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METHOD OF ANALYSIS OF QUALITY INDICATORS OF COMPUTER NETWORK OF INFORMATION SYSTEM OF CRITICAL APPLICATION

Abstract. Topicality. Our time is characterized by rapid growth in the number and quality of various computer systems. An increasing number of users, both in their daily and professional activities, need data transmission and Internet access services in addition to traditional communication services. The network infrastructure created as a result of convergence should provide the needs of preferably all service users. But in order to ensure the necessary quality of service, it is necessary not only to have information about the state of the telecommunications network, but also to be able to forecast it. This fact requires improving the analysis of quality indicators characterizing the state of the telecommunications network. **The purpose of this work** is to develop a method of analyzing quality indicators of a computer network of a critical application information system**. The object of the research** is the process of functioning of the computer network of the information system of critical application. **The subject of the study** is the methods of analyzing the quality indicators of the computer network of the information system. **The results**. In this work, an analysis of the main indicators of the quality of the telecommunications network, which are basic in the provision of services. **Based on the received** information, 6 states of the telecommunication network were formalized and classified, in which it is possible to provide non-essential quality of service for various types of traffic. The developed method of analyzing the quality indicators of the computer network of the information system of critical application is planned to be used in the future to ensure the quality and reliability of the functioning of computer systems of critical application.

Keywords: computer systems, computer network, critical application information system, traffic, quality indicators.

Introduction

Formulation of the problem. The modern period of development of the information society corresponds to an ever-increasing increase in demand for infocommunication services. These services are provided by a variety of computer systems, including critical ones. An increasing number of users, both in everyday and professional activities, along with traditional communication services, require data transmission and Internet access services. Today, it is generally accepted that circuit-switched and packet-switched networks are gradually evolving towards the creation of a common infrastructure. The infrastructure that emerged as a result of convergence must ensure the transport of telephone network traffic, television networks, and application traffic that traditionally uses Internet networks. Such a convergence scenario offers both economic gains from the convergence of technologies and determines the development of the telecommunications sector through the creation of new services. However, the process of convergence has so far proceeded rather slowly. And here one of the dominant ones is the problem of providing the necessary quality of service, which is one of the main inhibitory factors in the process of convergence of networks and services. In addition to the standard set of services that are currently provided by telecommunications networks, the following services can be distinguished: voice transmission; video calls; video conference; data transfer; Internet surfing; voice transmission for corporate subscribers.

In this regard, new requirements for the quality of service are put forward, involving the solution of the following tasks:

– prioritization and differentiation of traffic;
– providing information flows with the necessary network resources;
– improving transmission reliability;
– network congestion prevention;
– shaping network traffic for smoothing and creating a more uniform flow.

Different types of traffic have different network resource requirements and require different classes of service. In this regard, to ensure the required quality of service, it is necessary not only to have information about the state of the telecommunications network, but also to be able to predict it.

From the foregoing, we can conclude that the task of analyzing quality indicators that characterize the state of a telecommunications network and classifying the states of a telecommunications network is **relevant**.

Analysis of recent research and publications. The increasing rate of use of new information technologies leads to an increase in the number of services provided by the telecommunications network. According to studies [1], the dynamics of Internet traffic growth over the past ten years is 70–150% per year, i.e. On average, every year the amount of information transmitted over the network doubles.

Existing studies show that network technologies are far ahead of theoretical and analytical understanding of network interactions in their growth. Highly specialized and limited telecommunication problems of the past years are well studied and mathematically formalized, in particular, based on the provisions of the queuing theory [2]. Attempts to formalize and classify the states of a telecommunications network were carried out in [3, 4]. However, the traditional principles of theoretical analysis and existing methods do not meet the necessary requirements of the modern period of development of telecommunications. To date, information about the state of the network is represented by a set of values of quality indicators. This set of quality indicators is not standardized. For various reasons, the problem of classifying the states of a telecommunications network is currently not well understood. There is currently no single approach to grading the states of a telecommunications network. All this leads to the fact that providers evaluate their capabilities differently, which in turn leads to an unjustified decrease in the quality of service.

The purpose of this work is to analyze the quality indicators used to assess the state of a telecommunications network, formalize and classify the states of a telecommunication network, as well as analyze the requirements of various types of traffic to network resources.

Main part

The state of the telecommunications network is characterized by a certain set of quality indicators. According to ITU-T Y.1540 [5] and ITU-T Y.1541 [6], it is possible to identify the main indicators of the quality of the network, which are basic in the provision of services: number of lost IP packets (IPLR); number of packets with errors (IPER); delay (IPTD); jitter (IPDV).

Number of lost IP packets (IPLR). Packet loss occurs when the value of delays in the transmission of packets exceeds the normalized value T_{max}. The main reason for packet loss is the growth of queues in network nodes that occurs during congestion. The IPLR parameter for a certain time interval t is defined as follows:

$$IPLR = \sum_t LP \Big/ \sum_t RP \,, \qquad (1)$$

where LP − is the number of lost packets; RP − number of received packets.

Number of packets with errors. The IPER parameter for a certain period of time t is defined as follows:

$$\mathrm{IPER} = \sum_t \mathrm{RPE} \Big/ \sum_t \mathrm{RPS} + \mathrm{RPE} \,, \qquad (2)$$

where RPE − number of packets received with errors; RPS − number of successfully received packets.

IPER depends mainly on the data transmission systems used at the physical layer of the network. The use of cables with optical fibers makes it possible to achieve high reliability of the transmitted information.

Delay (IPTD). The IPTD parameter is defined as the time (t2 - t1) between two events - the packet entering the network input point at time t1 and the packet output from the network output point at time t2, where (t2 > t1) and (t2 > t1) <= Tmax, where Tmax is the maximum delay for various applications. IPTD includes the following components:

$$IPTD = T_{пак} + T_{среды} + T_{сети} + T_{буф}, \qquad (3)$$

where T_{pack} is the time to form a packet, depends on the type of traffic; T_{medium} - signal propagation in the transmission medium, does not depend on the type of traffic; $T_{networks}$ - transportation over a packet network (processing in network nodes), depends on the type of traffic; the most unpredictable and quality-reducing component of IPTD; T_{buf} - delay in the receive buffer, depends on the type of traffic, service discipline, traffic priority (SLA)

In general, the IPTD parameter is defined as the packet delivery time between source and destination for all packets, both successfully transmitted and those affected by errors. The average packet delivery delay is defined as the arithmetic mean of packet delays in a selected set of transmitted and received packets. The value of the average delay depends on the traffic transmitted in the network and the available network resources, in particular, on the bandwidth. An increase in load and a decrease in available network resources lead to an increase in queues at network nodes and, as a result, to an increase in average packet delivery delays. Voice information and partly video information are examples of traffic sensitive to delays, while data applications are generally less sensitive to delays. When the packet delivery delay exceeds certain Tmax values, such packets are dropped. In real-time applications (such as IP telephony), this leads to poor voice quality. The limitations associated with the average latency of IP packets play a key role in the successful implementation of Voice over IP (VoIP), video conferencing, and other real-time applications. This parameter largely determines the willingness of users to accept such applications.

Jitter (IPDV). This parameter began to be tracked quite recently, it is of the greatest importance for multimedia traffic. IPDV manifests itself in the fact that successive packets arrive at the recipient at irregular times. In IP telephony systems, this leads to sound distortion, and as a result, speech becomes unintelligible. Currently, there are several approaches for the numerical estimation of the parameter IPDV:

$$IPDV = X_k - d_{1,2}, \qquad (4)$$

where X_k the absolute value of the delay in the delivery of a packet with index k; $d_{1,2}$ is the reference value of the packet delivery delay for the same network points (input and output), defined as the absolute value of the first packet delivery delay between these network points;

$$IPDV = \mathrm{mean}\ (\mathrm{abs}\ (t_i - t_{i-1})); \qquad (5)$$

$$IPDV = \mathrm{mean}\ (\mathrm{abs}\ (t_i - a_i)), \qquad (6)$$

where t_i is the network delay of the i-th packet; a_i is the time of receipt of the i-th packet;

$$IPDV = \mathrm{mean}(P_i) + \mathrm{mean}(N_i), \qquad (7)$$

$$Pi = t_i - D_i \text{ at } t_i > D_i,\ Ni = D_i - t_i \text{ at } t_i < D_i,$$

where D_i – average delay estimate/

Based on the recommendation ITU-T Y.1540 [5], depending on the accepted values of the performance parameters of a telecommunications network, 6 classes of service can be distinguished (Table 1).

Table 1 – **Compliance of classes of service with performance parameters of a telecommunications network**

Param.	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5
IPTD	120мс	450мс	120мс	450мс	1с	–
IPDV	75мс	75мс	–	–	–	–
IPLR	0,001	0,001	0,001	0,001	0,001	–
IPER	0,0001	0,0001	0,0001	0,0001	0,0001	–

Based on the proposed classes of service, it is possible to consider the requirements of the main types of traffic to network resources and identify the states of the telecommunications network under which it is possible to provide the required quality of service for various types of traffic. The main types of traffic encountered in telecommunication networks are: interactive audio and video; streaming video; streaming audio; interactive data traffic; game traffic; administrative traffic; the rest of the traffic. The sensitivity of various types of traffic to the main indicators of the quality of the network, which are basic in the provision of services, is presented in Table 2.

Table 2 – **Sensitivity of different types of traffic to network characteristics**

Type traffic	Sensitivity level		
	Losses	*Delay*	*Jitter*
Voice	Medium	High	High
E-commerce	High	High	Low
Transactions	High	High	Low
Email	High	Low	Low
Telnet	High	Medium	Low
Web search	Medium й	Medium	Low
Constant web search	High	High	Low
File transfer	Medium	Low	Low
Video conference	Medium й	High	High
multicasting	High	High	High

Interactive audio and video. This type of service is mainly defined by telephone calls (VoIP), low-resolution audio and video conferences. According to studies [7], [8] and standards [9], [10], one-way delay should not exceed 150ms, jitter 30-50ms, since exceeding this threshold greatly affects the quality of speech perception. At delay values less than 100ms, service users do not notice it. Packet loss values should not exceed the threshold of 1 - 5% depending on the codec. The class of service that best meets these requirements according to [5] is 0.

Streaming video. First of all, these are such services as television broadcasting, video on demand (Video on Demand). The following general requirements are true for them [6]: good video quality; high readiness; average interactivity - determined for the reverse flow (from the user).

These requirements must be translated into values for the requirements for the transport of data over a telecommunications network. Due to the large size of a single stream (from 3.5 Mbps for MPEG-2 and from 2 Mbps for H.264) and the coding principles used, there are increased requirements for packet loss.

From recommendation J.241 [6] for excellent quality of service (ESQ), the IPLR value should be minimal. It should also be noted that the delay parameter can vary greatly depending on the video streaming service. In particular, for video on demand, “network video recorder” and services of similar architecture, these requirements increase, as more interactivity is required.

Since the latter services are more demanding on the performance parameters of telecommunication networks, it makes sense to determine the threshold value for these services as well. According to Addendum 3 dated 05.2008. to recommendation Y.1541 [11], the values of IPTD, IPDV, IPLR, IPER correspond to service class 5 of the recommendation itself. Studies [12, 13] confirm the values indicated in [14].

Streaming audio. First of all, these are such services as radio broadcasting, audio on demand (Music on Demand). For this type of traffic, the values of the performance parameters of the telecommunications network according to [5, 15] fit into classes 0, 1. For interactive services (Music on Demand), the service class is 0, for broadcasting services, the service class is 1

Interactive data traffic. This class of traffic should include web surfing, telnet, ssh, interactive messaging (for example, chat) [14]. The class of this traffic is 2.

Game traffic. The values of the performance parameters of a telecommunications network according to studies [16 − 18] correspond to service class 0, which contradicts the recommendation [19]. The delay value for good and excellent quality of service should not exceed 100 ms, the jitter value should be less than 50 ms, and the packet loss rate should remain below 1%.

The rest of the traffic. As a rule, this is the traffic of file sharing, mail services and other less important services. Service class - 4. Depending on the type of traffic, 6 states of the telecommunications network can be distinguished. In all states of the telecommunications network, it is possible to transfer both traffic specialized for a given state and traffic with a lower priority. The state of the telecommunication network "0" is intended for real-time information exchange (in particular, for speech using IP technology). It provides for the creation of a separate queue with priority processing of packets. The "0" state is characterized by restrictions on the principles of routing (the maximum number of transits) and the allowable distance between interacting terminals (the propagation time of signals). Interactivity (probability of using interactive mode) for state "0" is defined as "high".

The state of the telecommunications network "1" is also intended for real-time information exchange, but with less stringent requirements. Therefore, less significant restrictions are imposed on the principles of routing and the propagation time of signals than for the "0" state. It also provides for the creation of a separate queue with priority processing of packets.

The state of the telecommunications network "2" is focused on data exchange with a high degree of interactivity. As well as the state "0", the level of high interactivity is assigned. In this state, in particular, signaling information is transmitted. State "2" is characterized by the same restrictions on the principles of routing and propagation time of signals as for state "0". For packets in this state, their own processing queue is formed, which is carried out with the second priority. This means that packets in the telecom network state "0" and "1" have priority for processing.

Telecommunication network state "3" is intended for exchange with a lower level of interactivity, it has the same restrictions on the principles of routing and signal propagation time as state "1". Packets in this state should be serviced with second priority. This state is considered acceptable for interactive communication.

Telecom network state "4" is intended for the exchange of various information with a low probability of loss (short transactions, streaming video, etc.). Long queues of packets are allowed for processing, which is carried out with the second priority. There are no restrictions on the routing and delivery time of messages.

Telecom network state "5" is oriented towards those IP applications that do not require high quality of service. The corresponding packets form a separate queue; service is carried out with the lowest priority (in this case, it has the third number). There are no restrictions on the routing and delivery time of messages.

Conclusions

An analysis was made of the main indicators of the quality of the telecommunications network, which are basic in the provision of services, namely, the number of lost IP packets (IPLR), the number of packets with errors (IPER), delay (IPTD), jitter (IPDV). Depending on the accepted values of the performance parameters of the telecommunications network, based on the recommendation of ITU-T Y.1540, 6 classes of service have been identified. The main types of traffic encountered in telecommunication networks are considered, an analysis of the requirements of various types of traffic to network resources is carried out. Based on the information received, 6 states of the telecommunication network are formalized and classified, under which it is possible to provide the required quality of service for various types of traffic.

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Метод аналізу показників якості комп'ютерної мережі інформаційної системи критичного застосування

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Анотація. Актуальність. Наш час характеризується стрімкім зростанням кількості та якості різноманітних комп'ютерних систем. Все більшій кількості користувачів, як у повсякденній, так і в професійній діяльності, поряд з традиційними послугами зв'язку потрібні послуги передачі даних та доступу до Інтернету. Мережева інфраструктура, що виникла в результаті конвергенції, повинна забезпечувати потреби бажано всіх споживачів послуг. Але для забезпечення необхідної якості обслуговування необхідно не тільки мати інформацію про стан телекомунікаційної мережі, але і вміти його прогнозувати.. Цей факт потребує покращення аналізу показників якості, що характеризують стану телекомунікаційної мережі. **Метою даної роботи** є розробка методу аналізу показників якості комп'ютерної мережі інформаційної системи критичного застосування**. Об'єктом дослідження** є процес функціонування комп'ютерної мережі інформаційної системи критичного застосування. **Предметом дослідження** є методи аналізу показників якості комп'ютерної мережі інформаційної системи. **Результати.** У даній роботі проведено аналіз основних показників якості роботи телекомунікаційної мережі, які є базовими при наданні послуг.. Розглянуто основні типи трафіку, що зустрічаються в телекомунікаційних мережах, проведено аналіз вимог різного виду трафіку до ресурсів мережі.. **Висновок.** На основі отриманої інформації формалізовано та класифіковано 6 станів телекомунікаційної мережі, при яких можливе забезпечення необхідної якості обслуговування для різних видів трафіку. Розроблений метод аналізу показників якості комп'ютерної мережі інформаційної системи критичного застосування планується використовувати у подальшому для забезпечення якості та надійності функціонування комп'ютерних систем критичного застосування.

Ключові слова: комп'ютерні системи, комп'ютерної мережі, інформаційна система критичного застосування, трафік, показник якості.