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ABSTRACT

of the dissertation for the degree of Doctor of Philosophy

IMPROVING SERVICE QUALITY AND RESOURCE EFFICIENCY IN MULTISERVICE TELECOMMUNICATION NETWORKS

Specialty: 3325.01 - Telecommunication technology

Field of science: Technical sciences

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GENERAL CHARACTERISTICS OF THE WORK

The urgency and status of the problem. The main development traditions of modern communication systems and networks, the development of information communication processes has been an objective factor for the emergence of the global information society. The development of technologies, continuous quantitative and qualitative changes of telecommunication networks require the solution of a number of new urgent issues. Among such issues, systematization of the theoretical bases of the known approaches for the design of telecommunication networks and the creation of new methods require analysis and synthesis.

The main task of modern telecommunication networks is to deliver data from the source to one or more users with the desired quality of service. Based on scientific studies, it was found that the application of modern mathematical and optimization methods allows to reduce 20% of all costs during the planning and operation of telecommunication networks. The main factor for improving the quality of service in modern multiservice telecommunication networks is the effective use of network resources based on the agreement of issues of routing, traffic sorting and priority packet processing at network nodes.

For the long-term functioning of telecommunication networks, quality indicators are of particular importance. To increase the performance of telecommunication networks, it is required that the delay parameter (from 200 m/s to 10 sec.) and the probability of packet loss (from 10^{-12} to 10^{-3}) are strictly ensured. The theoretical methods in the development of mass service systems allow to solve many issues related to increasing the quality of work indicators and effective use of resources of various communication tracts, nodes and devices of global multiservice telecommunication networks. Nowadays, various multichannel and multinode priority and multiservice telecommunication networks and switching systems are developed and applied in many countries of the world.

During the design of telecommunication networks (using switching devices, digital communication lines, different transmission media for data transmission, voice and video information, various multiplexing methods), the necessity to take into account the number of parallel channels, the size of buffer memory, the number of nodes of the network and the type of priority service is an undeniable fact.

The available options of models and methods for calculating the indicators of telecommunication systems are not enough to fully apply the structural characteristics of the service systems and the rules for serving different types of traffic.

It is for this reason that the dissertation envisages the realization of a method and algorithm for calculating performance indicators of multi-service telecommunication networks, increasing service quality and effective use of resources.

Mathematical models that take into account the number of channels in network nodes, the number of waiting places and the number of nodes in the network have been developed in the dissertation work, which allow to evaluate the work performance of modern telecommunication networks. The calculations were carried out by obtaining numerical indicators, which are of great importance for the assessment of the work quality of the network and the switching node suitable for engineering use during the design of modern telecommunication networks.

Important quality of service protocols such as Resource reSerVation Protocol (RSVP) - resource reservation protocol, Real-time Transport Protocol (RTP) - real-time transport protocols are applied to combat network overloading and ensure packet streams are served at the required level.

Designing telecommunication networks and ensuring quality of service requires the priority of an actual issue Quality of Service (QoS), which consists in determining the change in the probability-time characteristic of packets (the average time of packet delivery, the determination of the average number of packets in the queue and the probability of loss). The development of methods for determining the change of these indicators in multi-service, limited queue and priority, single-channel and multi-channel telecommunication networks is an urgent issue.

Research object and subject. The object of the research is multi-service telecommunication networks working with priority mode. The subject of the research is analytical models of multi-service telecommunications networks working with priority mode, service quality and increasing the efficiency of resource use.

Research goals and objectives. The purpose of the research is to develop scientific analysis, methodology development, synthesis models and methods aimed at improving the quality of service provided by multi-service telecommunication networks and the effectiveness of their activities. The main issues are the following:

- development of a method that determines the probability-time characteristics (delivery time, number of packets in the queue, probability of loss) of packet streams in multi-service networks with different priority, limited queue, single-channel and multi-channel multi-node;
- development of an algorithm to calculate the optimal values of work quality indicators of single-channel and multi-channel multiservice telecommunication networks with limited queue absolute priority;
- development of a method for calculating the probability of packet loss and the probability of timely delivery in single-channel and multi-channel limited queue absolute priority multiservice telecommunication nodes;
- development of an algorithm and method of optimizing the parameters of the switching node working with the combined priority mode;
- development of an analytical model of multi-node and multi-channel multi-service telecommunication networks and development of an algorithm that allows calculating the quality of the network.

Research methods. Researches were carried out using the provisions of probability theory, mathematical statistics, methods of mass service networks theory, numerical calculation and computer modeling.

Main clauses defended. The main provisions defended are the following:

1. Mathematical models and calculation methods of probability-time characteristics of packet streams in single-channel and multi-channel multi-node multi-service, limited-queue, absolute-priority telecommunication networks, methods of increasing service quality and effective use of resources by determining the number of resources in network nodes and packets queued in buffer memories.

2. Calculation algorithm important for calculating the optimal values of performance indicators of single-channel and multi-channel multi-node multi-service, limited queue, absolute priority telecommunication networks.

3. The method of increasing service quality and resource efficiency by calculating the probability of packet loss and the probability of their timely delivery to the address in single-channel and multi-channel limited queue, absolute priority multiservice telecommunication nodes.

4. A method of improving the quality of service based on the optimization of the parameters of multi-service, priority and limited queue telecommunication nodes operating in different operating modes.

5. Method of selection and effective use of resources of local and global multi-service switching nodes and networks.

Scientific novelty of the research. The scientific innovations of the dissertation work are as follows:

1. A method has been developed that allows determining the probability-time characteristics of packet streams in telecommunication networks with absolute priority, limited queue, single-channel and multi-channel multi-node multi-service.

2. An algorithm was developed to calculate the optimal values of indicators characterizing the work quality of telecommunication networks with absolute priority, limited queue, single-channel and multi-channel multi-node multi-service.

3. An algorithm has been developed to calculate the optimal values of indicators that determine the work quality of single-channel

and multi-channel multi-node absolute priority, limited queue multi-service telecommunication nodes.

4. A method of calculating the probabilities of packet loss and timely delivery to addresses in single-channel absolute priority, limited-queue multiservice telecommunication nodes has been developed.

5. A proposal has been developed to calculate and select the optimal values of the parameters characterizing the activities of multi-service telecommunications nodes working with the combined mode of operation.

Theoretical and practical significance of research. The possibility of choosing one of the methods of effective use of resources during the creation of multi-service telecommunication networks, as well as the proposed calculation methods, allow to determine the delivery times of packets, the number of waiting places and the probabilities of loss of packet streams in single-channel and multi-channel multi-node multi-service systems with limited queue priority and to be able to choose. These methods can be used for both designed and operational communication networks and nodes. The proposed methodology allows determining the optimal number of places in the queues and the optimal number of packets that should wait in the queue, which is an important condition for increasing the quality of service and effective use of resources. The presented calculation methods allow determining the real values of the probabilities of packet loss and timely delivery to the address in real telecommunication systems. The optimization algorithm of the parameters of multi-service switching nodes operating with the combined mode of operation allows to choose the most optimal design method of the telecommunication network. The results of the dissertation work are applied in the educational process and scientific-research processes at AzTU**.**

The degree of honesty of the results. Dissertation work was performed by modeling in modern application software packages. The results of computer modeling confirm the integrity of the research.

Personal contribution of the author. The scientific issues raised in the dissertation work and the main results obtained were obtained directly by the author independently. A method for determining the probability-time characteristics of packet streams in absolute priority, limited queue, single-channel and multi-channel multi-node multi-service telecommunication networks, an algorithm for calculating the optimal values of indicators characterizing the work quality of absolute priority, limited queue, single-channel and multi-channel multi-node multi-service telecommunication networks, single-channel and the algorithm for calculating the optimal values of indicators that determine the work quality of multi-channel multi-node absolute priority, limited-queue multi-service telecommunication nodes, the method of calculating the probabilities of packet loss and timely delivery to addresses in single-channel absolute priority, limited-queue multi-service telecommunication nodes, multi-service telecommunication nodes operating in a combined mode The author proposed to calculate and choose the optimal values of the parameters characterizing their activities performed independently of Afin or with his participation as a responsible executive.

Implementation and application of work results. The results of the dissertation work are applied in the implementation of scientific-research works and in the teaching process at Azerbaijan Technical University.

Dissertation work approval. The main results of the dissertation were discussed in detail at the following conferences: 8th republican scientific conference of graduate students and young researchers held at AzTU, Baku, 2002; International scientific and technical conference held at AzTU, Baku, 2016; International Conference "Communication and Technology Management", Vienna, 2021; International conference on "Modern means of communication" held by the Belarus State Academy of Communications, Minsk, 2021.

The structure and scope of the work. The dissertation consists of 137 pages of computer text in A4 format, including an introduction, three chapters, main results, a list of used literature, appendices, a list of abbreviations and conventional signs. 102 names of literature were used in this work. The number of marks in the dissertation is equal to 221855. The number of signs in the abstract is equal to 37667.

MAIN CONTENT OF THE STUDY

In the introduction, the relevance of the researched topic is substantiated, the main issues of the research are formulated, the main scientific results of the dissertation are explained, the main propositions submitted to the defense are listed, the structure and scope of the dissertation are noted, the practical importance of the research results and areas of application are determined, information is given about the approval of the work and its its brief content has been interpreted.

In the first chapter, the technologies that provide quality services of modern telecommunication networks and the main requirements for them are analyzed. Here, the main methods of building classical telecommunication networks and their analysis were performed. Due to the rapid development in the field of telecommunications and information technologies, it was considered appropriate that the functional model of the new generation multiservice network generally consists of three levels.

These are considered as transport layer, switching management and information delivery layer and service management layers.

A multiservice network should be a universal, multipurpose medium, a network that provides the transmission of speech, images and data using packet switching technology. The option of building a multi-service telecommunication network according to the concept of a new generation communication network has been developed and proposed.

The following requirements are imposed on the new generation multiservice communication network:

- multi-service. This requirement means that services provided independently of transport technology are possible;

- broadband. This requirement makes it possible to dynamically and flexibly change the speed of information transmission in a wide range, depending on the current demand of the user;

- multimedia. This requirement ensures the ability of the network to provide synchronization of all types of information (speech, video, audio, text) in real time and using a complex connection structure;

- intellectuality. This requirement ensures that service providers are able to manage calls and connections by users;

- the presence of a connection option. This requirement means that it is possible to connect to services regardless of the technology used;

- multiperiodism. This requirement involves the participation of several operators in the provision of services, so that their responsibilities within the scope of the service are separated.

Along with these, when the requirements for the next generation networks are defined, the requirements that take into account the characteristics of the activities of the service providers are also set. It is known that there is a flood of calls (information, requests, requests, packets, etc.) entering the network from the load sources that drive communication networks. Calls enter the network at random times, one independently of the other. The order in which calls enter the network is determined according to a certain distribution law.

To express a stream of calls or packets, it is sufficient to know the distribution of the i-th intervals between each packet:

$$A_i(t) = P(\tau \le t),\ i \ge 1 \qquad (1)$$

The expression $A_i(t)$ means that the distribution function in any interval will be equal to or smaller than the interval of two adjacent calls of the time interval *t* between the input moments of two adjacent calls. $A_2(t) = A_3(t) = \cdots = A(t)$, whose distribution functions are the same starting from the second, is called a recurrent stream with limited result. In a recurrent stream with $A_1(t) = A(t)$ if $A(t)$=1-$e^{-\lambda t}$, such a flood is called a Poisson flood.

The Poisson flood is often called the simplest flood. In other words, the stationary ordinal and inconclusive flood is called the simplest flood. For such a flood, the probability of entering *n* calls in the time interval $P_n(t)$ is calculated by the following formula:

$$P_n(t) = W[(\lambda t), n], \qquad (2)$$

where λ is the intensity of the incoming flood.

The fact that the flood is stationary means that the call flood will have a constant value (λ=*const*) in a given interval. For the simplest flow, the mean interval between two neighboring input moments is $\lambda = 1/\tau_{or}$. Here τ_{or} is the average value between two adjacent input moments. In a simple call stream, its parameter coincides with the intensity λ. Calls (data) entering communication networks are serviced at communication nodes. In the simplest case, each call is served by connecting a certain input line of the communication system to the required output line. The connection time of the input and output lines depends on the duration of the information or information provided by the circuit created at this time.

Thus, call service times in communication networks have random values and can logically be explained by the distribution law.

Let us assume that *t* is the call serving time, λ is the call serving intensity (bit/s). The probability function of being served calls can be expressed by the following distribution law:

$$B(t,\lambda) = P\{\gamma < t,\lambda\},\ (t \geq 0,\lambda>0) \tag{3}$$

The function $B(t)$ determines the probability that the service time γ will be less than any given value *t*. It is clear from this that $B(t)$, like all distribution functions, must be a positive monotonically increasing function and must not be greater than unity.

The distribution law of serving time allows to determine the intensity of serving. The average call service time is calculated by the following formula:

$$E[T_{or}(t)] = \int_0^\infty t dB(t), \tag{4}$$

If the service time varies exponentially, $B(t)=1-e^{-\mu(t)}$, where $\mu(t)$ is called the service intensity. Information flows and their service times that can exist in telecommunication networks can be distributed separately by Poisson, Erlang, hyperexponential and constant law. Comparing the Poisson, Erlang, and hyperexponential distributions, it turns out that the Erlang distribution function becomes exponential (Poisson distribution) when k=0, and the Erlang distribution becomes a constant service time distribution function (regular distribution) when $k \to \infty$. The Erlang distribution function becomes very large at large values of *t* than in the case of the Poisson distribution. Since the traffic generated from multiservice services is of a hopping nature in a large time interval, this feature has a rather serious impact on the performance quality indicators of the network. At the present time, the study of the impact of traffic generated from multiservice services on the quality of work of telecommunication networks, their management and calculation is considered one of the urgent issues. In this regard, an engineering calculation that allows calculating the price of information flows entering an arbitrary node of a multi-service network from external load sources and individual nodes of the network, the intensity of information flows that can occur between two nodes of the network, as well as the values of information flows that leave one node of the network and are directed to other nodes formulas have been developed.

In the second chapter, the analysis of technologies providing information exchange in multi-service networks is explained. The main attention was paid to the analysis of protocols that provide quality service in multi-service telecommunication networks.

The standards that determine the form of data presentation and the method of sending, the rules of their change, and the sequence of joint operation of different equipment in the network are called protocols. Protocol means rules of interaction. A network protocol defines the rules of operation of computers connected to it. Standard protocols determine how different computers “speak the same language”. When different types of computers are connected to the Internet, certain operations are performed, which are controlled with the help of different operating systems.

There are several layers of protocols that work with each other in a network. At the low level, two basic protocols are used. One of them is called Internet Protocol (IP) - internet protocol, and the other is called Transmission Control Protocol (TCP) - the protocol that controls the transmission. Because both protocols are closely related to each other, they are often combined and called TCP/IP. TCP/IP is used as the base protocol on the Internet. All other multiple protocols are based on TCP/IP. The scheme of operation of the TCP/IP protocol in the Internet network is described in Figure 1. Here, for example, the word “Congratulations” is divided into packets and delivered to the address schematically.

The TCP protocol divides information into portions (parts or packets) during transmission and numbers all parts. Then, with the help of the IP protocol, those packets are sent to the required, that is, mandatory destination address via the Internet network. Packages are delivered to the required address through the Internet network in various ways. At this time, some of them reach the address with a delay, and the sequence of their reception is disturbed. At the receiving station, these packages are placed sequentially according to their numbers through the technical device, as required by the protocol, and arranged in the order of their original form. All these operations are performed by the TCP protocol. For the TCP protocol, it doesn't matter how the packets reached the address. The IP protocol deals with this issue. The IP protocol service adds information to each packet. Based on this service information, the address that sent the package and received it is known. If we analyze the delivery of packages to addresses in accordance with postal communication, we will see that packages are placed in an envelope and an address is written on it, entered into the network and sent to the address (Figure 1). In Figure 1, the data source (MM) is shown as the data recipient (MA).

The IP protocol ensures that all packets are delivered to addresses, just like regular mail. In this case, the shipping speed and route of individual parts may vary.

If individual envelopes have received certain distortions on their way and have been received incorrectly, it is ensured that they are re-received several times through the request.

The intensity of different types of traffic is increasingly different. Fast data transmission of all traffic has been increasing for several years now. This type of traffic surpassed the volume of speech traffic a few years ago, and the rate of growth of this type of traffic continues even now. Therefore, the future network will be defined by a fast and efficient data transfer network.

This is ensured by the IP protocol. This protocol ensures the rapid development of the Internet. This protocol also ensures the development of networks at lower levels in all possible variants. In this chapter, the operation of the converged IP network was analyzed and it was noted that the requirements imposed on the converged IP network: combating overload, reducing delay, ensuring high throughput and fair service are considered to be the main tasks of this network. At the same time, this chapter provides an explanation of integrated (Integrated Services, IS) and differentiated (Differentiated Services, DS) services.

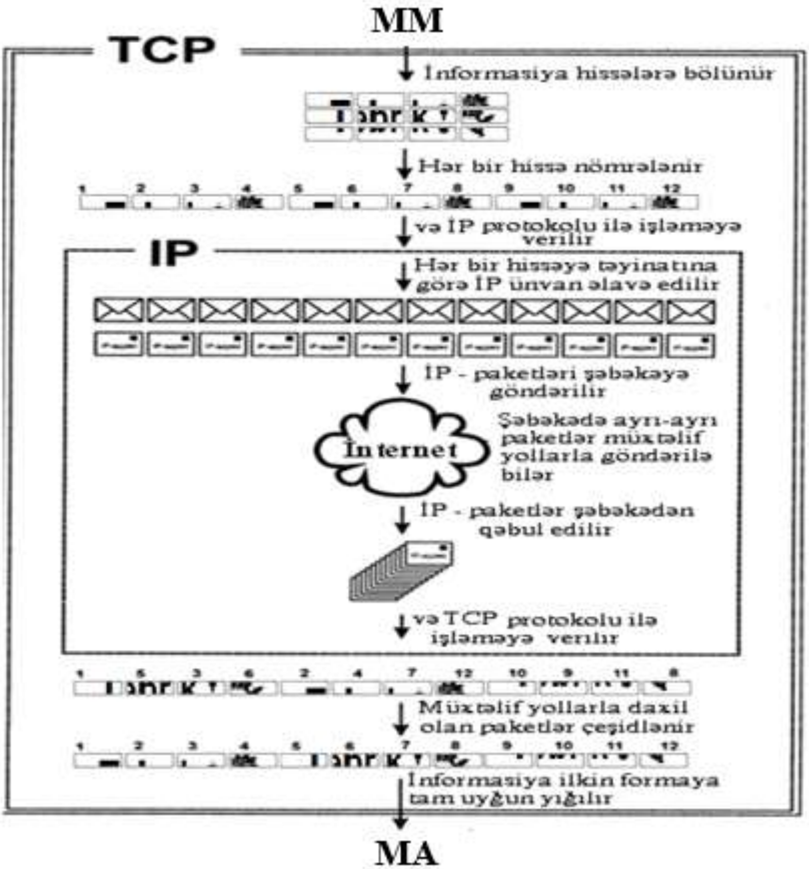

Figure 1. Working diagram of TCP/IP protocol

Resource Reservation Protocol (RSVP) - resource reservation protocol, Multi Protocol Label Switching (MPLS) - multi-protocol switching according to label and Real-time Transport Protocol (RTR) - real-time transport protocol protocol were analyzed.

In modern multi-service telecommunication networks equipped with the latest technologies, voice, text, video and other types of information are delivered to the required addresses in the form of digital packet streams, in various ways, in the form of electrical or optical signals. In such a network operating in packet switching mode, each information (data) converted into binary number form is divided into packets and delivered to the required address. Each packet, along with the data it carries, has a header and a field for writing some service information, which is placed either at the beginning or at the end of the packet. The division of data into packets is shown schematically in Figure 2.

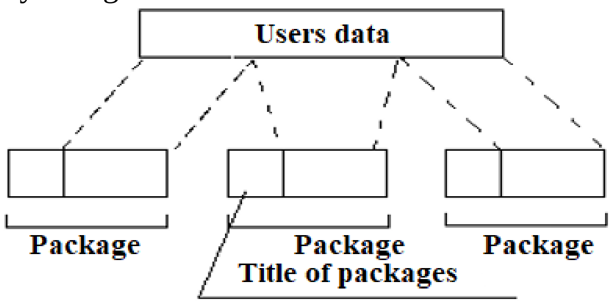

Figure 2. Schematic representation of the division of data into packets

In such a mode, each packet is transported through the network independently of the packet before and after it. When forwarding a packet, each node in the network selects the next node to retransmit the packet based on the packet header and information about the surrounding nodes. In this case, different packets that will reach the same final address can be transported from the sending node to the receiving node in different ways. Packets are restored in the correct sequence either at the end receiving node or at the receiving address, and the transmission of packets in this way is called datagram mode.

Asynchronous Transfer Mode (ATM) - an asynchronous mode of data transmission network is sometimes described as a retransmission of slots, where the information flow is described in the form of fixed size packets called cells in each logical connection. In the ATM network, 53-byte cells are used, of which 5 bytes are the header and 48 bytes are the information field. The architecture of protocols developed by ITU-T for ATM is illustrated in Figure 3.

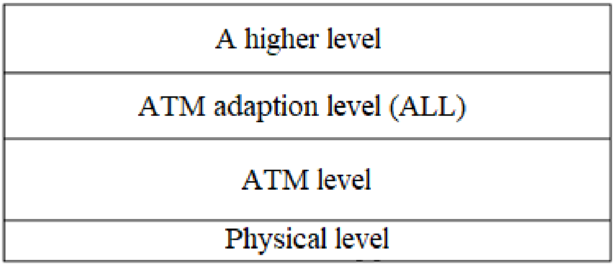

Figure 3. The architecture of ATM protocols

The transmission speed specified in this standard is 155,52 Mbit/s and 622,68 Mbit/s. Also, this protocol can work at higher and lower transmission speeds. The ATM layer ensures the transmission of a data stream, thanks to the creation of logical connections of slots with a specific length. Here, the AAL sublayer is used to translate high-level information into ATM slots so that it can be transmitted over the ATM network. Then, the information from the ATM slots is collected and transferred to a higher level.

In the third chapter, the issues of increasing the service quality and resource efficiency of the multi-service telecommunications system were studied, and an analytical model of the queue formed in front of a server was developed. Packet reception rate (λ) and service time (T_s) were used as input information for the queue. The purpose of queue analysis is to get the following information at the output according to such input information: the number of waiting packets (w), the waiting time (T_w), the number of packets in the system (r), the time the packet is in the system (T_r). First of all, we need to know their average values (w, T_w, r, T_r). The calculation of the main indicators of a single-channel multiservice communication system working with limited waiting and priority mode is based on the following conditions based on Two streams of packets with respective intensities λ_1 and λ_2 enter the input of the system. The frequency of flooding is determined by the Poisson distribution. The distribution function of service times for the first and second priority streams is determined by the exponential law. In this case, the load created by the first and second priority currents is ρ_1, ρ_2, respectively. Numerical reports are performed with analytical models to calculate the average number of first-priority packets waiting in the queue, the average number of second-priority packets waiting, the average duration of the first-priority packet in the service system, and the average duration of the second-priority packet in the service system.

The average delay time of packets varies dramatically depending on the variation of the distribution law of service times. The largest lag time occurs in the hyperexponential distribution, and the smallest lag occurs in the regular (stationary) distribution law.

Calculations show that the quality of system work and effective use of resources depend on the average delivery time of packets, as well as the load created by packets and the buffer memory allocated for waiting packets.

The analysis of the obtained curves shows that high-priority packets are served faster than low-priority packets (Figure 4).

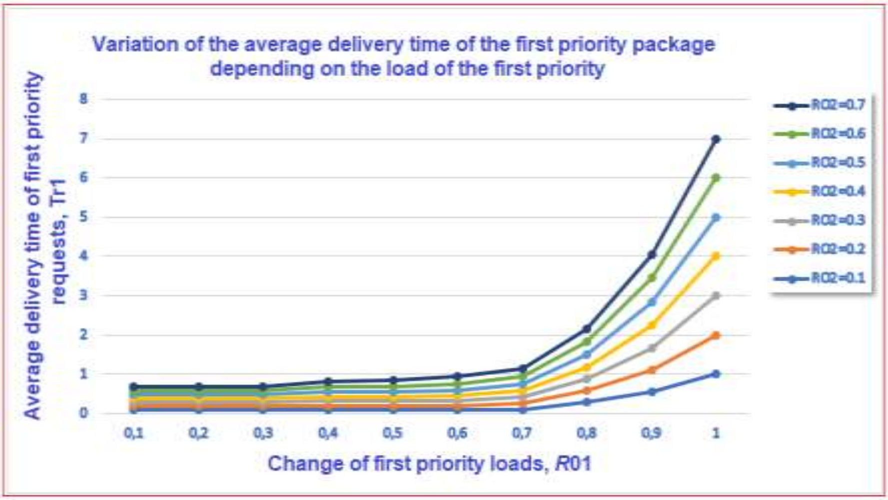

Figure 4. Variation curves of the average transportation (delivery) time (T_{r1}) of the first priority package depending on the load of the first priority (ρ_1)

Figure 5 depicts the graph corresponding to the loss probabilities depending on the load change (ρ) caused by the first-priority packets at different values (k=4-16) of the first-priority packet waiting places in a single-node single-channel network (N=1).

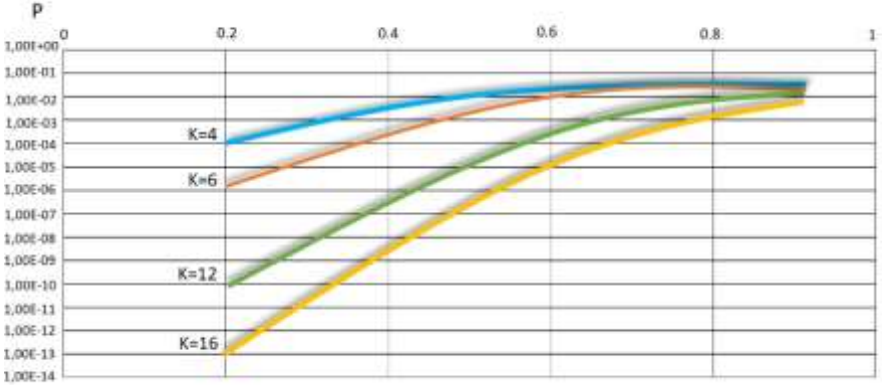

Figure 5. The graph of the probability of loss depending on the load change (ρ) caused by the first-priority packets at different values of waiting places (k=4-16)

The multi-channel and priority mass service system model was used to increase the service quality and effective use of resources of the multi-service communication node serving speech and data packets. A system of equations determining the state of the node was developed and various options of work quality indicators depending on its incoming traffic, resources, loss rate were calculated and analyzed, and an option was created to increase its efficiency.

In order to increase the work quality and effective use of resources of a multi-service telecommunication network with a complex structure, studies were conducted using its analytical model. Many literatures have used the provisions of probability theory and mathematical statistics to obtain a mathematical model of a multi-node system.

Mathematical models proposed in the mentioned literature were used for the study of multi-node networks.

The mathematical model for calculating the probability of loss in multi-channel and multi-node networks, since there is a limited queue in the service for first-priority packets, is as follows:

$$P_{i1} = \sum_{i=1}^{N} \frac{\rho_{i1}^{s_i+k_i}}{s_i^{k_i} s_i!} \left[\sum_{\nu=0}^{s_i} \frac{\rho_{i1}^{\nu}}{\nu_i!} + \frac{\rho_{i1}^{s}}{s_i!} \sum_{j=1}^{k_i} \left(\frac{\rho_{i1}}{s_i} \right)^j \right]^{-1}, \quad (5)$$

where N is the number of nodes in the network; ρ_{i1} - first priority load; s_i - number of channels; k_i is the number of queues at node i of the network.

The mathematical model for calculating the probability of denial in multi-channel and multi-node networks is as follows:

$$P_{i2} = \sum_{i=1}^{N} \frac{\left(\rho_{i1} + \rho_{i2}\right)^{s_i+k_i}}{s_i^{k_i} s_i!} \left[\sum_{\nu=0}^{s_i} \frac{\left(\rho_{i1} + \rho_{i2}\right)^{\nu}}{\nu!} + \frac{\left(\rho_{i1} + \rho_{i2}\right)^{s_i}}{s_i!} \sum_{j=1}^{k_i} \left(\frac{\rho_{i1} + \rho_{i2}}{s_i} \right)^j \right]^{-1}, (6)$$

where ρ_{i2} is the load of the second priority of the *i*-th node of the network.

An algorithm has been developed to calculate the probability of (loss) denial for packets of first and second priority in multi-channel and multi-node networks, taking into account the existence of a limited queue in the service. The numerical calculation of the probability of loss during the service of the first and second priority packets was performed. At this time, when calculating the probability of loss in multi-node networks, the rejection probabilities of the nodes are assumed to be the same. Thus, when designing a given communication network, it is necessary to take into account the real value of the probability of loss at the designed nodes of the network (Figure 6).

In Figure 6, the number of nodes is N=2, the load created by the second priority packets is $\rho_2 = 0{,}2$, the number of active channels in the first node is S_1=3. The number of channels in the second node reflects S_2=4 and the number of waiting places (buffer memories) k=4-10. It can be seen from the obtained characteristics that in order to increase the required quality of service and the effective use of resources, the number of parallel channels required between the nodes of the network can be adjusted by determining the optimal number of waiting places (memory capacity) in the corresponding communication nodes.

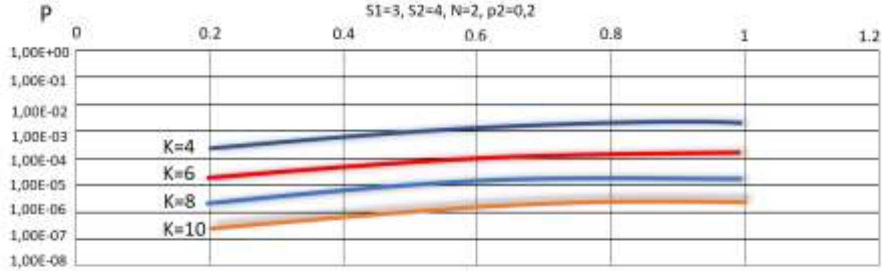

Figure 6. Graph of the dependence of the number of channels of the probability of loss on the number of waiting places in multi-node networks

The economic loss function was used to optimize the quality of service indicators and resources of a multiservice switching node operating in limited standby mode:

$$G(s)=[q_{n\kappa}(s-M_m)+q_{g\ddot{o}}.M_{g\ddot{o}z}+q_y.W_{(s+k)}.\lambda+q_k.M_m].T \qquad (7)$$

In this function, the probability of busyness of all serving communication channels and all available waiting places in the system, the average number of packets waiting in the queue, the intensity of the incoming packet stream, the cost of the communication channel being idle at one time, the cost of the packet waiting in the queue at one time cost and the economic value or income obtained through the use of a communication channel at the same time is not reflected.

Figure 7 shows the graphs of the economic loss value depending on the load for the number of waiting places k=4 and different values of the serving channels.

It can be seen from Figure 7 that at the same value of load ρ =4, s=1 and k=4 when G_{min}=1,4 man/min., when s=2, G_{min}= 1,04 man/min., when s=3, G_{min}=0,84 man/min., s=4, G_{min}=0,4 man/min. happens. It is clear from this that as the number of channels increases, some of them have to be idle. This causes the value function (revenues) to decrease.

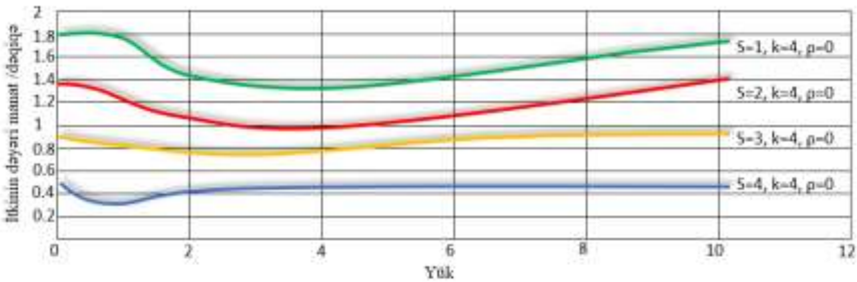

Figure 7. Graphs of changes of the economic loss value depending on the load for the number of waiting places *k*=4 and different values of the serving channels

By obtaining the optimal operating mode and the optimal number of resources of the telecommuting node working with the limited standby mode, the presented methodology can be used to increase their quality indicators.

MAIN RESULTS

1. The analysis of the technologies that ensure the quality operation of modern multi-service telecommunication networks shows that these protocols as a whole ensure the timely delivery of packet streams to their addresses and initially requires the calculation of the characteristics of the quality indicators of the telecommunication network. Analyzes show that due to the joint use of the advantages of ATM and IP technologies, it gives the best results in combating overload, reducing delays, ensuring high throughput and ensuring the quality of serving ordinary priority packets.

2. Increasing the number of waiting places in single-channel non-priority networks with limited waiting does not cause the probability of packet loss at low load values, while the probability of packet loss at the same values of waiting places increases significantly when the load increases.

3. Based on the analytical model of the one-channel multi-node telecommunication network with limited queue absolute priority, which depends on the number of nodes, the number of waiting places at individual nodes, and the loads created by the first and second priority packets, a report of the important probabilistic time characteristics of the system was made.

4. Algorithms have been developed to calculate the probabilities of loss during the service of the first and second priority packets. Calculations were made by means of the developed packet programs and the curves of packet loss probabilities depending on different values of loads, the number of nodes in the network and the number of waiting places in the nodes were constructed. It was determined that the quality of packet service deteriorates significantly as the number of nodes at the same price increases in single-channel multi-node, limited-queue and absolute-priority networks. To ensure the desired quality of service in such networks, it is important to increase the number of waiting places at individual nodes. The analysis of single-channel multi-node, limited queue and absolute priority networks has shown that the quality of serving packets in such networks varies depending on the loads created by packets with different priorities. The conducted numerical calculations and established graphical dependences allow to determine the required service quality of telecommunication networks with known values of the traffic created by the first and second priority packets with a known number of nodes and a known number of waiting places at each specific node.

5. An analytical model of the multi-channel multi-node telecommunication network with limited queue absolute priority, which depends on the number of nodes, the number of waiting places in individual nodes, and the loads generated by the first and second priority packets, has been developed. It is shown that in order to ensure the required quality of work in multi-node multi-channel telecommunication networks, it is important to know the optimal number of service communication channels (S), the number of waiting places in nodes (K) and the traffic generated in individual nodes ρ_1, ρ_2. It was determined that in order to ensure the normal operation of the network, as the number of nodes in the network increases (N), the

number of serving channels (S) and the number of waiting places in the nodes (K) should be increased accordingly. Derived schedules and functional dependencies ensure the design of telecommunication networks with the required quality of service.

6. When designing telecommunication nodes working with the combined mode, it is appropriate to calculate the minimum values of economic value loss depending on the specific price of traffic, in order to optimize the number of communication channels and the number of waiting places required at the node.

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Personal participation of the author in scientific works published with co-authors:

- [1, 2, 12, 14, 15] - works were performed independently by the author;
- [3-11, 13] - method that allows determining the probability-time characteristics of packet streams in absolute priority, limited queue, single-channel and multi-channel multi-node multi-service telecommunication networks, to calculate the optimal values of indicators characterizing the work quality of absolute priority, limited queue, single-channel and multi-channel multi-node multi-service telecommunication networks algorithm for calculating the optimal values of the indicators determining the work quality of single-channel and multi-channel multi-node absolute priority, limited-queue multi-service telecommunication nodes, a method for calculating the probabilities of packet loss and timely delivery to addresses in single-channel absolute priority, limited-queue multi-service telecommunication nodes, with a combined mode of operation single to calculate and select the optimal values of the parameters characterizing the activities of working multiservice telecommunications nodes fiber performed by the author.

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