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AUDIO SIGNAL TRANSMISSION METHOD IN NETWORK-BASED AUDIO ANALYTICS SYSTEM

The **subject matter** of the article is audio signal transmission method in network-based audio analytics system. The creation of a network-based audio analytics system leads to the emergence of new classes of load sources that transmit packetized sound data. Therefore, without constructing adequate mathematical models, it is impossible to build a well-functioning network-based audio analytics system. A fundamental question in traffic theory is the question of load source models. The development of an method for transmitting audio signals in a network-based audio analytics system becomes necessary. Based on this, the **goal** of the work is to create methods an method for transmitting audio signals in a network-based audio analytics system to ensure efficiency and accuracy in audio analytics. The following **tasks** were solved in the article: the formation of a model for the system's load sources, investigation of connection and traffic management, implementation of control and traffic monitoring functions in the network, research of methods to ensure the quality of audio signal transmission and the development of a method of transmitting an audio signal by virtual routes switching. To achieve these goals, the following **methods** are used: mathematical signal processing, data compression algorithms, optimization of network protocols, and the use of high-speed network connections. The obtained **results** include modeling of the system's load sources, examination of connection and traffic management, investigation of methods to ensure the quality of audio signal transmission and a method of transmitting an audio signal by virtual routes switching was proposed. In **conclusion**, the possibilities of using simulation modeling of nodes in the network-based audio analytics system are highly limited. This is explained by the fact that the acceptable level of information loss in data centers is very low. The use of the developed method enables effective control and processing of sound information in real-time. This method can find broad applications in various fields, including security, healthcare, management systems, and other industries where the analysis of audio signals is a crucial element.

Keywords: audio analytics; audio signal transmission method; traffic management; virtual routes; delay.

Introduction

In the era of continuous development of audio technology and network systems, the importance of audio signal transmission in a networked audio analytics system is becoming indisputable. The method of audio signal transmission in a networked audio analytics system plays an important role in the development and implementation of modern technologies aimed at improving the safety, convenience, and efficiency of various areas of our lives.

Networked audio analytics systems are used to monitor and detect events in public places, offices, transportation, etc. The effectiveness of these systems depends on the accuracy and speed of audio signal transmission, which allows for real-time response to danger or violations. In the field of building and infrastructure management, network systems are used that analyze audio signals to detect changes in the environment. This allows you to automatically respond to changes in the use of premises and energy efficiency. By transmitting audio signals in a networked system, it is possible to implement interactive control and interaction systems, such as voice assistants or voice recognition systems, which make the use of technology more convenient and accessible. In the medical field, the transmission of audio signals in a networked system is used to record and analyze sounds, which can be useful in diagnosing various conditions or monitoring patients.

The ability of the transmission method to operate in real time is of great importance. Delay in the transmission of audio signals can negatively affect the system's response time, which is critical in security or instant control scenarios. Therefore, the audio transmission method in a networked audio analytics system becomes a key tool for improving various aspects of everyday life, providing the ability to use audio information with maximum efficiency and safety.

Analysis of recent research and publications

The efficiency of an audio analytics system depends on the accuracy of audio signal recognition. Papers [1–4] consider modern methods for processing various audio signals, including human speech. The analysis of the dataset for training the audio signal recognition model is presented in [5]. The task of evaluating the accuracy of an audio signal processing method is presented in [6]. The transmission method must ensure stable transmission

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of high-quality audio data so that the system can correctly identify and analyze audio information.

Delay and real-time capability are critical parameters for the effectiveness of the audio transmission method in a networked audio analytics system, especially in demanding applications such as security, medicine, and other areas where rapid response to events is important. Additionally, the transmission method should be optimized to minimize the bandwidth consumed, especially in the face of large amounts of data being transmitted. This makes the system more efficient and cost-effective. These issues are discussed in [7–9].

The transmission method must demonstrate resistance to errors and noise that may occur during the transmission of audio signals over the network. Ensuring reliable transmission even in the presence of various interferences is a key aspect of efficiency. The issues of audio signal compression and audio codecs are considered in [10–12].

The ability of the transmission method to adapt to an increase in the amount of data and the scale of the system is an important factor in the context of the development and expansion of applications of the networked audio analytics system. The issues of scalability of complex networks are discussed in [13, 14] and the results of these works can be applied to the construction of networked audio analytics systems.

Due to the increase in the number of connected devices and the volume of audio data exchange, it is important to develop methods to protect against potential attacks and ensure the confidentiality of transmitted data. Security issues are discussed in [15, 16].

Identifying parts of a common problem

In the digital era, the transmission of audio signals in a networked audio analytics system is proving to be a key aspect for the development and improvement of audio technology. Methods of transmitting audio data over a network are becoming increasingly important in ensuring not only the quality of sound reproduction, but also in analyzing, processing and understanding large amounts of audio information. The creation of a network audio analytics system leads to the emergence of new classes of load sources that transmit packetized audio data. Therefore, without building sufficiently adequate mathematical models, it is impossible to build a well-functioning network audio analytics system. A fundamental issue in the theory of telecommunication traffic is the issue of load source models.

Thus, it becomes necessary to develop a method for audio signal transmission in a networked audio analytics system. Based on this, the purpose of this paper is to develop a method for transmitting audio signals in a networked audio analytics system by switching virtual routes.

Modeling of system load sources

The main feature of the load sources of network audio analytics systems is that they are characterized by different information transmission rates. The introduction of new models of load sources requires new descriptions for the load itself. When describing the load models that arise in networked audio analytics systems, a three-level load processing model is used, as shown in Fig. 1.

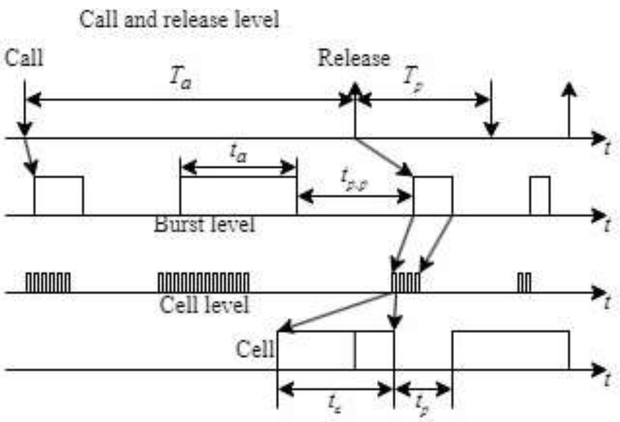

Fig. 1. Three-level load model:
T_a – time of the user's active state;
T_p – time of the user's passive state;
t_a – time of the user's active information transmission;
t_p – duration of the pause in information transmission;
$t_{p,p}$ – duration of the pause between cell transmissions;
t_c – time of cell transmission.

The first level – the Call Level – corresponds to the duration of a communication session. At this level, it is necessary to take into account the activity of individual load sources, to know the intensity of calls from them and the intensity of their service.

The second level (Burst Level) determines the nature of the bitstream at the time of user service and corresponds to the moments of activity and passivity. The distribution of pause durations between bit streams determines the burstiness of the message.

The third level is the Cell Level, which is the level of transmission of cells transmitted to the transport network. Transmitted cells from all load sources form

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the traffic of the audio analytics network system. This level directly determines the capacity of buffer drives, the probability of cell loss, and the time they wait in the queue.

A networked audio analytics system is a system of high structural complexity, for which analytical models have been developed relatively recently. Simulation modeling is widely used to verify analytical models of user load and quality of service (GoS). In networked systems, audio analysts estimate GoS not only by the probability of loss of information cells, but also by the probability of their delay, and even by the value of delay jitter.

Call admission rules and the service strategy itself have a significant impact on these service quality characteristics. A large number of studies have been devoted to solving these three fundamental problems when creating networked audio analytics systems [17, 18].

A certain flexibility is required from the traffic management system because it is necessary to maintain the specified QoS and GoS levels. At the same time, the control system is required, on the one hand, to be relatively simple to implement at the ATM level and, on the other hand, to be robust, i.e., to ensure high efficiency of resource management when the traffic situation on the network changes. When organizing access to the network system, audio analysts distinguish two aspects, namely the rule for call access in the network (Call Admission) and the strategy for servicing a call when the call has already been accepted for service (Source Policing).

The first aspect is related to the study of the impact of priorities on the quality of service, and the second aspect is related to the uniformity of information cells in the network. The rules for admitting a call to the network are based on knowledge of the link state, i.e. the number of calls of different types accepted for service. The decision to accept the next call is made if the probability of loss of information cells is within acceptable limits.

To satisfy these aspects, the network audio analytics system needs to implement two functions of managing and controlling traffic in the network: user access control (Connection Admission Control) and Usage Parameter Control (UPC).

Managing connections in a networked audio analytics system

The Connection Admission Control (CAC) algorithm defines the actions performed by the network that allow a connection to be established. With the CAC algorithm, a connection is established only if the network can support this connection with the required QoS and does not degrade the quality of service of already established connections. The algorithm allows you to block the admission of new connections to the network if the accumulation of cells in the queues increases. The ACS algorithm performs actions both at the initial stage of call service and at the stage of renegotiation of call service conditions. The ACS algorithm decides whether to provide a virtual channel connection (VC) or a virtual path connection (VP) to the consumer. This algorithm strikes a balance between efficiency and QoS.

To decide whether to accept or reject a connection, the ACS algorithm must determine the acceptable traffic volume limits and the required QoS class (values of cell delays, cell delay jitter, cell losses and bursts).

At the stage of implementing the CAC algorithm, the so-called Traffic contract is determined. This contract requires compliance with a set of service parameters, including QoS parameters. According to the ITU recommendations, the contract is monitored by checking it for compliance with the accepted service rule and network QoS. The contract identifies the user with whom it is possible to change its further service. The set of parameters describing the traffic is called the traffic descriptor of the connection being established. The traffic descriptor is regulated by the ITU and is associated with the establishment of a single BP connection or a connection BC. According to the ITU recommendations: the traffic descriptor defines the traffic parameters, the compliance of the traffic parameters with the validation rule, and the network QoS.

Thus, the CAC algorithm represents a set of actions that must be performed at the connection establishment stage in order to accept or reject user service. A connection request is accepted for service only if there are sufficient resources to realize a connection throughout the network with the required GoS. At the same time, it is a mandatory requirement to maintain the accepted quality of service of already established connections.

Traffic management in a networked audio analytics system

User-generated load policing and Usage/Network Parameter Control (UPC/NPC) are performed at the User-Network Interface (UNI) and Network-Node Interface (NNI) level. This control is a set of actions

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that must be performed by the network to monitor and manage the traffic generated by ATM connections. The main purpose of managing the parameters of the traffic being served is to monitor each established connection for its stricter compliance with the traffic contract. An ideal UPC/NPC algorithm should be able to detect any unacceptable traffic situations, respond quickly to deteriorating traffic parameters, and have a simple implementation.

Among the methods proposed to meet these requirements, considerable attention is paid to the method that has received the conventional name of "Leaky bucket". Another method widely used in data transmission networks is the so-called Sliding Window Protocol.

Two approaches are used to build a control procedure: statistical and based on the use of some a priori rules (the so-called operational approach). The disadvantage of the statistical approach is the need for long-term observation and, as a result, a slow management response. Therefore, in practice, operational approaches are used.

ITU Recommendation 1.371 presents two equivalent versions of the operational algorithm – the so-called Generic Cell Rate Algorithm (GCRA). The first version is called the Virtual Scheduling Algorithm (VSA), and the second version is called the Leaky Backet Algorithm (LBA).

Both versions of GCRA, for any sequence of cell arrival moments $(t_a \geq 1)$, determine the cells that can be transmitted along the link or their transmission will be delayed.

The VSA virtual scheduling algorithm uses two parameters: I – increment; L – some limit of its increment and a set of intermediate variables $GCRA(I,L)$. The theoretically predicted moment of TAT cell appearance is calculated under the assumption that the cells are uniformly distributed in time, and the distance between two consecutive cells is equal to I at the moment of source activity. If the actual arrival time exceeds $t = \mathrm{TAT} - L$, where L is some acceptable value, then the cell is allowed, otherwise the cell is delayed.

A diagram of the virtual scheduling algorithm is shown in Fig. 2.

The LBA algorithm (Fig. 3) shows how the burst flow of cells fills bucket B1. If bucket B1 is not overflowing, then cells enter the transmission medium at a guaranteed rate. If bucket B1 is full, the excess flow goes to bucket B2. If bucket B2 is not overflowing, then from B2 the flow of cells enters the transmission medium at an arbitrary rate, but less than the guaranteed rate. If B2 is full, the excess flow is discarded. The described flow control scheme corresponds to the algorithm shown in Fig. 4.

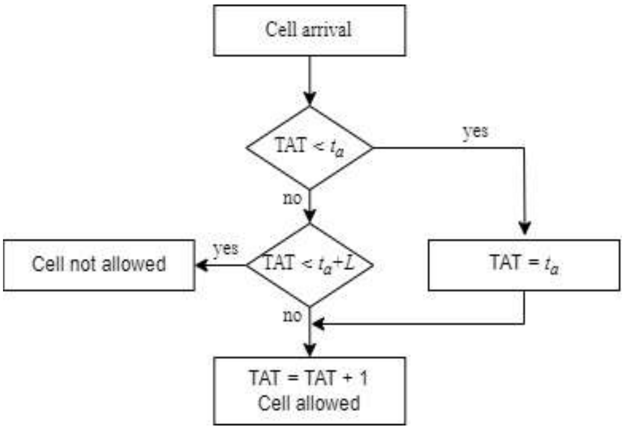

Fig. 2. Virtual scheduling algorithm

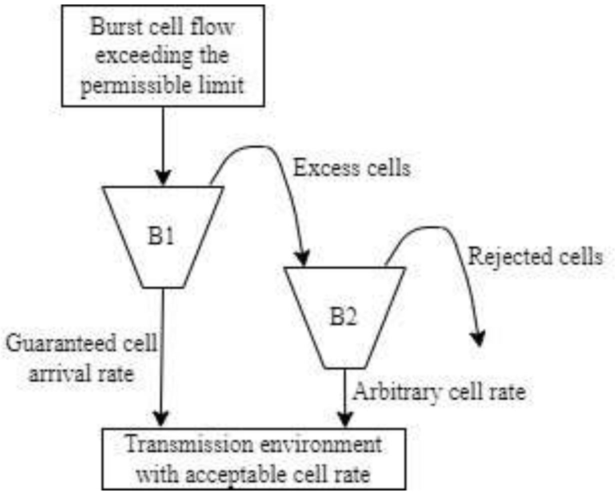

Fig. 3. Physical interpretation of the "leaky bucket" method

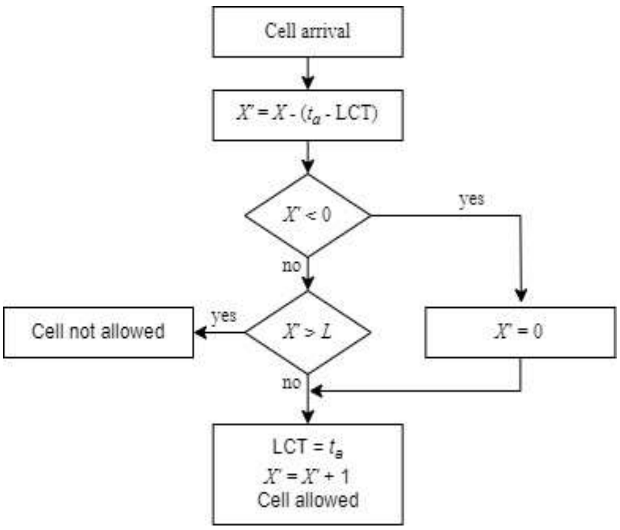

Fig. 4. "Leaky bucket" bitstream control algorithm

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It should be noted that unlike the sliding window mechanism, the leaky bucket mechanism operates with individual cells. The flow of cells is copied to the so-called pseudo-queue ("leaky bucket"). The queue is serviced at a certain speed (the rate of the bucket). When the buffer is full, the cells of the outgoing stream are delayed.

Each established connection is assigned a counter, the content of which increases by one when an information cell arrives and decreases accordingly if the current bit rate is acceptable in the transmission link. The counter has a certain threshold L. If the counter is equal to L, the access of the cells to the network is terminated, and they are transferred to the buffer drive. Thus, this method of service controls two parameters: the current information transfer rate (the number of cells transmitted per unit of time) and the threshold L (bucket capacity).

Estimates of the probability of cell loss and waiting time in the buffer drive during statistical multiplexing, taking into account the above factors, are important system characteristics.

Ensuring transmission reliability

Semantic transparency (ST) of a network is defined as the ability of a network to transport messages from a source to a destination with an acceptable number of errors. With a given transmission system, network transparency is ensured by encoding the message, repeating the message at the receiver's request, and a combination of these techniques. The most important characteristic that defines the ST is the possibility of distortion of bits of information – the Bit Error Rate (BER). BER is defined as the total number of erroneous bits divided by the total number of transmitted bits.

For different transmission systems, the BER measurement is practically reduced to a set of statistics. Since the probability of bit distortion is different in the transmission systems and the transmission medium, it takes different time to obtain representative statistics. In modern fiber-optic communication lines, the value of $\mathrm{BER} = 10^{-9}$. As the BER probability increases, the number of retransmissions of damaged packets increases. The increase in load can be estimated by the following formula, which takes into account the use of the sliding window control protocol:

$$R(n) = \frac{W}{2} \times \frac{1-(1-\mathrm{BER})^{nL}}{(1-\mathrm{BER})^{nL}}, \qquad (1)$$

where W – window size;

n – number of transmission links;

L – packet length, bits.

Figure 5 shows the dependence of traffic growth on the change in BER on the transmission link. The graph is plotted for the case of five transmission links $(n = 5)$. In this figure, the traffic is evaluated relative to the original traffic. For example, if $R(\mathrm{BER}) = 1$, it means that the number of transmitted packets doubles.

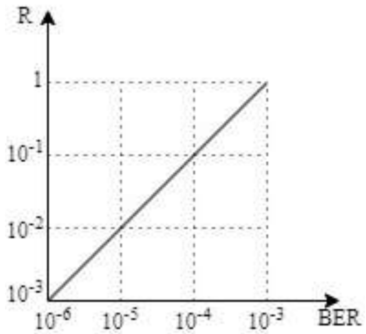

Fig. 5. Dependence of load increase on the probability of BER error on the transmission link

At a low BER value, link-to-link control is not very effective, because the probability of errors is extremely low. In this case, end-to-end control is the most effective.

Figures 6 and 7 illustrate the case of link control and end-to-end control. They show the dependence of the load increase on the change in the BER error probability. For end-to-end control, the case of five links is considered. Fig. 6 corresponds to the case when the packet length is 53 octets, Fig. 7 corresponds to the case when the packet length is 1000 octets.

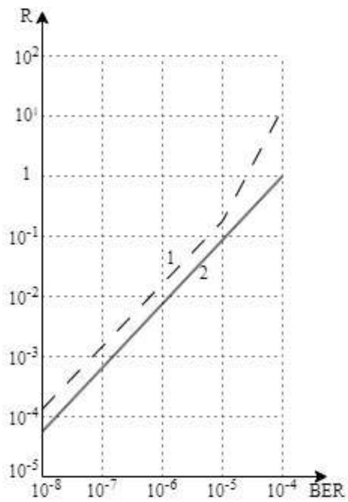

Fig. 6. Dependence of the change in link utilization on the change in the error probability on the link at a packet length of 53 octets: 1 – end-to-end control, five links; 2 – control on one transmission link

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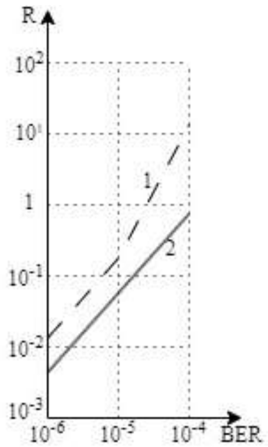

Fig. 7. Dependence of the change in link utilization on the change in the error probability on the link at a packet length of 1000 octets:
1 – end-to-end control, five transmission links;
2 – control on one transmission link

The lower the BER value in the channel, the more efficient the end-to-end control is compared to the link control. Fig. 8 shows the dependence of the efficiency of end-to-end control compared to link control for a standard cell of length $53 \times 8 = 424$. The effectiveness will be evaluated by the ratio of the loads of the path with n-link control and one-link control:

$$C = R(n)/R(1). \qquad (2)$$

It is assumed that there are five transmission links between users $(n = 5)$, i.e.:

$$C = R(n)/R(5). \qquad (3)$$

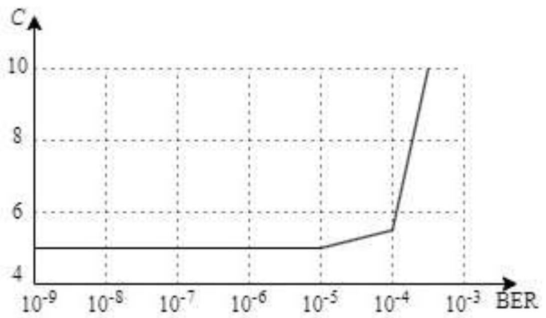

Fig. 8. Dependence of efficiency change on error probability

From the graph shown in Fig. 8, it can be determined that up to a value $\mathrm{BER} \le 10^{-5}$, "end-to-end" control is practically indistinguishable from link control. Indeed, if $\mathrm{BER} \le 10^{-9}$ then $C = 5{,}0$, and if $\mathrm{BER} \le 10^{-5}$ then $C = 5{,}04$.

In packet-oriented networks, a different assessment is used – PER (Packets Error Rate). The PER value is determined for a certain normalized time and significantly depends on the properties of the transmission medium. It is calculated as the number of erroneously received packets to the total number of transmitted packets.

Buffer overflow can also lead to an increase in the load on the link. However, in practice, the probability of cell loss is extremely low $\left(P_{cell} \le 10^{-9}\right)$. The capacity of buffer drives that ensure a low probability of cell loss can be seen in Fig. 9, which was calculated using the model $M/D/1/L$ $(L \ne \infty)$. The curve is plotted at the load intensity applied to the link $a = 0{,}8$ Erl.

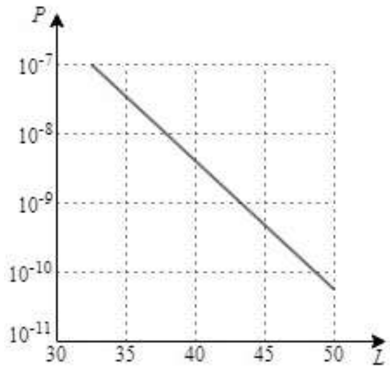

Fig. 9. Dependence of the probability of cell loss on the change in the buffer storage capacity at $a = 0{,}8$ Erl

Compliance with time requirements

The time transparency of the network is measured by the message delay and the end-to-end jitter of the message delay. Delay significantly affects the quality of communication for users who transmit real-time audio or video. Long delays for audio messages are manifested in the form of an echo effect. The amount of signal delay during end-to-end transmission is normalized.

The delay time consists of two components, namely, the delay caused by the transmission medium and the delay caused by the processing of messages (packets). In digital switching systems, the amount of delay is limited by the value of $T_{csk} \le 450$ µs.

The delay time caused by message processing can be estimated as a first approximation as the probability that the actual waiting time will exceed the permissible time t_a and can be defined as follows:

$$P(>t_a) = 1 - (1-\lambda)\left[\sum_{k=0}^{[t]} \frac{\left(\lambda(k-t)\right)^k}{k!} e^{\lambda(t-k)}\right]. \qquad (6)$$

The form of the function $P(>t_a)$ for $\lambda =$ 0,2, 0,4 and 0,8 as a function of waiting time is shown in Fig. 10.

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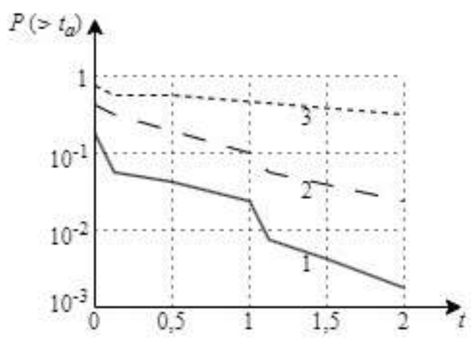

Fig. 10. Change of probability $P(>t_a)$ for three load values: 0,2 Erl; 0,4 Erl; 0,8 Erl

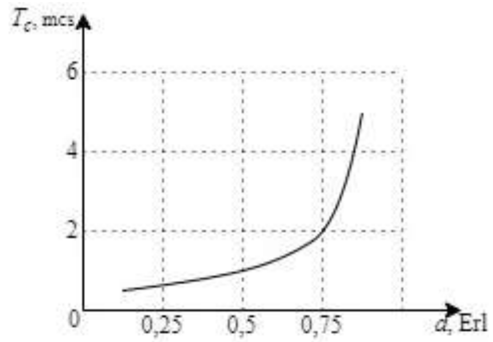

Fig. 11. Dependence of the average delay time on changes in load intensity

Dependence of the average time spent in the queue T_c of the delayed cells from the change in load a is shown in Fig. 11.

The nature of connection delays for different transmission speeds on the transmission link (1 and 10 Gbps) can be seen in Table 1. The data in Table 1 are obtained with the probability of packet loss and the intensity of the incoming load $a = 0,8$ Erl.

Table 1. *The nature of connection delays at different bit rates*

Delays, µs	**Transmission speed on the link, Gbit/s**					
	1			**10**		
	Packet size, bytes					
	256	512	1024	256	512	1024
Transmissions	4000	4000	4000	4000	4000	4000
Fixed on the switching node	16	32	64	4	8	16
In the queue and during depacketization	100	200	400	10	20	40
During packetization	500	1000	2000	100	200	300
In a synchronous network	900	900	900	900	900	900

Method of audio signal transmission by switching virtual routes

The method of audio signal transmission in an audio analytics network system by switching virtual routes is based on the principle of fast packet switching. This method is designed to optimize the transmission of audio signals in a network audio analytics system. The main goal is to ensure the efficiency and accuracy of real-time audio analytics. When creating this method, mathematical models were developed to adequately reflect new classes of load sources that transmit packetized audio data and analysis and improvement of connection and traffic management methods in the network system for resource utilization were performed. As a result, mechanisms for managing user access to the network and managing resource utilization parameters were implemented, and an efficient switching method based on the use of virtual routes for routing audio packets was created.

The principle of fast packet switching is shown in Fig. 12. A cell arriving at a switching node (SN) contains an information field and a header. At the switching node, using a special routing process, the cell is assigned bits of an additional address representing the address of the route of the cell through the switching system. As the cell passes through the cascades of the switching system, the bits of the additional address are used to select a route. The output cell, as before, contains only the information field and the header.

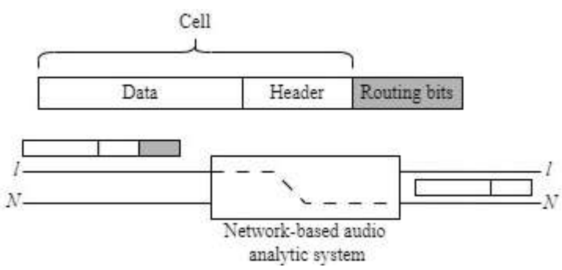

Fig. 12. The principle of fast packet switching

To connect two terminals to each other, you need to switch not only virtual channels, but also virtual paths. A VP switch is a virtual path switch, a VP/VC switch is a virtual path switch, and a virtual channel switch.

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A VP/VC switch consists of a virtual path switch (VP switch) and a virtual channel switch (VC switch). The switching scheme of virtual paths and channels is shown in Fig. 13.

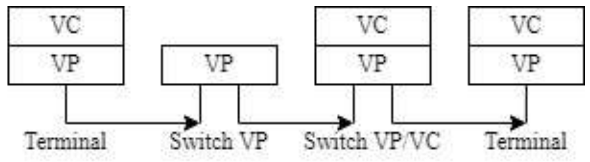

Fig. 13. Switching scheme for virtual paths and channels on the network

Assigning a route address and routing a cell by a switching system requires certain management resources. Fig. 14 shows the conditional dependence of control costs on the transmission rate for different types of switching. As follows from Fig. 14, the control costs of the FPS are significantly higher than in the case of channel switching and multichannel switching, but lower than in the case of packet switching in virtual channel mode and datagram mode.

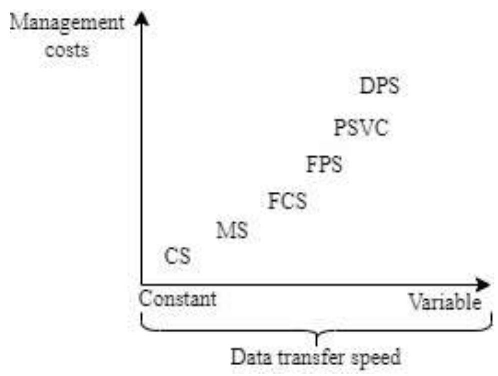

Fig. 14. Dependence of management costs on the transmission rate for different switching modes:
CS – channel switching; MS – multichannel switching;
FCS – fast channel switching; FPS – fast packet switching;
PSVC – packet switching with virtual channels;
DPS – datagram packet switching

The possibilities of using simulation modeling of the nodes of the audio analytics network system are very limited. This is due to the fact that the acceptable level of loss of information cells is very low. When constructing probabilistic models of audio analytics network systems, various approximations are used: Markov processes, combinatorial approaches, methods based on decomposing the system into subsystems with a simpler structure. In some cases, the Wiener-Hopf approximation is used to estimate the delays of information centers, which sometimes allows obtaining fairly simple formulas when the incoming flow of requests is a Markov modulated Poisson flow.

The quality of service of information cells in a networked audio analytics system is significantly affected by routing. The so-called FPS mode is implemented at the AC of the audio analytics network system. This switching mode has both the properties of the channel switching mode and the packet switching mode.

When creating networked audio analytics systems, the search for networks with an optimal structure is continuously underway, as well as the development and research of methods for calculating their probabilistic and time characteristics. Intensive research is currently underway to create new classes of networks. In particular, networks with so-called "variable" routing are of interest. In such networks, an information cell can pass through only a part of the system's cascades, not all of them. To create such opportunities, a special switching element with a capacity of 2×4. Special classes of networks are being studied that allow for the selection of a shared buffer. Due to the rapid improvement of the technology, it is interesting to study networks built with the use of high-capacity switches: 8×8, 16×16, 32×32 inputs and outputs.

An important problem when building networked audio analytics systems is the problem of network management. When creating a sufficiently branched network audio analytics system, the problem of distributing the total transmission rate along the path between individual network nodes arises. Network management also affects the admission of calls to the network and routing. These issues are of great importance and are still poorly understood.

Conclusions

The main requirements for the efficiency of the audio transmission method in a networked audio analytics system include minimizing latency to meet high standards of interaction and security. Research in the field of optimizing audio data processing algorithms and the use of fast network connections are identified as important factors in achieving these goals.

Consideration of scalability and error tolerance requirements is also important for the successful implementation of the audio transmission method at various levels. A balanced approach to the development and improvement of this technology allows to take into account the diverse requirements of different areas and opens the way for innovative solutions.

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Thus, the following results were obtained in this article:

1. Modeling of the system load sources was performed.
2. Connection and traffic management in a networked audio analytics system is considered.
3. Methods for ensuring the quality of audio signal transmission are investigated.
4. A method of audio signal transmission by switching virtual routes is proposed.

Future research in the field of audio signal transmission methods in a networked audio analytics system may focus on the development of data compression technologies, the use of artificial intelligence to improve recognition accuracy, and the study of integration with other technologies, which will help to increase the functionality and versatility of the method.

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МЕТОД ПЕРЕДАЧІ АУДІОСИГНАЛУ У МЕРЕЖНІЙ СИСТЕМІ АУДІОАНАЛІТИКИ

Предметом дослідження в статті є метод передачі аудіосигналу в мережній системі аудіоаналітики. Створення зазначеної системи приводить до появи нових класів джерел навантаження, що передають пакетизовані звукові дані. Тому без побудови достатньо адекватних математичних моделей не можна створити мережну систему аудіоаналітики, яка ефективно працює. Фундаментальним у теорії телетрафіку є питання моделей джерел навантаження. Отже, стає необхідним розроблення методу передачі аудіосигналу в мережній системі аудіоаналітики. З огляду на сказане **метою статі** є створення методу передачі аудіосигналу в мережній системі аудіоаналітики з метою забезпечення ефективності та точності аудіоаналітики. У статті вирішуються такі **завдання**: формування моделі джерел навантаження системи, дослідження управління з’єднаннями і трафіком у мережній системі аудіоаналітики, реалізація функцій управління та контролю трафіком у мережі (управління доступом користувача до мережі та управління параметрами використання), дослідження методів забезпечення якості передачі аудіосигналу та розроблення методу передачі аудіосигналу з допомогою комутації віртуальних маршрутів. Для досягнення поставлених завдань використовуються такі **методи**: математичне оброблення сигналів, алгоритми компресії даних, оптимізація мережних протоколів і використання швидких мережних з’єднань. Досягнуто таких **результатів**: змодельовано джерела навантаження системи, розглянуто управління з’єднаннями і трафіком у мережній системі аудіоаналітики, досліджено методи забезпечення якості передачі аудіосигналу та запропоновано метод передачі аудіосигналу з допомогою комутації віртуальних маршрутів. **Висновки**. Можливості використання імітаційного моделювання вузлів мережної системи аудіоаналітики дуже обмежені. Це пояснюється тим, що допустимий рівень втрат інформаційних осередків дуже низький. Упровадження розробленого методу передачі аудіосигналу в мережній системі аудіоаналітики дає змогу досягти ефективного контролю та оброблення звукової інформації в режимі реального часу. Цей метод може широко застосовуватися у сферах безпеки, медицини, систем управління та інших галузях, де аналіз аудіосигналів є важливим складником.

Ключові слова: аудіоаналітика; метод передачі аудіосигналу; управління трафіком; віртуальні маршрути; затримка.

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