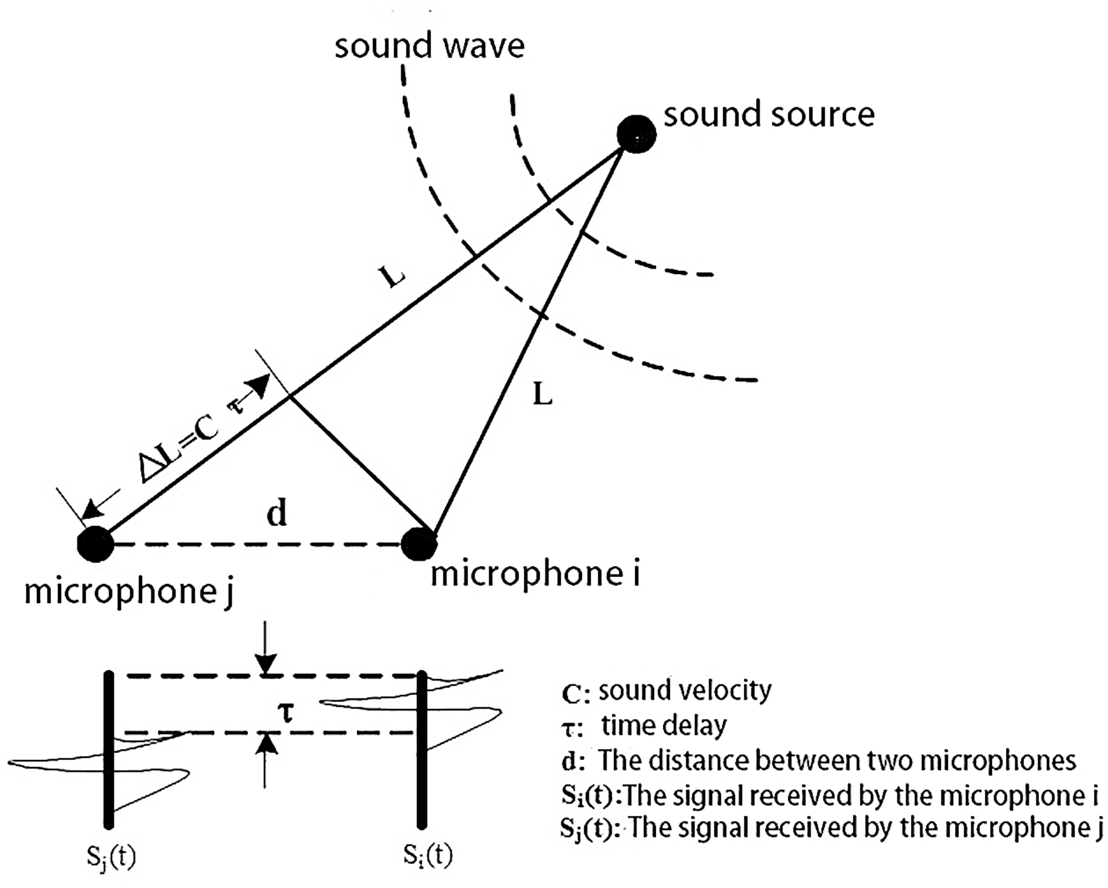

Fig. 2 Principle diagram of sound source localization [2]

Sound source localization technology employing microphone arrays can be broadly categorized into four distinct approaches: controllable beamforming technology based on high output power, high-resolution spectral estimation technology, positioning technology based on sound pressure amplitude ratio, and positioning technology based on Time Delay of Arrival (TDOA) of sound.

Beamforming Technology:

Beamforming serves as a direct method for sound source localization. The fundamental concept involves the formation of a beam through the weighted summation of sound signals received by the microphone array. By adjusting the weights, the microphone array's output power is maximized, pinpointing the location of the sound source where the beam exhibits high output power. In the context of a traditional beamformer, the weight is contingent upon the phase delay of the signal on each element, a parameter linked to the delay and arrival time difference of the sound, thus earning the designation of a delay summation beamformer.

Let the number of microphones be denoted as M, and the signal received by the i_{th} microphone, after time delay alignment, is represented as:

$$y(t,q) = \sum_{i=1}^{M} x_i\,(t+\tau_i) \tag{1}$$

Here, τ signifies the controllable delay when the array is directed towards the search point. This delay is proportional to the number of microphones, array aperture, incidence angle of the sound source, and sampling frequency. Conversely, it is inversely proportional to the propagation speed of sound. The cumulative output power, denoted as $P(q)$, or the power of the beam, is expressed as:

$$P(q) = \int_{-\infty}^{\infty} |Y(\omega,q)|^2 d\omega \tag{2}$$

The frequency-domain representation is then computed, and the sound source's location can be determined by maximizing the output power, as per the following expression:

$$\overline{q} = \arg max_q\, P(q) \tag{3}$$

Through control of the array's direction, the beam is directed, facilitating the identification of the sound source's location where the beam exhibits heightened output power. [2]

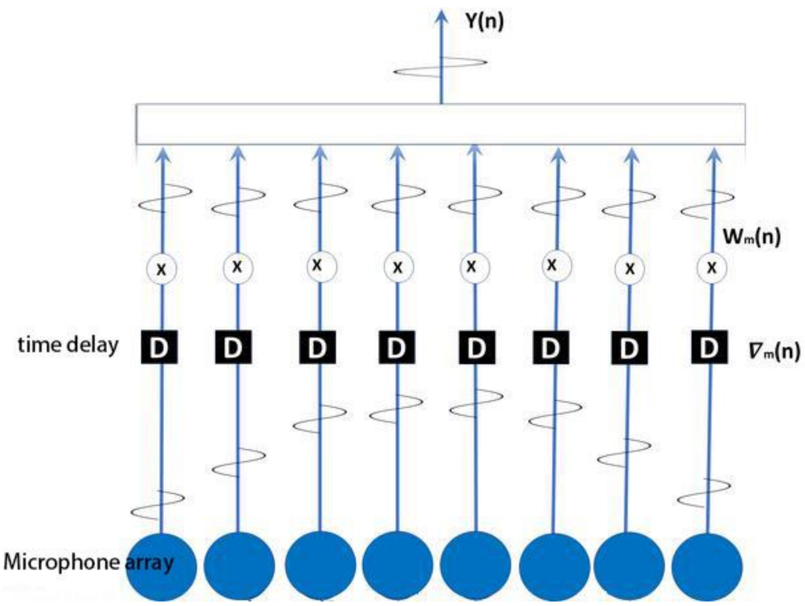

Fig. 3 Determining the position of sound source through microphone array [2]

High-Resolution Spectral Estimation Techniques:

The high-resolution spectral estimation technique relies on the spatial spectrum of the correlation matrix of the received signal to ascertain the direction angle, subsequently determining the sound source's position. This approach encompasses methodologies such as the autocorrelation Augmented Reality (AR) model, the small variance spectrum estimation, and the eigenvalue decomposition algorithm. While adept at handling multiple sound sources, these techniques necessitate the computation of the correlation matrix of the spatial spectrum from the microphone array signal. Challenges arise when the matrix is unknown, requiring estimation from acquired data, demanding stability and constancy of sound sources or noise, a condition often challenging to achieve in practical scenarios. Additionally, the computational intensity of this method limits its prevalence in sound source localization systems.

Localization Method Based on Sound Pressure Amplitude Ratio:

This method leverages differences in the intensity of sound signals received by diverse microphones from a common source for sound source localization. A constraint is derived from the voltage output of sound pressure at the microphone and the distance to the corresponding sound source, delineating a sphere in three-dimensional space. Each microphone establishes such constraints, either independently or in tandem with those derived from a time-difference-based approach.

Sound Source Localization Technology Based on Time Difference of Sound Arrival:

Positioning reliant on Time Difference of Arrival (TDOA) estimation boasts high accuracy and computational efficiency, rendering it apt for real-time implementation.

This method adopts a two-step approach: TDOA estimation to determine time differences between opposing array elements, employing techniques such as cross-correlation, generalized cross-correlation, adaptive filter, mutual power spectrum phase, and higher-order statistical methods. Subsequently, the sound source's location is determined by integrating the estimated time differences and the known spatial geometric relationships of the microphone array. Despite its commendable real-time performance, challenges persist, including error transmission amplification and limitations in localizing multiple sound sources.

2.1 Huygens' Principle

The Huygens' Principle, also known as the Huygens Principle or the Huygens-Fresnel Principle, is an analytical method for studying wave propagation problems. It was named after the Dutch physicist Christian Huygens and the French physicist Augustin Fresnel. This principle applies to the diffraction phenomenon of waves during propagation, whether it is far-field limit or near-field diffraction.

Huygens's principle reveals the characteristics of wave propagation, that is, in the process of wave propagation, each wavefront can be regarded as a new wave source, and the envelope of these wavelets at any time after that is a new wave front. This principle provides a theoretical basis for sound source localization. [3]

By using the Huygens principle, the shape and propagation direction of sound waves radiated by different wave sources can be determined. This enables us to determine the location of the sound source by measuring and analyzing the propagation characteristics of sound waves.

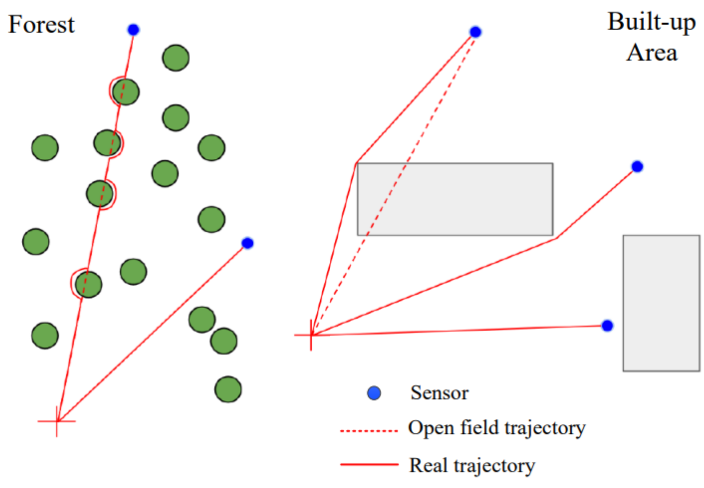

Fig. 4 Sound propagation through different environments [3]

The Huygens principle can explain many phenomena of sound wave propagation in both homogeneous and non-uniform media. This enables us to further optimize the algorithm and model for sound source localization based on the propagation characteristics of sound waves in the medium. The Huygens principle provides important theoretical support and practical guidance for sound source localization, which helps to improve the accuracy and efficiency of sound source localization.

2.2 Dual-Microphone Sound Source Localization

The dual-microphone real-time sound source localization system operates by capturing and analyzing sound signals through acoustic sensors. This system determines the sound source's location by measuring time and amplitude differences between two microphone sensors. Utilizing triangulation based on the speed of sound and distance between sensors, the azimuth and pitch angles of the sound source are calculated. [5][6]

2.2.1 Angle of Arrival (AoA) Computation

The Angle of Arrival (AoA) computation forms the foundational logic for this acoustic detection system. Assuming a planar acoustic wave reaching the device, non-directional sound alters time differences between timestamps received by two microphones. AoA is defined as the angle between the sound signal's propagation path upon reaching the device and the line connecting the event position and the device. Triangulation calculations using the known device length and time differences result in the AoA .[4]

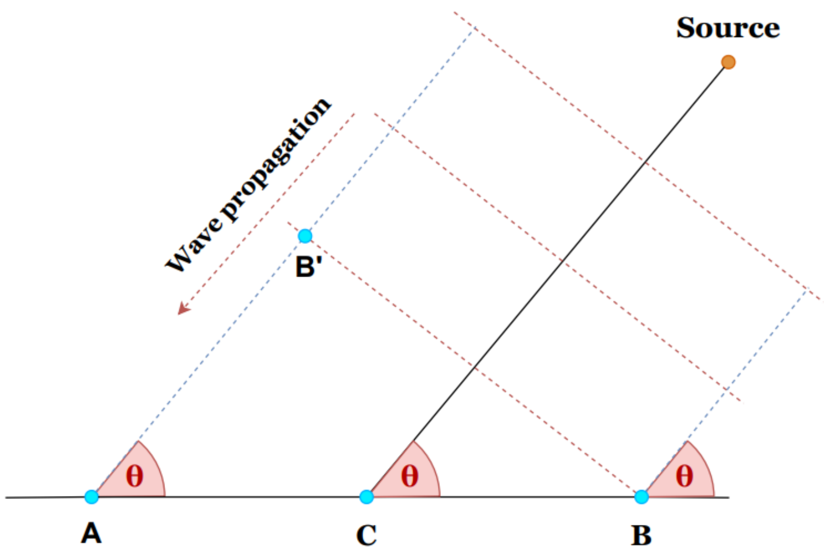

Fig. 5 Angle of Arrival [4]

2.2.2 Data Obtained from Single Set of Dual-Microphone Sensor Detection

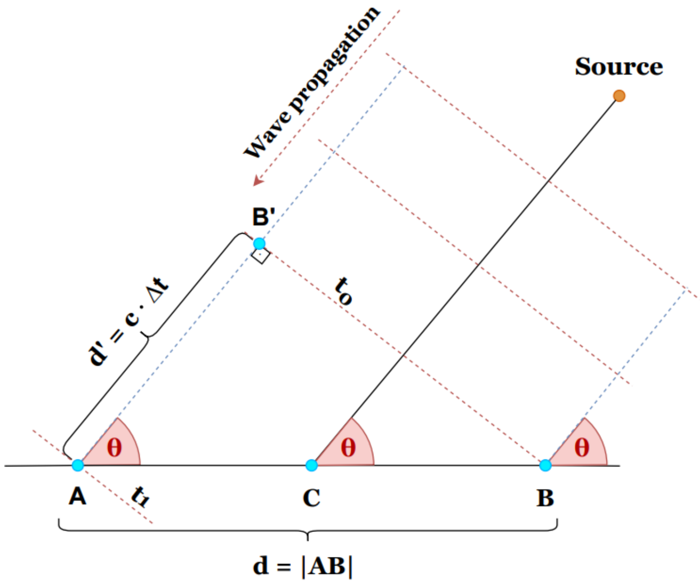

Fig. 6 Angle of Arrival - computation visualization [4]

To apply the AoA method, a specific formula for the acoustic detection system with two microphones is derived. Assuming planar acoustic waves, the distance r of the sound source from the microphone is governed by:

$$r > \frac{2d^2 f_{max}}{c} \tag{4}$$

where:

d represents the distance between the microphones,
f is the frequency, and
c denotes the speed of sound.

Under these conditions, the acoustic wave is assumed to be planar, simplifying the computation required to determine the angle of arrival of the acoustic signal at the unit.

Referring to Fig.6, two microphones, A and B, are assumed to be separated by a distance $d = |AB|$. Let the planar acoustic signal arrive at microphone B at time t_0. The distance |AB| is denoted by d', and this distance is equal to $\mathrm{d}' = \mathrm{c} \cdot \Delta\mathrm{t}$, where c is the speed of sound, and Δt equals the difference in times t_1 and t_2 when the signal reaches microphones A and B, respectively. [4]

At this point, the angle of arrival θ can be expressed by the following equation:

$$\cos(\theta) = \frac{d'}{d} \tag{5}$$

which can be rewritten to express θ as:

$$\theta = \arccos\left(\frac{d'}{d}\right) = \arccos\left(\frac{c \cdot \Delta t}{d}\right) \tag{6}$$

Once the Angle of Arrival has been obtained, we can proceed to define the linear function corresponding to the Angle of Arrival. For ease of computation, we opt to describe the linear function using an analytical expression.

$$y = k \cdot x + b \tag{7}$$

Firstly, we can derive the slope θ of the line using trigonometric formulas:

$$k = \tan\left(\theta \pm \arccos\left(\frac{c \cdot (t_1 - t_{2)}}{d}\right)\right) \tag{8}$$

Subsequently, utilizing the slope and the position of the device in the Cartesian coordinate system $[D_x, D_y]$, we can deduce the intercept of the linear function:

$$b = y - k \cdot x = D_y - \tan\left(\theta \pm \arccos\left(\frac{c \cdot (t_1 - t_{2)}}{d}\right)\right) \cdot D_x \tag{9}$$

This approach facilitates the analytical representation of the linear function, providing a convenient way for further calculations in the context of the sound source localization system.

3. Complex Real-World Scenarios

In the pursuit of accurate event localization, the preceding chapter successfully integrated data from multiple sensors. However, these analyses were conducted under controlled conditions, and the complexities of real-world environments introduce uncertainties that can impact the precision of calculations.

3.1 Timestamp Shift

Uncertainty arises from timestamp shifts, where the precision and alignment of timestamps are critical [7]. Deviations in timestamp synchronization or shifts can introduce errors in Angle of Arrival (AoA) determination, influencing the accuracy of sound source localization.

Timestamp Precision: Divergent timestamp precision among devices can lead to alignment issues when combining data from systems with different timestamp resolutions. [8]

Timestamp Synchronization: Achieving precise timestamp synchronization across devices in a distributed system is challenging due to network delays and device discrepancies, contributing to timestamp discrepancies. [9]

AoA Determination: Errors or offsets in timestamps directly impact AoA calculations, thereby affecting the accuracy of sound source localization. [10]

Impact on Sound Source Localization: Inaccuracies or resynchronizations in timestamps can disrupt the precise alignment of signal data from multiple sensors, impacting the overall accuracy of sound source localization [7]

Mitigation Strategies:

1. Utilize high-precision timestamps and clock sources. [11]
2. Regularly calibrate and synchronize timestamps across all devices. [12]
3. Employ algorithms in the data processing stage to compensate for known timestamp errors. [11]
4. Implement hardware devices or protocols, such as IEEE 1588 (PTP), to ensure precise timestamp synchronization. [12]

Ensuring timestamp precision and synchronization is imperative for accurate data collection and processing, especially in applications requiring high-precision measurements.

3.2 Device demensions

The length of the device introduces an additional layer of uncertainty. While idealized calculations assume a known device length, variations due to manufacturing tolerances or wear and tear can impact the precision of sound source localization. [7]

Mitigation Strategies:

1. Implement regular maintenance and calibration to ensure equipment accuracy. [12]
2. Utilize mathematical models or algorithms to correct data for equipment length deviations.[13]
3. Verify equipment accuracy by comparing measurement results from multiple devices. [14]

Regular maintenance and data correction techniques are essential to mitigate the impact of equipment length on accuracy.

3.3 Environmental Changes — Variable Speed of Sound

In real-world scenarios, the assumed constant speed of sound becomes uncertain due to environmental changes. Variations in air pressure, humidity, or gas density can result in fluctuations in the speed of sound, necessitating consideration for diverse environmental conditions. [7]

Definition of Speed of Sound:
In an ideal gas, the speed of sound v_s can be defined by the following formula:

$$v_s = \sqrt{\gamma \cdot R \cdot T} \tag{10}$$

Here:
v_s denotes the speed of sound,
γ represents the adiabatic index, also known as the heat capacity ratio,
R stands for the gas constant,
T signifies the temperature of the gas in Kelvin.

Relationship between Speed of Sound and Temperature:
In the air, the speed of sound and temperature exhibit a positive correlation. As temperature increases, the average thermal motion of molecules intensifies, leading to an augmentation of the speed of sound. The relationship between speed of sound and temperature can be expressed using the aforementioned formula.

Relationship between Speed of Sound and Density:
The speed of sound is also contingent upon the density of the medium. In the case of air, the relationship between the speed of sound and air density ρ is articulated as:

$$v_s = \sqrt{\frac{\gamma \cdot P}{\rho}} \tag{11}$$

Here:
P denotes the pressure of the gas.

Impact Factors:

1. Air Pressure: Decreases with altitude, impacting sound speed. Consideration is crucial for accurate acoustic measurements at varying altitudes. [15]
2. Humidity: Affects sound speed; higher humidity leads to decreased sound speed. Consideration is vital for accurate measurements in humid environments. [16]

In summary, real-world conditions introduce complexities, making it challenging to pinpoint an exact location accurately. Identifying potential outcomes, eliminating erroneous results, and determining a plausible range that encompasses the sound source's location emerge as prudent strategies in the face of uncertainties.

4. Algorithm analysis

Building upon the fundamental principle of sound source localization based on the angle of arrival, an innovative application is designed with the aim of accurately calculating the location of acoustic events. This application is specifically tailored for deployment in a system comprised of units, each equipped with two microphones. The objective is to enhance the precision of sound source localization by leveraging geographical information and speed of sound considerations.

The input Data for this Complex Application comprises GPS Coordinates (Latitude and Longitude) of each Unit, the Speed of Sound in the given environment, and the Anticipated Location of Sound Events. These units are strategically positioned, forming a sensor network that creates a distributed system capable of triangulating acoustic event sources with higher precision. However, following discussions with my supervisor, we decided to initiate the project in a simplified manner: replacing GPS coordinates with Cartesian coordinates.

The operational mechanism of the application involves the systematic calculation of the likelihood of the position of all potential sound sources within the sensor network. By analyzing the time delay between the reception of sound signals at the two microphones on each unit, the system determines potential angles of arrival for the acoustic events. This information, combined with the GPS coordinates and the known speed of sound, is used to estimate the potential location of the sound source.

However, recognizing the challenges inherent in real-world scenarios, the application goes beyond simple calculations. It incorporates advanced algorithms to discriminate and filter the calculated positions, systematically excluding false results. The discrimination process is crucial for mitigating the impact of environmental factors, such as echoes, obstructions, and background noise, which can introduce inaccuracies in the localization process.

Once the discrimination process is complete, the application derives logically correct position information for the detected sound sources. This logical refinement ensures that the final output is not only accurate but also reliable, offering valuable insights into the precise location of acoustic events within the monitored environment.

In essence, this application represents a significant advancement in sound source localization, combining geographical data, speed of sound considerations, and advanced algorithms to provide a robust and accurate solution. Its potential applications extend across various domains, including security, surveillance, and environmental monitoring, where the accurate identification of sound sources is of paramount importance.

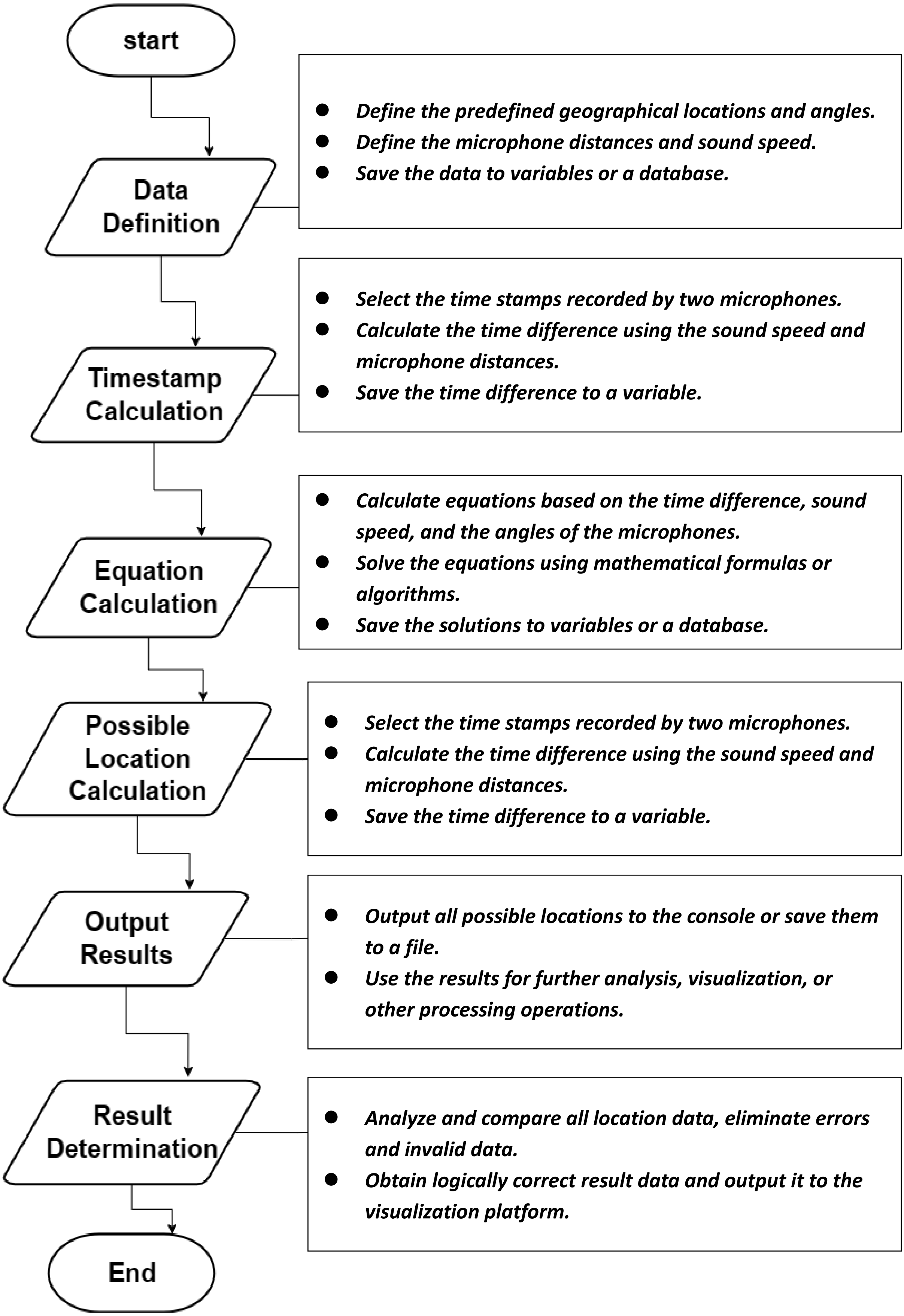

Fig. 7 Sound Localization Algorithm Flow

4.1 Timestamp Calculation

Calculate the ideal time difference using sound source (event location), device location, device setting angle, device length (distance between two microphones), and ambient sound speed. The ideal situation means that we believe that the distance between the sound source (event) and the device is much greater than the length of the device (the distance between two microphones). Therefore, we only use the position data in the same coordinate system to calculate the relative position azimuth relationship, regardless of the coordinate distance between the sound source (event) and the device. This also means that the data used in testing can be used to construct any suitable two-dimensional coordinate system within an appropriate range, independent of the units of other data. The appropriate range means that the data used should be all real numbers greater than 0. The limitation of greater than zero is that in subsequent calculations, the sign of the coordinates (positive or negative) will be used as one of the basis for determining orientation. Therefore, for the set test range and position data, both must be greater than 0. If the testing environment does not meet the requirements for suitable binary coordinates, it can be mapped and transformed into suitable data that preserves positional relationships without any impact on the calculation results.

```
% Calculate the time difference of arrival
% Input: sound source location, device information, speed of sound
% Output: time_difference: The calculated time difference of arrival

function time_difference = calculate_td(event location, device data, speed of sound)
    % Calculate angle of arrival using sound source location and device data
    % Calculate the difference in timestamps using the angle of arrival
    % Difference in formatting timestamps to make them easier to use later

    If The sound source is closer to the main microphone
        time_difference = abs(the difference in timestamps)
    else The sound source is further away from the main microphone
        time_difference = -abs(the difference in timestamps)
end
```

This code defines a function which calculates the time difference between microphones. This function generates time data based on the sound source location, device location, device angle, microphone spacing, and ambient sound speed.

By analyzing the positional relationship between the sound source (event) and the sensor device, the expected time difference (the difference between the timestamps returned by the two microphones) can be calculated. Based on the microphone on one side, set its return timestamp to 0ms, and display the order of the two microphone return timestamps in a positive or negative form in the timestamp returned by the other microphone. In this way, in the calculation of the difference in subsequent timestamps, there is no need for additional data to determine which microphone is closer to the sound source (event) location. Due to the need to simulate actual situations, uncertainties are inserted when calculating timestamps (the difference between timestamps), including timestamps offset caused by frequency, device length errors, and constantly changing environmental sound speeds.

4.2 Exploration of Potential Results Using Timestamps

After obtaining the necessary data, the location of the sound source (event) can be calculated backwards. Due to the fact that the deviation (uncertainty) between sensor length and environmental sound velocity objectively exists in reality and cannot be known during measurement, preset data that are known and known are used for reverse calculation of sound source (event) location. Thus, the impact of uncertainty can be compared in the results, the accuracy of this algorithm can be evaluated, and the error in the obtained results can be estimated.

```
% Calculate the slope and intercept of the straight line where the angle of arrival lies,
   and determine its domain.
% Input: timestamps, device information, speed of sound
% Output: Matrix containing slopes, intercepts, and domains of two straight-line functions

function set_fun = calculate_function(timestamps, device data, speed of sound)
      % Calculate angle of arrival using timestamps
      % Calculate the slope of a straight line function
      % Calculate the intercept of a straight line function
      % Use the order of returned timestamps
         to determine the domain of the straight line function
      % Each device can find two such straight-line functions with defined domains,
      % Store the slope, intercept and domain of the straight-line functions.

      set_fun = [F1_slope, F1_intercept, F1_domain; F2_slope, F2_intercept, F2_domain]
end
```

Firstly, by utilizing the position of the sensor, sensor angle, sensor length, ambient sound velocity, and the two timestamps returned, the calculation of the line function can be achieved. Two line functions intersecting at the sensor position will be obtained, and there will be deflection angles (slope changes) with the same numerical value but different directions (clockwise and counterclockwise) as the sensor angle. Simultaneously using data such as sensor angle and two timestamps returned, determine the directional relationship between the sensor device and the sound source (event) location, and use it to determine the effective domain of the two line functions found.

```
% Calculate the intersection point of two straight line functions
% input: two sets of equations
% Output: list of intersection

function psb_point = calcul_point(f1,f2)
    % f1 , f2: The set_fun determined in the previous step each contains two sets of
    straight line function data (slope, intercept, and domain )
    %Make four judgments on the intersection of the two sets of straight lines;

    If: the intersection is within the definition domain, store the intersection data;
    else: it exceeds the definition domain, use [0, 0] for placeholder;
    % psb_point(n,1:2) = [x,y] or [0,0];
end
```

Due to the uncertainty of the data, it is almost impossible to find a point where two line functions intersect within their domain. Therefore, when searching for possible points, the intersection point of the line functions of two sensors within the defined domain will be calculated using the set of line functions of each sensor. The number of results obtained should be one, two, or four. And among these results, only one is consistent with the logic of computational theory. So far, all results will be recorded and checked in the following section to eliminate erroneous results. And the remaining result, due to the presence of uncertainty, is likely to not coincide with the position of the sound source (event), so we call it the logically correct result (position).

4.3 Exclusion of Erroneous Results

This section is divided into three parts to conduct three checks on the results obtained in the previous section.

The first Checkpoint inspection is to check whether the results obtained are within the scope defined by the test. Due to the effective measurement range of sensors in reality, a range is defined during initial setup, and all test data is within its defined range. Therefore, if the calculation result is outside the range, it is directly excluded.

```
% Checkpoint 1: Confirm whether the result point is within the working area
% Workspace: minimum maximum value of the boundary
% Input: list of result points and working area
% Output: list of result points after updated

function cpl1 = cp1(list of points, min and max of area)
    % Traverse the list of points
    if: the result point is in the work area
        % store the result point location
        cpl1(n, 1:2) =[result point]
    else: continue
end
```

The second Checkpoint inspection is to check the distance relationship between the calculation result and the sensor. This distance relationship refers to the relationship between the position of the sound source and the position of the two sensors, which must be relatively close to one of the sensors and relatively far from the other. This can be obtained by comparing the order in which the timestamps are returned (the size of the timestamps). However, if the result of the calculation is contrary (not consistent), it means that the result does not meet the known conditions and should be excluded. In practice, this part of the check should be done using timestamps. At this stage, for the time being, the location of the test sound source (event) and the position of each sensor are used for distance comparison. This is due to the timestamps in the test data so far, making it impossible to complete this part of the comparison. Therefore, as an alternative, the distance relationship is directly compared using position.

```
% Checkpoint 2: Check whether the distance relationship matches
% Input 1: list of result points
% Input 2: sound source location (SSL)
% Input 3: All device data location [Dv1, Dv2, Dv3, …]
% Output: list of result points after updated

function cpl2 = cp2(list of points, SSL, [Dv1, Dv2, Dv3, …])
    % Group all devices into groups of two [[Dv1, Dv2], [Dv1, Dv3], [Dv2, Dv3], …]
    % Compare whether the two sets of comparison results are the same
    If: they are the same, save the result point.
        cpl2(n, 1:2) =[result point]
    else: continue
end
```

The third Checkpoint inspection is to check the azimuth (angle) relationship

between the calculation result and the sensor. For the two sensors used to calculate the results, the azimuth relationship between them must be correct. However, if other sensors are present, the orientation of these sensors will be used as a basis for judging whether this result is consistent. There are three possible ways to check:
1) Use the location of the calculated result to calculate the difference between its timestamp and the timestamp of other sensors, and compare it with the difference of the known timestamp;
2) Calculate the angle of arrival to other sensors using the position of the result, and compare it with the known angle of arrival;
3) Use the linear equation with a defined domain obtained from other sensor data to determine whether the calculated result is crossed.
Methods 1) and 2) are largely the same, while method 3 is relatively complex to calculate.

```
% Checkpoint 3: Confirm whether the result point is suitable for other devices
% Input: list of result points and Devices data
% Output: list of result points after updated

function cpl3 = cp3(list of points, Other Devices data)
    % Other Devices: If the result point is calculated from the data of Dv_1 and Dv_2,
      then the other devices are all devices other than this [Dv_3, Dv_4, Dv_5, ...]

    If: The resulting point coincides with the angle of arrival of all other devices
        cpl3(n, 1:2) =[result point]
    else: continue
end
```

Due to uncertainties in the test environment and the need to calculate and validate specific data without accounting for these uncertainties, different deviation parameters (uncertainty parameters) are required for various detection methods. The deviation parameters hold distinct meanings in different inspection methods:

1) Since even logically correct results are subject to uncertainty and deviations from the value of the difference between the known timestamps, a parameter is needed to determine the maximum value (maximum range) of the deviation.

2) Since even logically correct results are subject to uncertainty and deviate from the value of the known angle of arrival, a parameter is required to determine the maximum value (maximum range) of the deviation.

3) Since even logically correct results are subject to uncertainty, there is a high probability that a straight-line equation with a defined domain will not (almost) cross the calculated result. Therefore, a distance parameter is required, which is used as the radius to determine whether the linear equation with a defined

domain obtained by other sensor data passes through the region within the parameter.

Since the actual parameters are not used in the test phase, these parameters are temporarily estimated by backward extrapolation, such as the function based on the check method 1) that has been completed at this stage, and the parameter used is 0.05ms, which means that the calculation result with a deviation greater than 0.05ms is not considered to be a logically correct result. However, this parameter is not accurate enough at the moment, and more analytical testing is needed to ensure that the only logically correct result can be filtered out of each set of calculations using such parameters.

As for the sequence of the second and third part checks, after logical analysis and practical testing, we do not consider it to have an impact on the results. As to whether the second part of the inspection and the third part of the examination are necessary at the same time, after logical analysis and practical testing, the answer is yes. In tests, there may be cases where non-logically correct results can be checked by the second or third, especially if the number of sensors is small. Therefore, it is necessary to retain both parts of the test at the present stage of the test.

5. MATLAB application with User Interface

The user interface of this application is created using MATLAB application design tools. The design of the graphical user interface (GUI) emphasizes simplicity and functionality, allowing users to seamlessly interact with the underlying computing process.

This application has comprehensive functionality aimed at providing precise and in-depth results. Users can input relevant parameters such as sensor timestamp, location, and environmental factors through specified fields. Then, data validation is used for data detection to ensure correct data entry. During calculation, the application uses advanced algorithms to determine the angle, distance, and potential intersection points of the sound source relative to the sensor array.

Configurable parameters: This application allows users to configure parameters such as sensor position, sound source characteristics, and verification tolerances, providing flexibility for different scenarios.

Verification module: Three different verification modules (position, distance, and angle alignment) ensure the accuracy and reliability of the calculation results. Users will receive feedback on the validity of each result.

Real time computing: Users can initiate real-time computing based on the provided sensor and sound source data. This application uses advanced algorithms to determine potential sound source locations.

Interactive visualization: Coordinate system display diagram visualization illustrates the spatial relationship between sensors and potential sound sources. Users can interact with these visual elements to gain deeper insights.

5.1 Introduction to Software Operation Interface

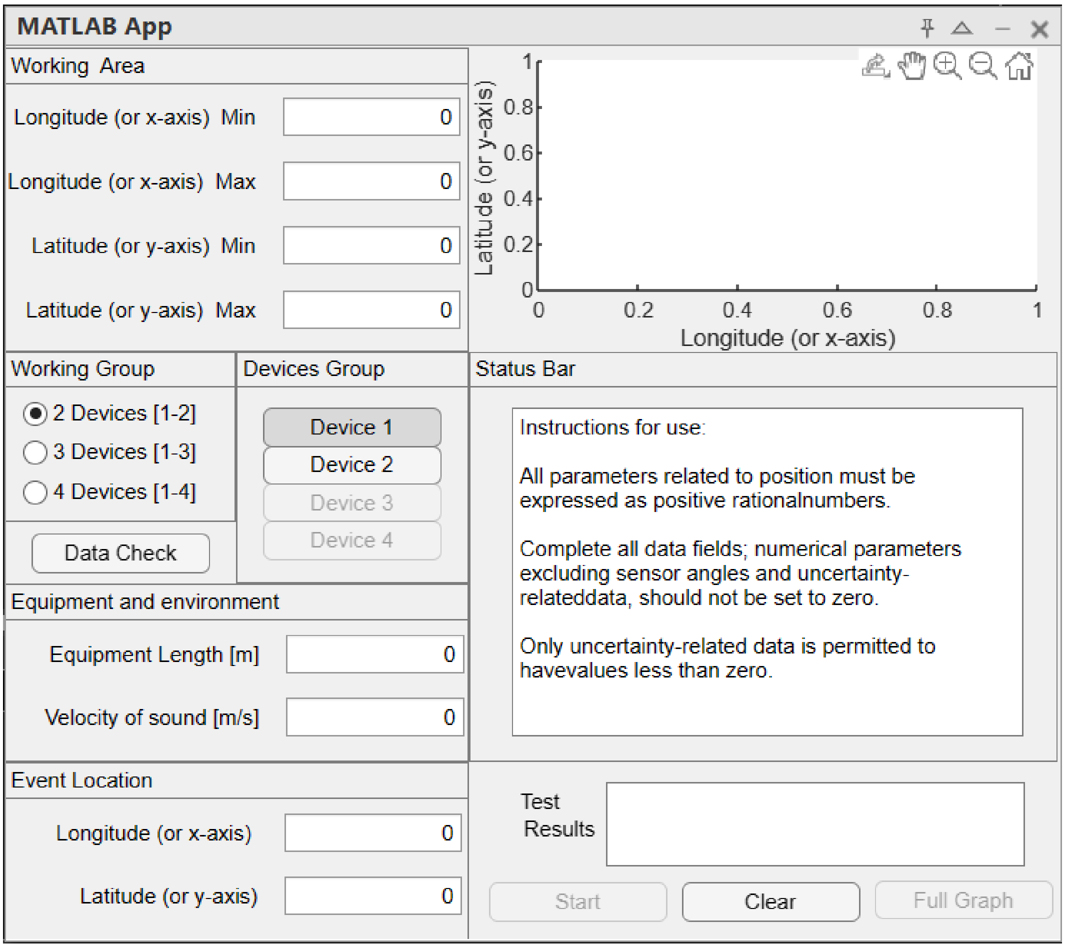

Fig. 8 Software Operation Interface

On the left side are the basic information settings, including:

Working Area: Defines the testing range within the Cartesian coordinate system. It specifies the maximum and minimum values for the horizontal and vertical coordinates, and all subsequent coordinate-related data must fall within this defined domain.

Working Group: Determines the number of sensors used in the test. Currently supports an optional range from 2 to 4 sensors.

Device Group: When selecting 3 or 4 devices in the Working Group, buttons for Device 3 and Device 4 will be unlocked. Pressing these buttons in this area will switch the right side to the corresponding device data page for configuring data for each device.

Data Check Button: Used for checking data before performing calculations, an essential step in the process.

Equipment and Environment: Specifies parameters for sensor device length (distance between two microphones) and the ambient speed of sound used in the test.

Event Location: Selects the location of the test sound source (event) within the previously defined test area.

On the right side from top to bottom:

Thumbnail: After completing calculations on the data, a thumbnail with the computed results will be generated in this area.

Status Page: Initially displays prompts for entering input data. When selecting different devices, it switches to the corresponding data input page for that device. After initiating Data Check, it shows the detection results.

Test Results: Displays the results of the test after the calculations are completed.

Start Button: Available after Data Check, initiates the calculation part of the test.

Clear Button: Clears all user input data, image data, and calculation results.

Full Graph Button: Available after completing calculations and generating a thumbnail, opens another window to display a clearer image.

5.2 Application User Guide

In order to facilitate a rapid and proficient initiation for users, this section endeavors to elucidate the procedural guidelines pertaining to the application. The elucidation will commence with an overview of input data and extend to the scrutiny of computational outcomes, providing a comprehensive delineation of sequential operational steps.

Commencing with the input data, the instructional discourse will systematically expound upon the procedural nuances, progressively guiding users through each step. The aim is to furnish users with a thorough understanding of the application's operational intricacies, ensuring a seamless transition from data input to the examination of computational results.

This instructional segment is devised to serve as a pedagogical tool, enabling users to navigate through the intricacies of the application with ease. Emphasis will be placed on elucidating each procedural facet comprehensively, fostering a user-friendly experience.

5.2.1 Enter various parameters

Before entering the parameters, users must acquaint themselves with the prescribed data usage conventions of this program. All parameters related to spatial coordinates must be rational numbers greater than zero. Users are required to populate all data fields, except those about sensor angles and uncertainty-related data. If sensor angles are left unspecified, the default is set to zero; however, it is recommended to input values for at least one less than the chosen sensor count.

Except for sensor angles and uncertainty-related data, numerical parameters other than zero are mandatory (as the default value is zero, users must input these values). Specifically, only sensor angles and uncertainty-related data are permitted to be less than zero.

While inputting parameters, users are earnestly advised to follow a basic left-to-right, top-to-bottom sequence, as this approach effectively mitigates the possibility of data omissions. Initiate the process by inputting parameters for the "Working area" section, which determines the test range and establishes the coordinate system for all positional systems. Users have the flexibility to input longitude and latitude limits for a coordinate system that closely mirrors real-life scenarios or opt for a meter-based coordinate system for swift implementation.

Subsequently, users are encouraged to select the "Working group" to potentially unlock additional sensors for use. Following this, users can configure devices in the "Devices Group." Clicking on different numbered device buttons will transition the information panel in the middle-right section to the corresponding device's data input interface.

Upon completing the data entry for all devices in the "Working group," users are advised to first fill in hardware information (Equipment Length) and environmental details (Velocity of sound). This signifies the completion of the test environment configuration. Except for the Velocity of sound and uncertainty-related data, the majority of these data remain constant in real environments. Changes in the Velocity of sound will be provided by other devices in real-world scenarios, rendering these data as background environmental parameters.

Finally, only the data for "Event Location" remains to be filled. Users can determine device placement, monitor changes in the measurement environment, and estimate the impact of uncertainties. However, it is only after an event occurs that relevant data can be obtained and calculations for the event location can commence.
Additionally, users can employ the "Data Check" button at any stage of data entry to inspect the inputted data for accuracy.

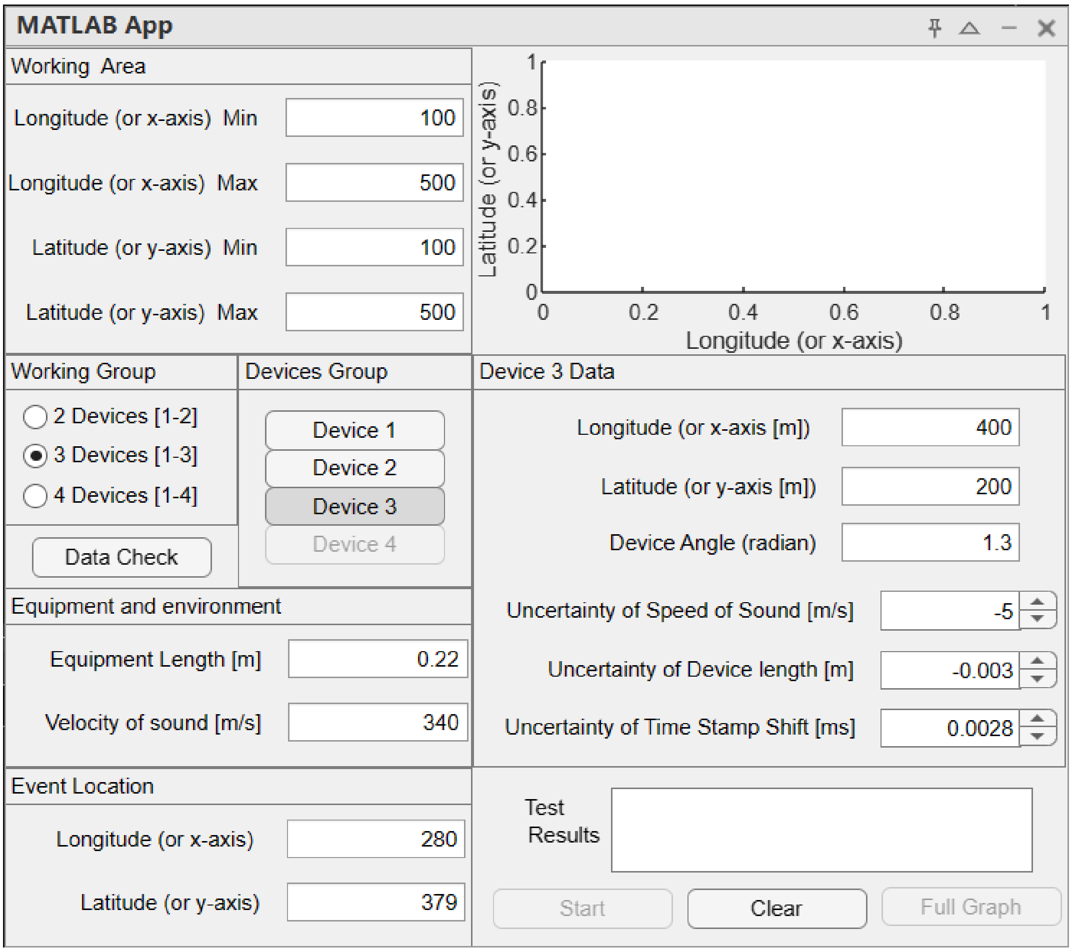

Fig. 9 Enter various parameters

5.2.2 Data detection

Upon completion of input of various parameters, it is imperative to perform data verification. In the event of errors, a notification specifying the nature of the error will prompt users to rectify and retest. Once data verification is successfully conducted, the information page will indicate readiness for subsequent operations, presenting systematically organized data for user manual scrutiny. Subsequently, users may proceed by clicking the "Start" button to commence the testing computation.

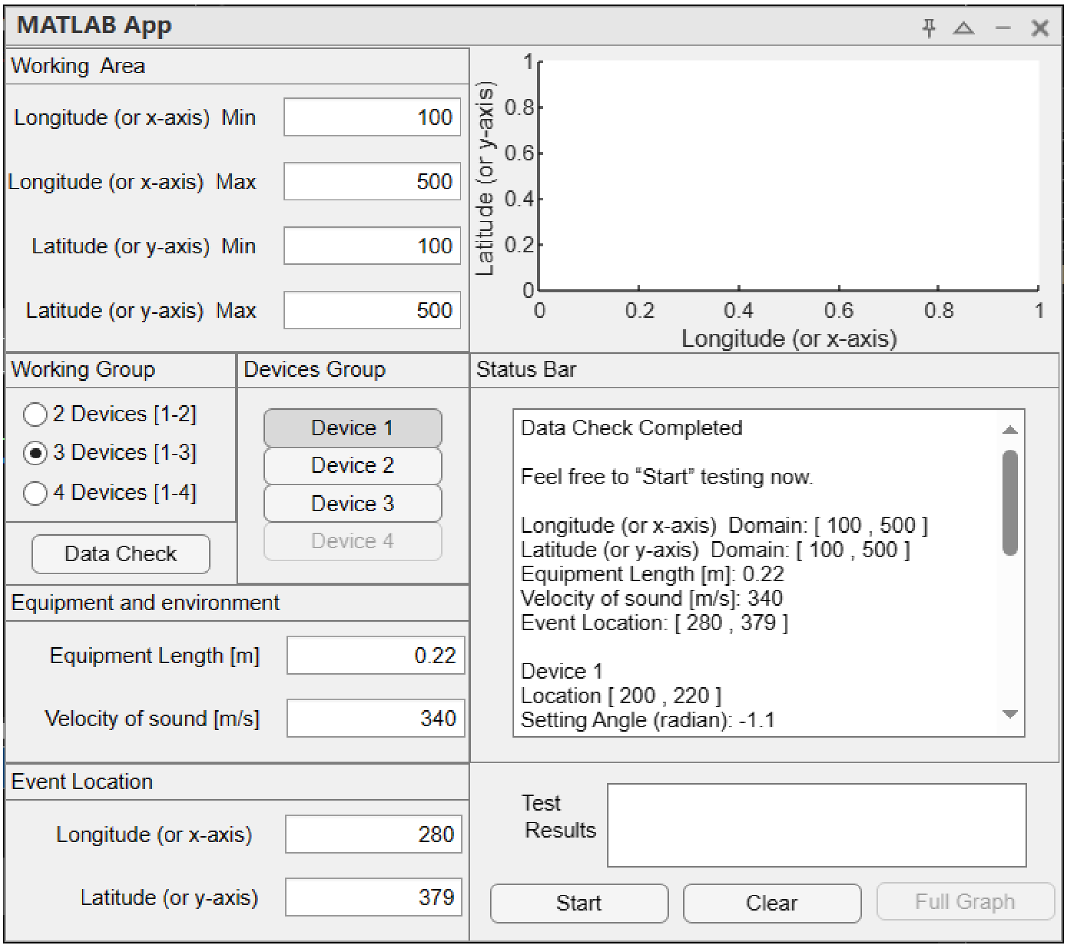

Fig. 10 Data Detection

5.2.3 Display of calculation results

After the data passes the detection, click the start button to perform the calculation. The calculation results will be displayed in the coordinate system. Then you can click the “Full Graph” button to zoom in and display the coordinate results.

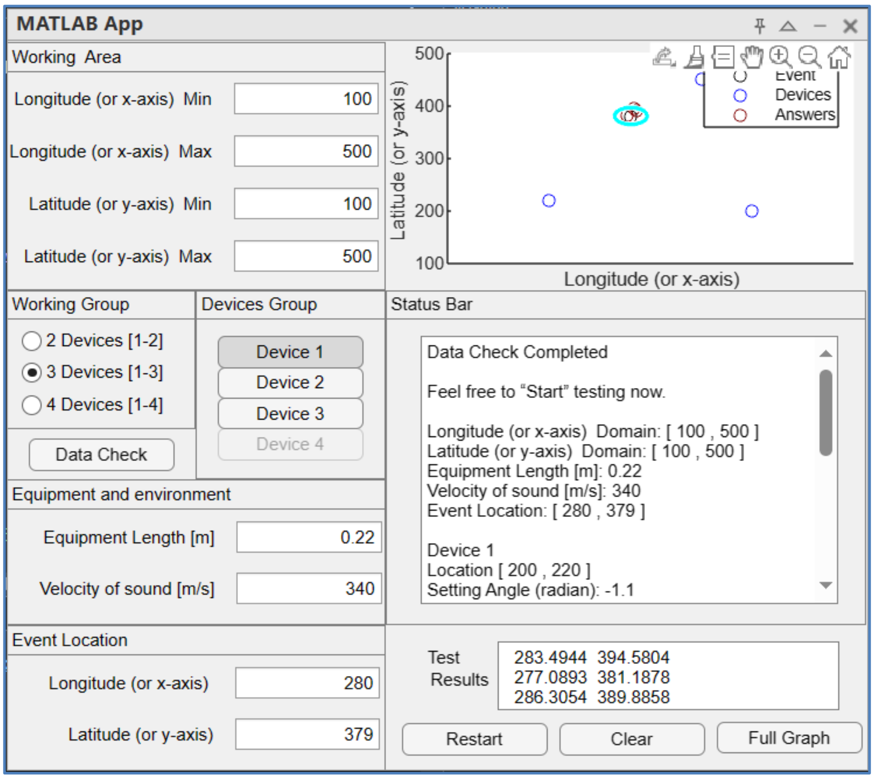

Fig. 11 Display calculation results

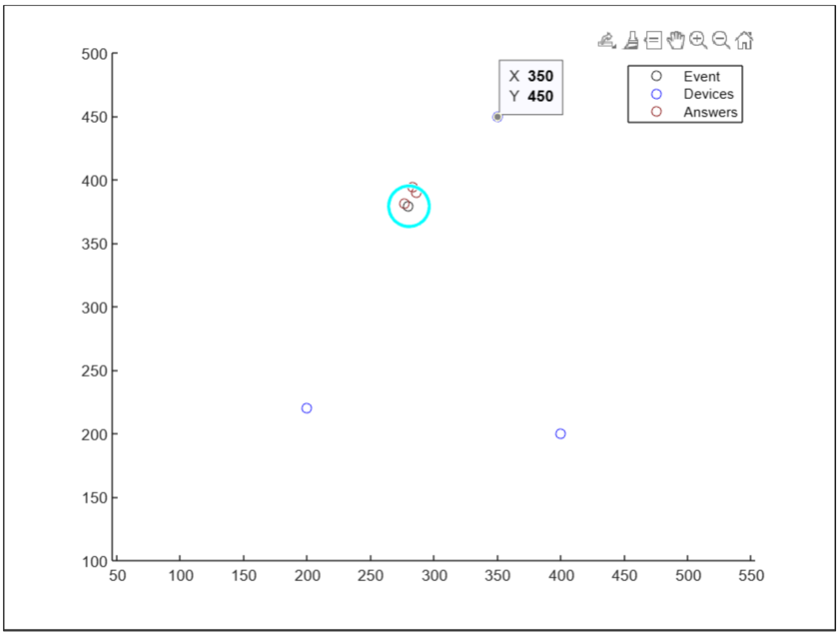

Fig. 12 Display full graph

5.3 Data Verification Process

Before initiating the calculation process, a comprehensive data verification is mandatory to ensure the accuracy and reliability of input parameters. The data check process is divided into three sequential layers:

5.3.1 Working Area Validation:

Confirm the defined working area parameters, including the boundaries of the Cartesian coordinate system (maximum and minimum values for horizontal and vertical coordinates), ensuring consistency with the intended testing environment (minimum values must be positive rational numbers, and maximum values must be greater than their corresponding minimum values).

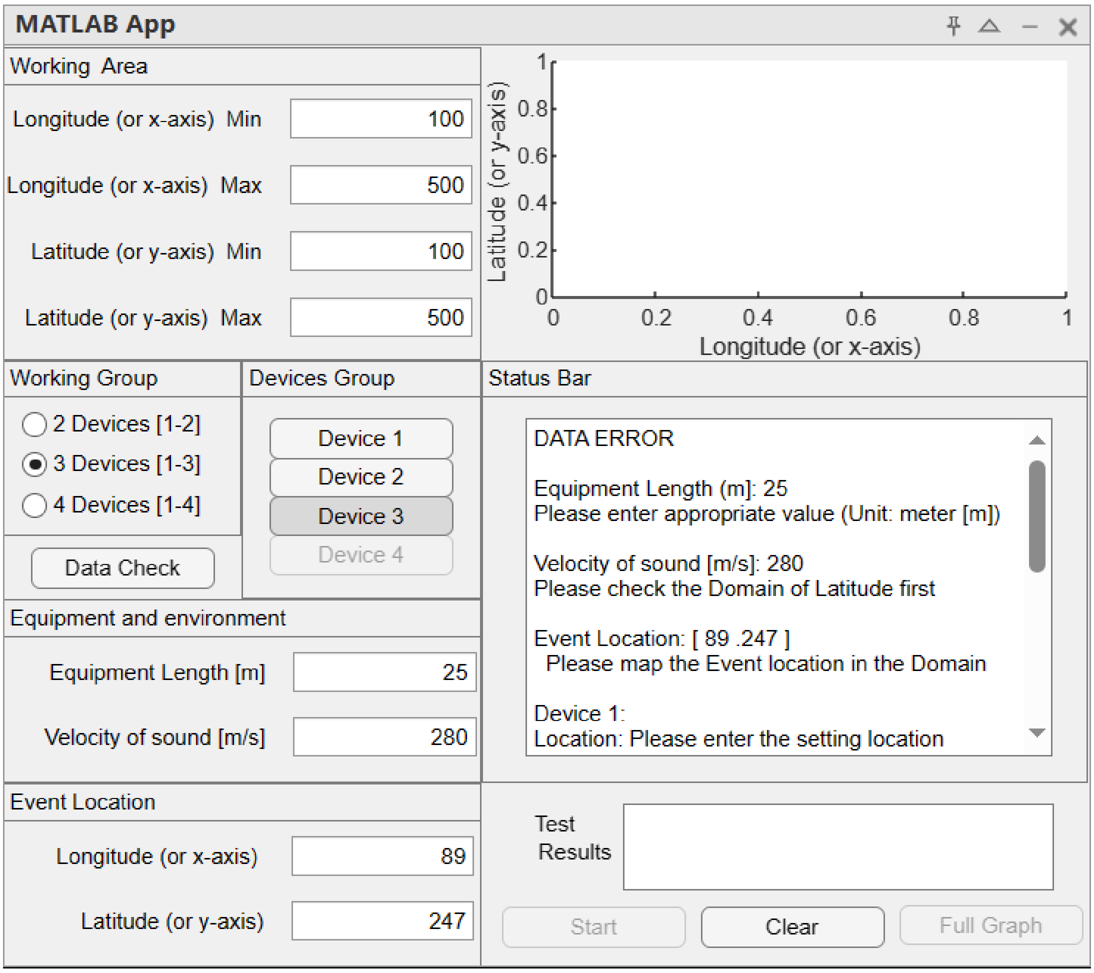

Fig. 13 Working Area Validation

5.3.2 Event and Devices Location:

Following the completion of Working Area Validation, validate the chosen location for the test sound source (event) and the positions of devices, ensuring they fall within the specified coordinates. However, feedback is not provided for situations where users set these positions on the Working Area boundaries or very close to the boundaries at the current stage.

5.3.3 Verification of Other User-Set Parameters:

Check other parameters set by the user and provide feedback for data significantly deviating from the simulated testing environment, such as a sound speed of 280 m/s. Users will be informed of acceptable data ranges.

By systematically conducting these data verification steps, users can enhance the reliability and precision of the testing simulations, ensuring that input parameters are within acceptable and meaningful ranges.

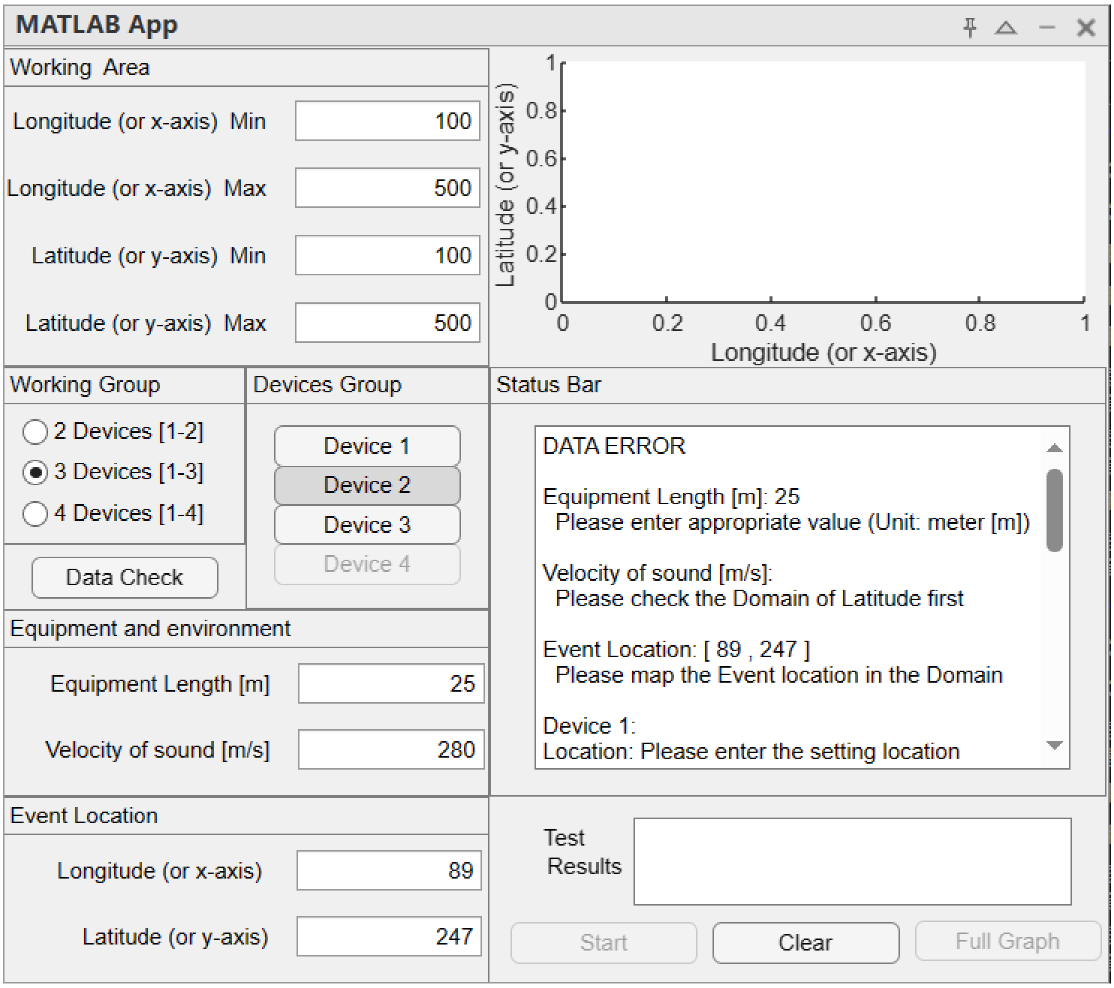

Fig. 14 Data Detection

6. Test simulation and analysis

6.1 2-Sensor Test and Analysis

We commence the testing phase with an analysis of results from scenarios involving two sensors. Previous analyses indicated that, for two arbitrarily positioned sensors and a sound source (event), initial intersection calculations may yield 1, 2, or 4 intersection points.

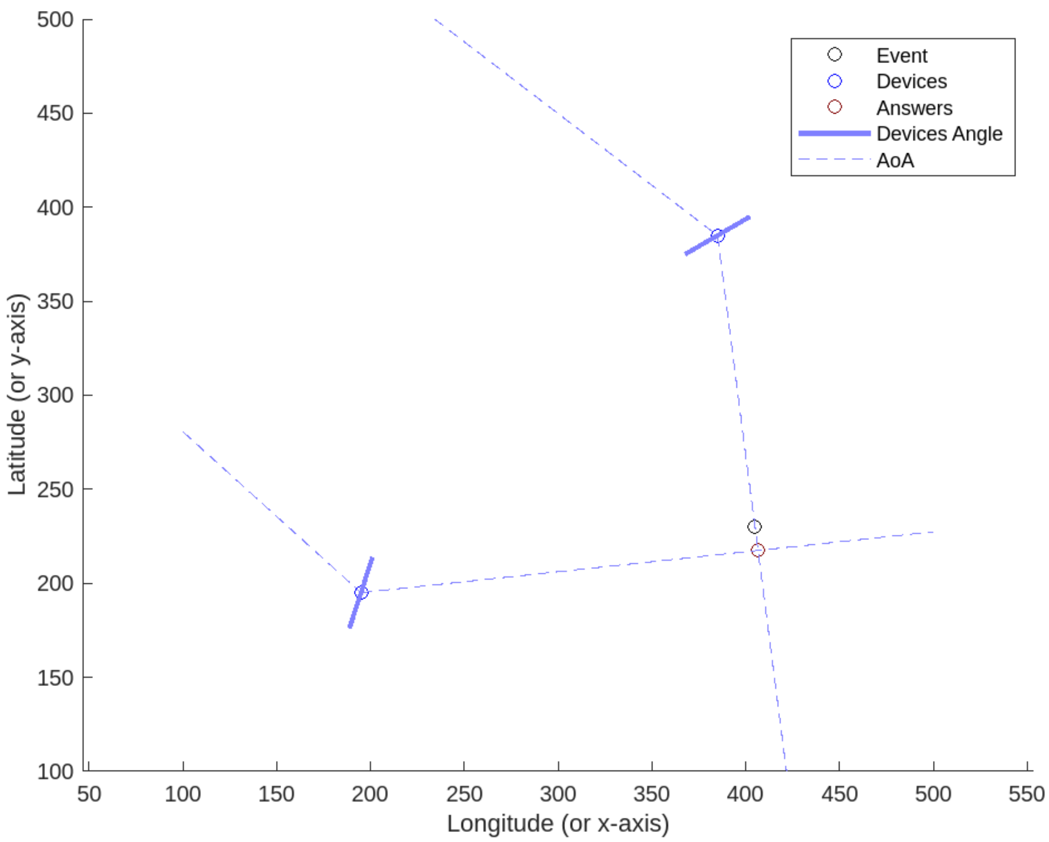

Fig. 15 2-Sensor Test and Analysis

However, in cases where more than one result is obtained, it becomes challenging to eliminate erroneous outcomes, hindering the confirmation of logically correct results, irrespective of uncertainties. As illustrated in the three figures above, using different sound source positions while keeping other parameters constant resulted in varying numbers of outcomes.

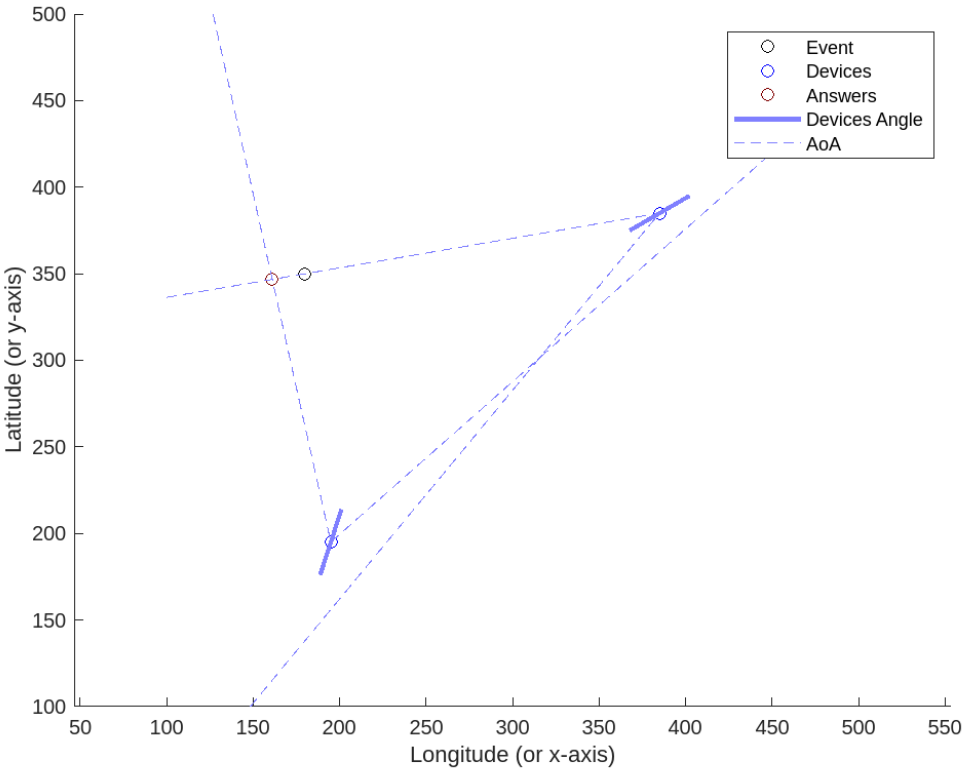

Fig.16 2-Sensor Test and Analysis

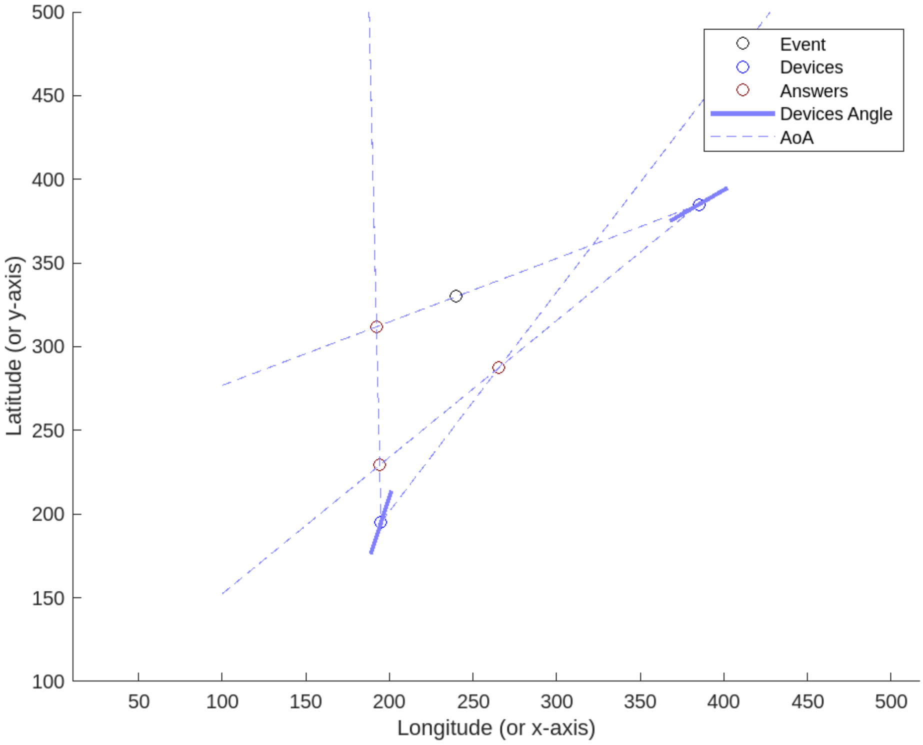

Fig. 17 2-Sensor Test and Analysis

We observe that although the third-part check system may not be effective, the second-part check system can exclude a certain number of erroneous outcomes, barring specific circumstances.

Moreover, in scenarios with only two sensors, a significant uncertainty blind zone exists in velocity measurement. This zone is characterized by logical correct results being erroneously discarded due to the introduction of uncertainty. As depicted below, the relative distance relationship between sensors and the comparison of sound source (event) positions may contradict logical correct results due to uncertainty.

Insert Figure illustrating the uncertainty blind zone

By adjusting uncertainty settings, we can visually observe the impact of uncertainty on results in this phase. By introducing uncertainty only to the data received by one sensor, while the other sensor's uncertainty remains at 0 (perfect data), we gain a more intuitive assessment of the "impact of uncertainty on results."

6.2 3-Sensor Test and Analysis

With three sensors, the algorithm's ability to pursue correct results improves. The count of logically correct results increases to three, making it more challenging to mistakenly eliminate them. The algorithm is likely to locate the sound source within the region delineated by the three correct results.

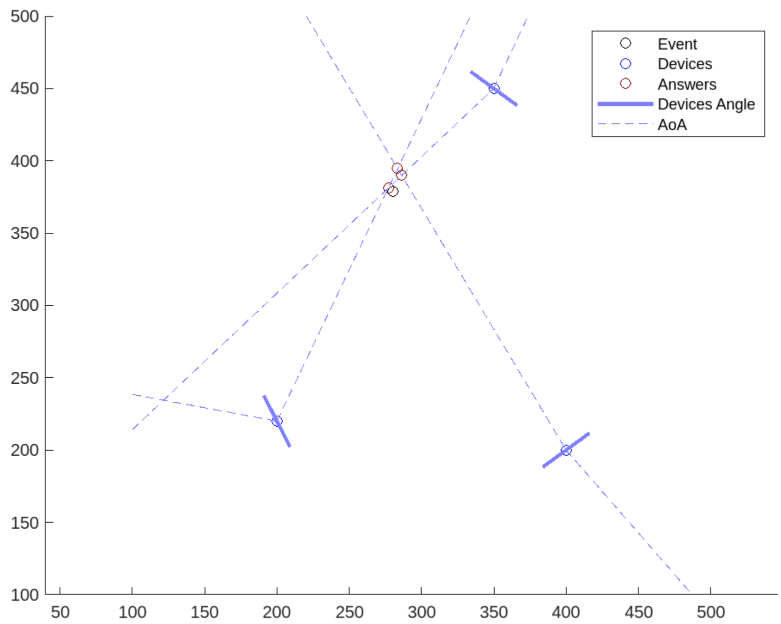

Fig. 18 3-Sensor Test and Analysis

6.3 4-Sensor Test and Analysis

When four sensors are employed, there seems to be minimal improvement compared to the three-sensor scenario. However, having an additional sensor can address several practical issues. The current timestamps are computed using the arrival angles to each sensor, assuming the validity of Huygens' principle throughout. Yet, in real scenarios where the sound source is too close to certain sensors, the reliability of data from those sensors decreases significantly. Using four sensors, each satisfying the conditions of Huygens' principle, ensures the accuracy of the detection, even if some sensors are damaged or under maintenance.

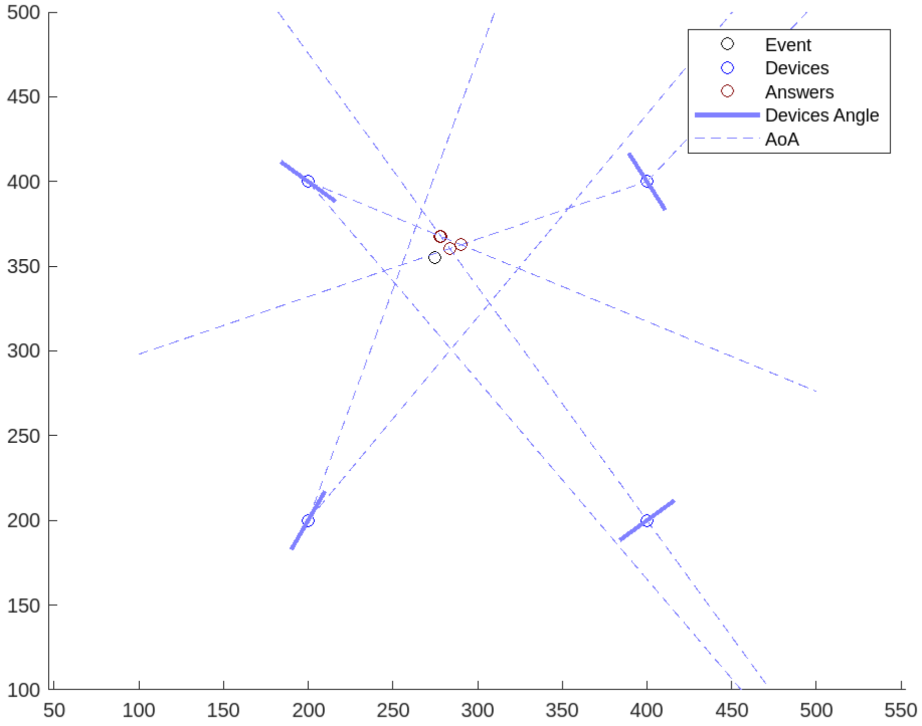

Fig. 19 4-Sensor Test and Analysis

7. Sound Source Localization Optimization

In the preceding chapter, an in-depth analysis of the testing results shed light on the intricacies and challenges associated with the existing checkpoint system. The examination revealed notable limitations, particularly in the face of uncertainties that extend beyond the anticipated ranges. Building upon this understanding, the subsequent sections outline strategies and considerations for optimizing the sound source localization algorithm.

7.1 Refining the Checkpoint Functions

Within the realm of the current checkpoint functions, specific challenges have been identified across its distinct stages. These challenges are addressed individually to enhance the overall efficacy of the algorithm.

7.1.1 Enhancements to the First Checkpoint Function

The initial checkpoint stage, designed to eliminate results beyond predefined ranges, is subjected to critical scrutiny. Despite its perceived redundancy, this stage proves crucial in scenarios where the sound source approaches predefined boundaries under the influence of stronger uncertainty influences. Strategies for refining this stage are explored to mitigate potential errors and ensure the inclusion of logically correct results.

7.1.2 Improvements to the Second Checkpoint Function

The second checkpoint function, based on relative distances, encounters challenges related to uncertainties and minimal distances between the sound source and sensors. The introduction of a range parameter sensitive to differences in relative distance is proposed as a potential solution. Considering sensor data in timestamps, this range parameter is meticulously defined to represent a minimal time interval, addressing exclusion concerns and enhancing the system's robustness.

7.1.3 Optimization of the Third Checkpoint Function

The third checkpoint function, leveraging arrival angles and a corresponding range parameter in tandem with the second checkpoint, is scrutinized for efficacy. The relative range parameter's potential definitions—time-based or distance-based—are evaluated in terms of simplicity and accuracy. This optimization aims to further filter out erroneous results and bolster the overall accuracy of the sound source localization algorithm.

While acknowledging existing optimization opportunities, the logical filtering approach demonstrated effectiveness, particularly in scenarios with fewer sensors.

7.2 Adaptive Strategies: Sensors and Error Iteration

As the number of sensors increases, a discernible trend emerges in the density of logically correct results around predefined sound source locations. This revelation prompts consideration for adaptive strategies that leverage the benefits of a higher sensor count, potentially rendering the existing checkpoint system obsolete.

Introducing the concept of an acceptable error, defined as the maximum tolerated error in detection results during practical use, offers an innovative alternative. An iterative search method is proposed, wherein results within the acceptable error range are systematically analyzed, grouped, and regions with the highest density of logically correct results identified. This approach, emphasizing adaptability to varying sensor counts, marks a departure from the conventional checkpoint system, setting the stage for a more nuanced and effective sound source localization methodology.

8. Conclusion

In conclusion, the present application and its algorithm, albeit in an early developmental stage, have revealed notable imperfections during testing, necessitating significant enhancements for alignment with real-world scenarios. The existing checkpoint system exhibits limitations, particularly under the influence of uncertainties beyond expected ranges. A logical filtering approach has been implemented through various checkpoint stages, each with its associated shortcomings. Despite optimization opportunities, this approach has proven effective, especially in scenarios with fewer sensors.

An observed trend in scenarios with an increased number of sensors suggests the potential obsolescence of the existing checkpoint system in favor of a more exhaustive approach. Introducing the concept of an acceptable error provides an alternative method, where an iterative search considers results within the acceptable range, allowing for the identification of regions with the highest density of logically correct results.

In conclusion and looking towards future directions, while the current application and algorithm exhibit promising features, ongoing efforts should prioritize the refinement of the checkpoint system, exploration of adaptive range parameters, and implementation of an iterative search method based on acceptable error. Incorporating real-world data is crucial for algorithm validation and continuous improvement.

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