Study of the system of the main functions of Schauder as a means of presenting and compressing sound information for wireless sensor networks

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Abstract

An analysis of the technical possibilities of using the Schauder basis function system for presenting and compressing information, for example, network data, is provided. Advantages compared to other bases have been revealed. On the basis of experimental studies, recommendations are provided for their use for digital processing of audio information at a representative level in computer systems and networks. From our point of view, digital systems that use the representation of sound signals by a system of basic functions have great potential. Further considerations will be directed to justifying the expediency of their use in engineering developments. At the same time, computer modeling is an effective tool for finding rational methods of building digital signal processing equipment. Preliminary modeling of signal analysis and synthesis processes allows at the research stage to determine the complexity of hardware solutions, possible parameters of digital processing (accuracy of transmission and reproduction, degree of elimination of redundancy (compression), speed, and others).

Keywords

system of basic functions of Schauder, presentation and compression of information, digital processing of speech signals

1. Introduction

A characteristic feature of modernity is the sharp excess of the growth of the flow of information over the expansion of data transmission channels [1, 2, 3]. Therefore, the search for rational methods and technical means of information processing is essential [4, 5, 6, 7]. One of the stages of processing is information compression. At the modern level, digital transmission systems have become widespread, the relevance of which is, first of all, connected with the rapid development of information computer technologies that provide transformation, storage and transmission of information, as well as modeling of presentation and processing processes signals, systems and devices. Among the important areas of application of digital methods are the presentation and processing of broadcast signals, audio information in the broad sense [8, 9, 10].

The ever-growing volume of useful information obtained from sources of various physical origins requires the search for effective preprocessing methods at the stage of presentation, storage, and determination of channel transmission methods [11, 12, 13].

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The most important task of this problem is to reduce the redundancy while maintaining the desired transmission quality [14, 15, 16]. Traditional experiments have become methods of reducing the redundancy of sound signals, which involve coding of acoustic oscillations or analysis/synthesis based on hardware and software tools [17, 18, 19]. Despite the fact that a significant amount of research has been conducted in the field of digital processing of sound signals, especially speech signals, there are still many unsolved problems [20, 21, 22]. One of the reserves of improving systems for reducing the redundancy of sound signals is the search for new methods of presentation and processing using programming computing tools of computer technologies [23, 24, 25].

2. Review of previous studies, literary sources and research methods

In 1927, Schauder proposed a system of linearly independent basis functions. The use of Schauder functions in digital signal processing systems is the main topic of research. The distribution in a row according to the system of basic functions is generally presented as:

$$x(t) = \sum_{n=0}^{N} b_n \, \psi_n(t),$$

where $x(t)$ is the realization of the signal as a random process; $\psi_n(t)$ – system of basic functions; b_n are the coefficients of the schedule.

From our point of view, it is appropriate to present the Schauder function in the form [26]:

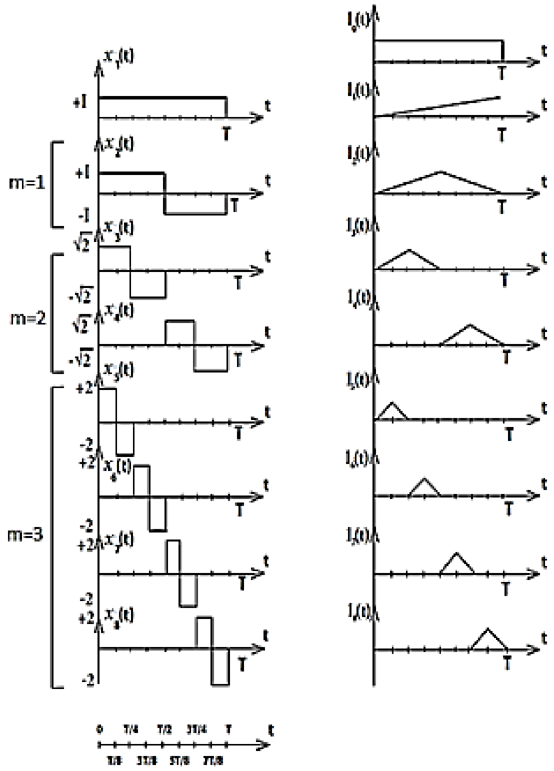
$$e(t) = \begin{cases} 1, & k = 0, \quad t \in [0, T], \\ \dfrac{1}{T} \displaystyle\int_0^t \mathcal{X}_1(r) \, dr, & k = 1, \quad t \in [0, T], \\ \dfrac{2^{\frac{m+1}{2}}}{T} \displaystyle\int_0^t \mathcal{X}_k(r) \, k(r) \, dr, & k = 2, 3, \ldots, \quad t \in [0, T], \\ 0, & t \notin [0, T], \end{cases}$$

where $\mathcal{X}_k(r)$ is a system of Haar functions defined on the orthogonality interval as follows:

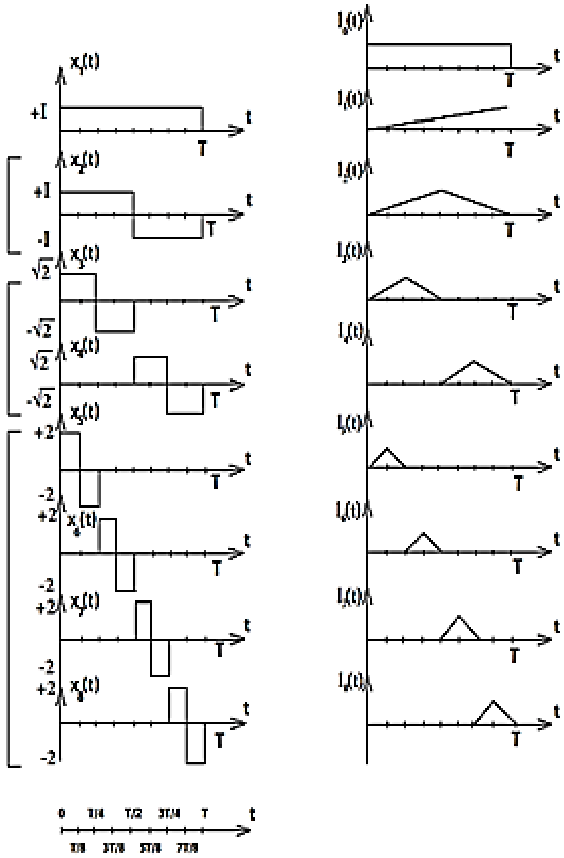
$$\mathcal{X}_1 \, (t) = 1, \;\; t \epsilon \, [0, T] \, ,$$

$$\mathcal{X}_k(r) = \mathcal{X}_{mn}(r) = \begin{cases} 2^{\frac{m-1}{2}}, & r \in \left[\dfrac{(n-1)T}{2^{m-1}}, \dfrac{(n-0.5)T}{2^{m-1}} \right], \\ -2^{\frac{m-1}{2}}, & r \in \left[\dfrac{(n-0.5)T}{2^{m-1}}, \dfrac{nT}{2^{m-1}} \right], \\ 0, & r \in \left[\dfrac{(n-1)T}{2^{m-1}}, \dfrac{nT}{2^{m-1}} \right], \end{cases}$$

where $k = 2^{m-1} + n$; $m = 1, 2, 3, \ldots$; $n = 1, 2, \ldots, 2^{m-1}$ — the number of the Haar function group — sets the duration of the zero value of the Haar function, which is equal to $T/2^{m-1}$, and its amplitude, which is equal to $2^{(m-1)/2}$. The index n determines the position of a non-zero value on the segment $[0, T]$. One group includes Haar functions with the same duration of a non-zero value. The relationship between single and double numbering of functions is as follows: $m = 1, 2, 3, \ldots, M$; $n = 1, 2, 3, \ldots, 2^{m-1}$; $k = n + 2^{m-1}$. Figure 1 presents graphical images of the Haar (a) and Schauder (b) functions, which are locally defined on the interval $[0, T]$.



(a)



(b)

Figure 1: Graphic representation of Haar (a) and Schauder (b) functions.

The formulas for calculating the expansion coefficients for $m = 1, 2, 3$ have the following form [26, 27, 28]:

$$
\begin{aligned}
&C_0 = x(0);\\
&C_1 = -[x(0) - x(T)];\\
&C_2 = -0.5\left[x(0) - 2x\left(\tfrac{T}{2}\right) + x(T)\right];\\
&C_3 = -0.5\left[x(0) - 2x\left(\tfrac{T}{4}\right) + x\left(\tfrac{T}{2}\right)\right];\\
&C_4 = -0.5\left[x\left(\tfrac{T}{2}\right) - 2x\left(\tfrac{3T}{4}\right) + x(T)\right];\\
&C_5 = -0.5\left[x(0) - 2x\left(\tfrac{T}{8}\right) + x\left(\tfrac{T}{4}\right)\right];\\
&C_6 = -0.5\left[x\left(\tfrac{T}{4}\right) - 2x\left(\tfrac{3T}{8}\right) + x\left(\tfrac{T}{2}\right)\right];\\
&C_7 = -0.5\left[x\left(\tfrac{T}{2}\right) - 2x\left(\tfrac{5T}{8}\right) + x\left(\tfrac{3T}{4}\right)\right];\\
&C_8 = -0.5\left[x\left(\tfrac{3T}{4}\right) - 2x\left(\tfrac{7T}{8}\right) + x(T)\right].
\end{aligned}
$$

The works [26, 27, 28] show the perspective of using non-orthogonal functions in digital broadcast transmission systems and develop a device for analyzing signals based on Schauder functions, which can be implemented as an analog, as well as on the digital integral element base [29, 30, 31]. In these works, the results of research into the procedure for calculating the coefficients of the series expansion of signals according to the system of Schauder functions, taking into account the rational algorithm of using the values of the realization of the signal $x(t)$ at the points $l\Delta$ t are given $[0, T]$.

Analyzing the expression for calculating coefficients:

$$c_k = x\left(\frac{n-0.5}{2^{m-1}}T\right) - \frac{1}{2}\left[x\left(\frac{n-1}{2^{m-1}}T\right) + x\left(\frac{n}{2^{m-1}}T\right)\right],$$

where $k = 2^{m-1}+n; n = 1, 2, \dots, 2^{m-1}; m = 1, 2, \dots, M; T = 2^M\Delta$ t; Δ t is the signal quantization interval over time, it can be established that for $i = 0, 2, \dots, 2^m$ there is a regular repeatability, the same $x(t)$ values are used many times to calculate other coefficients.

So, for example, the value of $x(0)$ is used to calculate the coefficients $C_2, C_3, C_5, C_9, \dots, C_{2^{(m-1)}+1}$, value of $x(T)$ respectively for $C_2, C_4, C_8, \dots, C_{2^m}$; coefficients $C_4, C_5, c_9, \dots, C_{7\cdot 2^{m-3}+1}$, and $C_1, C_4, C_8, \dots, C_{7\cdot 2^{m-3}}$.

During the research, it was established that the values of $x(t)$ at the points $t = 2^{m-i} + r$ (where $i = 1, 2, 3, \dots$, $r = 0, 1, \dots, 2^i$) are used for calculation of a series of coefficients with serial numbers $k = 2^i + r$. When using this approach, the calculation time can be significantly reduced. Without taking into account the regularity, each time when calculating the next coefficient, it would be necessary to recalculate the coordinates of the $x(t)$ values on the segment $[0, T]$.

3. The method of operation of the audio codec according to the Schauder basis function system for the sound recognition system for the terrain monitoring sensor network

In this way, it is proposed to use the discrete Schauder transformation for the design and research of network encoding/decoding means, which can be implemented at the algorithmic-program level or in a hardware-technological design based on FPGA-type integrated circuits. The advantage of this method is the use of the Schauder transformation, which has certain advantages compared to trigonometric bases, for example, the possibility of local processing on the time and frequency interval when segmenting the incoming sound stream, reducing the time for mathematical calculations of the expansion coefficients, minimizing the amount of memory. All this significantly affects the speed of data delivery to the user. A variant of the structural diagram of the application of this method in the information transmission channel consisting of a Schauder encoder, a transmission channel and a Schauder decoder has been developed.

A method of dynamic sound recognition for a multi-sensor Internet of Things location monitoring sensor network, characterized in that modules are placed on each device or node of the Internet of Things that needs monitoring, which configure and receive operational data and data about the environment of the device or node of the Internet of Things (Figure 2).

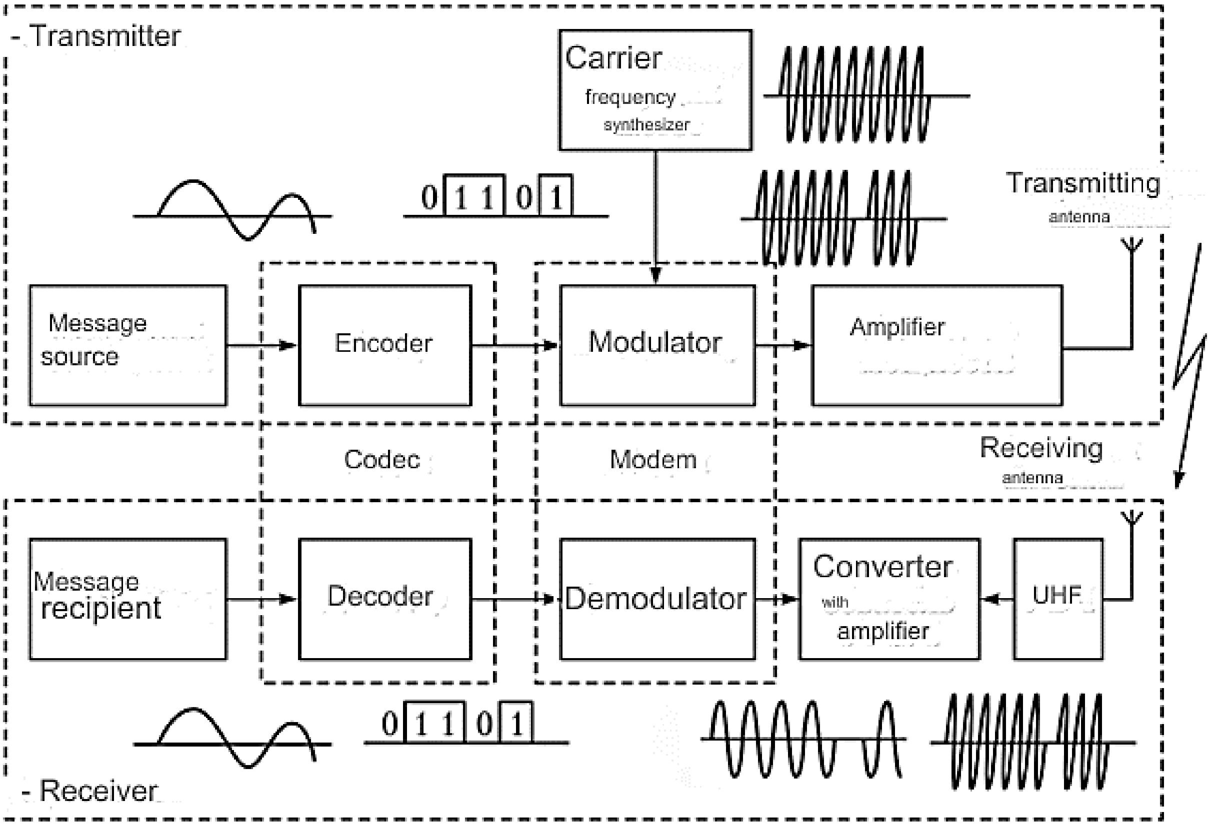


Figure 2: Structural diagram of the application of this method in the information transmission channel of the sensor network, consisting of a Schauder encoder, a transmission channel, and a Schauder decoder.

These modules consist of a transmitter and a receiver.

The transmitter contains the following modules and blocks:

- Message source: Generates an output message, which can be in the form of an analog or digital signal;
- Encoder: Converts a message into a digital code that is convenient for transmission. This may include data compression or error correction;
- Modulator: Uses a carrier frequency signal generated by a synthesizer to superimpose information (modulation) on a high-frequency signal;
- Power Amplifier: Amplifies the modulation signal to a level sufficient for transmission; • Transmitting antenna: Radiates an amplified signal into space.

The receiver contains the following modules and blocks:

- Receiving antenna: Receives a signal from space;
- UHF converter: The received signal is often attenuated and mixed with an intermediate frequency (UHF) for further processing;
- Demodulator: Extracts the information signal from the modulation signal;
- Decoder: Converts the digital code back to the original form of the message, performs decoding and error correction;
- Message Recipient: The end device receiving the recovered message.

A codec using the method of Schauder basis functions performed in the encoder and decoder for the audio recognition system works as follows:

Coding (encoder):

1. Analog-to-digital conversion (ADC): The audio signal is first converted from analog to digital using an ADC.
2. Decomposition into Schauder basis functions: The digital signal is decomposed into basic Schauder functions. These functions form a set of orthogonal bases that can be used to represent the signal. Decomposition is carried out by projecting the signal onto these basis functions. The result is a set of coefficients describing the signal in the space of basis functions.
3. Quantization of coefficients: The coefficients obtained as a result of the decomposition are quantized to reduce the amount of data. This means that the values of the coefficients are rounded to the nearest permissible value from a limited set of levels.
4. Coding of coefficients: Quantized coefficients are encoded for further transmission or storage. This may include the use of compression algorithms such as Huffman coding or other entropy coding techniques.

Decoding (decoder):

1. Decoding coefficients: The received coded coefficients are decoded to restore the quantized coefficients.
2. Interpolation and dequantization: The quantized coefficients are converted back to continuous form (dequantized) to improve the accuracy of the reconstructed signal.
3. Signal reconstruction: The reconstructed signal is formed by a linear combination of Schauder basis functions using dequantized coefficients. This process is the reverse of decomposition and allows the signal to be restored to its original form (or with allowable losses if it is a lossy system).
4. Digital-to-analog conversion (DAC): The recovered digital signal is converted back to analog form using a DAC if analog output is required.

The working principle of the method is based on the use of Schauder basis functions, which are orthogonal and enable the efficient representation of any signal, including audio signals, with minimal information loss. These basis functions yield coefficients that provide a compact representation of the signal while preserving its essential characteristics. By quantizing and encoding these coefficients, the data volume required for transmission is significantly reduced. The decoding and reconstruction processes then apply inverse operations to those used during encoding, allowing for the accurate recovery of the original signal.

4. Results of the study of speech signal processing

Figures 3–7 illustrate implementations of a fragment of a speech signal processed using the Schauder transformation. The signal is presented in blocks of equal size (2048 samples each). For each block, the decomposition coefficients were calculated with a spacing of 64 samples. The number of calculated decomposition coefficients was 64, 32, 16, 8, and 4. Accordingly, the synthesis of the output signal was performed using the specified number of coefficients (marked as "b" in the figure). In addition, an option for selecting informative coefficients from among the calculated ones based on the "|max|" or "max, min" principles was investigated, as shown in Figs. c and d, respectively.

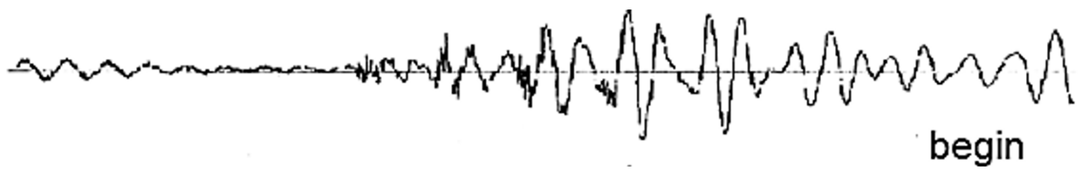
By comparing the obtained signal implementations, we can conclude that the transformation based on Schauder's functions reproduces the speech signal well. The calculation of consecutive 64 and 32 coefficients ensures a fairly accurate reproduction of signals.

When reproducing the signal by 16 coefficients, the implementation differs in the smoothing of local peaks in the areas of the high-frequency interval, while the low-frequency areas are reproduced quite well. If 8 coefficients out of 64 possible are used for signal synthesis, then the signalgrams show a clear difference between the output signal and the input signal.

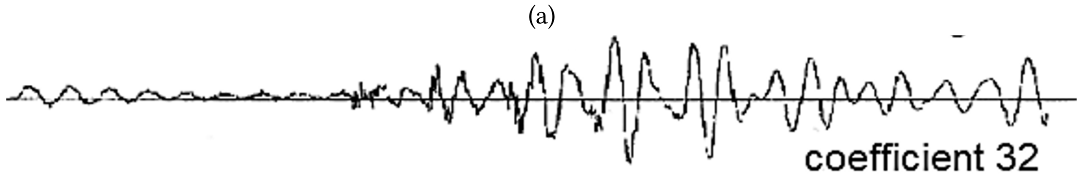
(a)

(b)

Figure 3: Signalgrams of a fragment of a speech signal: (a) - initial; (b) - restored by 64 calculated coefficients.



(a)



(b)

(c)

(d)

Figure 4: Signalgrams of a fragment of a speech signal: (a) - initial; (b) - restored by 32 calculated coefficients; (c), (d) restored by 32 selected from 64 calculated.

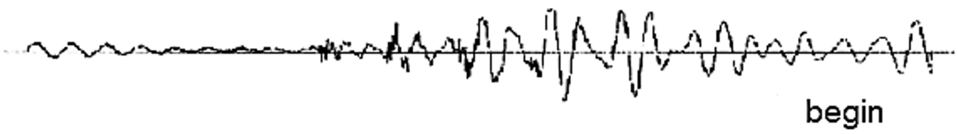
(a)

(b)

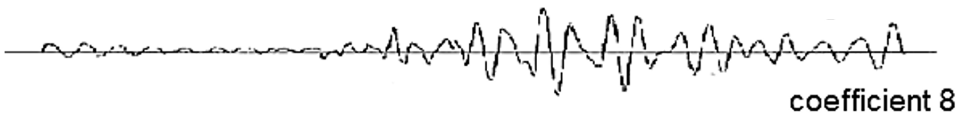
(c)

(d)

Figure 5: Signalgrams of a fragment of a speech signal: (a) - initial; (b) - restored by 16 calculated coefficients; (c), (d) restored by 16 selected from 64 calculated.

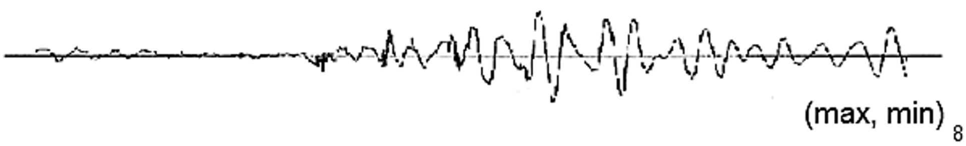


(a)



(b)

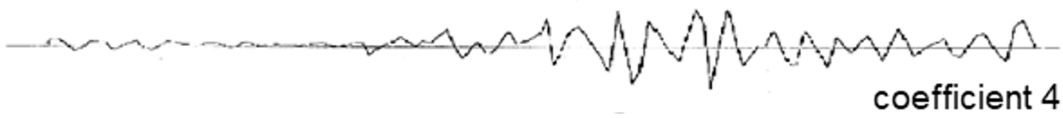
(c)



(d)

Figure 6: Signalgrams of a fragment of a speech signal: (a) - initial; (b) - restored by 8 calculated coefficients; (c), (d) restored by 8 selected from 64 calculated.

(a)



(b)

(c)

(d)

Figure 7: Signalgrams of a fragment of a speech signal: (a) - initial; (b) - restored by 4 calculated coefficients; (c), (d) restored by 4 selected from 64 calculated.

This difference is especially clearly expressed when using 4 coefficients of the decomposition. Similar conclusions can be drawn regarding the number of informative coefficients selected according to the "|max|" principle. or "max, min".

Comparing the images of the output signal obtained by synthesis with the same number of coefficients but selected in different ways, we come to the conclusion that these images differ from each other with a certain difference. In order to clarify these changes, it is necessary to conduct an auditory subjective analysis in the future. Carrying out only a visual comparison of the images of the output signals, it is difficult to determine an objective criterion in which case the output signal is of higher quality. In order to make a qualitative assessment of the output signals, you should listen to these signals and compare their static characteristics for different options.

5. Conclusions and discussion

In the theory and practice of signal processing, researchers increasingly turn to their representation by a finite system of basis functions, orthogonal or non-orthogonal.

The solution to this problem is relevant in connection with the need to improve the means of analysis, processing and synthesis of real physical processes.

Systems of non-orthogonal linearly independent functions have a number of advantages, for example, currently known PL-functions and Schauder functions [29, 11, 12, 14]. So, for example, the calculation of the decomposition coefficients in this case is reduced to the summation with the weights of the readings of the aliasing signal on the approximation segment, which for speech signals is chosen in the range of 8-22 ms.

Among the advantages of non-orthogonal decompositions should also be added the simplicity of their hardware implementations in comparison with analyzers and synthesizers built on the principle of orthogonal transformations [29, 32, 33].

The method is to include in each group of sensors sound sensors and ultrasound sensors that detect and recognize different acoustic waves, while using at least three sensors of each type in such a way that one sensor of each type is in working condition at the same time, and the other sensors were in an inactive state, after a certain period of time, the sensor in the working state is put into the rest state, and the other sensors in the rest state are put into the working state, and each sensor will store the environmental and operational data that it senses.

Information from the sensors is processed and transmitted to the anomaly analysis unit using a system based on the Schauder algorithm and containing information transmission channels consisting of a Schauder encoder, a transmission channel and a Schauder decoder, the data anomaly analysis unit is configured to perform data anomaly analysis based on the received pre-processed data and obtaining the result of the analysis of anomalies, as well as correcting the errors of the result of the analysis of anomalies to obtain the final result of the analysis.

Declaration on Generative AI

The authors have not employed any Generative AI tools.

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