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RESEARCH ARTICLE

A device for calling emergency operational services to provide voice communication between the driver of a two-wheeled vehicle and the operator of the ERA-GLONASS system

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Abstract
Objectives. The aim of the study is to improve road safety by developing an emergency call device for drivers of two-wheeled vehicles, as the most vulnerable road users, and improving their technical equipment.
Methods. In the course of the study, the characteristics of the acoustic signal transmission channel and the processes accompanying its propagation were analyzed. When studying the parameters of voice communication, noise reduction, echo cancellation and echo compensation methods were used, as well as algorithms for converting acoustic information implemented in the hardware and software of the device.
Results. The results of practical implementation are presented: the design of a prototype device, its integration into the dashboard of a two-wheeled vehicle. During the design of the device, the control features of a two-wheeled vehicle, the influence of external factors and climatic conditions were taken into account. An implementation of the interface of interaction between the driver of a two-wheeled vehicle and the operator of the ERA-GLONASS system is proposed, taking into account the specifics of its use. Structural schemes of an echo compensator and a dual speech signal detector using an adaptive filter are presented. The algorithms implementing these processes and the possibility of their adaptation to the tasks of the emergency call device are considered. The procedure for automatically adjusting the amplification of the acoustic signal of the speech range is described, an analytical description of the technical problem and the applied methods of digital processing are given. A structural diagram of the test stand, software for qualitative analysis of the acoustic signal, visualization of the test results of the prototype are presented, and the effectiveness of the proposed solution is evaluated.
Conclusions. The results of a study on the design of an emergency call device have shown that the use of analog and digital speech signal processing algorithms implemented in the device's codec and modem will ensure a high-quality level of voice communication between the driver and the emergency services operator.

Keywords: two-wheeled vehicle, acoustic signal, duplex voice communication, echo cancellation algorithm, noise reduction algorithm, normalized least squares algorithm, digital signal processor, LMS algorithm, NLMS

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НАУЧНАЯ СТАТЬЯ

Устройство вызова экстренных оперативных служб для обеспечения голосовой связи водителя двухколесного транспортного средства и оператора системы «ЭРА-ГЛОНАСС»

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Резюме
Цели. Целью исследования является повышение безопасности дорожного движения за счет разработки устройства вызова экстренных оперативных служб для водителей двухколесных транспортных средств, как наиболее уязвимых участников дорожного движения, и улучшения их технической оснащенности.
Методы. В ходе исследования проанализированы характеристики канала передачи акустического сигнала и процессов, сопровождающих его распространение. При исследовании параметров голосовой связи применялись методы шумоподавления, эхоподавления и эхокомпенсации, а также алгоритмы преобразования акустической информации, реализованные в аппаратно-программной части устройства.
Результаты. В ходе проектирования устройства учтены особенности управления двухколесным транспортным средством, влияние внешних воздействующих факторов и климатических условий. Предложена реализация интерфейса взаимодействия водителя двухколесного транспортного средства с оператором системы «ЭРА-ГЛОНАСС», учитывающая специфику его использования. Приведены структурные схемы эхокомпенсатора и детектора двойного речевого сигнала с использованием адаптивного фильтра. Описана процедура автоматической регулировки усиления акустического сигнала речевого диапазона. Рассмотрены алгоритмы, реализующие эти процессы, и возможность их адаптации к задачам устройства вызова экстренных оперативных служб. Показаны результаты практической реализации опытного образца устройства: конструкция, его интеграция в приборную панель двухколесного транспортного средства. Приведены структурная схема тестового стенда, программное обеспечение для качественного анализа акустического сигнала, оценена эффективность предложенного решения.
Выводы. Результаты исследования по конструированию устройства вызова экстренных оперативных служб показали, что применение алгоритмов аналоговой и цифровой обработки речевого сигнала, реализуемых в кодеке и модеме устройства, позволит обеспечить качественный уровень голосовой связи водителя с оператором экстренных оперативных служб.

Ключевые слова: двухколесное транспортное средство, акустический сигнал, дуплексная голосовая связь, алгоритм эхокомпенсации, алгоритм шумоподавления, нормализованный алгоритм наименьших квадратов, цифровой сигнальный процессор, алгоритм LMS, алгоритм NLMS

Для цитирования: Никитин В.В., Увайсов С.У., Басов Д.В. Устройство вызова экстренных оперативных служб для обеспечения голосовой связи водителя двухколесного транспортного средства и оператора системы «ЭРА-ГЛОНАСС». *Russian Technological Journal*. 2025;13(6):63–77. https://doi.org/10.32362/2500-316X-2025-13-6-63-77, https://www.elibrary.ru/FGTHOC

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INTRODUCTION

Ensuring road safety remains a critical and pressing concern, a challenge amplified by the evolving road network and infrastructure. This complexity is further heightened by the growing variety and number of vehicles using the roads on a daily basis. A crucial element of road safety lies in comprehending the nature of accidents, including the number of casualties and injured individuals, in order to provide the best possible aid and support for those affected. Accidents involving two-wheeled vehicles are especially high risk due to the lack of a protective structure for riders, unlike car drivers. As a consequence, eyewitness accounts and, if possible, statements from those involved offer valuable insights, beyond automatically detecting accidents. To this end, two-wheeled vehicles should be equipped with emergency communication systems (ECS) allowing for two-way communication with a designated operator.

The unique characteristics of the human-vehicle system (consisting of a driver and a two-wheeled vehicle) pose specific challenges for implementing ECS. During development, the significantly smaller size and mass of the vehicle compared to a car needs to be considered, as well as the driver's helmet which muffles speech and may be difficult to remove in an emergency. Additionally, the system must function in all weather conditions and account for variable distances between the speaker and microphone. The high noise levels inherent in traffic situations also pose a challenge.

In order to ensure a clear and reliable voice connection with emergency operators, the use of advanced analog-to-digital voice processing systems is crucial (GOST 34788-2021[1]). This includes effectively minimizing noise, eliminating echo, and dynamically adjusting microphone and speaker gain levels based on the surrounding environment (GOST 33468-2015[2]).

Due to the inherent delay in signal propagation caused by digital processing, a balance must be found between the desired processing quality and the acceptable signal transmission time [1].

Contemporary speech processing algorithms typically use digital signal processors (DSP) which are often integrated into cellular modems. Improving processing efficiency and reducing latency can be accomplished by distributing individual calculations across other circuit components, such as the codec and microcontroller.

KEY MEASURES OF EFFECTIVE TWO-WAY VOICE COMMUNICATION

The ECS employs a duplex hands-free communication method when interacting with emergency operators from the subscriber's end. In order to ensure the clarity of voice data, the terminal device must possess the following attributes [2]:

1) speaker signal level sufficient to exceed the total noise level;
2) microphone signal level sufficient to exceed the total noise level;
3) wide dynamic microphone range, in order to ensure stable operation in a noisy public road environment;
4) low in-channel noise;
5) automatic gain control;
6) noise reduction;
7) echo cancellation;
8) addition of comfortable noise to the communication channel.

Items 1 to 4 are accompanied by a circuit design which includes a speaker of suitable power, an audio amplifier, a high-sensitivity microphone featuring a broad dynamic

[1] GOST 34788-2021. Interstate Standard. *Motor vehicles. Call emergency services systems. Speakerphone quality. Technical requirements and test methods*. Moscow: Russian Institute of Standardization; 2021. 20 p. (in Russ.).

[2] GOST 33468-2015. Interstate Standard. *Global navigation satellite system. Road accident emergency response system. In-vehicle emergency call device/system. General technical requirements*. Moscow: Standartinform; 2016. 74 p. (in Russ.).

range and built-in amplification. Additionally, there is a bandpass filter which only allows the transmission of useful speech frequencies (300–3400 Hz), as well as filters to suppress narrowband interference (harmonics from electrical equipment and GSM[3]).

Items 5 to 8 are implemented using digital speech processing algorithms. The ECS specific configuration determines whether DSP is an independent component or integrated into a cellular modem.

The device is designed using a printing unit meticulously crafted with advanced computer-aided design software.

AUTOMATIC VOICE GAIN CONTROL

Automatic voice gain control is a difficult task to implement using digital signal processing. Thus, in most cases, the method involves transferring analog circuitry algorithms into a digital signal. However, modern speech processing systems use digital signal processing techniques in automatic gain control systems by calculating based on the analytic signal. Each sample of the digital analytic signal, $x(n)$, is a complex number:

$$x(n) = \mathrm{Re}\{x(n)\} + i\mathrm{Im}\{x(n)\},$$

wherein n is the sample number.

The normalized digital waveform has the following form:

$$x_{\mathrm{norm}}[n] = \frac{x(n)}{|x(n)|}. \tag{1}$$

When dealing with signals composed of harmonic oscillations and noise, or a combination of harmonic oscillations, normalization can lead to signal distortion. In order to ensure that the signal aligns with formula (1), the denominator should not represent a fixed reference point, but rather the average value of the signal envelope. This average amplitude can be determined using a finite impulse response (FIR) filter, in order to average the absolute value of the signal rather than the signal itself. Following this procedure, the normalization formula ultimately assumes the following form:

$$x_{\mathrm{norm}}(n) = \frac{x(n)}{x_{\mathrm{av}}(n)},$$

wherein $x_{\mathrm{av}}(n)$ is the average signal modulus value calculated using the moving average method:

$$\dot{x}_{\mathrm{av}}(n) = \frac{1}{N}\sum_{k=0}^{N-1} |\dot{x}(n-k)|,$$

[3] Global System for Mobile Communication (GSM) is the second-generation mobile communication standard.

wherein N is the length of the averaging window and k is the summation index.

Several key considerations arise when this speech processing technique is applied. Firstly, the pauses inherent in speech between words and phrases can lead to elevated noise levels in the digitized audio signal. This noise includes in-channel interference from electrical circuitry and external noise, along with quantization noise generated during the conversion process. In order to address this issue, two strategies can be implemented. One approach is to deactivate automatic gain control when the speech is absent. A speech detector is used to identify these pauses. This prevents the gain control from amplifying noise during these silent periods. The second strategy entails incorporating a DC component into the average signal envelope, with its parameter values set above the maximum noise levels observed during pauses. This guarantees that the calculated average signal remains uncontaminated by noise present in those areas. In this scenario, the equation to calculate the average signal would be:

$$x_{\mathrm{norm}}(n) = \frac{x(n)}{x_{\mathrm{av}}(n) + \mathrm{L}}.$$

The L constant is chosen empirically during the system tuning process for specific noise conditions.

NOISE REDUCTION

The ERA-GLONASS system operator frequently encounters noisy speech signals due to the vehicle's design. As a two-wheeled vehicle lacking an enclosed cabin, there is a significant level of ambient noise intrusion. Intense background noise can significantly disrupt the accuracy of speech analysis and recognition processes. Several noise reduction methods are utilized, in order to eliminate unwanted noise. While some techniques can work in real time with minimal impact on the signal's delay, a key difficulty in noise reduction lies in the fluctuating volume and character of noise over time. The most prevalent noise reduction techniques operate under the assumption that speech and noise signals are independent of each other [3]. Their widespread adoption stems from their proven ability to effectively diminish background noise and enhance speech recognition performance.

Let speech signal $x(n)$ be distorted by additive noise $v(n)$. Then, the noisy signal $y(n)$ can be expressed as follows:

$$y(n) = x(n) + v(n).$$

The general goal of noise reduction is to reconstruct the signal $\hat{x}(n)$ as close as possible to the original signal $x(n)$, based on the observed noisy signal $y(n)$. In order to achieve this, adaptive algorithms have been developed which utilize noise estimation and suppression techniques.

When segmenting a signal into windows ranging from 10 to 30 ms, it can be reasonably assumed that both the speech signal and background noise become stationary. Therefore, noise reduction techniques are developed under this premise [4, 5]. Within contemporary modem digital signal processors, the Kochen–Berdugo algorithms for minima controlled recursive averaging (MCRA), quantile estimation, and its diverse variations are predominantly employed for noise estimation [6, 7]. A contemporary adaptation of Wiener filtering is employed specifically for noise reduction.

ECHO CANCELLATION

In telephony, there are two types of echo:

- acoustic echo caused by the reflection of sound waves; and
- electrical echo due to problems with the line matching (which is currently not a common issue).

For ECS, it is essential to eliminate acoustic echo. This phenomenon results from the acoustic connection between the microphone and speaker of the user interface unit (UIU) in the telecommunications operator system. It is also be a product of sound waves reflecting off surrounding objects and causing reverberation.

Echo cancellation techniques can be categorized into two types: those operating in the time domain; and those in the frequency domain [8, 9]. In the context of low-frequency signals, such as those found in telephone lines, time-domain algorithms have demonstrated superior efficiency in terms of computational resource utilization.

The adaptive echo cancellation algorithm involves the following steps [6]:

1. Echo path modeling.
   A model of the true acoustic path is created to describe the characteristics of the echo path $\boldsymbol{\theta}$. This model is a filter with adjustable coefficients $\hat{\boldsymbol{\theta}}$.
2. Estimating the echo signal.
   At each step n, an estimate of the echo $\hat{z}(n)$ is calculated by convolving a reference signal $\mathbf{x}$ with the model coefficients vector $\hat{\boldsymbol{\theta}}$ as follows:
   $$\hat{z}(n) = \hat{\boldsymbol{\theta}}^{\mathrm{T}} \cdot \mathbf{x},$$
   wherein T indicates transposition, and the vector length is dictated by the digital filter length.
3. Model parameter adaptation.
   The $\hat{\boldsymbol{\theta}}$ model parameters are continuously adjusted based on the analysis of the error signal $e(n)$, which represents the difference between the actual echo signal $z(n)$, and its estimated value $\hat{z}(n)$. This adaptation mechanism employs a coefficient adjustment algorithm to effectively reduce this error.
4. Echo cancellation.
   The final output signal with echo cancellation is formed by subtracting the estimated echo $\hat{z}(n)$ from the total microphone signal $m(n)$, as follows:
   $$y(n) = m(n) - \hat{z}(n),$$
   wherein $m(n) = \omega(n) + z(n)$, $\omega(n)$ is the useful near-end speech signal and $z(n)$ represents the real echo.

The operation of the echo canceller is illustrated in Fig. 1. In this figure, the far-end signal represents the voice of the operator, while the near-end signal indicates the voice of the user transmitted through the speakerphone system.

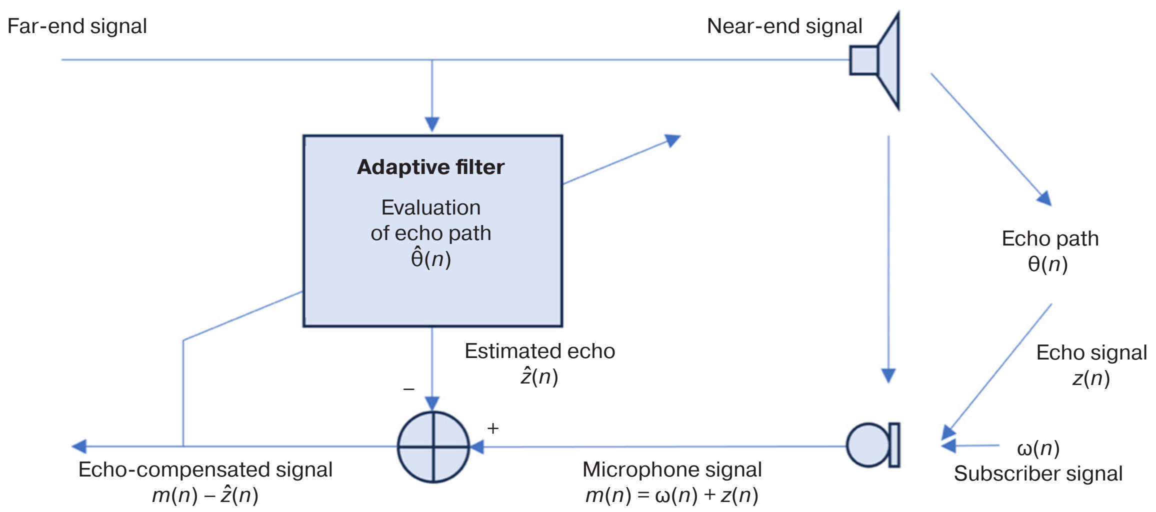

Fig. 1. Echo canceller circuit using adaptive filter

LEAST MEAN-SQUARES ALGORITHM

The Widrow–Hoff algorithm [10], also known as the Least Mean-Squares (LMS) algorithm, remains one of the most effective adaptive filtering algorithms in use today. This algorithm is widely employed in the development of echo cancellation systems, since it adjusts the filter coefficients $\hat{\boldsymbol{\theta}}$ in order to minimize the difference between the desired (target) input signal and the output of the filter. Due to its simplicity in terms of computational requirements, this algorithm has gained widespread adoption.

NORMALIZED LEAST MEAN-SQUARES ALGORITHM

The Normalized Least Mean-Squares (NLMS) algorithm [11] is derived from the LMS algorithm. The NLMS algorithm addresses the issue of the step size changing with the input power of the adaptive filter. When the input power changes over time, the step size between adjacent filter coefficients also changes which can affect the convergence rate of the algorithm. At low signal levels, the convergence rate slows down due to the small signal strength. Conversely, at high signal levels, the convergence rate increases and can cause errors. In order to address this issue, the step size needs to be adjusted based on the input signal level. This new step size is called normalized.

DOUBLE-TALK DETECTOR

Acoustic echo cancellation faces a significant challenge in addressing the double-talk (DT) effect [12] which arises when both far-end and near-end speech signals are active simultaneously. In this scenario, the signal from the remote participant becomes distorted by overlapping local speech. In order to mitigate this issue, a double talk detector is incorporated into the system. Its role is to halt the adaptation process of the filtering algorithm temporarily when a speech signal is detected at the near end, preventing inconsistencies in the performance of the adaptive algorithm.

The implementation of the Double-Talk Detector (DTD) together with the adaptive filter is shown in Fig. 2. This arrangement relies on the DTD to evaluate three critical parameters:

1) the far-end signal $x(n)$, which serves as the echo reference signal;
2) the near-end signal $m(n)$, representing a combination of the desired signal and the echo;
3) the error signal $e(n) = m(n) - \hat{z}(n)$, where $\hat{z}(n)$ is the estimated echo produced by the adaptive filter.

The error signal $e(n)$ represents an echo-compensated output generated by subtracting the simulated echo from the microphone input. This signal typically arises in two key scenarios:

- during the initial phase of operation when the adaptive filter has not yet converged;
- when there is a change in the room's acoustic properties, such as repositioning the microphone.

The DTD assesses the signal strength of $x(n)$, $m(n)$, and $e(n)$, evaluates the signal-to-echo ratio, and determines whether to update the filter coefficients. This ensures adaptation when no double talk is present or halts updates when overlapping speech is detected [13].

Various DTD algorithms exist, with the most widely used being the Geigel algorithm and those reliant on

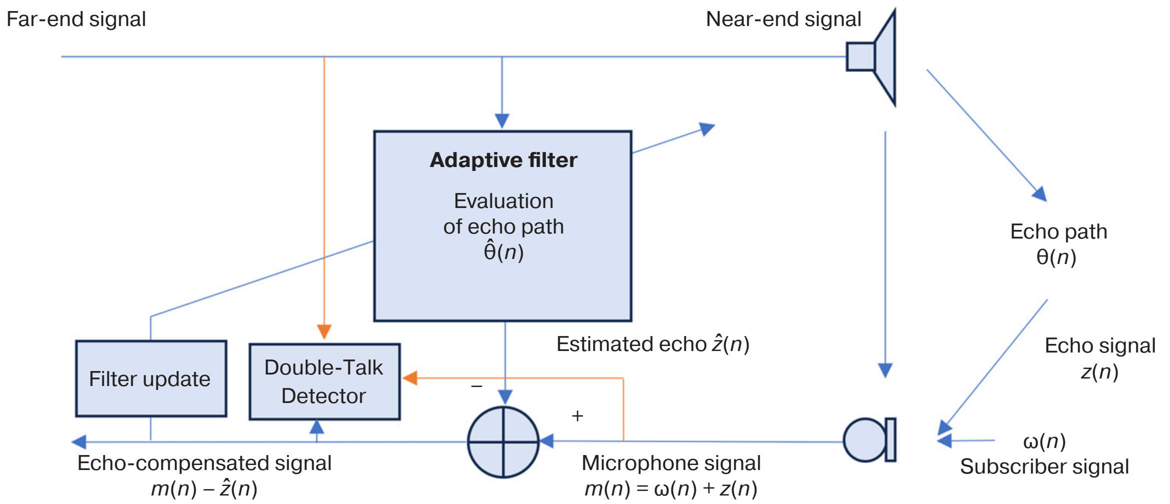

Fig. 2. Connection diagram of the double-talk detector in conjunction with an adaptive filter

cross-correlation calculations, such as Benesty and Normalized Cross-Correlation (NCC) [14].

The Geigel algorithm fundamentally works by analyzing and comparing the power levels of two signals: the microphone signal, which may include echo and/or near-end speech; and the far-end reference signal transmitted to the loudspeaker. When the microphone signal consists solely of echo without near-end speech, its power is lower than the reference signal due to attenuation along the acoustic path. However, when near-end speech is introduced alongside the echo, the overall power of the microphone signal rises substantially. This increase in signal strength enables the algorithm to identify effectively the presence of double talk.

In contrast to the Geigel energy method, the Benesty algorithm employs a normalized cross-correlation coefficient to analyze the relationship between the far-end signal (reference echo) and the microphone signal (a combination of echo and near-end speech). A higher correlation coefficient signifies that the microphone signal is predominantly influenced by echo, whereas a drop in the correlation suggests the emergence of near-end speech. This technique delivers more precise detection when compared to basic power comparison methods.

The normalized cross-correlation algorithm computes decision statistics by examining the interplay between the microphone signal and the error signal which represents deviation between the actual echo and its estimated counterpart. This approach relies on assessing the dispersion of the near-end signal (the useful signal) and evaluating the cross-correlation between the microphone and error signals. By normalizing with respect to signal power, the algorithm maintains robustness against fluctuations in signal levels, a crucial attribute in dynamic acoustic environments.

PRACTICAL IMPLEMENTATION OF THE PROTOTYPE USER INTERFACE

A prototype UIU has been designed to facilitate the modeling and testing of algorithms. The appearance of the device is shown in Fig. 3.

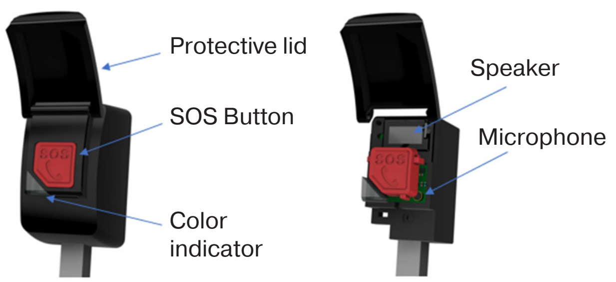

Fig. 3. Appearance of the designed UIU

The device is positioned on the steering apparatus of a two-wheeled vehicle, within easy reach of the driver. It features a microphone, speaker, an SOS button for contacting emergency services, a status indicator for the device, and a protective cover to avoid unintentional activation. The device layout and the UIU components are shown in Fig. 4.

Fig. 4. ECS placement on a two-wheeled vehicle

Since the speaker and microphone are housed within the same unit and positioned closely together, acoustic echo tends to manifest quite prominently [15]. In order to mitigate this issue, the use of a digital signal processor within the modem is suggested, while auxiliary computations can be handled by the microcontroller in the main unit. The UIU functional diagram is shown in Fig. 5.

The pathway of the acoustic speech signal as it passes through the UIU is shown in Fig. 6. Within the codec housed in the UIU, the analog signals from the microphone and speaker are transformed into digital data. The reverse process (via the I^2S interface[4]) is also performed at this stage. The signal is then transmitted via the RS-485 interface[5] to the main unit, the primary function of which is to ensure robust noise immunity. Subsequently, the main unit microcontroller transfers the signal to the I^2S interface, directing it toward the modem. Given that the I^2S interface is designed for short-range transmission, integrating the RS-485 interface proves to be a highly effective solution [16].

The ECS main unit consists of the Telit LE910 modem (manufactured by Telit Cinterion, Italy) and the NAU8810 codec (produced by Nuvoton Technology Corporation, Taiwan). The modem operates on a Qualcomm chipset and integrates a digital signal processor developed by the same company. Configuration of the digital processor parameters is managed via the user interface of the *Qualcomm QACT* software (Qualcomm, USA).

[4] I^2S (Inter-Integrated Circuit Sound) is a standardized electrical serial bus interface for connecting digital audio devices.

[5] Recommended Standard 485 is the physical layer standard for an asynchronous interface.

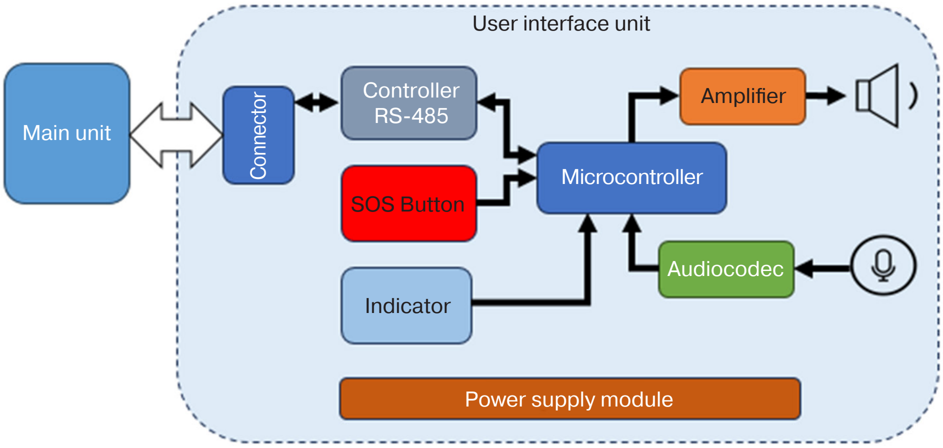

Fig. 5. UIU functional diagram

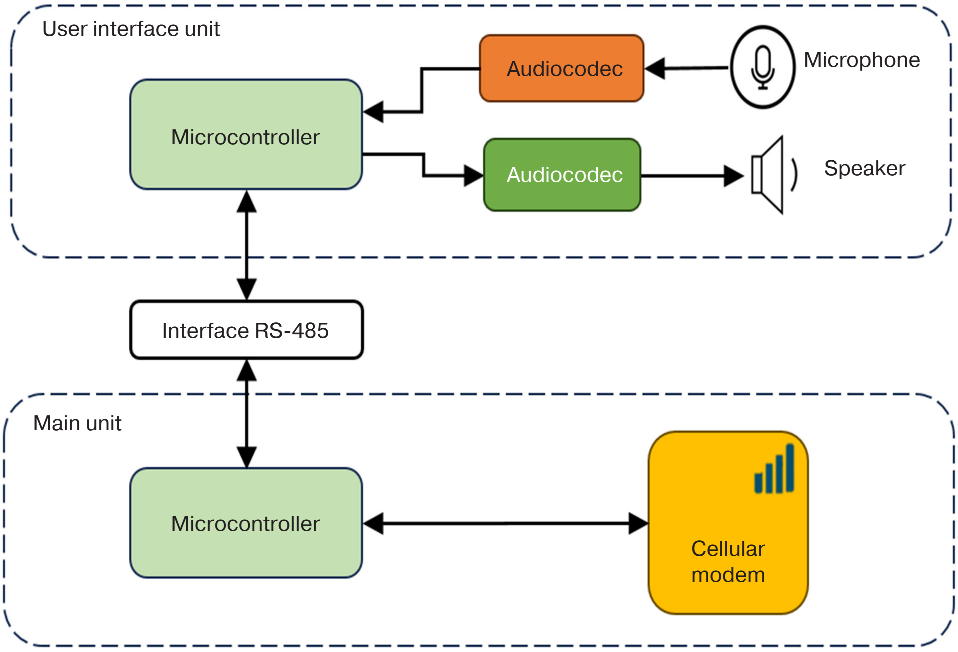

Fig. 6. Pathway of the acoustic speech signal passing through UIU

DEVICE TESTING

Figure 7 shows the block diagram of the test bench designed for evaluating and analyzing voice quality. The emergency communication system, mounted on a two-wheeled vehicle, is linked to a radio communication tester via cellular network connectivity. Additionally, the audio input of the tester is connected to a computer equipped with the *HEAD Analyzer ACQUA* software (HEAD Acoustics GmbH, Germany). This software provides a comprehensive suite of audio tests for assessing the quality of speech signals. Its operational algorithm enables voice quality testing from both the operator's perspective and the vehicle driver's viewpoint.

During operator trials, calibrated audio signals are sent to the radio communication tester, passed through the ECS modem, and routed along the speech path to the UIU speaker. From there, these signals are received by the mannequin's artificial hearing aid and analyzed using the *HEAD Analyzer ACQUA* software.

Testing on the vehicle's driver side involves transmitting calibrated audio signals from the mannequin's artificial hearing aid via the UIU microphone to the modem. These signals are then routed through the radio communication tester and analyzed using dedicated software.

To replicate street ambient noise, the noise scenarios management workplace is employed within a soundproof test chamber.

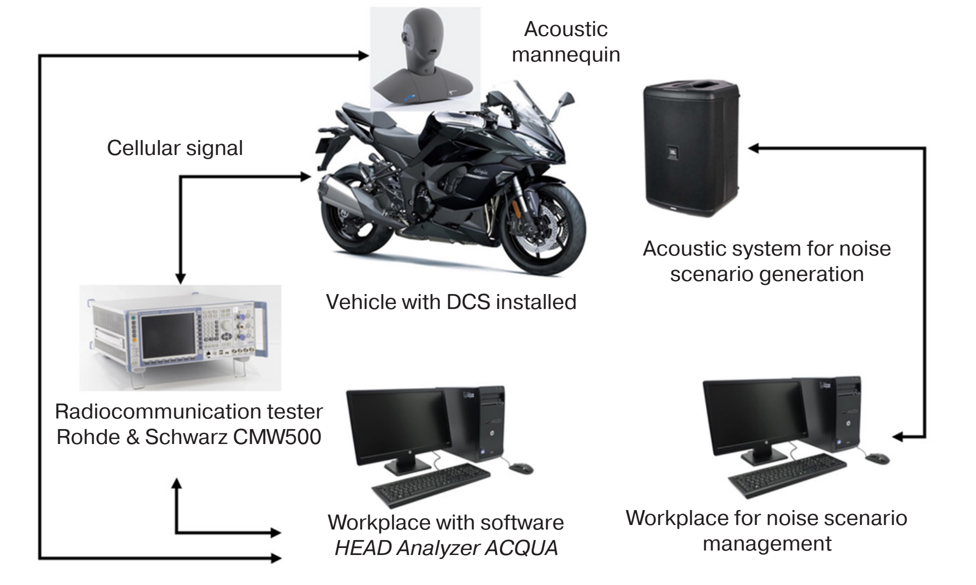

Fig. 7. Structural diagram of the DCS test bench

Table. Duplex DCS performance type parameters

Parameter	Quality class				
	1	2a	2b	2c	3
	Full duplex communication	Partial duplex communication			Half duplex communication
$A_{H, S, dt}$, dB	≤3	≤6	≤9	≤12	>12

Configuring the digital signal processor and microcontroller algorithms allows voice communication performance to be attained between the emergency operator and the vehicle driver in compliance with GOST 33464-2015[6]. Although the methods outlined in this standard are primarily designed for in-car testing, they can also be adapted for two-wheeled vehicles. Below is an overview of the tests conducted on the most critical performance indicators.

ATTENUATION IN THE TRANSMISSION CHANNEL IN DOUBLE-TALK MODE

While subscribers engage in simultaneous conversations, the maximum allowable decay $A_{H, S, dt}$ introduced by the DCS into the transmission channel S is regulated by GOST 33464-2015. This is verified using the procedures outlined in GOST 33468-2015, specifically in paragraph 7.9.2. The level of attenuation is dependent on the DCS performance type, categorized by quality class for duplex communication, and must align with the specified values detailed in the table.

The requirements must be fulfilled under both normal signal conditions, and when there is an imbalance in signal levels. The following two specific signal level scenarios should be tested:

- standard signal levels for both reception and transmission;
- a scenario where the transmission signal level is increased by 6 dB, while the reception signal level is decreased by 6 dB.

The test involves two sequences of uncorrelated composite source signals (CSS) transmitted and received simultaneously, with a partial temporal overlap to simulate the dynamics of concurrent communication.

The initial segment of each CSS period (representing a voiced sound, displayed in black in Fig. 8) moving in one direction overlaps with the concluding segment of each CSS period (characterized by pseudo noise, displayed in white) traveling in the opposite direction. The analysis is conducted during active signal instances aligned with the transmission direction.

Prior to initiating testing, the echo canceller should be configured to achieve maximum echo cancellation. This is accomplished using a training sequence on the receive channel, comprising 10-second recordings of both male and female voices.

6 GOST 33464-2015. Interstate standard. *Global navigation satellite system. Road accident emergency response system. In-vehicle emergency call device/system. General technical requirements*. Moscow: Standartinform; 2017. 86 p. (in Russ.).

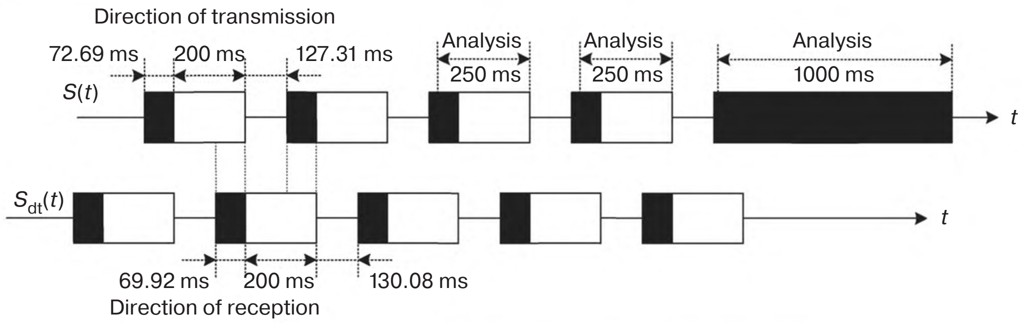

Fig. 8. Test signals used to determine the attenuation range in the transmit direction during a simultaneous conversation. The signal in the transmit direction is denoted as $S(t)$, and the signal in the receive direction is denoted as $S_{dt}(t)$

The signal strength of the transmission channel is evaluated in the time domain using an integration time constant of 5 ms. A time-based relationship for signal strength is established. Signal attenuation within the transmission channel is assessed by comparing the signal strength during simultaneous double talk or two-way conversation to that observed during one-way conversation (specifically during reception signal pauses), assuming that the transmission channel is fully active. This analysis covers the complete test sequence, beginning with the second period of the CSS signal.

Figures 9 and 10 present the measurement results. Figure 9 illustrates near-end recordings overlaid on a single timeline, both during silence and in the presence of the far-end signal emitted through the loudspeaker system. In Fig. 10, the volume level is displayed on a linear scale with values provided in volts: these readings were captured after processing through the measuring system's vocoder. This figure represents the signal ratios derived from Fig. 9, highlighting that the near-end signal, when accompanied by the far-end signal, is consistently quieter compared to its level during silence. The curve values are expressed as negative numbers.

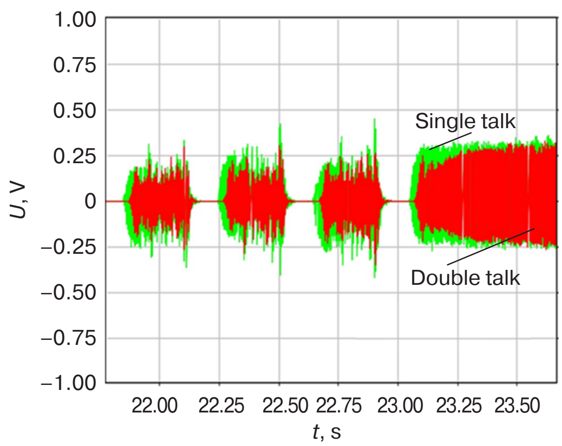

Fig. 9. Signal strength versus time for a one-way and two-way conversation in the presence of a far-end signal, integrated with a time constant of 5 ms

As a final estimate, the value of the maximum permissible attenuation is calculated in the following way:

$$A_{\mathrm{H,S,dt}} = \frac{5}{t_{\mathrm{end}} - t_{\mathrm{start}}} \int_{t_{\mathrm{start}}}^{t_{\mathrm{end}}} |L_{\mathrm{dB}}(\mathrm{t})| dt,$$

wherein $L_{\mathrm{dB}}(t)$ is the instantaneous attenuation at time t in dB, $(t_{\mathrm{end}} - t_{\mathrm{start}})$ is the duration of the time interval, and 5 is the sampling step in ms.

Attenuation is defined as the average value of the area under the curve, obtained by integrating a signal with a time constant of 5 ms. This value is calculated by dividing the total area across all intervals by the number of samples used for integration, as shown in Fig. 10. In addition, the level curve is also integrated with a 5 ms time constant, in order to minimize the risk of distortion in the results caused by random short-term fluctuations.

The results obtained and analyzed using the *HEAD Analyzer ACQUA* software correspond to class 2a and meet the requirements of GOST 33464-2015.

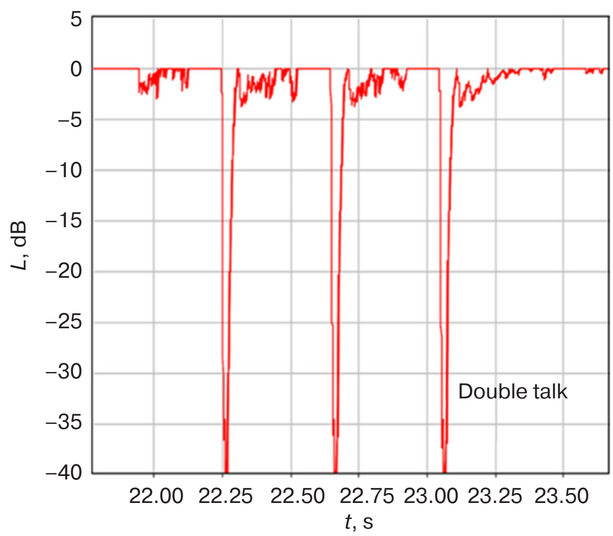

Fig. 10. Ratio of the signal transmitted during a two-way conversation scenario to that of a one-way conversation scenario over time, integrated with a time constant of 5 ms

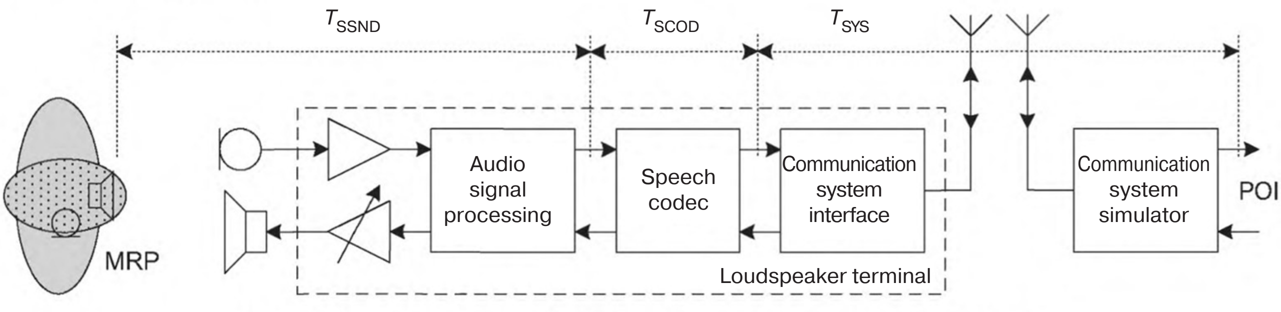

Fig. 11. Acoustic signal processing and propagation delays

Similar tests are conducted under conditions where the transmit signal level is increased by 6 dB and the receive signal level is decreased by 6 dB. In other words, this scenario corresponds to the near-end sound being 6 dB louder while the far-end sound, played through the device's speaker, is 6 dB quieter. Even at standard volume levels, the performance measurements stay within acceptable boundaries.

ECHO ATTENUATION STABILITY OVER TIME

When a test signal, combined with an artificial voice signal at a nominal level, is transmitted through the DCS reception channel, the echo signal attenuation in the DCS transmission channel should remain stable over an extended measurement period. Specifically, this attenuation should not decrease by more than 6 dB from its peak value, as outlined in the standards set forth in GOST 33464-2015, as well as the verification procedures detailed in GOST 33468-2015, paragraph 7.7.3.

The first test signal consists of a periodically repeated combined CSS signal, evaluated at two average signal levels: −5 dBm0 and −25 dBm0.

For the second test signal, a 10-second sequence is used, featuring both male and female voices, each having an average level of −16 dBm0. The analysis is conducted across the entire duration of the signals.

The echo cancellation rate is plotted against time. With an integration time constant of 35 ms, the system effectively smoothes random fluctuations and reduces the influence of short-term interference, ensuring more accurate measurement outcomes.

The CSS signal plot represents the curve of the ratio between the strength of the recorded signal in the transmit direction, while the strength of the reproduced signal in the receive direction. A key consideration is that one of these ratio signals should be time-offset to account for physical and software delays inherent in acoustic signal processing and propagation. These include: sound signal propagation delays T_{SSND} (where S stands for signal and SND signifies sound); delays stemming from the software processing of the audio signal S within the T_{SCOD} speech codec (where S stands for signal and COD signifies codec); and delays introduced by software handling during transmission over the communication system network T_{SYS} (SYS referring to system). The cumulative propagation delay across the transmission channel, $T_{SSUM} = T_{SSND} + T_{SCOD} + T_{SYS}$, is measured as the total time taken for the acoustic speech signal to travel from the sound pressure level measurement point—located 25 mm ahead of the human lips or the emission ring of an artificial mouth device (referred to as the mouth reference point, MRP)—to the point of interface (POI). The POI marks the electrical speech signal's connection and level measurement reference within the receiving and transmitting channels of a mobile communication system simulator, following decoding. This process is illustrated in Fig. 11.

For a CSS signal, the time-dependent behavior of echo cancellation is illustrated in Fig. 12. The overall curve level remains minimal because the reproduced far-end signal contains only a low-level echo component. From the perspective of assessing stability in echo cancellation over time, the system is deemed stable if the difference between the minimum and maximum values during evaluation stays within 6 dB. It is important to highlight that this evaluation focuses solely on eight active sections of the CSS signal, excluding pauses, giving the measurement mask a toothed appearance. According to the *HEAD Analyzer ACQUA*'s results, the echo attenuation stability test is successfully passed across different time intervals.

Figure 13 illustrates the time-dependent variation of the echo level for an artificial voice signal. Instead of directly comparing the artificial voice signal with the reproduced far-end signal, the focus is on evaluating the transmission signal stability (echo-opening, curve *1*) when artificial voice signals, representing male and female voices (curve *2*), are reproduced on the far-end side. The system is deemed stable, if the difference between the minimum and maximum values observed during the evaluation remains within 6 dB.

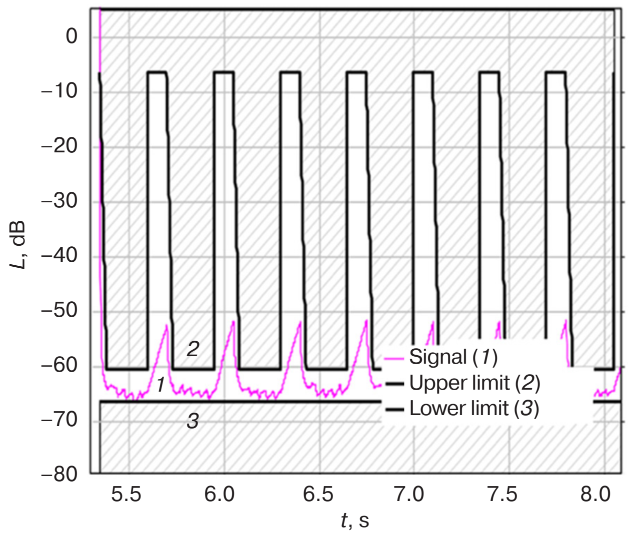

Fig. 12. CSS echo cancellation rate versus time with 35 ms integration time constant

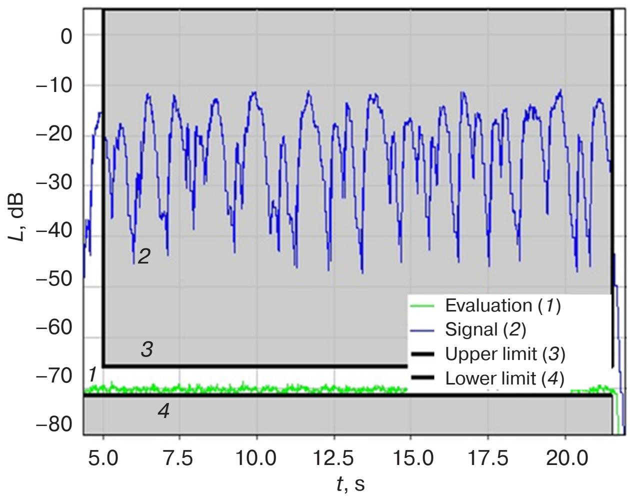

Fig. 13. Artificial voice echo level versus time with 35 ms integration time constant

OPERATION OF TRANSMISSION CHANNEL IN ACOUSTIC NOISE

For near-end speech in background acoustic noise conditions, the signal-to-noise ratio (SNR) at the output of the transmission channel should be at least 6 dB. The recommended SNR is 12 dB, as specified by GOST 33464-2015 and GOST 33468-2015, section 7.10.1.

Background acoustic noise around the two-wheeled vehicle are reproduced for both normal and worst-case noise scenarios.

When tested in noise levels greater than 50 dB (A), the output level of speech signals should increase by 3 dB for every 10 dB increase in noise, averaged over time. This reflects the effect of a person in noisy surroundings increasing the volume of their voice. The maximum increment is 8 dB.

An artificial voice serves as the input acoustic test signal, introduced through the mannequin's artificial mouth. Two sequences are employed: one representing a male voice and the other a female voice, both containing pauses.

The processed signal is extracted from the electrical output of the speech codec within the radio communication tester. In order to estimate signal and noise levels, an integration time constant of 35 ms is utilized. The signal strength envelope and pause noise are then calculated to determine SNR within the transmission channel.

The measurement outcome is represented as a graph depicting the envelope of the signal level and the noise in the transmission path. For active speech regions, an integrated assessment is performed, as follows:

$$S_s = \int_{t_{s_{start}}}^{t_{s_{end}}} L_{dB}(t)dt,$$

wherein $L_{dB}(t)$ is the signal level in decibels and $t_{s_{start}}$, $t_{s_{end}}$ are time limits for the speech segment.

This value is equal to the area under the curve in Fig. 14 that corresponds to all active speech segments.

The score value is calculated as follows:

$$S_n = \int_{t_{n_{start}}}^{t_{n_{end}}} L_{dB}(t)dt,$$

wherein $t_{n_{start}}$, $t_{n_{end}}$ are time limits for the speech pause noise. The estimate value is determined by the area under the curve shown in Fig. 14, which represents all noise segments. Both estimates incorporate an integration time constant of 35 ms, in order to filter out random

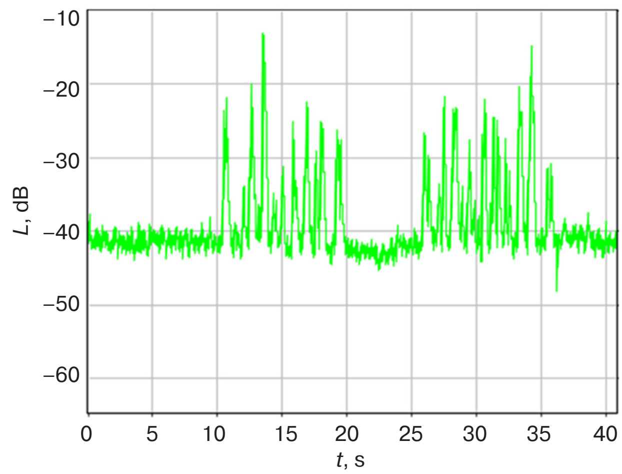

Fig. 14. The envelope level of the signal and the pause noise in the transmit direction with a 35 ms integration time constant

fluctuations in the signal. The operation of the transmission channel in acoustic noise is deemed satisfactory when the ratio of integral estimates exceeds 6 dB. In practical terms, this corresponds to a relatively soft signal, particularly when operating within the GSM standard acoustic frequency range of 300–3400 Hz. Therefore, achieving SNR of at least 12 dB is recommended. Naturally, in scenarios with more severe noise conditions, the estimate will be lower due to elevated background noise levels, which creates additional challenges for the system being tested.

The DCS data channel undergoes testing under different noise conditions derived from recorded environmental noise. For this evaluation, a street noise recording with a volume of 68 dB is utilized. Following the calibration of the microphone gain, speaker settings, and the DSP algorithms, a signal-to-noise ratio of 11 dB is achieved, thus successfully meeting the standards outlined in GOST 33464-2015.

CONCLUSIONS

As part of the project, the DCS prototype has been developed specifically for installation upon two-wheeled vehicles. Its primary function is to enable voice communication between the driver and the ERA-GLONASS system operator in cases of emergency or accidents. During the development process, the unique aspects of the “human-two-wheeled vehicle” interaction system were carefully examined, highlighting its distinctive characteristics compared to other dynamic systems within the broader road transport network. A variety of system factors, operational features, and constraints were also considered, such as weight and size requirements, installation dimensions, and optimal placement on the dashboard.

In the design of the device, significant focus was placed on addressing challenges related to acoustic signal processing, in order to ensure dependable and high-quality voice communication. Given the close proximity between the speaker and microphone within the device, a crucial element was to overcome issues such as mitigating and compensating for acoustic effects that accompany voice signals in the communication channel. In order to address these challenges, noise reduction, echo cancellation, and echo compensation algorithms were thoroughly evaluated for their applicability to the system under development. Subsequent trials and testing confirm that implementing both analog and digital speech signal processing algorithms—integrated into the codec and modem of the device—will deliver superior voice communication quality between the driver and the emergency operator.

Authors’ contribution
All authors contributed equally to the research work.

REFERENCES

1. Chuchupal V.Ya., Chichagov A.S., Makovkin K.A. *Tsifrovaya fil’tratsiya zashumlennykh rechevykh signalov* (*Digital Filtering of Noisy Speech Signals*). Moscow: Vychislitel’nyi tsentr RAN; 1998; 1998. 52 p. (in Russ.).
2. Kropotov Y.U., Paramonov A. *Metody proektirovaniya algoritmov obrabotki informatsii telekommunikatsionnykh sistem audioobmena* (*Methods of Designing Algorithms for Information Processing of Telecommunication Audio Exchange Systems*). Moscow: Direct-Media; 2015. 228 p. (in Russ.). ISBN 978-5-4475-4106-4.
3. Vishnyakov I.EH., Masyagin M.M., Odintsov O.A., Slyusar V.V. Methods and algorithms for real time voise noise cleaning. *Izvestiya vysshikh uchebnykh zavedenii. Ehlektronika = Proceedings of Universities. Electronics*. 2021;26(2): 184–196 (in Russ.). https://doi.org/10.24151/1561-5405-2021-26-2-184-196
4. Lyalin S.G. Noise reduction method in speech signals using neural network. *Adv. Sci.* 2019;2:32–38 (in Russ.). https://doi.org/10.25730/VSU.0536.19.021
5. Tikhonov V.I., Kharisov V.N. *Statisticheskii analiz i sintez radiotekhnicheskikh ustroistv i sistem* (*Statistical Analysis and Synthesis of Radio Engineering Devices and Systems*). Moscow: Radio i svyaz’; 1991. 608 p. (in Russ.). ISBN 5-256-00789-0
6. Azarov I.S., Vashkevich M.I., Likhachev D.S., Petrovsky A.A. Reducting complex noises in speech using filtering in modulation domain. *Tsifrovaya obrabotka signalov = Digital Signal Processing*. 2013;4:25–31 (in Russ.). https://elibrary.ru/rvwqdz
7. Zheleznyak V.K., Rahanov K.J., Adamovskiy Y.R. Evaluation of Information Indicators of Quantization Noise Generated by Speech Signal Analogue-Digital Conversion. *Doklady BGUIR*. 2023;21(2):32–40 (in Russ.). http://doi.org/10.35596/1729-7648-2023-21-2-32-40
8. Nguyen H.N., Dowlatnia M., Sarfraz A. *Implementation of the LMS and NLMS algorithms for Acoustic Echo Cancellation in teleconference system using MATLAB*. Vaxjoe: MSI Växjo University; 2009. 69 p. Available from URL: https://www.academia.edu/94741525/Implementation_of_the_LMS_and_NLMS_algorithms_for_Acoustic_Echo_Cancellationin_teleconference_systemusing_MATLAB
9. Carline M.J., Diana D.C. An Investigation on Adaptive Algorithms for Echo Cancellation of Acoustic Signals. In: *2nd Int. Conf. Sustain. Comput. Data Commun. Syst. ICSCDS 2023 – Proc.* 2023. P. 960–966. https://doi.org/10.1109/ICSCDS56580.2023.10105081

10. Haykin S. *Adaptive Filter Theory*: 5th ed. New York: Pearson; 2014. 960 p.
11. La Rosa A.B., Pereira P.T.L., Ücker P., Paim G., da Costa E.A.C., Bampi S., Almeida S. Exploring NLMS-Based Adaptive Filter Hardware Architectures for Eliminating Power Line Interference in EEG Signals. *Circuits, Systems, and Signal Processing*. 2021;40(5):3305–3337. https://doi.org/10.1007/s00034-020-01620-6
12. Zhang H., Tan K., Wang D.L. Deep learning for joint acoustic echo and noise cancellation with nonlinear distortions. In: *Proceedings of INTERSPEECH*. 2019. P. 4255–4259. http://doi.org/10.21437/Interspeech.2019-2651
13. Zhang H., Wang D.L. Deep learning for acoustic echo cancellation in noisy and double-talk scenarios. In: *Proceedings of INTERSPEECH*. 2018. P. 3239–3243. http://doi.org/10.21437/Interspeech.2018-1484
14. Hänsler E., Schmidt G. *Acoustic Echo and Noise Control: A Practical Approach*. Wiley-IEEE Press; 2005. 472 p. ISBN 978-0-471-67839-7
15. Constantini G., Casali D. *New Advances in Audio Signal Processing.* MDPI Books; 2024. 206 p. ISBN 978-3-7258-1059-8.
16. Demidenko A.V. *Mikrokontrollery dlya nachinayushchikh: Interfeisy i upravlenie* (*Microcontrollers for Beginners: Interfaces and Control*). LitRes; 2022. 100 p. (in Russ.).

СПИСОК ЛИТЕРАТУРЫ

1. Чучупал В.Я., Чичагов А.С., Маковкин К.А. *Цифровая фильтрация зашумленных речевых сигналов*. М.: ВЦ РАН; 1998. 52 с.
2. Кропотов Ю., Парамонов А. *Методы проектирования алгоритмов обработки информации телекоммуникационных систем аудиообмена*. М.: Директ-Медиа; 2015. 228 с. ISBN 978-5-4475-4106-4
3. Вишняков И.Э., Масягин М.М., Одинцов О.А., Слюсарь В.В. Методы и алгоритмы шумоочистки звука в реальном времени. *Известия высших учебных заведений. Электроника.* 2021;26(2):184–196. https://doi.org/10.24151/1561-5405-2021-26-2-184-196
4. Лялин С.Г. Метод шумоподавления в речевых сигналах с помощью нейронной сети. *Adv. Sci*. 2019;2:32–38. https://doi.org/10.25730/VSU.0536.19.021
5. Тихонов В.И., Харисов В.Н. *Статистический анализ и синтез радиотехнических устройств и систем*. М.: Радио и связь; 1991. 608 с. ISBN 5-256-00789-0
6. Азаров И.С., Вашкевич М.И., Лихачев Д.С., Петровский А.А. Алгоритм очистки речевого сигнала от сложных помех путем фильтрации в модуляционной области. *Цифровая обработка сигналов*. 2013;4:25–31. https://elibrary.ru/rvwqdz
7. Железняк В.К., Раханов К.Я., Адамовский Е.Р. Оценка информационных показателей шума квантования аналогоцифрового преобразования речевого сигнала. *Доклады БГУИР.* 2023;21(2):32–40. http://doi.org/10.35596/1729-7648-2023-21-2-32-40
8. Nguyen H.N., Dowlatnia M., Sarfraz A. *Implementation of the LMS and NLMS algorithms for Acoustic Echo Cancellation in teleconference system using MATLAB*. Vaxjoe: MSI Växjö University; 2009. 69 p. URL: https://www.academia.edu/94741525/Implementation_of_the_LMS_and_NLMS_algorithms_for_Acoustic_Echo_Cancellationin_teleconference_systemusing_MATLAB
9. Carline M.J., Diana D.C. An Investigation on Adaptive Algorithms for Echo Cancellation of Acoustic Signals. In: *2nd Int. Conf. Sustain. Comput. Data Commun. Syst. ICSCDS 2023 – Proc.* 2023. P. 960–966. https://doi.org/10.1109/ICSCDS56580.2023.10105081
10. Haykin S. *Adaptive Filter Theory*: 5th ed. New York: Pearson; 2014. 960 p.
11. La Rosa A.B., Pereira P.T.L., Ücker P., Paim G., da Costa E.A.C., Bampi S., Almeida S. Exploring NLMS-Based Adaptive Filter Hardware Architectures for Eliminating Power Line Interference in EEG Signals. *Circuits, Systems, and Signal Processing*. 2021;40(5):3305–3337. https://doi.org/10.1007/s00034-020-01620-6
12. Zhang H., Tan K., Wang D.L. Deep learning for joint acoustic echo and noise cancellation with nonlinear distortions. In: *Proceedings of INTERSPEECH*. 2019. P. 4255–4259. http://doi.org/10.21437/Interspeech.2019-2651
13. Zhang H., Wang D.L. Deep learning for acoustic echo cancellation in noisy and double-talk scenarios. In: *Proceedings of INTERSPEECH*. 2018. P. 3239–3243. http://doi.org/10.21437/Interspeech.2018-1484
14. Hänsler E., Schmidt G. *Acoustic Echo and Noise Control: A Practical Approach*. Wiley-IEEE Press; 2005. 472 p. ISBN 978-0-471-67839-7
15. Constantini G., Casali D. *New Advances in Audio Signal Processing.* MDPI Books; 2024. 206 p. ISBN 978-3-7258-1059-8.
16. Демиденко А.В. *Микроконтроллеры для начинающих: Интерфейсы и управление*. ЛитРес; 2025. 100 с.

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