

The sound series are considered as an addition to visual and thermal imaging information flows when using computerized monitoring systems (CS). A minimum complete structure of spaced microphones for collecting data on sound rows, which is suitable for calibrating, isolating and transmitting data on sound anomalies (SA), is proposed. Duplication of the data transmission channel by wire and Wi-Fi module for recording and determining the type and coordinates of the SA is provided.

An experimental receiving module has been assembled, which includes microphones, amplifiers and signals matching boards for digital and analog forms, an ARDUINO UNO WIFI REV2 controller with an integrated Wi-Fi module. It is presented that its addition with a personal computer and a smartphone with the Android operating system forms a CS for remote wireless control of the course of the experimental analysis of sound series. It has been confirmed experimentally that its structure is minimally complete. An algorithm was developed and a software package was written in C/C++ languages. It is shown that the number of microphones is selected from the conditions of the problem from 1 to 5, but their number is limited to five digital inputs of the ARDUINO UNO WIFI REV2 board. A wave representation of the law of temporal changes in intensity and the integral norm of SA is applied. The possibilities of calibrating all data of sound series in analog and digital form are demonstrated. The article presents the suitability of testing the algorithms for determining the phases of echograms from time series data, containing SAs of different origins and recorded by three different microphones. The effect of connecting a Wi-Fi module on reducing the voltage drop by 0.5–1 V is shown. The necessity of an additional registration condition for all microphones is demonstrated. The software interfaces for the calibration of the receiving module and the operation of the mobile application have been developed

Keywords: computerized system, modular structure, reception algorithm, software, system testing

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INCREASING QUALITY OF THE WIRELESS MODULE FOR MONITORING AND SUPERVISION OF SOUND SERIES OF THE EXPANDED PURPOSE

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1. Introduction

The innovative attractiveness of sound series [1, 2], as an information supplement [3–5] as part of mobile devices [6–9] to visual and thermal imaging streams [10–12], is demonstrated by more and more examples of practical application [13–15] of computerized systems (CS) monitoring [16, 17]. The development of geoinformation [18], sensor [19], video and wireless [20–22], unmanned technologies [23–25] and miniaturization of single-chip implementations of the element base of devices create prerequisites for viewing and searching for new block diagrams of modules. However, the implementation of the theoretical premises presented in [26, 27] requires extensive functional verification and proof. Such devices will meet the requirements of their use in a new quality, including as part of unmanned vehicles [3–5], medical rehabilitation modules [26, 28] and general-purpose automated systems [6, 7, 12]. In addition, the development of their ap-

plication for solving problems of remote monitoring requires a detailed experimental study of the characteristic features of the sound sequence. Models of recording events based on soundtracks, as events with a jump in sound intensity, do not give unambiguous results. The formation of a detailed description of such events requires a preliminary experimental study and search for methods of unambiguous representation of them as a set of prototypes and corresponding images. The features of the temporal course of changes in the intensity of the sound stream demonstrate the need for detailing. A special problem is the overlap of sound streams of different origins, which complicates the formulation and solution of the problem of identical recording of an event by different microphones. In this regard, the formulation and solution of the problem of automated studies of sound features for known events of groups of anomalies becomes relevant.

Thus, the creation of modules for conducting and fixing a detailed experiment will open up opportunities for a com-

prehensive study of both the CS themselves for determining the coordinates of sound anomalies (SA) and their joint use in other devices. Based on the above, the development of a module layout as a tool for checking models and software tools (software) becomes relevant. Their presence, in turn, will create conditions for experimental research and the formation of the minimum structure of a wireless module for monitoring and observing sound series of extended use [18, 28–31].

2. Literature review and problem statement

In works [1–3], attempts are made to create small-sized CSs operating on the principle of registration and further analysis of the set of sound signals. Signals or individual audio time series recorded by synchronously spaced microphones supplement the decision-making information base. The latter is the reason for the development and expansion of functionality. Thus, in [1], based on the analysis results, information about the acoustic range of fire from small arms was formed and demonstrated how to track target movements using a single sensor unit and hand movement. However, the peculiarities of the SA nature as a random and sometimes a one-time signal complicate the applicability of the results [1]. Also important are the results of finding the coordinates and forming the directions of acoustic rays, the cross section of which determines the spatial position of the points of fire [2]. However, the reason for the complications in applying the results of [2], as in [1], is the need to analyze impulse and non-repetitive audio signals. In addition, as indicated in [2], an increase in the accuracy of determining the coordinates of the software is achieved by increasing the number of microphones. The need to increase the number of microphones conflicts with the need for a general miniaturization of the CS. The requirement for miniaturization is also dictated by the introduction of one of the ways to increase the efficiency of sound control of equipment by placing it on mobile aircraft [3]. However, the features of placement in space form a specific influence on the structural differences in the structure of acoustic reconnaissance means, especially with small overall dimensions [4]. In addition, as shown in [5], for the optimal organization of computations, the relative position of a group of microphones is important, for example, at the vertices of Platonic solids. An equally important example of consideration of the acoustic flow of information observed and processed by stochastic methods is modeling, evaluation and application in air control systems [6]. The work develops methods of assessment and forecasting, allowing to present a predictive control model for a hybrid system [7]. The advantages of combined systems are shown in [8]. Such a system includes FLIR, CCD with continuous zoom, laser rangefinder, GPS, commander control system and several “subscribers” located with the unit’s snipers [8]. SPOTLITE analyzes the detected fire sources, checks if each source is actually enemy fire. After verification, the data is transferred to the supported unit, which manages the fire support [8]. In work [9] it is shown that providing systems with means of remote control fundamentally expands their functions and capabilities. The work [10] also confirmed this conclusion. Thus, Metravib Defense solutions, which has been in service of widespread world security devices for more than 25 years, demonstrates their functionality and reliability [10]. Boomerang III determines the

location of small arms shot [11], and calculates the shooter’s coordinates based on the principle of passive acoustic detection and computer signal processing [11]. Coordinate determination time, reached less than a second, regardless of location on a stationary or mobile vehicle. Features and trends in the development of sound systems as additional sources of information about the state of the environment, recorded by means of video and thermal imaging observation, are presented in [12]. These works [8–12] have a general tendency, which is manifested in an increase in the total number of microphones to 6–8. The simultaneous use of indirect methods for formulating the problem of determining the coordinates of the sound source and the use of gradient methods increases the error of its solution. An increase in the number of microphones increases the number of terms in the functional and decreases the solution error, therefore, practical systems increased the number of microphones. In [13], a multilevel training approach based on B-splines for localizing sound sources is shown, and in [14], an automatic determination of the coordinates of microphones is presented. The quality of the created hardware, which traces the trends towards miniaturization and integration of single-chip single-board computers, leads to the proliferation of examples of use in the rehabilitation of military personnel working in high-noise areas [15]. The second success is the application in flexible robotic systems [16]. A special role was played by the successes in the development of CS elements in robotic systems of navigation and positioning units in production elements of flexible manufacturing systems [17]. The improvement of methods and tools for constructing dynamic scenarios in the operation of geoinformational navigation systems based on the principles of sound location has also been improved on the basis of the development of CS elements [18].

However, despite the positive results of the development of CS as a whole, the expansion of their functions and the improvement of qualitative characteristics require the search for new hypotheses about the SA composition, the nature of the wave and the features of its propagation. Of course, the latter requires physical modeling tools to build and test models [19]. In the same work [19], the analysis of the reasons for the error was theoretically carried out and it was shown that the irrationality of the functional is the main one. The use of the rebuilt functionality and the method of recurrent approximation made it possible to increase the accuracy and, accordingly, reduce the required number of microphones to 3–4. However, the implementation of these results and algorithms was hampered by the absence of a CS that could monitor the acoustic field and determine the moment of time at which one phase of the wave enters various microphones. A review and comparative analysis of the CS for identifying the position of the sniper was carried out in [20]. In [21], the authors were able to determine the parameters of the sound of a shot that affect the performance of specialized fire missions by the security forces. It is also indicated there that these parameters will be subject to regulation [21]. Work on navigation based on non-precious quadcopter cameras have shown that they are of increased interest [22]. There are two features of this reason: motion stabilization and video fixation and the study of the mutual influence of the systems operation [22]. The development of a method for determining the area of operation of unmanned vehicles using graph theory has demonstrated the expected negative consequences of the joint operation of systems [23]. It was also noted there that additional loading of the power

supply requires control, and in the event of a voltage drop, a change in the signal calibration conditions is required [23]. The use of sensor modules of adaptive robots as a principle of increasing efficiency was introduced for heterogeneous intelligent sensors on the basis of mutual influences in the work [24]. Viewing the requirements for the SA structure for additional study of the mutual influence of systems is one of the tasks, without solving which it is impossible to solve the problem of quadcopter reconnaissance [22, 23].

In connection with the above, the recurrent network as a tool for calibration in automated systems and interactive simulators will show its advantages [25]. However, the problems of such a study, as well as of the previous study on the mutual influence of systems, are presented in [25]. Its solution, as suggested there, is to create a physical model for carrying out model experiments. The proof of this conclusion is the work that uses the recurrent approximation in the problems of neural network synthesis to control the phototherapy process [26]. As proved in [27], recurrent approximation under special conditions of hardware design acts simultaneously as a tool for expanding the functions and modes of operation of the neural network [27]. The problem of the size of the transmitted information is one of the problems of the choice of elements and the structure of the hardware of the system and the algorithm of the processing software. However, as the authors have shown in [28], when analyzing the development of the innovation industry, fluctuations turn the process into complex measurements, which requires the use of the ateb-forecasting method. However, this solves the problem only for a stable trend [28]. The use of ateb-forecasting for other cases of the trend when using the Dirac function is subject to additional study and proof [29]. In this regard, another approach to compression is of interest, which was proposed for another practical application [30]. According to this approach, the parameters are formed, but as long as their values are within acceptable limits, only data on one of their generalized indicators are transmitted. If the value of at least one of the parameters exceeds the permissible limits, data on all parameters is transmitted. This algorithm leads to significant reductions in the amount of data transmitted without information loss [30]. As shown in [31], the analysis of acoustic means of detecting a shot from small arms and their classification requires model experiments, and hence the formation of a physical computerized model of reception, registration and processing of signals by a microphone system [31]. The work [32] demonstrated the need to study two types of waves: muzzle and supersonic. Each of them needs further research into physical characteristics. The first – the muzzle wave formed by the powder gases, has an equidistant wave front shape. The second is supersonic, which is formed by a ball or projectile, the speed of which is greater than the sound speed and has a variable wave front shape. For such a study, the creation and improvement of CS microphones for wireless recording of sound rows is required. Work [33] confirms the fruitfulness of the idea of isolating a muzzle wave and shows its ability to determine the coordinates and direction of the enemy's firing position by a group of sound receivers. Thus, the works [32, 33] emphasize the need to create equipment capable of accurately recording the moment of transition of the sound barrier and the moment of time when the propellant gases emerge. In turn, this requires the creation of a combined observation CS for carrying out an automated experiment and physical modeling. In addition, the main contradiction identified in [8–11] is the

need to increase the number of microphones to 6–8, which significantly increases the accuracy, and the requirement for miniaturization of the equipment [3, 4, 20–24]. The implementation of the results of work [19], which theoretically and numerical experiments demonstrated the possibility of reducing the number of microphones to 3–4, also requires the creation of a CS that meets the requirements of miniaturization of equipment as a whole. Achievement of the miniaturization requirements can be carried out by forming a minimum structure and by a special choice of functionally complete components, which increases the CS quality.

Thus, the main unsolved problem holding back the physical laboratory modeling of different types of SA is the structural imperfection of the equipment and the algorithm of its operation and the absence of remote control of the experiment.

3. The aim and objectives of research

The aim of research is to improve the quality of the CS equipment of microphones for monitoring and recording sound signals by forming and substantiating its minimum structure. This will allow realizing the functions of remote wireless control of the course of the experimental study of sound rows and will open up opportunities for further improvement of the elements of the entire CS.

To achieve this goal, the following tasks were set:

- to form the minimum structure of the CS for carrying out physical modeling for the SA simultaneous registration by spaced microphones, as a solution to the problem of dynamic programming with constraints;
- to build an algorithm for determining the phases of echograms according to time series data recorded by different microphones;
- to test the CS, calibrated and suitable for monitoring and observing sound series in the presence of SA fixation.

4. Materials and methods of research

The basis for the formation and creation of the minimum structure of the wireless module for monitoring and observing sound sequences of extended use, first of all, was the SA definition as a separate phenomenon. It was believed that the properties are sharply different from the general constant sound background. It was also assumed that the idea of creating a monitoring and observation module was based on the idea of the possibility of isolating SA of a known description, and its appearance in the observation area is abnormal. The second version of the SA is cases when a previously unknown sound contains rapidly similar changes in intensity that are not inherent in a given area of observation, appears accidentally and disappears. The study was based on the following hypotheses that the SA satisfies the principle of superposition, and the sequence of the identified phases is not violated. Under these conditions, the following assumptions are made:

- SA is a point source of an acoustic spherical wave;
- the microphones of the system are point receivers of acoustic waves;
- the distance between the microphones is stable and is two orders of magnitude larger than the size of the microphones.

Based on the formulated hypotheses and assumptions that were formed, the concept of quality was applied to solve the first problem. Further, by quality let's mean: compliance with the requirements for the specified list, including reliability and durability. As a quantitative indicator, it was determined by an integrated method by comparing the beneficial effect of consumption with the reference one. Rationing according to one of the Euclidean norms reduced the problem to a quantitative description and made it possible to apply the methods of operations research, namely dynamic programming, multiple and functional analysis.

In addition, it was assumed that the time flow of the SA intensity is a continuous function, admitting a schedule in a Fourier series and integrated with a square. Methods of expansion in Fourier series, methods of finite integral transformations, functional analysis and search for the root of nonlinear algebraic equations are applied.

To solve the third problem, it was assumed that the simulator generates a typical SA that imitates its main properties: continuity, time duration, and integrated with square. Methods of calibration, determination of accuracy classes, and the theory of errors were used to determine the estimate of the error in indirect measurements. The main function of the module for monitoring and observing sound sequences is to provide information completeness and intellectualized processing of information flows. To control the quality of the interface for monitoring smartphone data and display it on-demand, the methods of forming a generalized integral indicator of the interface quality were applied.

5. Research of the module, methods and tools for calibration and functioning of automated devices for extended use

5.1. The minimum structure for monitoring and observing sound streams

Analysis of the structure of systems for practical observation, reconnaissance and monitoring for various purposes [6–20] asserts that the efficiency of constructing a CS for determining the coordinates of software increases with an increase in the number of microphones to 3–5 units. However, it was stated that this conclusion of the indicated works was in contradiction with the need and tendency towards the general minimization of such systems. In this regard, the task was set to find the minimum structure analytically. For its setting, it is assumed that the signal at the output of each of the microphones must be amplified and brought into the normal range, and transmitted to a single-crystal controller. The number of amplification and adduction modules cannot be less than the number of microphones. Therefore, minimization of the module structure due to these elements is possible both by reducing the number of microphones and the number of single-chip controllers. To form a minimal structure, a dynamic programming problem was set to minimize the functions and elements of the block diagram. A block diagram of the structure in the maximum version of N lines was presented and it was assumed that the initial composition of the line contains the maximum number of blocks M (Fig. 1). The number of lines is equal to the maximum number of microphones N .

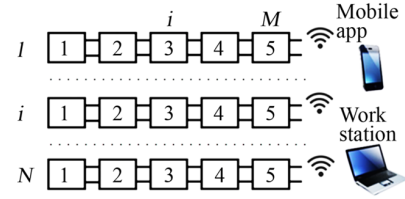


Fig. 1. Block diagram of a computerized system for determining the coordinates of a sound anomaly

Each of the lines contains functional blocks that implement the full list of M functions, from temporarily fixing an acoustic wave, transforming it to transmitting a signal to a smartphone. So, block 1 converts the vibrations in the microphone into an electrical signal. Block 2 amplifies it, in block 3 the signal is coordinated in accordance with the requirements of the input channel (input) of the ADC controller – block 4. Then the signal is transmitted by block 5 to the smartphone 6 or workstation 7. If the initial state is represented by a block diagram of the maximum possible structure, and the description of each of its elements is given by the functions of quantitative or qualitative measurement, then the problem is reduced to a problem with a known beginning. Also, applying the normalization according to one of the Euclidean norms, the problem of finding the minimum structure is reduced to the problem of optimal design – dynamic programming with inequality constraints.

The objective function of this task is formed as additive:

$$F(i, j, k) = \sum_{i=1}^N \sum_{j=1}^M f_{i,j}(k), \tag{1}$$

where $F(i, j, k)$ – the objective function, i – the line number, j – the number of the block of the initial block diagram, and k – the number of the new version of the transition step. In addition, $f_{i,j}(k)$ a normalized objective function of one variable is introduced for each element i, j , from which step k is carried out, then the problem will be written as follows:

$$\begin{aligned} & \min_{i,j,k} \sum_{i=1}^N \sum_{j=1}^M f_{i,j}(k); \\ & N\Delta t + b_5 \leq T; \quad i = \overline{1, N}; \\ & N_{out} \leq b_4; \quad j = \overline{1, M}, \quad k = \overline{4, 5}. \end{aligned} \tag{2}$$

Here N_{out} – the number of outputs from the matching blocks, Δt – the generation time of the amplified and matched signal in one channel, b_4 – the maximum number of inputs of the controller's ADC, b_5 – the interoperation time spent on preparing for data transmission of the communication channel, T – the permissible time for determining the data about the anomaly. The solution to problem (2) will be presented:

$$\min_{i,j,k} F(i, j, k) = \sum_{i=1}^{N_{\min}} \left(\sum_{j=1}^3 f_{ij} \right) + F_4, \tag{3}$$

where F_4 is a function defined as the product of the union of objective functions with k (4) and (5) by the solution function under conditions of comparison by several standards according to the algorithm [34]. If all blocks (microphone, amplifier and signal conditioner) of different lines are identical, then the last one will be written:

$$\begin{aligned} \min_{i,j,k} F(i,j,k) &= \sum_{i=1}^{N_{\min}} \left(\sum_{j=1}^3 f_{ij} \right) + \sum_{i=1}^{N_{\min}} (f_{i4} \cap f_{i5} \cdot L(i)) = \\ &= N_{\min} \left(\sum_{j=1}^3 f_{ij} \right) + f_{i4} \cap f_{i5} \cdot L(i). \end{aligned} \tag{4}$$

The function $L(i)$ is calculated from the property functions according to the productive rule:

$$L(i) = \begin{cases} 0, & \text{if } f_{i4} \cap f_{i5} < 1 \\ 1, & \text{if } f_{i4} \cap f_{i5} = 1 \end{cases} \tag{5}$$

Thus, solution (4), obtained by the Bellman principle, is presented as a two-step strategy. On the first one, the minimum number of microphones N_{\min} is selected. On the second, from the database of modern and available controllers, an element is selected that combines the functions of a controller and a communication channel, satisfying the list of requirements for openness of the code, availability of programs, reliability and the number of inputs, which must be more than N_{\min} . That is, the element with the highest quality is selected. The application of this technique made it possible to form and manufacture a CS with a minimum structure.

In addition, taking into account the requirement of data transmission using a Wi-Fi module, ARDUINO UNO WIFI REV2 (RS Components, Arduino manufacturer, Manufacturer number ABX00021) was chosen. The board has an integrated Wi-Fi module, the characteristics of which and the communication channel are presented in Table 1.

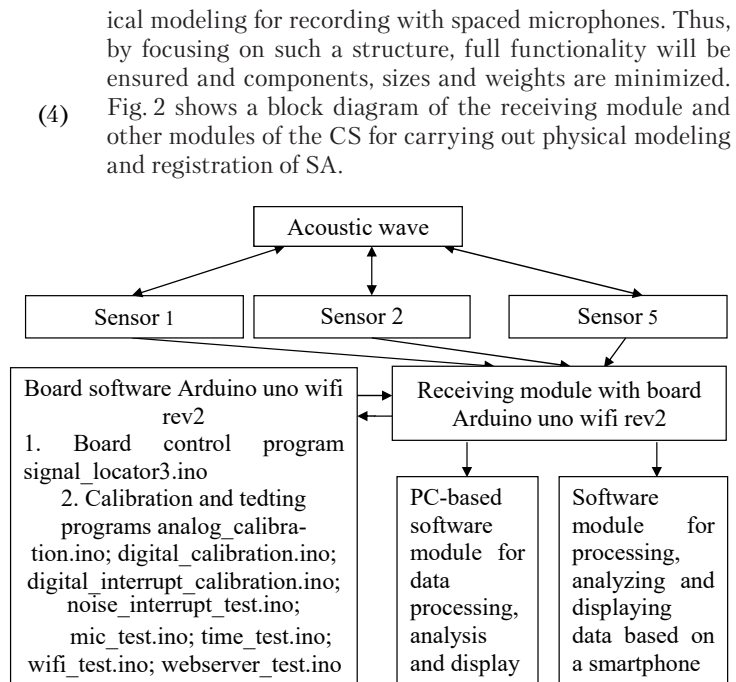


Fig. 2. Block diagram of a computerized system for determining the coordinates of a sound anomaly

The block diagram is presented in the form of a single module, which includes sensors, an Arduino board, software modules for controlling the board, and calibration and testing. Also, the CS includes software modules for processing, analyzing and displaying data based on a PC and processing, analyzing and displaying data based on a smartphone. Acoustic, operating and dimensional characteristics of the sensors are presented in Table 2.

Table 1
Channel and communication module characteristics

Name	NINA-W102
Manufacturer	u-blox
Communication module type	IoT
Network type	Bluetooth Low Energy, Wi-Fi
Supply voltage	3...3.6 V
Communication protocol	IEEE 802.11b/g/n
Transmitter power	14 dBm
Receiver sensitivity	90 dBm
Installation	SMD
Communication module properties	antenna
Dimensions (edit)	14×10×3,8 mm
Working temperature	-40...85 °C
Transmission speed	150 Mbps
Fog	2.412...2.484 GHz
Range	300 m
Number of inputs / outputs	24

The type of communication module, type of network, supply voltage, communication protocol, transmitter power and receiver sensitivity indicate that ARDUINO UNO WIFI REV2 will work with multiple microphones. The rest of the characteristics of the transmission rate, frequency bands, range and the number of inputs convinces of the fundamental possibility of forming a minimum CS structure, carrying out phys-

Table 2
Characteristics of microphone models:

Name	KY-037	MAX4466	Mic+AMP
Supply voltage, V	3.3...5	2.4-5.5	4 V-12 V
Working temperature, °C	0...+70	40 °C...85+ °C	0...+70
Mounting hole size, mm	3	3	absent
Module dimensions, mm	34×16	20×14×10	41×13×13.5

Each sensor includes an amplifier and signal range matching circuit. The number of sensors can vary up to five. All sensors are powered by a single power supply together with a receiving module based on ARDUINO UNO WIFI REV2. A separate element is a smartphone. In this diagram, the smartphone is conventionally not shown, and the connection with it is also not shown. It is proposed to use smartphones with a processor. When connected in parallel, the sensors will be polled one by one. The latter causes a delay. Therefore, the number of microphones should be limited to the minimum required. It was theoretically shown in [19] that the number of microphones can be reduced to 3. In the future, this conclusion will be taken into account and the experiments were carried out with a reduced number of microphones: three.

Thus, with the aim of further introducing the elements of an integral system, an acceptance module was built and

its study was carried out. The main purpose of this experiment is to prove the ability of the practical construction of a module that provides modes and work as part of an integral system for determining the SA coordinates.

5.2. Algorithm for determining the phases of echograms according to time series data recorded by different microphones

The formation of an algorithm for determining the moment of time of the phases of echograms from the data of sound time series recorded by different microphones is the main source of errors [19]. In this regard, it was determined to develop a key element of the general algorithm of the module program. The task of the module is to establish conditions that ensure the unambiguity and uniformity of fixing an identical moment in time in different echograms of the same SA. Consideration of the SA, in the form of a time-extended change in intensity, leads to a hypothesis about the phase invariance of any of its points in the course of propagation. Based on the foregoing, it was assumed that such a point of the echogram was found and there was an unambiguous relationship for its determination. Based on one of the conclusions of [27] about the calibration procedure as a tool to expand the functions and capabilities of the monitoring system, a four-point schedule was applied using the method of recurrent approximation (MRA). This is how the intensity of the acoustic wave SA $i(t)$ is presented as a function of time t :

$$i(t) = \sum_{\omega_i}^{om} I_{\omega_i} \cos(\omega_i t - k\bar{r} + \alpha_i), \quad (6)$$

and introduced the norm

$$\|i(t)\| = \left[\int \left(\sum_{\omega_i}^{om} I_{\omega_i} \cos(\omega_i t - k\bar{r} + \alpha_i) \right)^2 dt \right]^{1/2}. \quad (7)$$

Here I_{ω_i} – the amplitude value of the current strength, ω_i – the cyclic frequency from the components of the set of frequencies, k – the wave vector of the corresponding wavelength, r – the radius vector, and α_i – the initial phase. If, during the calibration of this type of software for a fixed set of frequencies ω_i , the coefficients λ_{ω_i} are established, then the basis for the construction of the algorithm is the following justification. According to the calibration of a SA fragment from N points for a set of frequencies ω_i and, by definition, the coefficients $\omega_i, \lambda_{\omega_i}$, are a solution to the equation:

$$i(t) - \lambda_{\omega_i} \|i(t)\| = 0. \quad (8)$$

(8) is expanded due to the continuity of its terms in a series in the MRA:

$$i(t_n) - \lambda_{\omega_i} \|i(t_n)\| + i'(t_n) \Delta t + \lambda_{\omega_i} A(t_n, \omega_i, \lambda_{\omega_i}) \Delta \omega + \dots R(t_n) = 0. \quad (9)$$

Then the recurrent approximation of the time instant of one phase of any microphone will be calculated:

$$t_{n+1} = t_n - \frac{i(t_n) - \lambda_{\omega_i} \|i(t_n)\| + \dots R(t_n)}{i'(t_n) + \lambda_{\omega_i} A(t_n, \omega_i, \lambda_{\omega_i}) \omega' + \dots R'(t_n)}. \quad (10)$$

In expressions (9), (10), respectively, $R(t_n)$ denotes the remainder as final term of the expansion in terms of MRA and the differentiated terms $A(t_n, \omega_i, \lambda_{\omega_i})$:

$$A(t_n, \omega_i, \lambda_{\omega_i}) = \left[\int \left(\sum_{\omega_i}^{om} I_{\omega_i} \cos(\omega_i t_n - k\bar{r} + \alpha_i) \right)^2 \right]^{-1/2} \times \int \left(\sum_{\omega_i}^{om} I_{\omega_i} \cos(\omega_i t_n - k\bar{r} + \alpha_i) \right) \times \omega_i \sin(\omega_i t_n - k\bar{r} + \alpha_i) dt. \quad (11)$$

Thus, based on the substantiation of the time point of the phase t_{n+1} , an algorithm for calculating the time point is formulated.

For any moment in time, four values of the intensity of the acoustic wave $i(t_n)$ were measured using a four-point scheme.

By expression (7), the norm was calculated for each frequency and from the set for a given type of SA.

The numerical values of the first and second derivatives are calculated from the values of the intensity of the acoustic wave $i(t_n)$.

The value of the constant $A(t_n, \omega_i, \lambda_{\omega_i})$ is calculated according to (11).

By expression (10), the value of the time of entry of the acoustic wave was calculated. In the case when the divergence between two consecutive points in time does not satisfy the required accuracy, then the approximation is repeated until the required convergence is achieved.

Thus, the introduction of the SA intensity as a continuous function (6), integrated with a square, the norm of which (7), allows several ways to determine the characteristic phase from the calibration data of the factors λ_{ω_i} and as the recording time of the SA arrival at the microphone. The analysis of different possibilities leads to a comparison of the wave amplitude and the norm, the center of gravity or the relative location of the point of extension as an unambiguous criterion on which the algorithm for fixing the software can be based.

The algorithm of the signal_locator3.ino program (Ukraine) [35] is shown in Fig. 3. At its core, it is an endless loop and consists in the fact that the program, after starting and running the loop () method, constantly works in the mode of an external loop.

At each iteration of the outer loop, the counter increases the iteration number, and every 10,000 iterations a check is made to ensure that all microphones have received a “one” signal (1, 1). If the fact of software registration is recorded by all microphones, then the time of entry of the sound signal is stored in each of the three microphones in the form of an array. After that, the operation of the signal_locator3.ino [3, 5] receiving module control installed on the ARDUINO UNO WIFI REV2 board is checked for the presence of an array.

Further, the array of the sound wave arrival time to each microphone stored on the ARDUINO board is sent to the smartphone via a wireless communication channel using the built-in WI-FI module.

During the development of the algorithm and its test tests, the data recorded by the microphones are read at each iteration, regardless of whether a positive “one” signal is received or not. Such steps of the algorithm led to an overload of ARDUINO, which occurred due to the limited resources of its RAM and processor performance. In this regard, in the mode of reading data recorded by microphones, it was proposed to programmatically disable the Wi-Fi module.

Disabling reduced the additional load on the ARDUINO board. The next step is to connect the Wi-Fi module and send information to the smartphone. This decision has a positive effect on the performance of the board. However, the next problem arising from the switching of the Wi-Fi module is the voltage drop sag on the power supply. As a result, the microphone calibration data becomes invalid and the measurement results become unstable and distorted. In order to avoid destabilizing the operating voltage in the future, the Wi-Fi module is not turned off. In addition, the state of the microphones is not checked at every pass of the outer loop, but after every 10,000 iterations. The fact that a sound wave has entered is confirmed and recorded only if there is a signal on all three microphones. In addition, the number of variables and their type have been optimized, due to which the work of the ARDUINO board has been stabilized in terms of power supply. The final calibration of the microphones with the Wi-Fi module turned on provided a high-quality cutoff of the so-called natural “noise” by the system. Removal of natural noise made it possible to communicate freely in experiments without the risk of false fixation of extraneous sounds. During the experiment, the system responded only to sound vibrations generated by a sound signal simulator. The sound signal simulator provided stability, quality and constant loudness of the sound source during the study.

connected to it. The constructed device consists of a stand 1 (Fig. 4), an antenna array 2, convenient for fixing microphones 3 (Fig. 5). The device also includes the ARDUINO UNO WIFI REV2 9 board (Fig. 7) with an integrated Wi-Fi module 4.

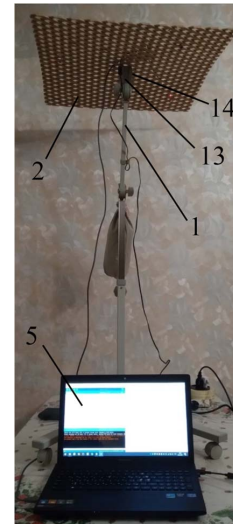


Fig. 4. General view of the receiving module

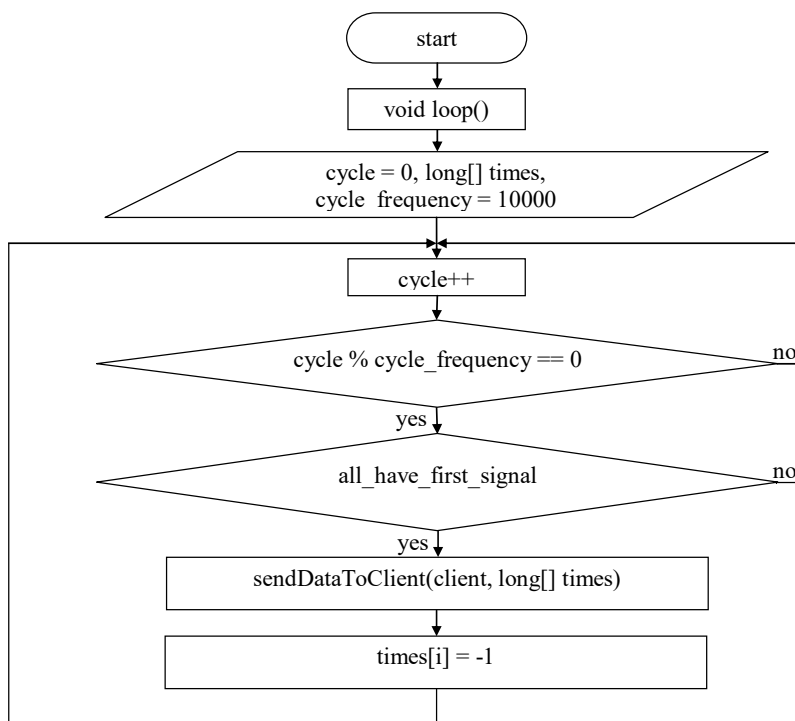


Fig. 3. Block diagram of the signal_locator3.ino program

5.3. Test of a computerized system calibrated and suitable for monitoring sound streams

To confirm the operability of the minimal structure of the wireless module for monitoring and observing sound series, the processes of performing a set of functions were experimentally investigated. Thus, in the course of an experimental test, a receiving module was built on the basis of the ARDUINO UNO WIFI REV2 board with an integrated Wi-Fi module and microphone modules

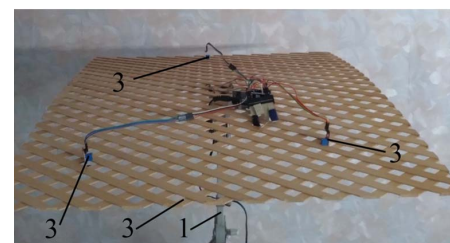


Fig. 5. Antenna array of microphones

Antenna array 2 is fixed on the top of the stand 1. On the antenna array 2, in turn, the main elements of the receiving module are located. To install the corresponding software previously developed by the authors [3, 5], this module, if necessary, is connected to the computer 5. In the operating mode, it exchanges data with a mobile device using a wireless LAN Wi-Fi (not shown in the photo). In Fig. 6, 7 show the elements fixed on the antenna array 2:

- microphone module (for example KY-037) with high sensitivity, has digital 6 and analog 7 outputs. The level of its operation is regulated by a replaceable resistor 8;
- board ARDUINO UNO WIFI REV2-9 (Fig. 7) with integrated Wi-Fi module;
- bus connector 10, expands the number of connection points to the “plus” and “ground” buses;
- a network of connecting wires 11;
- antenna array base 12.

The angle of inclination of the antenna array 2 is changed by means of a spherical hinge 13 with a locking screw 14, which is fixed on the upper part of the rack rod 1.

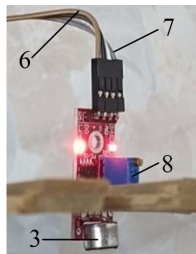


Fig. 6. Microphone module KY-037

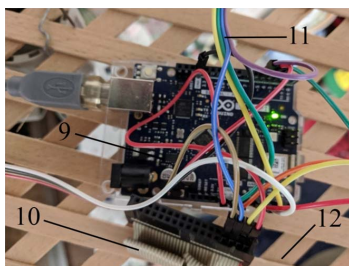


Fig. 7. ARDUINO UNO Wi-Fi REV2 board

The receiver module software set installed on the ARDUINO board consists of the following programs:

- the main program `signal_locator3.ino` performs the functions of receiving, processing, analyzing the sound signal and extracting information on the time of arrival of the sound wave to each of the microphones 3. It also forms an array of received data and transmits the latter to the mobile device 5 by means of the local Wi-Fi network;

- auxiliary application `analog_calibration.ino` provides setting of microphones to receive signal in analog mode. It was made it possible to reveal the peculiarities of the functioning of the ARDUINO board in the mode with the working and disabled Wi-Fi module.

When calibrating microphones, information about the recording of echograms is sent via a cable to the computer input, and the Wi-Fi module is not involved in this. The echogram recording is displayed on the computer monitor. During testing, information is transmitted via the Wi-Fi module. A fragment of the recording of echograms for three separate microphones and their total signal is shown (Fig. 8).

After comparing the results of work in different modes, it turned out that in the wireless mode, the data does not coincide with the previous and disfigured ones. It turned out that the main difference between these two modes is that the board itself, with the Wi-Fi module turned on, sags in power supply by 0.5–1 V. This, in turn, distorts the results of the preliminary calibration. Since it is not possible to refuse the data transmission channel via the Wi-Fi module for the period when the system is in observation mode, the microphones were recalibrated in the mode with the Wi-Fi module turned on. The results of calibration in the form of a fragment of echograms when fixing the software are shown in Fig. 9.

Calibration with the Wi-Fi module turned on gives results for which the time of arrival of the sound wave in the microphones completely coincides with the data in the test mode.

The `digital_calibration.ino` program is accordingly designed to calibrate microphones for receiving a signal in digital format (Fig. 10).

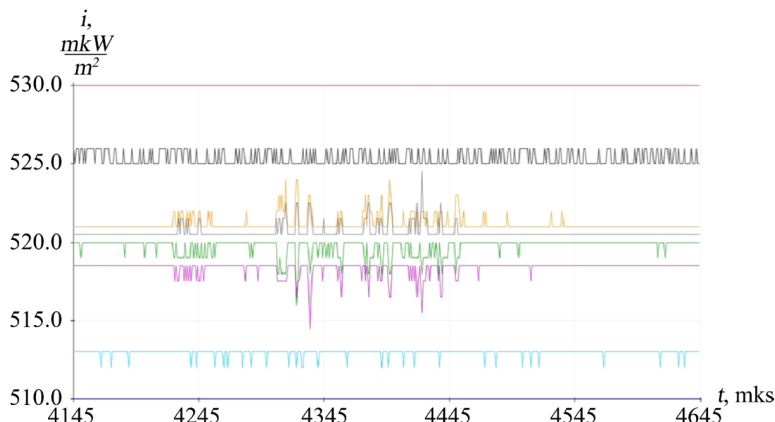


Fig. 8. Image of the echogram screen during microphone calibration:
 ● – mic_min0; ● – mic_max0; ● – mic_min1; ● – mic_max1;
 ● – mic_min2; ● – mic_max2; ● – mic_target_min; ● – mic_target_max

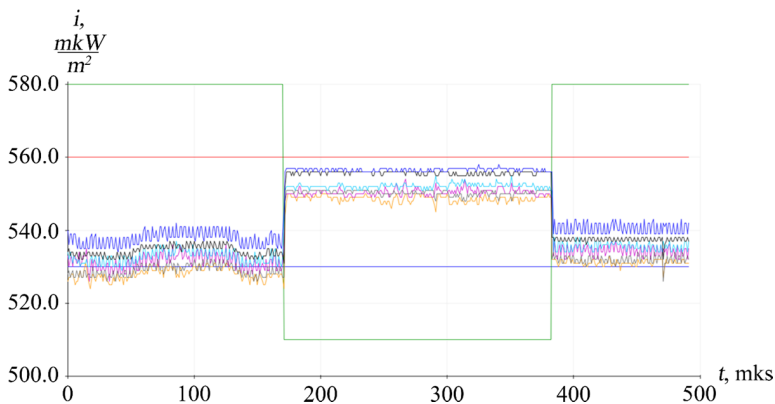


Fig. 9. Calibration of microphones with the included Wi-Fi module:
 ● – mic_min0; ● – mic_max0; ● – mic_min1; ● – mic_max1; ● – mic_min2;
 ● – mic_max2; ● – mic_target_min; ● – mic_target_max; ● – wifi_on

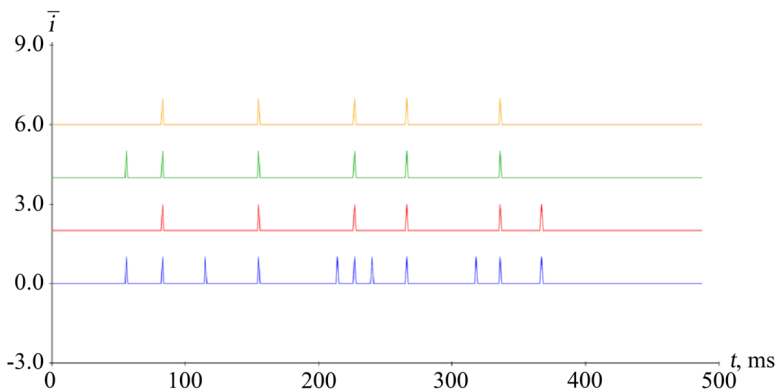


Fig. 10. Calibration of digital signal: ● – min0_has_detection; ● – min1_has_detection; ● – min2_has_detection; ● – all_have_detection

The digital_interrupt_calibration.ino program (Ukraine) is used to calibrate the sensitivity of microphones, that is, those analog signal values at which the threshold sensor on the microphone is triggered and generates a digital signal 1. This sensor is triggered only when the analog signal exceeds the threshold value adjusted by the control knob located directly on the microphone board. The program allows to keep track of the position of the handle of the microphone giving out digital signal 1 and, thus, allows to set the required intensity threshold. Adjusting the value of the threshold of the sound strength of the microphone allows it to be triggered when a shot is fired and not to react to noises, people talking, wind, etc. This setting adjusts the frequency of taking data about incoming sound signals. Thus, the goal is achieved to fix the full completeness of the signal spectra, and on the other hand, to prevent overloading the resource of the ARDUINO board.

The time_test.ino program in text format provides information on the time characteristics of the audio signal (Fig. 11), which is received and recorded by each of the microphones of the module.

Information presented in the form of a variable length array (Fig. 11) is not always convenient. The program also allows to check its accuracy and bring the information obtained to a tabular form, presented in Table 3.

```

Attempting to create access point with SSID: gan
SSID: gan
IP Adress: 192.168.1.147
cycle_delay_micros signal_offset_millis_0 signal_offset_millis_1 signal_offset_millis_2
430.5028 no signal detected
30.7614 no signal detected
30.7672 no signal detected
39.5200 0.000000 0.000052 0.001596
30.7614 no signal detected
40.2628 0.000000 0.000324 0.001288
30.9614 no signal detected
39.1464 no signal detected
40.0508 0.000000 0.000508 0.001392
39.2176 0.000000 0.000232 0.000121
    
```

Fig. 11. Time characteristics of the audio signal and the general signal

The first column (cycle_delay_micros) shows the cycle time for 10,000 iterations, the second, third and fourth columns show the time at which the signal enters each of the three microphones. It should be noted that if the signal about the presence of software is not registered, then all three microphones record a zero intensity value.

The mic_test.ino program (Ukraine) displays a diagram of incoming signals to each microphone on the monitor screen. It displays not only the output signals of each microphone, but also combined into one common signal (Fig. 12).

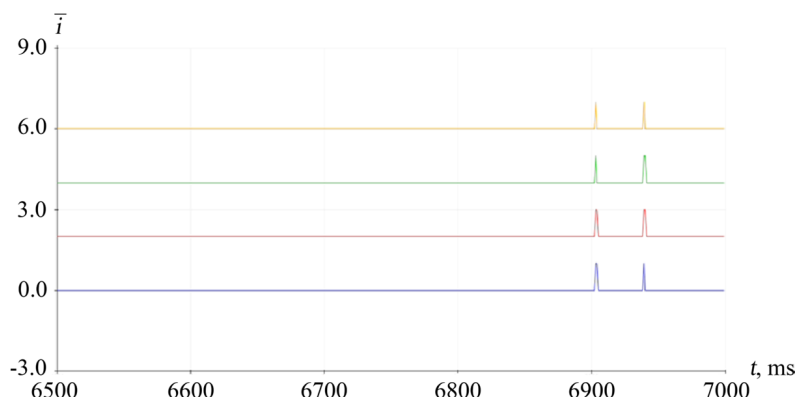


Fig. 12. Microphone signal diagram: ● – min0_has_detection; ● – min1_has_detection; ● – min2_has_detection; ● – all_have_detection

The webserver_test.ino program (Ukraine) tests the server installed on the ARDUINO UNO WIFI REV2 board. The program checks the connection of a mobile device to it and displays in text format the characteristics of this connection, including the name of the Wi-Fi point, its login, password and server IP address (Fig. 13).

```

Attempting to connect to WPA SSID: gan
You're connected to the networkSSID: gan
BSSID: E8:94:F6:04:06:74
signal strength (RSSI) :-42
Encryption Type:4
    
```

```

IP Address:192.168.1.147
192.168.1.147
MAC address: 84:OD:8E:34:C1:OC
SSID:gan
BSSID: E8:94:F6:04:06:74
signal strength (RSSI) :-42
Encryption Type:4
    
```

Fig. 13. Characteristics of Wi-Fi hotspot connection

After installing the developed software and launching the mobile device, the interface image is presented in the form (Fig. 14).

During the tests, SA simulator was used as a source of sound anomalies emitting signals, in which the sound was reproduced by the impact of two metal

Table 3

Results of experiments with temporary software fixation presented in tabular form

cycle_delay_micros	shot_offset_millis_0	shot_offset_millis_1	shot_offset_millis_2
68.8964	0.0	0.0	0.0
69.0860	0.0	0.0	0.0
67.0716	7.533212	7.534288	7.533788
69.5124	0.0	0.0	0.0
69.4640	0.0	0.0	0.0
65.4504	11.615592	11.616453	11.616161
69.3380	0.0	0.0	0.0
62.1892	12.837321	12.838344	12.837897
69.5268	0.0	0.0	0.0

objects. As a result of the impact of the latter in positions with different coordinates, the microphones of the receiving device recorded a sound wave passing through them as a function of time. The signal_locator3.ino program, in turn, recorded and memorized the input time of the wave through each microphone in the form of an array. Further, after the smartphone receives a signal that the receiving device is ready to transmit the array, the “RECEIVE DATA” button becomes active. After pressing the latter, information is sent to the smartphone via a wireless communication channel and is displayed on the screen. After pressing the “CALCULATE SOUND SOURCE COORDINATES” button, the calculation results appear at the bottom of the screen.

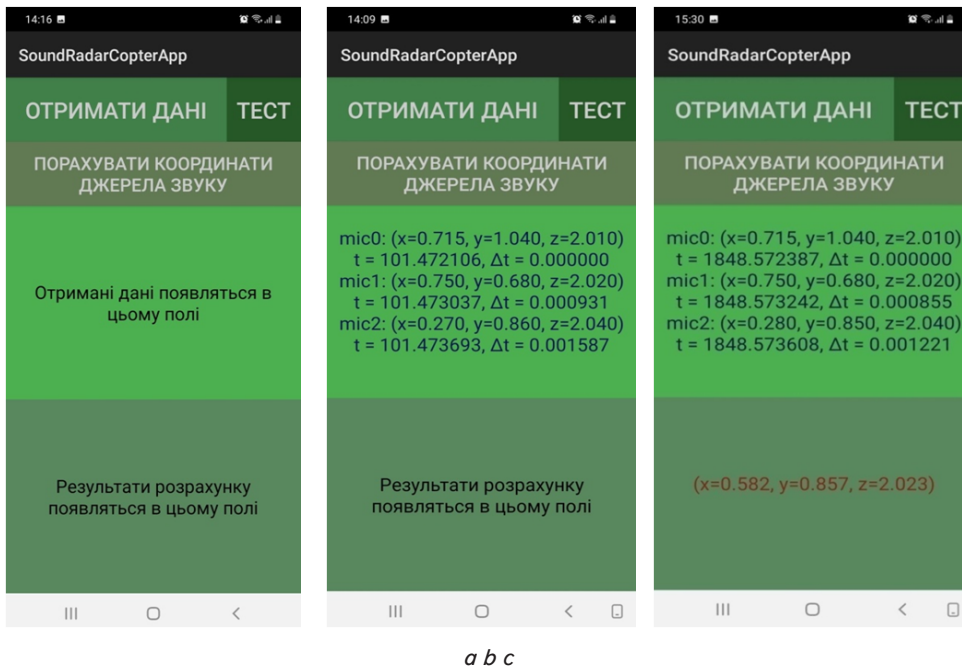


Fig. 14. Screen image of a mobile device when displaying the results of experiments: *a* – waiting mode for input data based on the results of the analysis of sound series; *b* – mode of outputting data on the coordinates of the location of the microphones and the time of fixation of sound anomalies; *c* – mode of output of coordinates of sound anomalies

Checking the data received by the smartphone from the receiving device showed that they completely coincide with the results obtained with independent calculations based on physical distance measurements using a conventional tape measure and calculator.

6. Discussion of the results of the study of the composition and operating modes

In the course of studying the functioning of the hardware and algorithmic software complex in various modes, the efficiency of the implemented idea of the minimum structure was proved. The ability of a module to research SA and determine its coordinates is especially important. The results of the experimental fixation of SA as a time series, as well as the determination and transmission to a mobile device of the values of identical points in time, recorded by different microphones, demonstrated the following possibilities.

The quality of the flowchart, which was considered to be the achievement of a complete list of its functions, based on

the complex method generally accepted in quality assurance technologies, is a key concept in the formation of a dynamic programming problem. The abstract representation of the maximum possible structure and the description of each of its elements by the functions of quantitative or qualitative measurement ensure the effectiveness of the approach. So, after normalization according to one of the Euclidean norms, the problem is reduced to finding a minimal structure. Another reason for efficiency is that the integer optimal design problem is turned into a dynamic programming problem with inequality constraints. The objective function of this task is formed as an additive one (1), and the constraints are presented through physical functional requirements (2). Solution (4), obtained according to the principle of R. Bellman, opened a new strategy of actions for the designers as a simple and reasonable strategy of two types of steps. They are applicable to the design of block-diagram structures in general and by a partial example of the minimum structure problem. At the first step, the minimum number of parallel links N_{min} was selected (for our case, N_{min} microphones). In the second step, the execution elements of the combined functions are selected from the database of modern and available execution units. So, for example, for this CS, an element was chosen that combined the functions of the controller and the communication channel, satisfying the list of requirements for the openness of the code, the availability of programs, reliability and the number

of inputs, which must be greater than N_{min} . In other words, the item with the highest quality is selected. The application of this technique made it possible to form and manufacture a CS with a minimum structure. However, a key role in this is played by the proposed new formulation of problem (1), (2) and its solution (3)–(5).

Ensuring the accuracy of determining the coordinates of sound sources, as works [20, 31–33] show, determines the accuracy of fixation by different microphones of the time moments of the same wave phase. An algorithm for determining the same phases of echograms based on time series data recorded by different microphones for SA with a complex time course is constructed. Based on the analysis of solution (10), it can be seen that the obtained invariance with respect to the SA amplitude waves. This dependence of the existing algorithms on the amplitude severely limited the operating range of the devices, since the amplitude of the acoustic wave intensity is inversely proportional to the square of the distance from the source to the receiver. This invariance with respect to the amplitude (the amplitude is the common factor of the numerator and denominator of formula (10))

makes it possible to use this algorithm to determine the time of one phase. Let's note that it is the dependence of the algorithms on the amplitude that contributed to the error in determining the coordinates. In addition, such invariance will allow collecting data from different sources to one controller, which opens up ways for the practical application of structural systems with an unlimited number of observation sources.

The equipment developed and manufactured according to this structure (Fig. 2) makes it possible to study software, calibrate and conduct a controlled experiment remotely. So, by reducing the number of single-chip controllers, choosing the block diagram of the signal_locator3.ino program (Fig. 3), a workable device for determining the coordinates of the SA was designed and manufactured (Fig. 4). The design of the structural elements of the device (Fig. 4–7) was carried out on the basis of the results of studies of methods for determining the coordinates of SA [19] and taking into account the development of the element base of electronic blocks. It should be noted that the input data for the algorithm [19] are the moments of time, which are proposed to be determined based on (10). The latter is a development of ideas and will increase the overall efficiency of such systems. This provided remote wireless control of the course of the experiment by analog (Fig. 8, 9) and digital (Fig. 10) calibration of microphones, fixation and study of sound sequences, and approbation of express computation algorithms. The number of microphones, their coordinates, location and orientation in space have a significant impact on the accuracy of determining the coordinates of the SA source. According to the data of [19], three microphones are their minimum number, which is necessary for the implementation of the algorithm for calculating coordinates. ARDUINO UNO WIFI REV2 board has 6 analog and 5 digital inputs. The maximum number of microphones is limited by the properties of the board and cannot be more than five. Under these conditions, after amplification and matching, information from microphones is read in analog and digital modes. When the number of microphones is increased to six, the information will be read in digital format only from five. In addition, since the channel numbers are determined each time depending on the order in which the sound wave arrives, they must be reconnected in the same order. The need for commutation will introduce significant complications into the structure of the CS. In this regard, for this board, ARDUINO UNO WIFI REV2, the decision to limit the number of microphones to five is reasonable. The use of the ARDUINO UNO WIFI REV 3 board allows to increase the number of microphones to 8 with 6 redundant PWM outputs. However, in this case, there is no recording of analog signals from the outputs, which means that the CS becomes incapable of conducting an experiment to study the properties and spectrum of SAs of different origins. Thus, the completeness of the functions of the minimal structure built on the proposed UNO WIFI REV2 is confirmed by the data in Table 3 and Fig. 11–14.

In addition, during the experiment, a decrease in the supply voltage was observed when the Wi-Fi module was connected, which formed ambiguous conditions for calibration and operation (Fig. 8). Elimination of this drawback can go in two ways: calibration with a connected Wi-Fi module, as is done in this work, and development of a source with a stabilized supply voltage. However, the second option will significantly complicate the CS scheme as a whole.

The developed CS of the minimal structure made it possible to investigate the algorithms for determining the characteristic phases of the SA, which are invariant for various microphones. The formation and testing of the hardware implementation of the CS allows one to study algorithms for choosing a set of conditions that uniquely distinguish the type and characteristic phases of the software, and are based on the wave representation and the integral norm. As proved in the course of the experiment, algorithms based on a simple comparison of intensity values are suitable only for processing simple single strokes. Naturally, its simplicity, which is necessary for testing the system's performance as a whole, will become a limitation for SA of a complex nature. For example, determining the facts of the work of dust, tractors during logging or when determining accidents on the highway gives complex sound sets. Under these conditions, the creation of other algorithms, including intelligent ones, will acquire relevance in the future. Further application of the improved algorithms will make it possible to conduct research on various types of SA, the verification and testing of which will ensure the correction and adjustment of the CS for remote wireless control of the experiment. The aforementioned problems of processing sound sequences, caused by the imposition and recognition of SA based on the combination and sequence of sounds, limit the use of such systems in production or on city streets. Of course, their solution will require theoretical and experimental studies, including those proposed by mobile CS. Together with new capabilities, such CSs determine the range of new research on the development and implementation of automated systems for determining the coordinates of the SA.

7. Conclusions

1. The CS structure of the SA registration system is formed, which includes microphones, amplifiers and adapters for digital and analog signals, the ARDUINO UNO WIFI REV2 board with an integrated Wi-Fi module and a personal computer and smartphone based on the Android processor is minimally complete. It implements the functions of remote wireless control of the course of the experience. The SA set of the receiving module installed on the ARDUINO board is transferred and installed from the PC. Calibration ensures the functionally complete operation of the system: data collection of sound sequences, calibration and software extraction, and data transfer to a smartphone or PC via a wire and a Wi-Fi channel where the type and coordinates of the SA are recorded and determined. The number of microphones is selected from 1 to 5 according to the conditions of the problem, but their number is limited by the number of digital inputs of the ARDUINO UNO WIFI REV2 board.

2. Wave representation of the law of temporal configurations of intensity and integral norm of acoustic wave. The software provides the latest capabilities of remote wireless control of the course of the experience of fixing and studying sound sequences. Approbation and comparison of the efficiency of the algorithm for determining the phases of echograms of various origins, recorded by three different microphones, allows to select a set of conditions that uniquely distinguish the type and characteristic phases of the SA. Expressions for determining the time instant of the same phase by different microphones is amplitude invariant, that is, independent of the distance to the source.

3. Testing of a computerized system, calibrated and suitable for monitoring and observing sound sequences in the presence of SA fixation, demonstrates the functionality of the proposed minimum structure. It allows the implementation of the algorithm for determining the SA coordinates in fact and the given time of its fixation by three different microphones, which, if necessary, expand to five.

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