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AUDIO COMMUNICATION QUALITY PROVISION IN A SELF-ORGANIZING NETWORK*

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Abstract

The quality of audio communication in self-organizing wireless networks is significantly lower than in the telecommunication networks with a fixed structure due to the negative effect of low signal transmission power, interference in channels, high dynamics of topology because of ad-hoc node mobility and the presence of destructive factors. It is possible to improve the transmission quality of voice streams provided that packet delivery delays, jitter and packet loss rate are minimized. The analysis of scientific and technical literature showed that the available theoretical developments and practical results focused on audio communication provision do not significantly reduce packet losses and delays, therefore, they do not guarantee the achievement of high-quality voice transmission in a wireless self-organizing network, which requires obtaining new scientific technical solutions in this subject area.

Keywords: wireless self-organizing networks, transmission, voice streams, audio communication quality.

1. Introduction

Self-organization procedures for software and hardware are used more and more often in modern wireless communication technologies. Self-organizing networks consist of nodes, the configuration of which is carried out automatically and does not require the functioning of additional infrastructure (central control element). The structure of such networks is formed in an arbitrary way, depending on a current situation. To ensure communication between the nodes of self-organizing networks, the technologies of the following wireless information transfer standards are mainly used: IEEE 802.11 (Wi-Fi) and Masrour, (2021). The choice of one or another technology for wireless information transmission should depend on the type and application

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features of a self-organizing network being created. There are the following main types of self-organizing networks:

1) mesh networks (Wireless Mesh Network, WMN);

2) wireless sensor networks (Wireless Sensor Networks, WSN);

3) ad-hoc networks, the varieties of which are wireless self-organizing networks (Mobile Ad hoc NETwork, MANET), automobile networks (Vehicular Ad-hoc NETworks, VANET), flying networks (Flying Ad-hoc NETworks, FANET).

Self-organizing networks can be used successfully in the absence of traditional telecommunication infrastructure and at possible destructive external influences to ensure the exchange of necessary information when solving the following tasks:

1) conducting of search and rescue operations;

2) organization of civil defense and counteraction to emergency situations;

3) the fight against terrorism and the elimination of terrorist act consequences;

4) provision of technical protection for the workers at hazardous industrial facilities (mines, power plants, chemical and oil refineries);

5) protection of important geographically distributed objects (bridges, subways, electricity, gas and oil pipelines) (Darmody and Bendis, 2021; Gardner, 2021; Konstantinov et al., 2016; Konstantinov et al., 2017; Singj etal., 2022).

In this case, the MANET packet transmission network has the greatest potential, all nodes of which are interchangeable and capable of performing the functions of not only terminal devices, but also of relay-routers. To ensure the efficiency of performing the abovementioned tasks, high-quality transmission of voice streams is required, on the basis of which the exchange of voice messages takes place containing:

1) the information about the current situation at the controlled facility, the incidents that have occurred, the risk of their occurrence, the hazardous areas of the facility requiring prompt intervention;

2) the information on the composition of the injured persons, their location, health status, and the necessary assistance;

3) the data on the scale and details of the material damage received;

4) information on the need for forces and means to counter the occurrence of emergencies in certain areas of the facility and eliminate their negative consequences.

In the process of performing the abovementioned tasks, in the event of complex or critical situations in which coordinated management of the activities of numerous participants (workers, rescuers, representatives of law enforcement agencies, medical workers, other specialists and volunteers) is required, periods of time are observed when the activity of MANET users is increased (Boni and Gunn, 2021; Ghebregiorgis and Negusse, 2022; Polshchykov et al., 2017; Pour et al., 2022). At the same time, the need for the transmission of voice messages increases sharply, which leads to a significant shortage of the required channel resources and to audio communication characteristic deterioration. At such moments, the awareness of the event participants, the correctness of the current situation understanding by them, the adequacy of their response depends on the quality of voice transmission which ultimately affects the saving of human lives and preservation of health, the amount of material damage, and the effectiveness of fulfilling the assigned tasks. Of course, the performance effectiveness of these tasks depends on many different objective and subjective factors, but it can be reduced due to poor communication in the control network. In order to avoid this efficiency decrease, the quality of voice stream transmission in the network should be such that the quality of speech received at the receiving nodes of the network corresponds to the highest class.

2. Materials and methods

Quality control of voice stream transmission can be based on the analysis of packet delivery characteristics over the network, which include: packet delivery delay; packet delivery delay variation (jitter); packet loss ratio The studies have shown that the quality of voice transmission in self-organizing wireless networks is significantly lower than in traditional communication networks. In accordance with ITU-T Y.1541 recommendations, it is required that the packet delivery characteristics do not exceed the values presented in Table 1 to ensure guaranteed transmission quality of voice streams.

Table 1. Package delivery performance requirements

Package delivery characteristics	***Values***
Package delivery delay	100 ms
Jitter	50 ms
Packet loss ratio	0.001

Audio streams are traffic sensitive to packet delays and jitter. The studies have shown that speech transmission delays over 150 ms degrades unacceptably the quality of a telephone conversation. When the difference in packet delivery delays exceeds several tens of milliseconds, sound distortions occur, leading to speech intelligibility. In the process of voice stream transmission, insignificant packet losses are allowed, the loss of which can be compensated for at a receiving node by approximating the values of previous and subsequent measurements of audio information (Polshchykov et al., 2019).

The main reason for packet delivery delays, jitter and packet loss increase is network congestion caused by a shortage of channel resources and leading to long waiting for packets in queues and overflowing the buffer memory of routers (Konstantinov et al., 2017; Polshchykov et al., 2018). A large number of publications are devoted to the problems of high-quality transmission provision for voice streams in MANET with a shortage of channel resources. The known approaches and methods aimed at solution of these problems are reduced to the creation of tools to decrease the level of network congestion, minimize losses and delays in transmitted packets, and reduce packet jitter. In order to transmit streaming information to MANET, a number of new routing protocols have been created (Ahmed et al., 2018; Rath et al., 2016; Seetharam et al., 2015; Tardioli et al., 2015; Xie et al., 2015), but their use does not allow channel congestion reduction to the required level. The solution to this problem is based on a traffic shaping mechanism for real-time applications based on the transfer of tokens in the process of packet routing in the transit nodes MANET (Rath et al., 2017), a hybrid scheme is developed to control the frequency of routing information update when streaming information in MANET. It should be noted that most of the abovementioned approaches and protocols do not significantly reduce packet loss and delay, therefore, they do not guarantee the achievement of a given level of voice transmission quality, and, consequently, the effectiveness of audio communication in MANET.

3. Results and discussion

The most preferred means of ensuring the voice stream transmission quality is channel performance reservation, which guarantees the fulfillment of the requirements to ensure acceptable values of packet delays and jitter, as well as the absence of packet losses associated with their long waiting in the buffer memory of routers or dropping due to congestion (Polshchykov et al., 2020). However, in this case, packet loss remains possible in MANET due to random changes in the network structure and the presence of interference in wireless channels.

Provided that the level of packet loss is acceptable, high quality of voice stream transmission can be provided based on the use of preliminary reservation of the network channel necessary performance. However, the implementation of channel capacity redundancy required for the transmission of high-quality voice streams cannot guarantee the absence of the following negative phenomena in MANET:

1) the occurrence of a temporary shortage and, on the contrary, an overabundance of channel performance due to their irrational use in the conditions of uneven receipt of requests for audio communication sessions;

2) lack of resources in a network to reserve the performance of channels necessary to satisfy (service) all incoming requests for audio communication sessions;

3) deterioration of audio information stream transmission quality due to unacceptably high packet loss caused by the influence of the dynamic MANET topology and the impact of interference in the wireless environment.

The impact of the first negative phenomenon can be minimized by buffering the requests for audio communication sessions from users. This will make it possible to smooth out irregularities in the time of arrival of these requests, thereby increasing the likelihood of their service and the load of wireless channels in the process of voice stream transmission. It should be borne in mind that buffering a significant number of requests at the same time can cause a significant increase of their waiting duration in queues, and this will invariably lead to too long delays in the start of audio communication sessions. This is confirmed by the results of computational experiments, which are obtained using mathematical models (Konstantinov et al., 2017) and which are shown in Fig. 1.

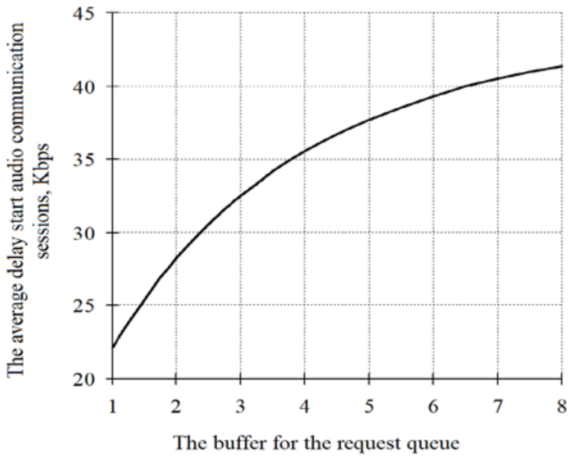

Fig. 1. Dependence of the average delay of audio communication session start on the buffer size for a request queue

Figure 1 shows that with the queue of request length increase for the transmission of voice streams over wireless channels, the average delay also increases during the beginning of audio communication sessions in MANET. A significant delay in the time of important voice information exchange reduces its relevance and increases the risks of non-fulfillment of certain specific management tasks. In this regard, it is advisable to limit the amount of delay in the beginning of audio communication sessions to a certain permissible value that minimizes the above risks. In order to obtain a rational value for the request buffer volume, at which the greatest load of MANET channels is achieved, but at the same time the average delay in the beginning of audio communication sessions does not exceed the permissible value, the development of an appropriate algorithm is required.

It is recommended to counteract the negative phenomenon, which consists in the lack of channel performance required to service as many incoming requests for the transmission of voice streams as possible, by increasing the bit rate of wireless channels rationally. Such a solution will increase the likelihood of request satisfactory service for audio sessions. This statement corresponds to the results of computational experiments, which were obtained using the mathematical model of request servicing for audio communication sessions in MANET (Polshchykov et al., 2018) and are presented on Fig. 2.

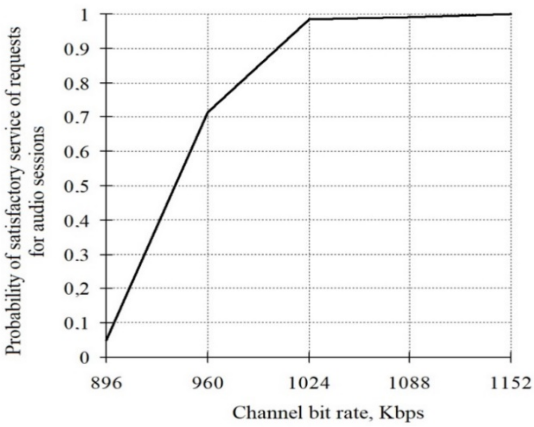

Fig. 2. Dependence of request satisfactory service probability for audio communication sessions on the channel bit rate

The analysis of the Fig. 2 allows us to verify that the probability of satisfactory service of requests for audio sessions in MANET increases with bit rate increase of the wireless channel. However, when designing a wireless self-organizing network, it should be borne in mind that the bit rate value increase will require the power of ad-hoc node module transmission increase. This will entail additional energy consumption, which is extremely undesirable for communication devices with autonomous rechargeable batteries. Therefore, the bit rate of the wireless MANET channels used for the transmission of voice streams should have the smallest values to ensure the required probability of satisfactory service of requests for audio communication sessions. It is necessary to develop an appropriate algorithm in order to calculate the values of the recommended channel bit rate necessary to increase the probability of request servicing for audio communication sessions.

Unfortunately, it is extremely difficult to eliminate the negative phenomenon of packet loss in a wireless self-organizing network. In this case, the longer the audio communication session under conditions of possible packet loss, the greater the likelihood of voice stream transmission quality unacceptable decrease during this session. This is confirmed by the results of computational experiments, which are obtained using the model for the audio information stream transmission in MANET (Polshchykov et al., 2019) and are shown on Fig. 3.

The analysis of the Fig. 3 allows us to make sure that the packet number increase contained in a voice stream and the packet loss probability increase during transmission of a voice stream leads to its high-quality transmission probability increase. In order to calculate the values of the recommended duration of audio communication sessions in MANET, necessary to increase the probability of a high-quality voice stream transmission, it is necessary to develop an appropriate algorithm.

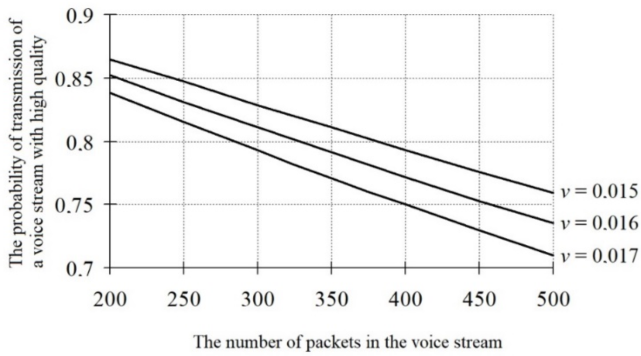

Fig. 3. Dependence of a high-quality stream transmission probability on the number of transmitted packets in a voice stream (v is the probability of packet loss during transmission)

4. Conclusions

Thus, to ensure the effective functioning of MANET in the process of audio communication, it is necessary to create algorithms for estimating the parameters of channels and audio communication sessions, which make it possible to justify decisions on limiting the number of places in the queues of requests for the transmission of voice streams, on the choice of channel bit rate values and on the duration of audio communication sessions. At the same time, the criteria for the effectiveness of MANET functioning are the following:

1) load of network channel increase in the process of voice stream transmission;

2) ensuring an acceptable delay in the beginning of audio communication sessions;

3) achievement of the required values of request satisfactory service probability for audio communication sessions;

4) achieving the required values of voice stream transmission probability with high quality.

In the course of the study, the previously developed models were used to estimate the following values:

1) the average channel load and the delay in the beginning of voice stream transmission, depending on the buffer size concerning the queue of requests for audio communication sessions;

2) the probability of request serving for audio communication sessions depending on the bit rate of the wireless channels;

3) the probability of a high-quality voice stream transmission depending on the number of packets it contains.

Based on the results obtained, it can be argued that the development of algorithms for estimating the parameters of channels and audio communication sessions, which make it possible to justify decision-making to ensure the effective functioning of MANET, is an urgent scientific and technical task. The solution to this problem will be the subject of further research.

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