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MICROPROCESSOR DEVICE DETERMINING DIRECTION OF SOUND TO IMPROVE TECHNICAL EQUIPMENT AT WEAPONRY

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The problem of designing microprocessor devices for determining the direction to the source of sound with high accuracy is considered. The basic developments of leading foreign and domestic researchers on similar solutions are analyzed. The microprocessor device has been designed to determine the direction to the sound source using a modern element base. The structural and functional schematic diagram of the proposed device is described in detail. The reliability of the scheme and the average time of device to failure are calculated. Several variants of the algorithms for calculating directions to the sound source are considered. The microphone amplifier circuit was simulated to test the device's performance.

Key words: microprocessor device, design, direction on sound source, reliability, microphone amplifier.

Introduction. With the modern development of military weapons there is an urgent need device to help determine enemy's concealed firing position on battlefield. Detection of sniper positions is very important to improve the survival of both advanced and damaged crews.

Traditionally, an acoustic system consisting of a base with three microphones and an electronics unit is used to detect the enemy's firing position [1,2]. Such device performs a continuous analysis of the sound environment on the battlefield and measures the delay of the arrival of the sound wave of a shot to each of the microphones, which depends on the position of the sound source.

To determine the firing position, the device determines the delay of the received audio signals on each pair of microphones. The delays between the signals determine the maximum correlation of the signal pair. Each pair of microphones gives the system a pair of solutions (one real and one phantom). Three pairs can be distinguished from three microphones, i.e. six

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solutions. Then, based on this data, the system filters out phantom solutions and determines the direction of sound arrival.

Two types of sensors are used to solve the problem of sound recording (pressure gauge or oscillator). An ordinary microphone and a human ear are pressure recorders. Practically only these types of recorders are used in the analysis of sound fields in gas environments. In hydroacoustics oscillatory speed recorders are often used [3]. In this regard, it is necessary to develop specialized microprocessor devices that will allow with greater accuracy to determine the direction of the sound source.

Analysis of developments and publications. The main works on determining the direction of a sound source are focused on the development of methods and means for determining the distance to the source of acoustic signals [4-6]. In [7] method of gradient processing of acoustic signals and determining the distance to the source of acoustic signals was proposed. Scatterplots of acoustic signals and their power attenuation in the propagation medium are also evaluated.

A number of researchers have focused their work on theoretical and model calculations of the possibilities of creating a sound source coordinate determination system [4,8]. Also, the method of gradient descent with step crushing as an algorithm for constructing mobile applications is proposed and implemented.

In [5,6,8] some cases of localization of sources of shot at parallel orientation of the trajectory of the ball and normal to the acoustic base are considered. A localization scheme was also proposed, which simplified the calculation algorithm, compared theoretical calculations and experimental studies.

The studies performed in [9,10] are aimed at developing the structure of a distributed automatic sound artillery reconnaissance system. It is implemented as a system of standalone speakers located in the area by known geographical coordinates, connected via wireless communication channels to the server. Such system reduces the time of detection and identification of targets compared to analogues. This system is also capable of being implemented on the basis of a cellular communication system.

However, despite the widespread use of similar systems, their primary purpose is the military. Therefore, most of the developments are focused on defense topics. The aim of this work is development of own cost efficient microprocessor device for determining the direction of the audio source with high accuracy, as well as to develop algorithms for calculating directions to the audio source.

Design of device. A number of similar circuit solutions for existing prototypes were used [11]. The core 32-bit microcontroller STM32 is selected (Fig. 1) in the form of an MCU, which performs all steps to implement the sound processing algorithms of the designed device. As input signals electrical oscillations in analog form were used at the outputs of three microphones (Fig. 1). The received signals are amplified by microphone amplifiers with program-controlled gear ratio up to the nominal ADC level. Using such a microphone amplifier with software-controlled amplification will allow to flexibly respond to the sound level of the medium and maintain a nominal input signal level at the ADC input. This approach allows to reduce both the noise level (at low input level) and the level of signal distortion (at high signal strength).

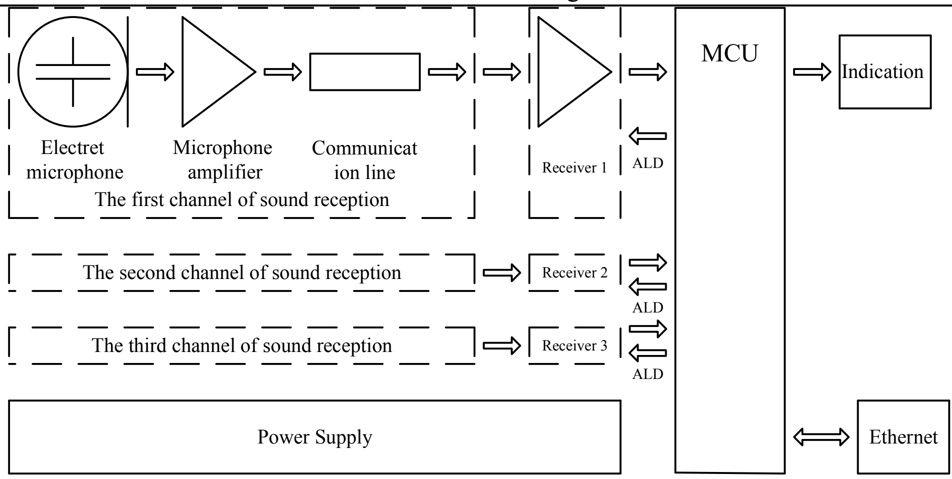

Fig. 1. Structural scheme of device for determining the direction of the sound source.

As noted above, the device contains three remote microphones that are attached via cables. The microphone amplifiers in each channel are split into two parts. The first part is located near the microphone and converts the output signal into a differential signal with low internal resistance. The second part of the microphone amplifiers is located inside the device itself, converting the differential input signals to the normal shape, removing the common-mode interference that could be caused by the communication line. The amplifiers of the second parts of the microphone amplifiers are program-controlled, allowing be programmatically adjusted to the nominal levels that require ADC inputs. It should be noted that in this development the built-in MCU ADC are used, which makes it easy to convert analog signals to digital form, and process them according to the necessary algorithms. The obtained results are displayed on the display panel and transmitted via Ethernet channel for further analysis. For the proper functioning of the device it is also necessary to have a standalone power supply that will provide energy to all the nodes.

Functional diagram of the device for determining the direction of the sound source is shown in Fig. 2. All functionality is implemented by MCU software and resources. The external elements of the circuit perform function of bringing the signal levels to nominal levels. It is known that modern microcontrollers have a well-developed structure and use specialized real-time operating systems to realize all their capabilities. In this case, the RTOC public key operating system will be used. It allows you to build applications using multiple threads that perform their functions in real time. The implementation of algorithms will be implemented as separate threads using an advanced interrupt system.

To receive nominal-level ADC inputs, individual streams will generate control signals for the amplitude gain control (AGP) system. This will allow to use the ADC readout values and generate such control signals to the ADC systems, that slowly change the transmissions of the microphone amplifiers. The ADC outputs receive digital data streams that are fed into the ring buffer for accumulation (necessary to implement the algorithm for finding mutual correlation). The ring buffers are constructed in such a way that each new data displaces the old obsolete data from the buffer.

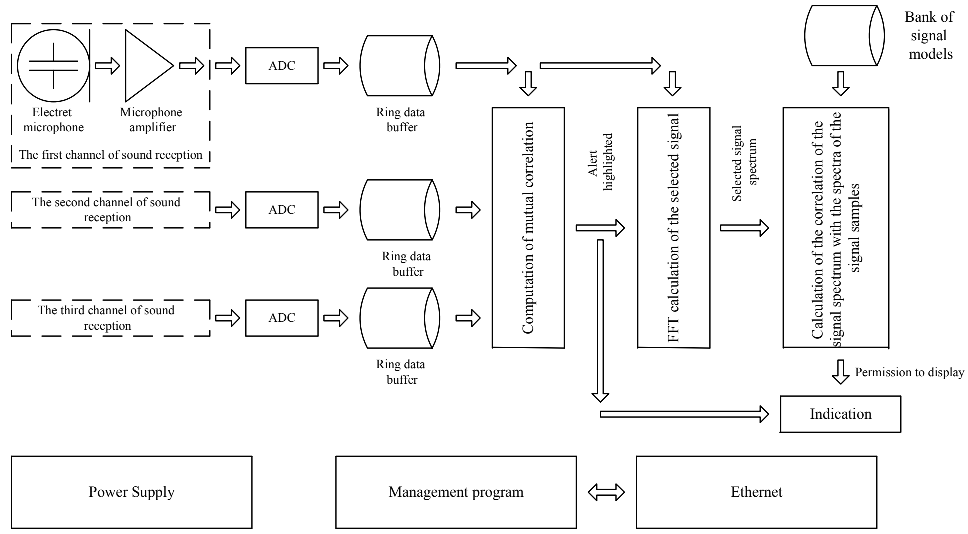

Fig. 2. Functional scheme of device determining the direction of the sound source.

In the first step, the correlation search algorithm has the ability for each of the receiving channels to find matching data with a delay. If such data fragments are detected, the mutual delay of the sound input in each of the ring buffers will be calculated. After comparing the data, the algorithm can uniquely calculate the direction to the sound source. In the second step, the selected signals are transformed by the FFT algorithm into a spectral representation [12], which can be interpreted as an array of data suitable for recognizing the type of sound source. In the third stage, the type of sound source is recognized. For this purpose, the array of spectral readings of the received signal is compared with the base of signal types. The comparison is made using the reciprocal correlation algorithm. The sample that gave the highest number of matching characteristics is considered to be the signal type.

In addition to determining the types of signals, the software with the ability to memorize the parameters of unknown signals in the database can be provide. The detected signals are shown on the display panel (built using three-color LEDs), which not only indicates the direction of the sound source, but also to quickly display the characteristic of the received sound, highlighting the shots of snipers in red.

The microcontroller program has a general control procedure, which ensures that the device performs all its functions, as well as ensures the smooth flow of data through all blocks and display the received information for indication. In addition, it transmits data to an external server for further analysis.

In the implementation of the electrical schematic diagram of the device used modern element base and circuit solutions. The STM32F4 microcontroller with integrated ADC and DAC makes it much easier to work with analog signals. The ADC setting allows for one-time and cyclic measurements. For the conversion at maximum speeds it is necessary to observe the voltage range of 2.4 ... 3.6 V. To control the internal temperature microcontroller built-in temperature sensor is used. The output of sensor through the multiplexer is connected to the ADC. The structure of the microcontroller includes integrated Embedded Trace Macrocell cells, which greatly expands debugging functions allowing to observe the flow of instructions and data in the middle of the CPU kernel in real time.

Electret microphones are used in this development. A typical scheme of input is shown in Fig. 3. However, its main disadvantage is the mismatch of parameters (the required level of amplification of the weak and the compression of loud sounds at short and long distances) for the tasks that must be solved by the microphone input of the projected apparatus.

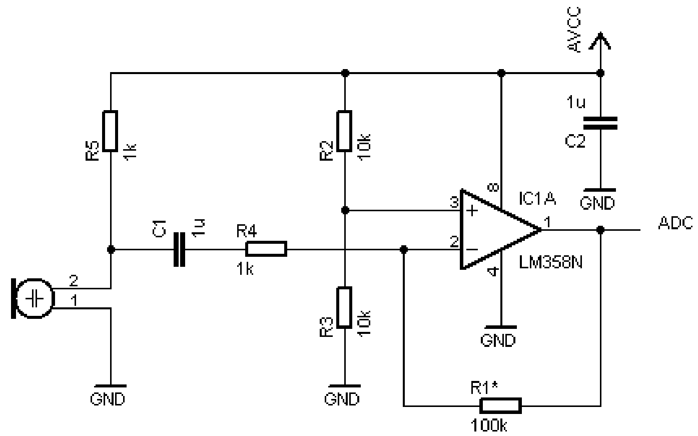

Fig. 3. The standard circuit for the input of electric microphone.

Using the Multisim program, the microphone amplifier path was simulated (Fig. 4). The microphone capsule is modeled by a Q1 field transistor and a R_1 resistor. The aim of resistor R1 is protect the output of the operational amplifier U1A from overload during operation of the device. The operational amplifier U1B is used in the signal inverter circuit. Resistors R6, R7 form a feedback circuit and set the gain to 1. The output of the operational amplifier U1B is fed through a resistor R9, which is designed to protect the output U1B from overload during operation. Since the cascade on the operational amplifier U1B inverts the signal, at the input of the model line, the differential signal pair is removed from the microphone. Long-distance signal transmission in the form of differential signal reduces the possibility of interference with the signal. The inductances L1… L4 and the capacitor C3 allow to simulate the effect of a long line on the signal parameters by selecting their values. Resistor R4 and capacitor C2 implement a voltage generating unit “0” for the cascade on the operational amplifier U1B. The constant voltage component at the microphone output depends on the supply voltage and ambient temperature. Resistor R4 is connected to the low-ohm output of the repeater on the U1A operational amplifier. Thus, the circuit element that forms the level “0” (microphone) is not overloaded and operates in the usual mode. The generated level “0” through the resistor R5 is fed to the positive input U1A. Because the repeater on the operating amplifier U1A repeats the voltage taken from the microphone, the inverted signal that forms the operational amplifier U1B will have a constant component close to the constant component signal at the output of the operational amplifier U1A with any changes in the temperature and supply voltage of the microphone. The work of the cascade on the U1B is stabilized at DC.

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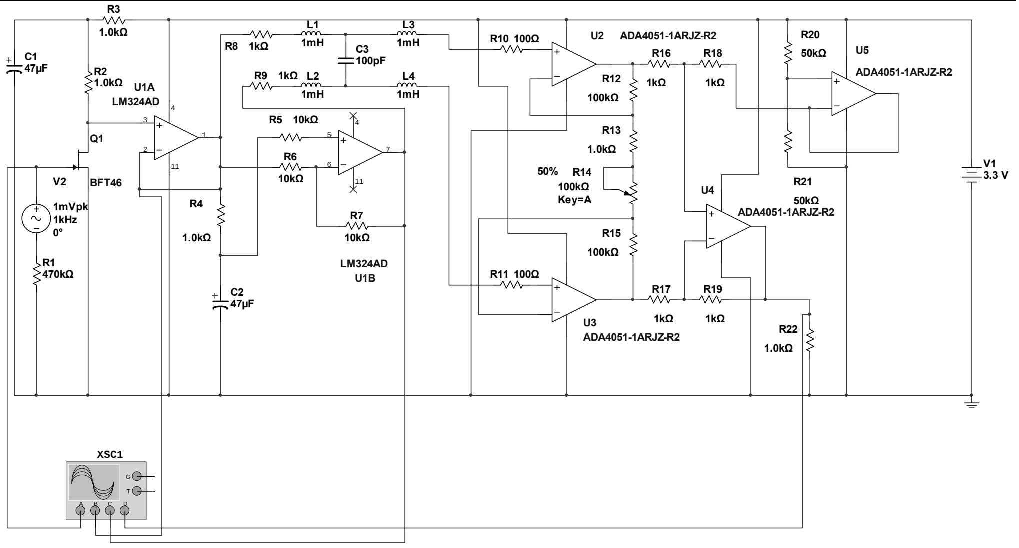

Fig. 4. Model of the microphone tract.

One of the negative points to eliminate is noise. Noise from interference (external sources) is added to the signal at the time it passes. Accordingly, the input of the amplifier receives a signal, which is the sum of noise and the initial signal. The noise level can be low that it cannot be heard on the background of the signal. With the arrival of the stroke to the differential input amplifier, to both differential signals, positive and negative, approximately the same noise signal is added. In this case, the differential pair is connected to a differential amplifier, which subtracts the phase-shifted signal. After processing by the amplifier, the noise is suppressed and the level of the useful signal doubles, going to zero.

The differential signal amplifier is built on the U2 ... U4 operational amplifiers, the input part is implemented on the U2 and U3 operational amplifiers, which are included in the repeater mode and provide high input impedance of the cascade. They provide all the amplification of the input signal. For the above scheme, the voltage gain is:

$$G = (1 + \frac{R_{12}}{R_{13} + R_{14}} + \frac{R_{15}}{R_{13} + R_{14}}) \cdot \frac{R_{18}}{R_{16}}, \tag{1}$$

at $\frac{R_{18}}{R_{16}} = \frac{R_{19}}{R_{17}}$. Because $R_{16} = R_{17} = R_{18} = R_{19}$, then $\frac{R_{18}}{R_{16}} = 1$.

Focusing all gain on the input of the differential stage is necessary to stabilize the “0” of the entire differential amplifier. The reference voltage is fed into the differential amplifier circuit through resistor R18. The transmission factor for the input voltage reference is one. The use of a buffer on the U5 OP with low power consumption between the voltage divider R20 and R21 and the input of the reference voltage (resistor R18) reduces the effect of the resistance of the voltage divider on the temperature stability of the differential amplifier. The low output impedance of the buffer cascade on the OP U5 eliminates the need for resistance selection and the problem of resistors with different TCS, and also allows to adjust the reference voltage of the differential amplifier. Resistor R22 simulates the input resistance of the ADC.

To test the performance of the scheme a simulation was performed. The ability of the circuit to operate in operating mode (charge capacitor C2) is shown in Fig. 5.

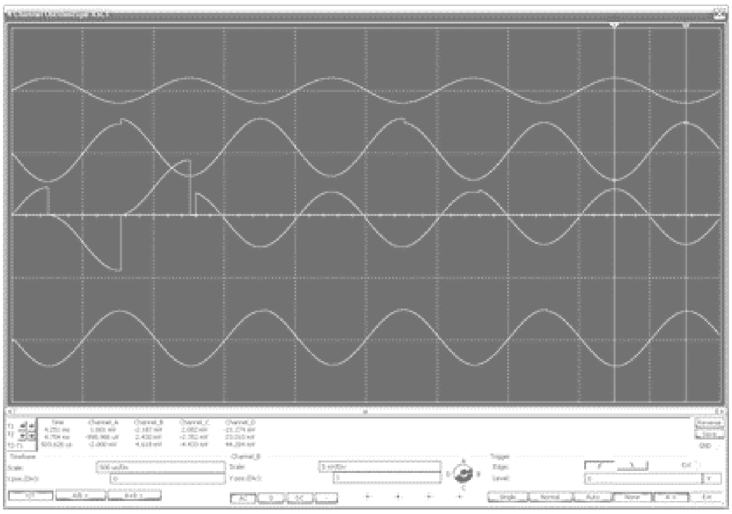

Fig. 5. The results of simulation of the transition process at power on.

Checking the gain of the microphone amplifier for voltage for different levels of the output signal is shown in Fig. 6 and Fig. 7. The maximum output signal level at load was 2.6 V, which is less than the full power range of 3.3 V at 0.7 V (Fig. 6). Further levels of the output signal are further distorted (Fig. 7). The output voltage span was 2.9 V, which is smaller than the full 3.3V by 0.4 V power supply.

Thus, the use of operational amplifiers (AD8603AUJZ type Rail-To-Rail) allows to receive on a load of 1 kOhm a signal with a magnitude of 2.5 V without distortion. However, the signal level will be higher due to the higher input resistance of the ADC.

The device contains three channels of microphone amplifiers. For the first channel, an electret microphone (Mic1.1) is loaded on a 1 kOhm resistor (R1.1).

The D1.1a operational amplifier is switched on with a high input impedance repeater, removes the signal from the microphone. The output of the D1.1a operational amplifier is fed through resistor R1.7 to terminal 2 of connector X1.1. Operational amplifier D1.1b is used in the signal inverter circuit. Resistors R1.4, R1.6 form a feedback circuit and set the gain to unity. Since the signal inverted on the D1.1b operational amplifier, we obtain at the terminals 2, 3 of the connector X1.1 the paraphase signal received from the microphone. Resistor R1.3 and capacitor C1.3 implement a voltage generating unit “0” for the cascade on operational amplifier D1.1b. The constant voltage component at the microphone output depends on the supply voltage and ambient temperature [12]. The cascade on the D1.1a operational amplifier uses it as a voltage of “0”. For the cascade on the D1.1b operational amplifier it must be formed.

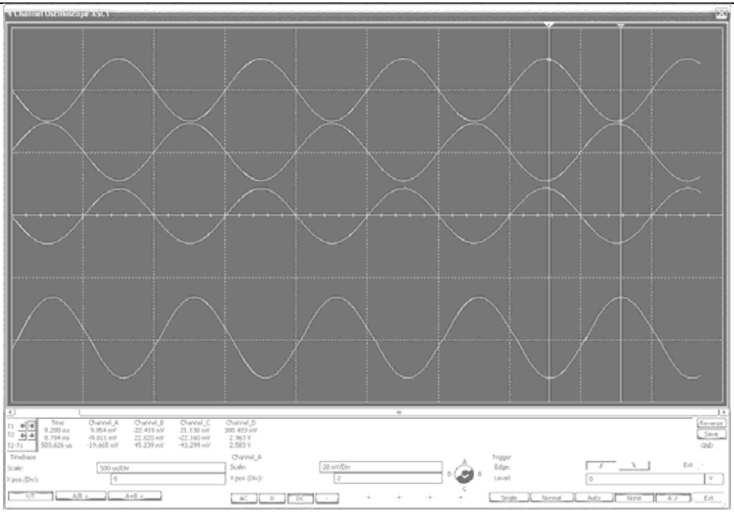

Fig. 6. Full signal span without distortion.

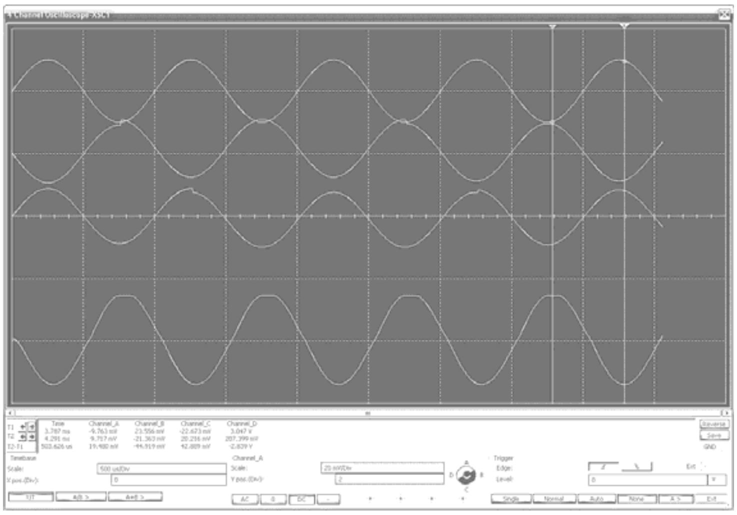

Fig. 7. Full swing, the beginning of distortions.

Couplings R1.3, C1.3 form a low pass filter, and a constant voltage is formed on the capacitor C1.3, which forms a level “0” for the cascade on the operational amplifier D1.1b. Resistor R1.3 is connected to the low-Ohm output of the repeater on the D1.1a operational amplifier. Thus, the circuit element that forms the level “0” (microphone) is not overloaded and operates in the usual mode. Generated level “0” through resistor R1.5 is fed to the positive input D1.1b. Since the repeater on the D1.1a operational amplifier repeats the voltage obtained from the microphone, the inverted signal forming the D1.1b operational amplifier will have a constant component close to the constant component signal at the output of the D1.1a operational amplifier with any changes in temperature and supply voltage microphone. Resistor R1.2 together with capacitor C1.1 forms a voltage filter for the microphone stage. Their use reduces the possibility of excitation of the microphone cascade.

According to the recommendations of manufacturers, the capacitor C1.2 (0.1 μF) should be physically located near pin 8 of chip D1.1, which will reduce the excitation of chip D1.1. Capacitors C1.4 and C1.5 filter the power supply of the entire "Output Microphone Amplifier 1". The C1.4 electrolyte eliminates low-frequency interference on the power line. The C1.5 ceramic capacitor blocks the high-frequency component of the interference on the power line. In the developed device a microphone signal is sent to the X1 connector.

The phase signal is applied to the inputs of the differential amplifier implemented on the D2 ... D4 circuits. The interference signal on the cable will be removed by the amplifier, with the noise suppressed and the useful signal level doubled. Operational amplifiers D2 and D3 are included in repeater mode and provide high input impedance to the cascade. They provide amplification of the input signal. Resistors R1 and R2 protect the inputs of the amplifiers from overcurrent.

At modeling with the Multisim package, the values of gain factors within 5 ... 200 for the entire path of the microphone amplifier are obtained, which fully meets the needs of amplifying the microphone signal to the input levels of the ADC.

On the chip D4 implemented the output of the differential amplifier. Its main task is to convert a single-phase signal into a unipolar one. The use of a buffer on OP D13 with low energy consumption between the voltage divider R28 and R29 and the input of the reference voltage (resistor R9) reduces the influence of the resistance of the voltage divider on the temperature stability of the differential amplifier. The low output impedance of the buffer cascade on OP D13 eliminates the need for resistance selection and the problem of resistors with different TCS, and also makes it easy to adjust the differential voltage of the differential amplifier.

The designed device contains three channels of microphone amplifiers, each of which has an input (X1… X3). The reference voltage generated by the OP D13 is fed into the circuits of all three channels of the microphone amplifiers. From the outputs of the output parts of the differential microphone amplifiers, amplified signals from three microphones are fed to the ADC inputs for converting analog signals to digital form.

The STM32F405VG (D16) microcontroller controls the device and performs all necessary data processing. The universal MCU has the standard binding required peripherals. The microcontroller synchronization unit is implemented on Q2 quartz with elements C30, C31 and R40. The mode of operation of the universal MCU is set by means of jumpers J2, J3. The hardware reset node is implemented using the S3 button and pulls up to the logic unit level of resistor R42.

S1 and S2 buttons are used to control the device, enabling the operator to change the program quickly. The results of the device and the racing mode are displayed by the RGB VD1 ... VD64 LEDs in the dynamic display mode.

The D15 (W5100) network controller chip has remote access to the developed device via Ethernet. The hardware implementation of the TCP/IP protocol stack provides data rates of up to 25 Mbps and provides easy connection to the Internet without the involvement of operating systems and external computers. The chip implements the following protocols of the OSI (Open System Interconnection) transport, network, and link layers: TCP, UDP, IPv4, ICMP, ARP, IGMP, and MAC.

Therefore, the proposed device contains 4 circuits, which are controlled by the STM32F405VG microcontroller using the SPI interface. These are digital potentiometer chips (D1, D5, and D9) and a D15 network controller chip. The microcontroller uses common SDO, SDI, and SCK signals to exchange data and commands with the chips, and to allow the operation of each of the four chips, the microcontroller generates CS1… CS4 resolution signals. The data exchange is fully managed by the MCU (Master), and all other SPI peripherals work in Slave mode.

The circuit of the projected device is powered by an external voltage using a dedicated charger.

One of the important characteristics of this device is its reliability, as this problem is associated with all stages of its creation and use. It is known that the performance of a device or system is evaluated as the product of the probabilities of failure-free operation of the elements [13]:

$$P(t) = \prod_{i=1}^{n} Pi(t), \tag{2}$$

$P_i(t)$ – the probability of failure of the i-th element.

The device can be in one of two incompatible states: failure or performance. Therefore,

$$P(t) + Q(t) = 1; Q(t) = P(t) - 1\,, \tag{3}$$

where $Q(t)$ - the probability of failure of the device, which is determined by equation:

$$Q(t) = 1 - \prod_{i=1}^{n} P_i(t) \tag{4}$$

In the case of an arbitrary law, the distribution of time to failure for each of the elements:

$$P_i(t) = e^{-\int_0^t \lambda_i(t)dt}\,, \tag{5}$$

where $\lambda_i(t)$- failure rate of the i-th element.

The probability of failure of the device will accordingly be written by:

$$P(t) = \prod_{i-1}^{n} e^{-\int_0^t \lambda_i(t)dt} \tag{6}$$

From expression (6) it is possible to determine the probability of failure-free operation of the device before the first failure with any law of change of the failure rate of each of elements in time.

For the most common condition $\lambda_i(t) = const$, the expression for system uptime is as follows:

$$P(t) = \prod_{i-1}^{n} e^{-\left(\sum_{i=1}^{n} \lambda_i\right)}\,, \tag{7}$$

where $\sum_{i=1}^{n} \lambda_i$ can be represented as the failure rate of a system reduced to an equivalent element with a failure rate:

$$\lambda_0 = \sum_{i=1}^{n} \lambda_i = const\,. \tag{8}$$

In our case, calculating the reliability of the device for 1 year with the use of nominal signifiers will produce at rate of 1 year. The nominal failure rate of the elements was used in the calculation. The data for determining the failure rate are given in Table. 1.

Table 1.

Determination of failure rate

No	Element	Number of elements	Failure rate in normal mode, $\lambda_0 \cdot 10^6$, 1/time	General failure rate, $\lambda_0 \cdot 10^6$, 1/time
1	Microcircuit	20	0.013	0.26
2	Transistor	32	0.84	26.88
3	Diode	64	0.2	12.8
4	Generator	2	0.35	0.7
5	Resistor	236	0.043	10.148
6	Capacitor	39	0.06	2.34
Total		53.128		

Thus, the probability of failure-free operation for a given time (8760 hours) for λ_0= 53.128:

$$P(t) = e^{-53,128 \cdot 10^{-6} \cdot 8760} = 0,6279 . \tag{8}$$

Average time to failure:

$$T = \frac{1}{53,128 \cdot 10^{-6}} = 1,8822 \cdot 10^4 \text{ (~2 year).}$$

Conclusions. Thus, the microprocessor-based device for determining the direction to a sound source developed in this work can be used in military tasks with the joint use of algorithms and approaches for determining the direction proposed by other authors. The conducted modelling of the path of the microphone amplifier in the Multisim medium-higher testifies to the correctness of the selected element base of the designed device. The calculations show the reliability of the device for a given time.

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МІКРОПРОЦЕСОРНИЙ ПРИСТРІЙ ВИЗНАЧЕННЯ НАПРЯМКУ НА ДЖЕРЕЛО ЗВУКУ ДЛЯ ПОКРАЩЕННЯ ТЕХНІЧНИХ ЗАСОБІВ ОЗБРОЄННЯ

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Розглянуто проблему розроблення мікропроцесорних пристроїв для визначення напрямку на джерело звуку з підвищеною точністю. Проаналізовані основні розробки провідних закордонних та вітчизняних дослідників щодо аналогічних рішень. Спроектовано мікропроцесорний пристрій для визначення напрямку на джерело звуку з використанням сучасної елементної бази. Детально охарактеризовано структурну та функціональну схеми запропонованого пристрою. Ядром обрано 32–розрядний мікроконтролер STM32, який здійснює всі дії щодо реалізації алгоритмів опрацювання звуку проектованим пристроєм. Як вхідні сигнали використано електричні коливання в аналоговій формі на виходах трьох мікрофонів. Одержані сигнали підсилюються мікрофонними підсилювачами із програмно керованим коефіцієнтом передачі до номінального рівня роботи АЦП. Використання такого мікрофонного підсилювача звуку із програмно керованим рівнем підсилення дозволило гнучко реагувати на рівень звукового сигналу середовища і підтримувати на вході АЦП номінальний рівень вхідного сигналу. Такий підхід дозволив зменшити як рівень шумів (при малому рівні вхідного сигналу), так і рівень спотворень сигналу (при потужному сигналі).

Розглянуто декілька варіантів алгоритмів обчислення напрямків на джерело звуку. Показано, що на першому етапі, алгоритм пошуку кореляції має можливість для кожного із каналів прийому звуків шукати співпадаючі дані із внесеною затримкою. При виявлені таких фрагментів даних буде обчислюватися взаємна затримка надходження звуку у кожному із кільцевих буферів. Після порівняння даних алгоритм зможе однозначно обчислити напрямок на джерело звуку. На другому етапі виділені сигнали перетворюються за допомогою алгоритму ШПФ в спектральне представлення, яке можна трактувати як масив даних, придатних для розпізнавання типу джерела звуку. На третьому етапі відбувається розпізнавання типу джерела звуку. Для цього масив спектральних відліків одержаного сигналу порівнюється із базою типів сигналів. Порівняння відбувається за допомогою алгоритму обчислення взаємної кореляції.

Для перевірки працездатності пристрою проведено моделювання тракту мікрофонного підсилювача у середовищі Multisim, що свідчить про коректність вибраної елементної бази проектованого пристрою. Запропоновані моделі реалізовані у принциповій схемі проектованого пристрою наведення на джерело звуку.

Розрахована надійність схеми та середній час напрацювання пристрою до відмови. Проведені розрахунки свідчать про надійність роботи пристрою упродовж заданого часу.

Ключові слова: мікропроцесорний пристрій, проектування, напрямок на джерело звуку, надійність, мікрофонний підсилювач.

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