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2-Dimensional Mapping of Reverberant Environment with Acoustic Array Sensors Using Decomposition of Unknown Signals into its Original and Echoes

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Abstract—**Determining the location of the source of unknown newly sensed acoustic signals in forest-like reverberant environments is crucial for both first-aid practices and security-based operations. It also, in direction finding, as in RF-based systems, studies in acoustics begun to gain importance over the years. While the usage of echo signals and the acoustic signal itself allows it to be possible to prepare a two-dimensional map of the reverberant environment, a time expansion in packet size contains information about the type (depth, softness, hardness) of that obstacle in the background. This information may crucially be helpful from the tactical point of view in military conflict zones. Besides the military importance, the proposed method has great potential applications in architecture and concerts (music industry). In this study, a 2D microphone array system is designed to collect data (unknown signal data packet) and those obtained. The proposed approach decomposes the received packet into its components, both as original and echoes, as a first step, then identifies the fingerprint signature of the original signal, matches the echo (reached from the obstacles) with the originals employing a predetermined signature, and allows two-dimensional mapping. It plots the location of any obstacle in the reverberant environment two-dimensionally with a success rate of 87% with ±2m resolution. Additionally, the length of the 10-ms packet extends from 13ms to 18ms depending on the type of vegetation canopy. Time expansion of the signal allows for a details 2D map of a reverberant environment in terms of obstacle types.**

Keywords— ***acoustic signals, acoustic sensors, direction finding, time difference of arrival.***

I. INTRODUCTION

The interaction of the acoustic signals with the environment in which it is located allows it possible to determine the physical properties of the obstacle and the direction and location of the acoustic sources. Based on this reality, acoustic signals have found a wide range of application areas, such as ultrasonography for biomedical imaging, underwater sonar systems for defense industries, etc. The propagation distance of signals (time of arrival of the signal to target), incoming angle of signal, and the content of signal packets are fundamental data to be used for. Regarding the tactical point of view, this information is crucial for first aid decisions and military conflict zones in the nearby forest. The energy of acoustic signals is weakened because trees reflect, absorb, and transfer. The fundamental forest parameters are the type, trunk diameter, number of trees, bark, leaf density, and canopy areas. They directly affect the propagation of the acoustic wave in the environment. As a reverberant environment, the forest can change the transmitted packet's size such that the signal packet's length does not vary when metallic reflectors or rock-like obstacles reflect it. Still, it widens when reflected by soft mirrors such as trees or bushes.

Both direction finding and localization studies in the forest require understanding the effect of a single tree on acoustic signals and the total forest effect. Behavior and measurement procedures of acoustic signals in the forest area are discussed in the literature [1-3].

An experimental study on acoustic attenuation in different forest types and tried to verify part of the Nord 2000 model is conducted in [4]. They showed that as the density and trunk dimensions of the trees increase, the weakening they cause increases; in most cases, the predicted reduction of the ground effect is consistent. On the other hand, it is proposed that time and frequency analysis of acoustic scattering by a bistatic method by an elastic cylindrical shell reveals that the arrival time of echoes and the elastic waves moving around the shell is a function of the bistatic angle [5]. Sound propagation modeling in forest areas and used the transmission line matrix method during the analysis is also studied in [6]. It is discussed that evolution of the theories and experiments for each aspect related to sound propagation in a forest as well as ground effects, scattering of sound, and meteorological effects.

There are different studies in the literature about the techniques to be implemented. A new broadband acoustic direction-finding method in reverberating environments is proposed using an acoustic vector sensor (AVS) [7]. They achieved to determine the source directions with a 5-degree deviation in each frequency of each received sound without limiting the frequency range. It is also studied on determining the angle and distance of the acoustic signals with plane microphone arrays [8]. The arrays they use to place the microphones are equilateral triangles and squares. They estimated the slope and spread to the microphones placed in the corners of the series by using the delay differences created by the sound coming from the source at different times. They underline that the increase in array elements results in higher sensitivity. They stated that their quasi-L1 autocorrelation algorithm and interpolation methods increased the delay rate estimation and accuracy.

The localization and counting method of multiple sound sources is discussed in [9]. They prefer a circular microphone array in one dimension rather than linear arrays not to observe uncertainties. The main pillars of their work are based on their prediction of the number and direction of arrival (DOA) of active sources by identifying the time-frequency regions where one source is predominant and matching histograms with other receivers. In their results, their success in estimating more than one source is more than 87%, and their overall success rates were 93.52%.

A new approach to the localization system of acoustic sources and a new method for security cameras with a microphone array in the external environment is presented in [10]. They introduced new approaches to filtering out incoming signals with cross-correlation, using a general windowing function to reduce environmental noise and improve arrival time delay using array microns. They provide a four-microphone surveillance camera to localize the direction of an audio source.

Acoustic source localization by using time delay estimation methods is discussed in [11]. Dominant Frequency Selection (DFSE) algorithm is implemented and applied. In their scenario, M acoustic sensors are in known locations and are to estimate two or three-dimensional coordinates of the acoustic sound source. They observed comparable results with those conventional time-delay methods such as GCC and GCC-PHAT.

It is assumed that the forest-like reverberant environment can be modeled as several dominant scatters (each producing an echo); all these scatters and the unknown source are located in the near-field of the array. In other words, passive ranging methods can obtain reliable range estimates, and the emitted signal from the unknown source is short-duration. The individual components of the received signal at each sensor are separate in time. With these assumptions, the particular features of the received signal at each sensor can easily be extracted. The source and obstacles (scatters) can be located, one at a time, using a TDOA-based passive localization method. When dragging a newly sensed unknown signal, the received packet is decomposed into its components both as original and echoes as a first, then identifies the fingerprint signature of the original signal, matches echoes (reflected from the obstacles) with originals employing predetermined signature.

In this study, the interaction of tree and forest-like structures with acoustic signals and the effect of this interaction on navigation success are also discussed. A 2-dimensional four elements Acoustic Array Sensor (AAS) is used to collect data. C++-based software, which uses angle transformation and phase correlation approaches from differential arrival time, was implemented to evaluate and interpret the collected data. The manuscript is organized as follows; Section II is for the problem statement and measurement set-up, Section III is for Mathematical Expression, Section IV is for Results, and Section V is for discussions about results and future works.

II. PROBLEM STATEMENT AND MEASUREMENT SET-UP

As well as in any reverberant environment, finding the location of unknown acoustic signals in a forest environment is critical for first aid and emergency cases. That also becomes much more vital during various operations. There will be much multipath propagation without knowing environmental and signal characteristics. Consequently, original unknown acoustic signals and their first and secondary echoes, as well as other combinations of echoes, will be received by the elements of AAS in that environment. Achievement of decomposition of an unknown signal packet allows us to determine the direction of each component, and this finally allows us to prepare a 2-D terrestrial map of an environment.

In order to achieve the proposed goal, measuring the time of arrival of any signal, determining the signature of newly sensed unknown signal, and doing the sub-classification of signal components as original, echo, and noise are predominant tasks. (1) expresses a signal reaching AAS from N different paths, including the original one, and echoes from the perfect reflector [3].

$$x(t)=\sum_{i=0}^{N} a(\theta_i,\varnothing_i)\,\Gamma_i e^{-\alpha t_i} s(t-\tau_i)+w(t) \qquad (1)$$

where s(t) and w(t) are the wideband acoustic source and noise signals at sensor outputs. Γ_i and α is the reflection coefficient at the ith path and the attenuation coefficient in the propagation. θ_i, φ_i and τ_i are the azimuth angle, elevation angle, and the time delay of the ith path. Since the direct path is the shortest between the source and an AAS, $\tau_i > \tau 0 = 0$ is valid for any non-zero i. The array response vector a (θ, φ) is defined by (2).

$$a(\theta,\varnothing)=[\cos(\theta).\cos(\varnothing)\sin(\theta).\cos(\varnothing)\sin(\varnothing)]^T \qquad (2)$$

A. System Verification Measurement

An external sound card with a 44.1 kHz sampling frequency and 16–24-bit adjustable resolution is used, and selected dynamic microphones as an element of AAS work between 80 Hz and 14 kHz. The measurement set-up selects the combination of 3 kHz, 6 kHz, and 9 kHz sinusoidal signals with equal weight to eliminate unintentional disrupting sounds like bird sounds. This combination is called a signal packet. In the set-up, the content/components of this intentionally formed signal packet are not known from the receiver side. Usually, in actual applications/sceneries, such systems must have signatures of targets as part of the lookup table in the storage to match them. The selected microphone as a sensor and its acoustic pattern in frequency is seen in Fig. 1. During the measurement set-up, the applied frequency is preferred as 10 kHz with a 10 ms packet length.

a) b)

Fig. 1 Elements (sensor) of AAS, a) selected microphone, b) the acoustic pattern of it.

Fully anechoic chamber measurements were carried out to ensure the system's accuracy. The first reference measurement is carried out related to elevation, in Fig. 2a., four array receiver set, and one transmitter is placed 4 m distance from each other in the same plane. The second measurement is carried out related to azimuth, in Fig. 2b., the source unit is hung orthogonally 3 m above the receivers.

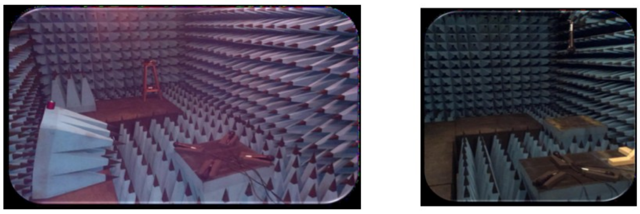
(a) (b)

Fig. 2. Calibration measurement

These measurements show that the proposed AAS can find the direction of the source within $\pm 2^0$ deviation angle). The calculated deviation angle of $\pm 2^0$ is the theoretical limit of the proposed AAS; its achievement cannot be better than this value.

B. *Forest Border Affect Measurements*

Sound direction estimation in forest area boundary measurements is examined in 2 dimensions. The alignment is overcome to distinguish echoes from the horizontal tree strings as seen in Fig. 3. Obstacles are determined with TDOA values of echo signals. As the receiver distance moves away from the boundary, the scattering amount in the high-frequency region decreases more than the scattering in the low-frequency region. The scattering expected from the body is remained below the perception threshold. It has been successfully detected in the direction and horizontal echoes of the sound source. As the distance of the echoes from the tree increases, the angle value between the trunk and the crown is considered 2° within the systematic error. The first received signal is due to the transmitter's line of sight (LOS) path.

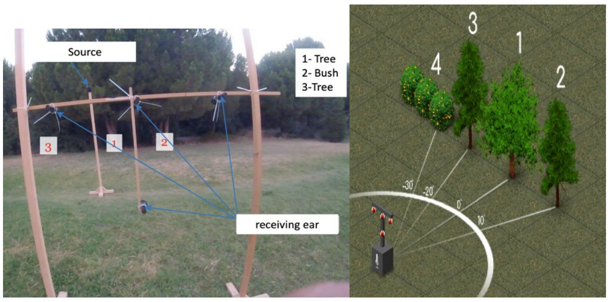

Fig. 3. Forest border set-up

C. *Measurement Set-up*

In order to make performance measurements, a region with multi-leaf but high-trunk trees are chosen as in Fig. 4. Ground effects and border effects are considered for model verification. That's why, before starting performance measurements, non-tree place measurements, single-tree measurements, and measurements close to dense forest borders were also carried out. Details of related verification and software/model updating measurements can be found in [13]. Receiving ears are placed 70 cm apart from each other. During the measurement set-up, temperature is 23°C ± 2°C and reported wind speed is 5 km/h.

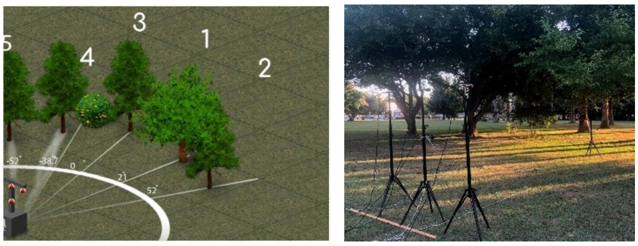

Fig. 4. Measurement set-up

III. METHODS AND MATHEMATICAL EXPRESSION

Sound waves travel at 343.2 m/s through the air at sea level and where the temperature is 21 °C. The waves transfer energy from the source of the sound to its surroundings. The ears detect sound waves when vibrating air particles cause the eardrum to vibrate. The bigger the vibrations, the louder the sound. The speed of a sound wave's propagation changes according to the temperature and density of the air; it decreases in cold weather. The sound changes its propagation direction from hot to cold air. The basic equation of the speed of sound can be expressed as in (3).

$$\vartheta_{air}=[331.5+(0.6*t)].m.s^{-1} \qquad (3)$$

where t is the temperature of the environment in °C.

A. *Progress of Execution*

The mathematical expression of locating the source depends on determining its fingerprint first. When the fingerprint is chosen, echoes in the signal's packets can easily be dug out. The final step will be calculating delays (times) between recorded fingerprints at each array element. Note that each obstacle in the environment will act as a reflector which means resource location. Phase difference calculation is critical in this model and is calculated using TDOA of signals which reach array elements independently. Interpolation, spectral consistency of amplitude, time difference of arrival (TDOA), and phase correlation techniques are applied to the fingerprint-extracted signal. The pre-processing and fingerprint removal process from the received signal until the direction determination can be summarized in the flow chart in Fig. 5.

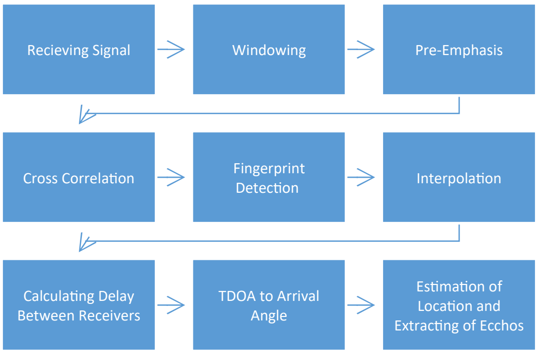

Fig. 5. Flowchart of the process

B. *Locating Algorithm*

The distance can be calculated by (4) by using the TDOA value of the original signal reflected from the objects. In the equation, l_m represents the estimated distance of the thing, the speed of propagation of the C_{air} audio signals in the air, and the TDOA value of the Δ_T signal.

$$l_m = C_{air} \times \Delta_T \tag{4}$$

This allows calculating the distance *x* between the transmitter and receiving ears at first glance. The total reflection path (TRP) transmitter-obstacle-receiver TRP (*y*+*z* in Fig. 6) is calculated using the same equation with echo. The angle α between the LOS path and TRP is found using the emerging phase difference between receiving ears. If two edges and the angle α between them are known for any triangle, the other edge can be calculated by (5).

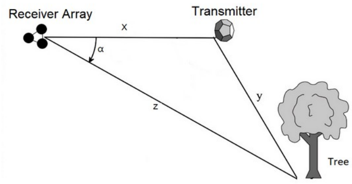

Fig. 6. Location estimation

$$y^2 = x^2 + z^2 - 2xz(\cos\alpha) \tag{5}$$

where this can be rewritten by known variables, as in (6). In the equation, *z* is already calculated and understood, and others are to be calculated. The related equation allows calculating *y* such that the next step is figuring by *z=TRP-y.* Using time of arrival, the incoming angle α can be calculated.

$$y^2 = x^2 + (TRP - y)^2 - 2x(TRP - y).(\cos\alpha) \tag{6}$$

C. Pre-Filtering and Processing

The first step is to determine the direction of the received signals; it is necessary to decide on the fingerprint of the signal. The related signal must be extracted from the continuous recording to do it. The methods of emphasis filter, Hamming Windowing, and Cross-Correlation are applied to convert the received signal's length, frequency, amplitude, and phase into the appropriate format.

D. Point-Based Acoustic Signal Recognition

For finding the direction of a newly sensed unknown source and 2-dimensional mapping, determining the signature of any signal is predominant for classifying the original signal and its echoes. The spectrogram is used in the frequency domain to identify the signal's signature created spectrogram; fingerprints are designed by keeping the vertex coordinates in a variable array [14]. The fingerprint extraction algorithm is executed in three steps: 1) Sorting peak points at the time-axis. 2) Identifying each peak as p_i and has coordinates as t_i – time and f_i-frequency. 3) At a point p_i, execute the condition: $if\ t_j > t_i, t_j < t_i + t + h, f_j < f_i + \left(\frac{f}{2}\right) \wedge f_i < f_j - \left(\frac{f}{2}\right)$, add the pair (p_i, p_j) to G array. h is a hop value that represents a gap between the anchor point and the left side of the target window. The G array is identified as the fingerprint of the signal. Fig. 7 is for the extraction of fingerprints. The spectrogram is obtained in the frequency-time domain by dividing the signals into small time packets and taking the Fourier transforms of each package. In this figure, axes represent time and frequency, respectively. The fingerprint of the signal is extracted from this spectrogram received in [15].

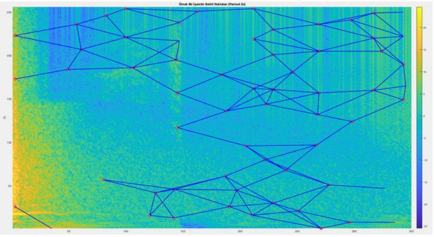

Fig. 7. A fingerprint of acoustic signal

E. Comparison of Packet Size of Original and Echo Signals

The packet size used as the actual mark in the measurement set-up is empirically chosen as 10 ms. This is used for the separation of echoes in the time axis and the recognition of the signal. When the echo extends up to 10 ms, the obstacle is not a perfect reflector, such as metal or plane rocks. Finally, the extension of packet size in time allows classifying obstacles.

IV. RESULTS

A. Single Tree Measurements

Microphones, positioned as a vertical linear array, provide reference information about how the signal was reflected, the reverberation time, and the reverb intensity, whether it resonates from the trunk or crown area of the tree. The crown area of the tree has an important place in the data obtained. Leaf density affects leaf structure, radius, and height. Depending on the distance of the acoustic signal sent to the tree trunk and the distance to the crown, the expected travel times and the direction calculated from the delay caused by the reach time of the signals reaching the recipients provide information about which part of the tree the sound reflects. Fig. 8 shows the response of single-tree in the case of the 10 ms length signal packet. Although the sent signal is 10 ms long, the echoes from the crown area take an average of 20 ms. Although the signal spreads in time, it maintains its spectrum feature.

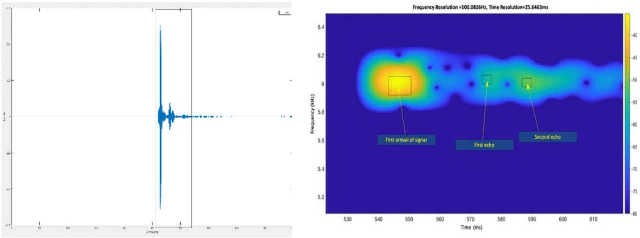

Fig. 8 Single-tree response

B. Classification of Type of Obstacle

In the set-up, 10 ms length signals are selected. It is observed using actual measurements that the length of eco signals varies from 11ms to 18 ms, but fingerprints are saved. Expansion of signal in time results from the type of vegetation, which acts as a reflector resource element. Almost no time extension is expected for obstacles like planar-metallic structures or planar rock. This analysis allows us to detail a 2D map of the reverberant environment present. Table 1 tabulates the time extension values of echo packets we measured in natural habitats due to different vegetation canopies.

TABLE I. RESULTANT EXTENDED PACKET SIZE

Type	Packet size (ms)	Keep fingerprint
Bush	13	Yes
Pine tree	13-16	Yes
Palm	11-13	Yes
Broad Leaf Trees	16-18	Yes

C. *Classification of Type of Obstacle*

Different region than the environment where calibration tests were conducted. That's because of to be sure about the proof of the system. The packet length was selected as 10ms. The performance measurements compared to actual data are shown in Fig. 9. The figure shows the exact and calculated angles between the transmitter and receiver. At this stage, note that echoes are not used in this calculation. Both measurements and calculated results are in coincide with systematic error. As a classic boxplot, the figure shows computed results with their distribution. Box limits are identified by the first quartile(Q1-minimum) and the third quartile(Q3-maximum). This box can be named the interquartile range (IQR), which ranges from 25%to 75% in the dataset. Horizontal lines are designated as whiskers. The minimum whisker represents the Q1-1.5IQR, and the maximum whisker represents the Q3+1.5IQR. Red lines are the medians of the datasets. Outliers are represented as red dots.

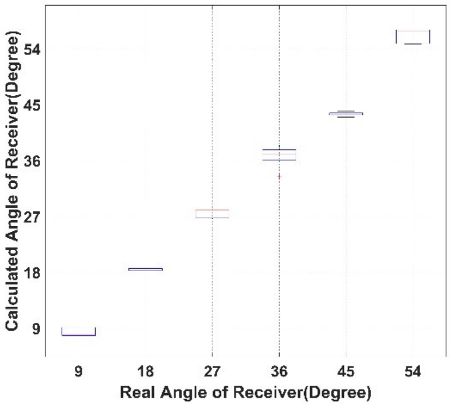

Fig. 9. Performance measurements

From the direction and distance information obtained for an echo, a 2-D obstacle map of the site is acquired by measurements in the tactical field. As mentioned, each obstacle tree in this scenario acts as a reflector resource element. The prepared 2-D map of the reverberant environment can be found in Fig. 10. In the figure, the actual and calculated positions of the obstacle (source of echo or equivalently resource) are shown in Fig. 4. Distance to the receiver as an origin is scaled 0 m and 20 m. The location of the source to the north is present at the outer circle. In the figure, a two-dimensional map of the forest environment, a deviation between actual and calculated tree locations is in the error of ±2 m, which is in the limit systematic error.

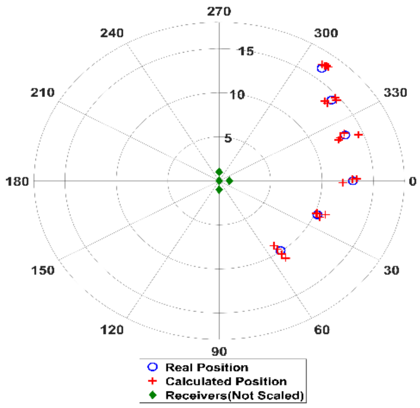

Fig. 10. 2-D map of the measurement set-up

This is the diffraction, scattering, and delayed scattering of the mark on the reflected object. From this point of view, the creep information of the sign in the time domain was used to classify the obstacle in any position on the obstacle map (bush, coniferous tree, high trunk tree, broadleaf tree, rock, wall or metal, etc.).

D. *Error Calculation*

During measurements, the transmitter is located in order. All points have 6 meters distance from the receiver array center. Angles are 9°, 18°, 27° and 36°, 45°, 54°, respectively. With these conditions, a location detection of the transmitter is held. And the results of the measurements are given in Fig. 9. Table 2. shows detail in Fig. 9. The maximum average angle error is determined as 2.16°. A single total error is detected in the first location as 3.2°. Detailed error calculations related to Fig. 10 for distance are tabulated in Table 3. The maximum distance error is determined for the 4th object as 1.2 m. Also, the maximum angle error is chosen for the 4th object as 19°. Average distance errors are less than 0.1875 m for all. Moderate angle errors are less than 0.925 degrees angles for all objects. In Table 3, d_{Real} and d_{Calc} are real and average calculated the distance of obstacle to the origin. θ_{Real} and θ_{Calc} are actual and average calculated angles of obstacles to the north of origin.

TABLE II. PERFORMANCE RESULTS FOR TRANSMITTER LOCATION

Location ID	Angle Between Transmitter and Receiving Ear Calculated in Degree			
	Real	Median	Average	RMSE
1	9	7.9	8.34	0.9066
2	18	18.4	18.44	0.4604
3	27	28	27.48	0.81
4	36	37	36.68	1.858
5	45	43.5	43.51	1.5233
6	54	56.9	56.16	2.4125

TABLE III. PERFORMANCE RESULTS OF OBSTACLE LOCATION i

Obst. ID	d_{Real}	d_{Calc}	RMSE	θ_{real}	θ_{calc}	RMSE
1	9.9	9.98	0.5612	53	53.93	0.7714
2	10.3	10.47	0.321	22	21.4	0.9074
3	13.04	12.81	0.4335	0	-0.4	0.7394
4	13.3	13.26	0.6301	337	337.61	1.0378
5	14.25	14.18	0.3967	320	319.96	0.7662
6	16.2	16.63	0.3611	308	308.33	0.7269

V. CONCLUSION

This study aims to determine the direction and location of a newly sensed unknown acoustic signal source and its echoes in a forest area. The next step is decomposing the received unknown signal packet into its components as original and echoes. By then, determining the position of obstacles that cause echo in the region is a third but crucial outcome.

As a result of the measurements made, it is observed that the 93% success rate obtained in the reference measurements made in the anechoic chamber decreased to 86.80% in the field. The measurement areas are heterogeneous, and the effects of independent noise sources appear. By the proposed approach, obtaining a 2-Dimensional map of the domain is possible within the error of ±2 m resolution.

The study will be applied as a model for solving acoustic problems in large closed-volume buildings such as theatres or concert saloons.

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