NETWORK TRAFFIC RESEARCH BETWEEN LAN AND ISP BASED ON OPNET MODELER AND WIRESHARK APPLICATION PROGRAMS

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ABSTRACT

Today, a person and a data processing system need quick access to any information on the Internet. To satisfy these needs, more and more Internet Service Providers are appearing on the market. Accordingly, any LAN should receive stable network traffic with minimal delays. The article is devoted to the study of traffic in the network between the LAN and the provider. The network under study was developed on OPNET Modeler with a complete step-by-step description. The logical parameters of the communication line are calculated: throughput and quality of service (QoS) depending on the codec used for voice transmission. The calculation of the information flow to the Internet was made, which allows identifying the shortcomings associated with the bandwidth and congestion of the channel. The congestion of the channel at the slightest increase in load can cause users large delays and packet loss, which will not allow users to work efficiently and use all the possibilities of accessing the Internet. To check the reliability of the network structure, a passive attack was carried out on the developed model using the WireShark program and the network security parameters between the LAN and the ISP were determined using the Net Doctor Module of the OPNET Modeler program. Link analysis between LAN and ISP solves the issues of optimal functional connectivity and productive user experience by analyzing passing traffic, analyzing and investigating the physical connection of the network to the ISP and further blocking unwanted load.

Keywords: *LAN, ISP (Internet service provider), OPNET Modeler, Wireshark.*

1. INTRODUCTION

The rapid growth in the number of local area networks, as well as networks of Internet providers, the rapid development in wired and wireless communications, the continuous emergence of new network technologies aimed at improving the speed and reliability of the network makes us think about the analysis and study of the communication network and improving the flow of traffic between different parts of the network. This direction is relevant and attempts to solve optimization problems are presented in unrepresented publications.

The article [1] is devoted to the characteristics of a local area network (LAN) and Internet equipment that is located between them and the speed of information transmission. At the same time, it is noted that the transfer rate between them can be large and equal to 1000 Mbit/s.

The article [2] is dedicated to the development of Internet service providers' points of presence. In this paper, a study is conducted that all Internet service providers are connected to each other through access points to the network, as well as through public network facilities on the Internet backbone. It was the growth of commercial Internet services and applications that contributed to the rapid commercialization of the Internet. An analysis of the functioning of Internet service providers showed that the number of LAN began to grow rapidly to 2.1 million in 2015, moreover, they no longer sought to become the leading providers of dial-up access services. Instead, I tried to become a free Internet portal.

The article [3] is dedicated to the fact that Cisco Internetwork Operating System (IOS) can support many different WAN protocols that can help expand LAN and other local area networks.

Connecting the websites of one company to exchange information is a prerequisite for the modern economy. However, in order to lay your own cable or connections to connect all the remote offices of the company, you will need a lot of money. Service providers allow you to rent or share connections that they have already established, which can save money and time. A wide area network (WAN) is similar to a local area network (LAN). The main difference is that the global network operates outside the geographical coverage of the local network.

The ever-increasing amount of information passing through the network creates a certain complexity for the underlying data of the transmission network in its processing area. On the other hand, modern society requires high transmission rates of processed information. The work [4] solves this problem by predicting traffic.

The paper [5] is devoted to the fact that currently this method of access to the network has practically lost its attractiveness due to low stability of connections and access speed, high costs for dedicated phone numbers, purchase of incoming telephone traffic, etc. Instead of it, everything Internet connection via a secure VPN tunnel is more often used, because today it is sometimes easier to connect to the Internet than through an analog landline phone. It is noted that the main disadvantage of dial–up access is the low data transfer rate. Often, this makes it uncomfortable or even impossible to work with corporate applications. Moreover, it makes it impossible to work remotely with large amounts of data. There is also another drawback, which is characterized by a higher cost of the channel compared to an L3 VPN of the same bandwidth. And as a result, the most versatile and affordable type of communication is the use of L3 MPLS/VPN solutions for networking at the IP level or L2 MPLS/VPN solutions for networking at the channel (usually Ethernet) level.

The article [6] provides information that difficulties arise in achieving the required Internet connection speed in the fundamental principles of building telephone networks, which by their nature are not designed for high-speed data transmission. As the cost of user equipment that allows access to the Internet gradually decreases, the bandwidth of the connection and its cost come to the fore. Telecommunication technologies of the future are based on providing each user with the possibility of high-speed data transmission. The question arises, how to transmit data at high speed over the critical "last mile"? In conclusion, we can note a significant trend in the evolution of broadband subscriber access networks, which follows from the general trend of increasing the bandwidth of the access network and consists in the emergence of optimal solutions that represent a combination within a single network and even a line of several access methods. Such technologies include, for example, mixed optical-radio-coaxial HFRC access technology, as well as VDSL technology, which offers the use of a mixed copper-optical transmission medium in a subscriber access network.

The paper [7] is devoted to the fact that total packet losses, determined and estimated, can provide operators with some valuable information that will help them accurately assess the overall performance of network paths and narrow down the range of network anomalies. The proposed method is thoroughly tested using simulation.

In [8] a study is being conducted that information processing systems need fast access to the Internet. In this regard, Internet service providers with high bandwidth appear on the market, where part of the bandwidth is free during off-peak hours. The main purpose of the study is to divide and distribute the excess bandwidth between users during non-working hours to achieve maximum user satisfaction. For this purpose, this publication proposes a scheme of the dynamic excess rate (DER) and its structure.

The paper [9] presents the implementation of a test bench. The objectives of the study are to identify problems associated with network congestion and propose a suitable method to overcome such problems and generate output data to compare the results before and after the proposed implementation.

After a comprehensive analysis of the relevant sources, it was proposed to ensure the quality of service with the help of additional rules, such as traffic control in the network. The proposed test bench also classified traffic. The results show that the proposed test bench is stable.

The work [10] is devoted to tunneling and translation, which are popular transition mechanisms for a hybrid network (IPv4/IPv6). ISPs configure both mechanisms to provide interconnection in a hybrid environment (IPv4/IPv6). This article provides an architecture independent of the Internet provider to ensure interconnection in a hybrid network, by deploying a tunneling mechanism and broadcasting through a decision-making object. The architecture allows the user to deploy the necessary tunneling and translation mechanism under the auspices of a decision-making object that defines the requirements for packages.

In [11] is talking about the cost of Internet provider services for obtaining high-quality information. Developing accurate models for the costs of communication network services has always been a difficult task for Internet service providers. The proposed new methodology significantly reduces the development time of the cost model, which leads to savings in operating costs of the Internet provider. In the study, a virtual private network service over the Internet Protocol (IPVPN) was considered. IPVPN creates a logical virtual network by providing an encrypted and dedicated data connection. In this paper, an experiment was also made with machine learning techniques for research, which gave additional results.

The work [12] is devoted to the use of SON algorithms to automate the operation of fourth-generation mobile networks based on OAM (Operation and Maintenance) operation, administration and management networks.

In [13] options for quality control of services are considered and alternatives are presented. The methods of implementing new protocols, their pros and cons are described. Considering that for large telecom operators, monitoring the quality of services provided is an important component of a stable business, the problem under consideration is relevant and requires new approaches at minimal cost.

According to the results of studies of existing LANs, which are covered in publications, the difficulties of achieving the required speed of connecting to the Internet have been identified, the solution of which can be optimization for various technical and economic parameters, the use of several types of access, as well as the optimal distribution of bandwidth, the use of tunneling mechanisms and broadcasting to improve not only the quality of the services provided, but also the formation of the cost of them. Based on their experience in building networks based on simulation modeling [14]-[20], the network under study and communication lines are modeled in the OPNET Modeler program. The main difference of this study is step-by-step instructions for designing a LAN and ISP network, setting up and selecting parameters for the study. a method for checking the reliability of the network structure and optimizing the functioning of users' work is also proposed based on the use of queuing theory.

2. DEVELOPMENT OF A NETWORK MODEL BETWEEN LAN AND ISP BASED ON OPNET MODELER AND EQUIPMENT CONFIGURATION

To carry out network modeling, it is necessary to configure the equipment (Figure 1). All workstations are configured according to the same principle, so it is necessary to give only one example of configuration. Only IP addresses are configured depending on the group, either traditional, or using a DHCP server, which are assigned automatically. Each workstation can be assigned a network VLAN number. The workstations must be connected to the switch. Optionally, you can specify the name and type of user. The workstation settings window in OPNET Modeler is shown in Figure 2.

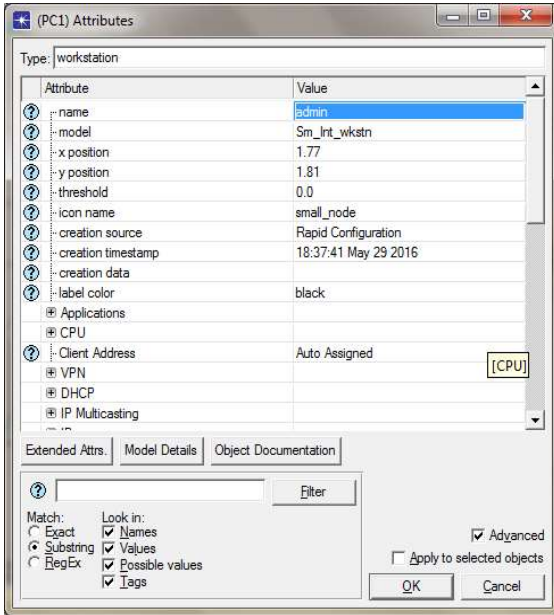*Figure 1: Diagram of the built network on the OPNET Modeler desktop*

Figure 2: Workstation settings window

2.1 Choosing the traffic type and codec

After the basic settings, you need to set the type of traffic in the network. To do this, go to the main menu and select the type of traffic you need in the position, as shown in Figure 3.
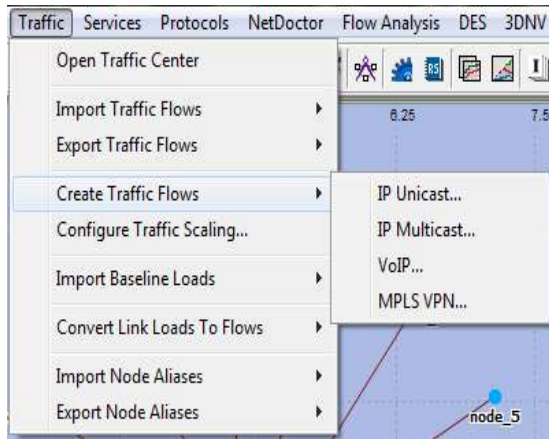

Figure 3: Highlighting the desired type of traffic

Two types of traffic are used in the work. Unicast IP traffic, which allows you to simulate traffic from an FTP server and Voice over IP (VoIP) traffic, which is responsible for voice transmission.

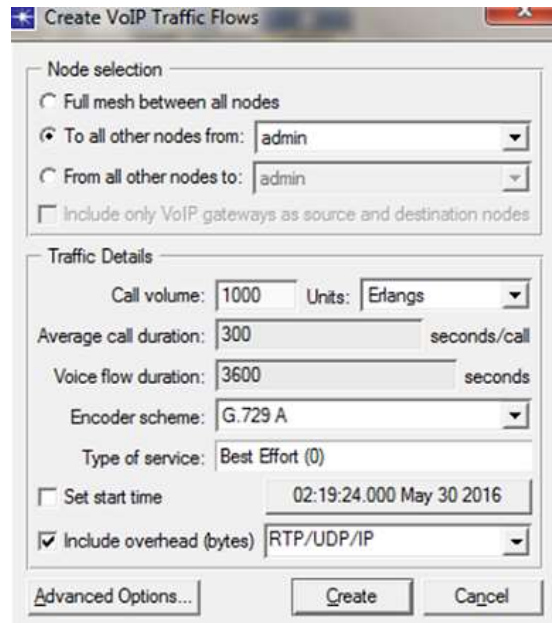We will select the type of codec depending on the bandwidth occupied (Figure 4).

Figure 4: Creation and modeling window VoIP traffic

To study the amount of information being transmitted, we select various types of codecs. It can be seen from Table 1 that the G.729 codec, in comparison with others, skips more information in magnitude.

Table 1: The amount of information transmitted using codecs for transmitting information over VoIP

Codec	G.711	G.726	G.729	G.723
Volume, bytes	2050000	2080000	4100000	4000000
Model time, msec	300	300	300	300

Based on the results shown in Table 1, a diagram is constructed, which is shown in Figure 5. The model time is located horizontally at the bottom of the diagram, and vertically the number of bytes that passed with different types of codecs participating in this experiment. Figure 5 shows that the largest number of bytes passes when using the G.729 codec, so we choose the G.729 codec.
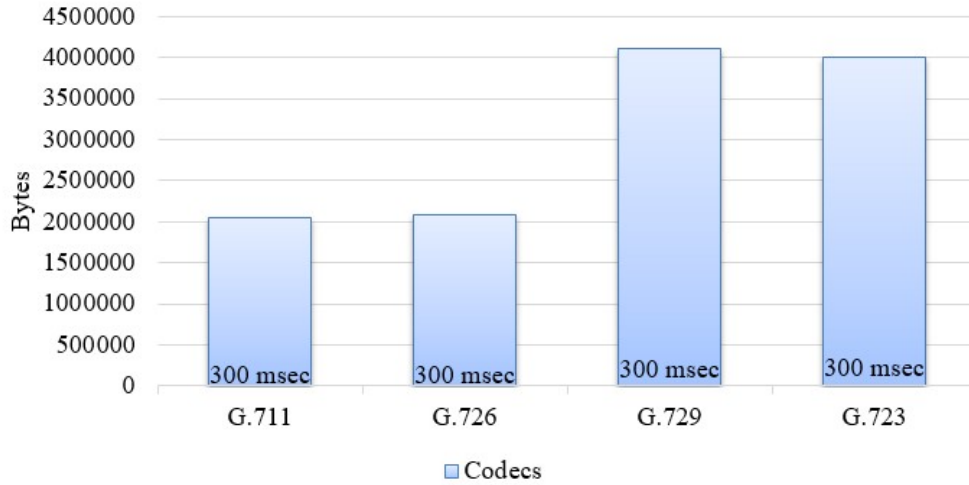
Figure 5: Diagrams of the distribution of transmitted information in the network depending on the type of codecs

3. SIMULATION AND STATISTICAL ANALYSIS OF THE OBTAINED PARAMETERS

During the simulation, it is necessary to analyze the data obtained. Before starting the process, you need to configure some simulation parameters. To do this, click configure/run simulation on the toolbar and enter simulation mode.

Next, we proceed to the stage of collecting and processing statistical data. Statistical data on the functioning of the network can be obtained from a specific node of the network or from the entire network (Global Statistics). Now it is possible to get statistics from a specific node (Router Load) and global statistics (Ethernet Delay). The load of the router connecting the LAN to the ISP is a key parameter, as it affects the network performance. To do this, right-click on the server and click "Choose Individual Statistics" from the context menu. The "Choose Results" window will appear. This window allows you to specify to the router the traffic for which you want to collect statistics.

In our case, since not only IP telephony works, but also other services, we indicate all the IP traffic that was received and sent to the Internet on this router. As well as the amount of buffer that was filled during data transfer [10].

To do this, follow these steps:

1. Click on the IP to open this branch;

2. Check the box next to Buffer Usage, Traffic Sent, Traffic Received.

All measurements are in bits/s.

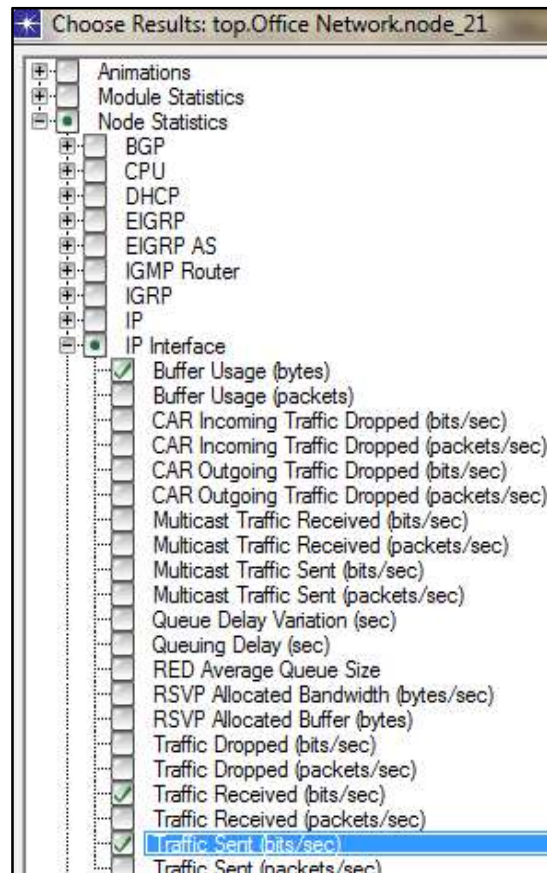Before each simulation of the model, it is recommended to save the network topology (Figure 6).

Figure 6: Results output selection window for the router

When creating a model, you must constantly refer to the properties of the "Library", since it is in the Library that all the network devices that will be needed when building the model are pre-installed. Now you can make a simulation of the

model and get the results of network modeling. To do this, right-click on the route connecting the LAN and ISP, and in the drop-down window that appears, click on the View Results button. After simulating the model, we get statistics on the flow passing through a communication line with Internet access and see that the workload is 78%, which is a good indicator (Figure 7).
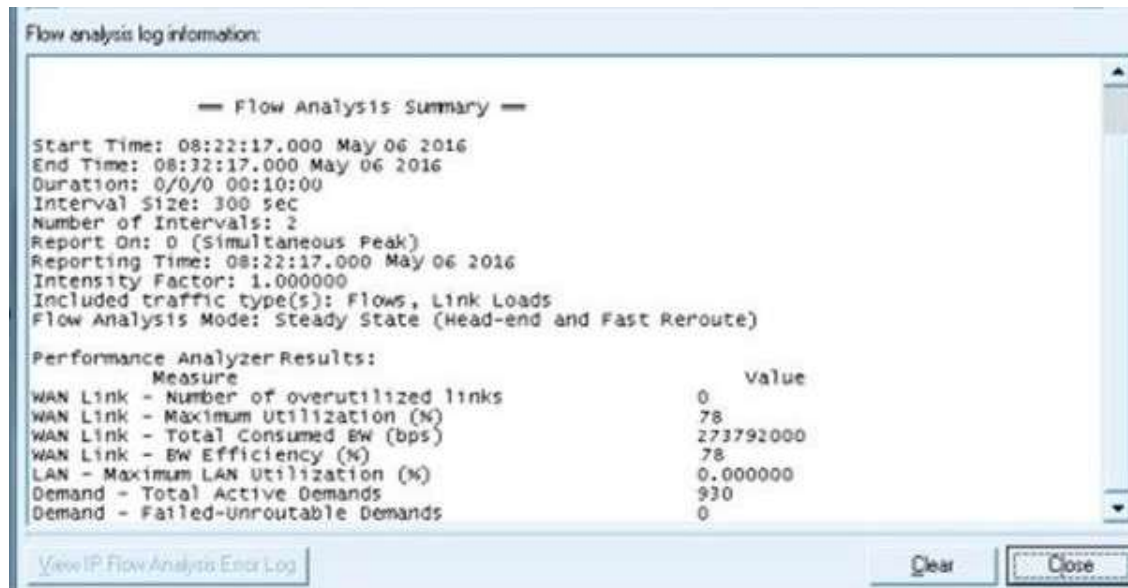

Figure 7. Simulation results of passing traffic on the router

HTTP and e-mail with parameters were used as Data traffic:
  − protocol version 1.1;
  − the time between page requests is 10 seconds;
  − the size of images on the page is from 2 MB to 10 MB;
  − number of images per page from 1 to 10;
  − the number of letters is 20.

The G.729 codec with parameters was used as voice traffic:
  − silence duration 0.65 s;
  − encoding time 0.02 s;
  − decoding time 0.02 s.
  − The VCR protocol was used for video traffic:
  − frame size 352 x 240 pixels;
  − the number of frames per second is 30 frames/s.

All traffic passing through the router was simulated for 10 minutes.

Voice and data transmission does not require a lot of bandwidth, but for this type of traffic like voice, constant bandwidth is required compared to video traffic. And for the Triple Play service, a bandwidth of 78% of the current possible bandwidth of the router is required. The utilization parameter itself shows the bandwidth usage as a percentage.

When modeling a network, it is necessary to consider delays, since they greatly affect the network. Figure 8 shows a graph of voice traffic latency with and without various quality of service parameters.

As can be seen from the graph, in the absence of quality of service, the delay is much higher than when using one of the mechanisms for providing quality of service, which may eventually affect the transmission of voice traffic.
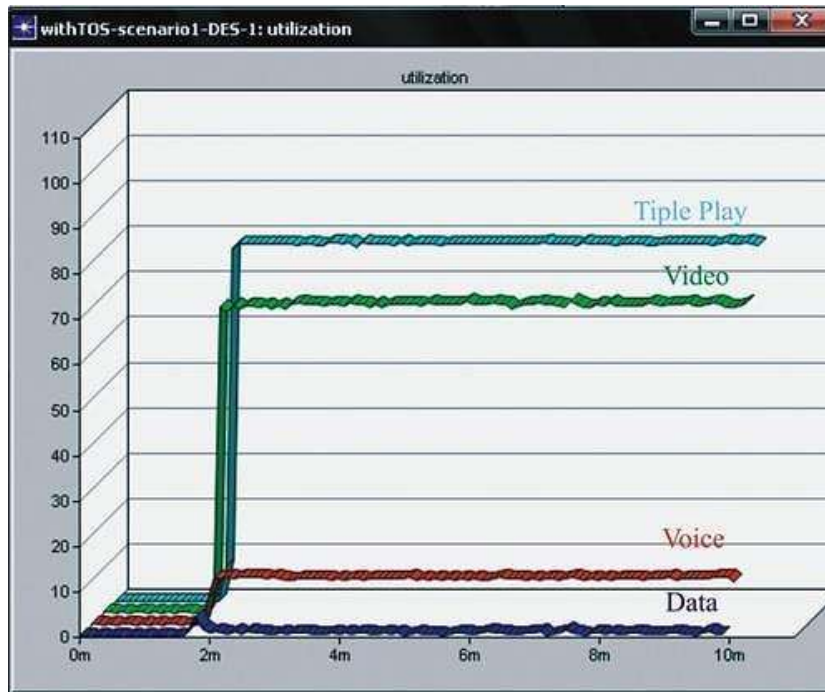
Figure 8: Channel bandwidth usage

For voice traffic, the best quality of service mechanism is Int-DiffServ. A feature of this mechanism is traffic classification and traffic prioritization, which allows you to transmit packets of the highest traffic without queues and delays, since bandwidth will always be allocated for a certain type of traffic (Figure 9).
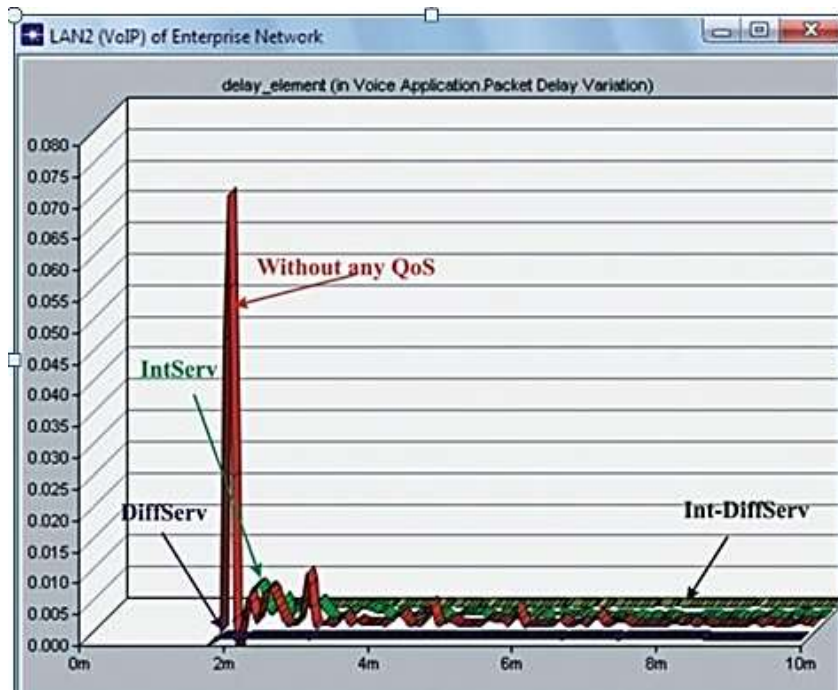

Figure 9: Delays in the transmission of voice traffic

From the simulation results obtained, the load of the communication line is from 1% to 78%, which allows users to work as comfortably as possible with this bandwidth. These values are similar to statistical data. The delay in the transmission of voice traffic is up to 0.070, and with the quality of network service, delays are reduced to 0, which is a very good indicator and indicates high network performance and the absence of congestion (Figure 10).

Figure 10: Delays in the transmission of video traffic

4. CALCULATION OF THE INFORMATION TRANSFER RATE PER CALL DEPENDS ON THE FRAME LENGTH, FRAME PROCESSING SPEED AND OTHER PARAMETERS

We will calculate the transmission rate of voice frames for the G.729 codec, considering additional service information, for one call to the interface channel with the Internet provider. In this case, the voice frame has a length from 10 to 240 bytes.

$$L = L_{RTP} + L_{UDP} + L_{IP} + L2_{OVH} + L_{VPS} \qquad (1)$$

here L_{RTP} – the length of the RTP protocol header, bytes;

L_{UDP} – the length of the UDP protocol header, bytes;

L_{IP} – the length of the IP protocol header, bytes;

$L2_{OVH}$ – the length of the frame at the second level, bytes;

L_{VPS} – the length of the voice frame, bytes. For instance,

$L = 12 + 8 + 20 + 18 + 20 = 78$ (bytes).

One voice packet by default contains a payload of 20 ms of voice (uncompressed or compressed). Since sampling occurs 8000 times per second, 20 ms gives us 160 samples. If we divide 8000 by 160, then 50 packets with 160 bytes of payload per second will be generated for a one-way voice stream [6].

$$BW/call = Payload + \frac{Protocol\ Payload}{Protocol} \cdot CodecBW \text{ ,kbit/s.} \qquad (2)$$

Here CodecBW – codec band.

$$BW/call = \frac{10 + 58}{10} \cdot 8 = 54{,}4 \text{ kbit/s}$$

Table 2 summarizes the bandwidth calculations per call.

Table 2: Calculation of the bandwidth per call

Frame length	10	20	30	40	50	60	70	80
Speed	54,4	31,2	23,47	19,6	17,28	15,73	14,6	13,8
Frame length	90	100	110	120	130	140	150	160
Speed	13,16	12,64	12,22	11,86	11,56	11,31	11,09	10,9
Frame length	170	180	190	200	210	220	230	240
Speed	10,72	10,57	10,44	10,32	10,2	10,11	10,02	9,93

Figure 11 shows a graph of the dependence of the bandwidth per call on the length of the frame.

We will calculate the information flow in the Internet access network for 35 LAN users, each of whom has a PC with Internet access. At the same time, the main load falls on the router of the Internet provider, since all data transmitted to the Internet first passes through it. It is necessary to know the size of the files to determine the required bandwidth. Thanks to the SNMP logs, it is possible to extract data for analyzing the volume of traffic, and for applications transmitted within an hour from users.
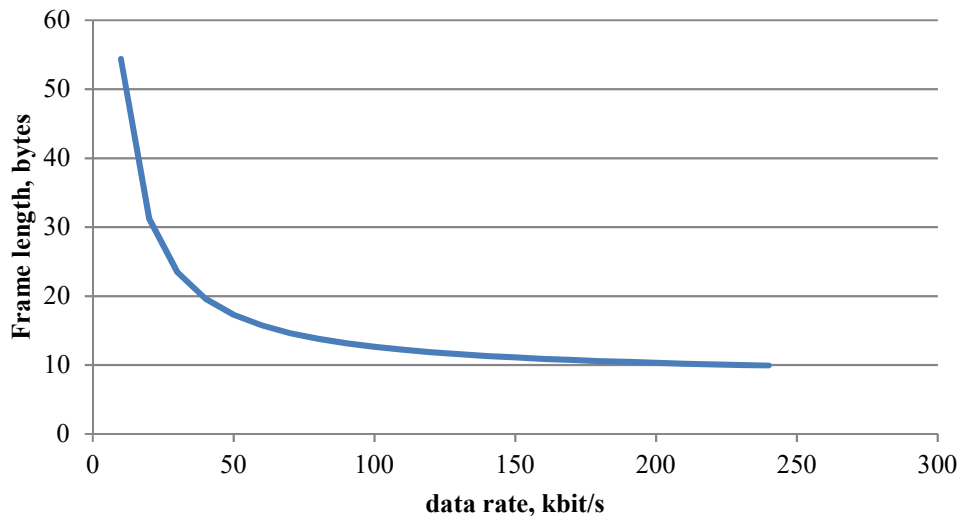

Figure 11: The dependence of the bandwidth per call on the length of the frame

When calculating the information flow to the Internet, the main load falls on the ISP router, since all data transmitted to the Internet first passes through it. You need to know the size of the files to determine the required bandwidth. Thanks to the SNMP protocol, it is possible to analyze how much traffic flows from each user within an hour and what data is transmitted.

Each user has an IP phone and has a connection in the internal network, as well as communication with remote branches. Users work with mail services and work intensively with remote servers (FTP, DNS). Table 3 shows the number of users and the services they demand. Figure 12 shows the applications in a three-dimensional pie chart.

Table 3: Services

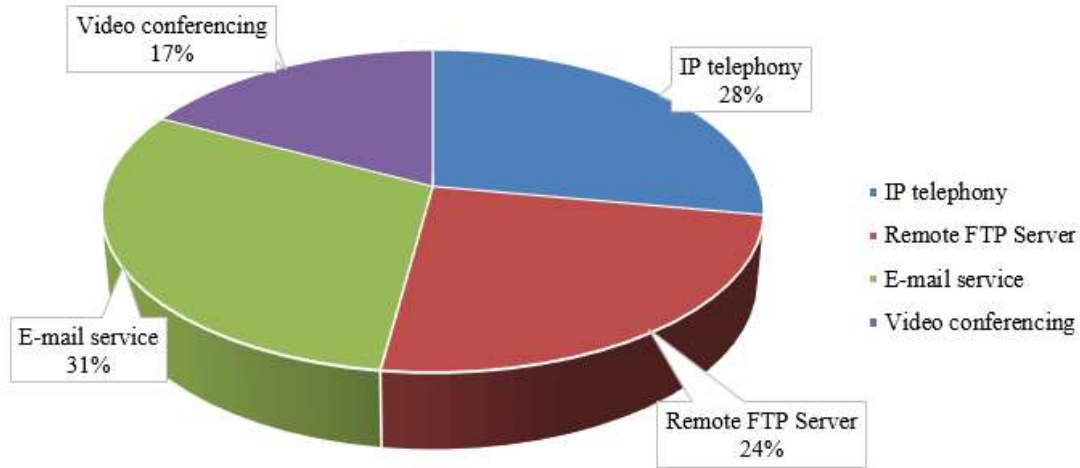| Name of the service	Quantity
IP telephony	35
Remote FTP Server	28
E-mail service	35
Video conferencing	20

Figure 12: Services

In order to determine the amount of traffic passing through the router, we use:

$$P = \sum P_i \cdot N_i.$$ (3)

here P_i – the amount of data transmitted by i category users;

N_i – number of users.

The amounts of data transmitted through the router during one hour of the working day are taken from the logs stored in the SNMP protocol as a result of certain settings on the router:

PIP-Phone = 556693696 bits = 69586712 bytes;
Pftp = 4225933992 bits =528174249 bytes;
Pmail = 18982584 bits =2372823 bytes;
Pv = 1827471360 bits =228433920 bytes.
Total volume per working day:

P = 8·(7086756·35 + 52817429·28 +
+2372823·35 + 228433920·20)

P = 19605593937 (bytes).

To find out enough bandwidth, you need to find out how many frames the router sends to the network every day. At the same time, the number of voice frames is:

$$N_{voice\ frames} = \frac{P}{68} = \frac{69586712 \cdot 35 \cdot 8}{68}$$

$$N_{voice\ frames} = 286533520 \text{ (frames per day)};$$

here P – the volume of voice information transmitted;

68 – the length of the voice frame in bytes.

The number of frames of file data and email data is:

$$N_{FTP,mail} = \frac{(528174249 + 2372823) \cdot 63 \cdot 8}{1518}$$

$$N_{FTP,mail} = 176150016 \text{ (frames per day)};$$

The number of video frames is:

$$N_{video\ frames} = \frac{228433920}{352 \cdot 240 \cdot 8}$$

$$N_{video\ frames} = 338 \text{(frames per day).}$$

The total number of frames will be:

$$N_{frames} = N_{voice\ frames} + N_{FTP,mail} + N_{video\ frames}$$ (4)

$$N_{frames} = 286533520 + 176150016 + 338 = 462683874.$$

For further calculation of the frame processing speed, we will use the mathematical apparatus of queuing theory. The initial data will be the number of transmitted frames found above and the length of the information part of one frame.

To use the theory of queuing, it is necessary to know the relationship between the rate of receipt of frames and the speed of service.

$$V = \frac{N_{frames}}{T \cdot 3600}. \qquad (5)$$

где N_{frames} - the number of frames transmitted during the working day;

T – working day duration, (hours).

$$V_{voice\ frames} = \frac{286533520}{8 \cdot 3600}$$

$V_{voice\ frames} = 9949$ (frames per second);

$$V_{FTP,mail} = \frac{176150016}{8 \cdot 3600}$$

$V_{FTP,mail} = 6116$ (frames per second);

$$V_{voice\ frames} = \frac{338}{8 \cdot 3600}$$

$V_{voice\ frames} = 0,0117$ (frames per second).

Since the data is transmitted to the Internet, address information is added to the information packets, which takes up 18 bytes, if labeled frames, then 4 bytes are added and, considering MPLS technology, another 4 bytes are added. The total frame length will be 1526 bytes for files and email, 72 bytes for voice and 675844 bytes for video.

Next, we will find the service time of one frame with a fixed bandwidth installed on the router:

$$t_{frame\ maintenance} = \frac{L_{frame}}{V_{channel}} \qquad (6)$$

где L_{frame} – the length of the transmitted frame;

$V_{channel}$ – the speed of information exchange in the channel between LAN and ISP (bit/s), set on the router.

The frame transfer time is related to the service time. The service speed is the inverse of the service time:

$$V_{maintenance\ FTP,mail} = \frac{V_{channel}}{L_{frame}} \qquad (7)$$

$$t_{maintenance\ FTP,mail} = \frac{1526}{40000000}$$

$t_{maintenance\ FTP,mail} = 0,00003815$ (sec);

$$V_{maintenance\ FTP,mail} = \frac{1}{0,00003815}$$

$V_{maintenance\ FTP,mail} = 26212$ (frame/sec);

$$t_{omaintenance\ voice} = \frac{72}{40000000}$$

$t_{omaintenance\ voice} = 0,0000018$ (sec);

$$V_{maintenance\ voice} = \frac{1}{0,0000018}$$

$V_{maintenance\ voice} = 555555$ (frame/sec);

$$t_{maintenance\ voice} = \frac{675844}{40000000}$$

$t_{maintenance\ voice} = 0,0168961$ (sec);

$$V_{maintenance\ voice} = \frac{1}{0,0168961}$$

$V_{maintenance\ voice} = 59$ (frame/sec).

Now using these values, it is possible to calculate the channel congestion between LAN and ISP:

$$P = \frac{V}{V_{maintenance}} \qquad (8)$$

где V – frame rate;

$V_{maintenance}$ – frame maintenance speed.

$$P_{voice} = \frac{6116}{555555} = 0,011;$$

$$P_{FTP,mail} = \frac{6116}{26212} = 0,23$$

$$P_{video} = \frac{0,0117}{59} = 0,000198$$

When analyzing the communication line and calculating its bandwidth, it was revealed that with the standard operation of the company's office, the bandwidth to the Internet is loaded by 23% by file and email exchange, 1.1% by voice transmission and 0.02% by video data exchange, which is not a bad indicator.

5. CHECKING THE NEED FOR SECURITY AND NETWORK ARCHITECTURE

Next, consider the need to ensure network security. Recently, the number of LAN is becoming more and more, and information with a large volume passes through them [11-13], therefore, performing the procedure for investigating such a process based on modeling a network experiment is becoming one of the important tasks. Next, consider this problem.

To do this, in the built network, we will conduct an experiment on the hacker's impact on the server of this network. To conduct such an experiment, it is necessary to study the Application Capture Manager (ACM) module and the WireShark program. The ACM program is necessary to organize a passive attack on the network server. To do this, select Application Capture Manager (Figure 13).
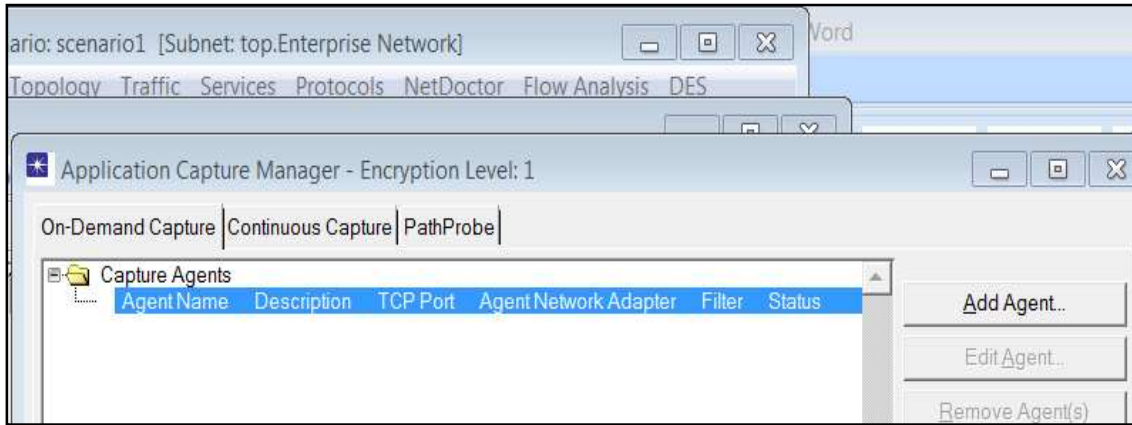

Figure 13: The window of the beginning of the WireShark impact on the network server

Next, by clicking on the Add Agent that allocates the IP address of the computer on which we work (192.168.1.106), we prepare to capture packets transmitted over a network built on OPNET Modeler, and launch WireShark, which is shown in Figure 14. After launching it, we get a window of the graph shown in Figure 15.
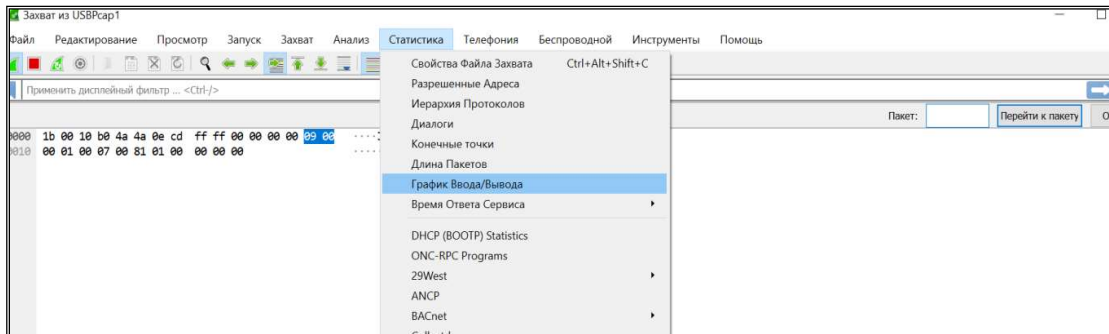

Figure 14: WireShark program window in packet capture mode from the network

Figure 15: Time-dependent packets captured by WireShark

Sometimes there are problems checking the correctness of the network construction during its modeling. In this case, the OPNET Modeler module program called NetDoctor is used. Starting the program and getting the results are shown in Figure 16. The figure shows that the network is built correctly as there are no indications of errors.
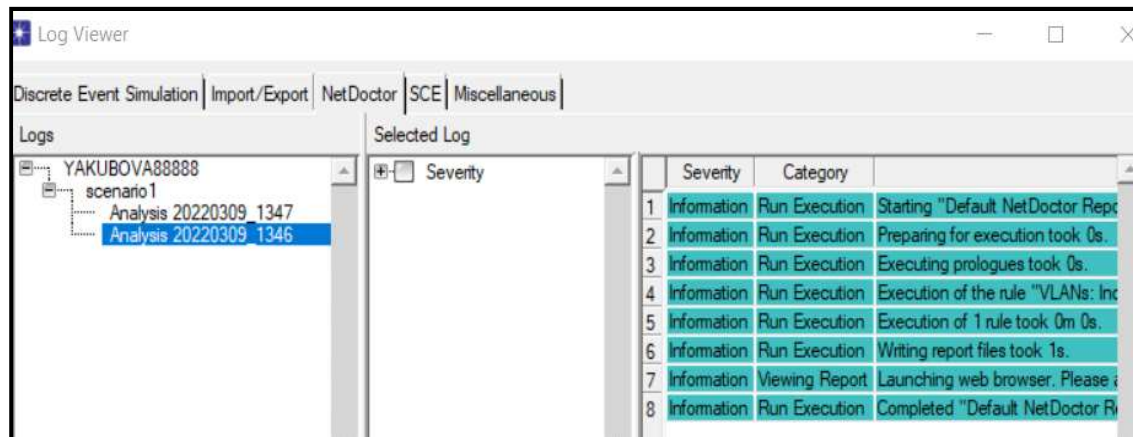

Figure 16: The window of the validation result of checking the built network

CONCLUSION

Based on the study of the communication line between the LAN and the ISP, it can be concluded that the optimal physical connection and the use of a cost-effective and technically functional connection. This was made possible by building a network simulation model between LAN and ISP based on OPNET Modeler.

Analysis of the results of modeling the network between the LAN and the provider when constructing the diagram shows that among the G.711, G.726, G.729 and G.723 codecs, when transmitting information over VoIP, the G.729 codec transmits the largest amount of information, and voice traffic and traffic data takes up less bandwidth than video traffic. Voice traffic transmission delays are up to 0.070, and when using the network quality of service mechanism, delays are reduced to 0, which is a very good indicator and indicates high network performance and no congestion.

Thanks to the calculations made in the article, we can choose the required codec and provide the required quality of service for it. The calculations also show that when analyzing the communication line and calculating its bandwidth, it was revealed that during the normal operation of the company's office, the bandwidth on the Internet was loaded by 23% when exchanging files and e-mail, by

1.1% when transmitting voice and by 0 .02%. % for video data exchange, which is a normal indicator and allows you to eliminate the shortcomings associated with the functioning of the provider and work effectively and use all the possibilities of accessing the Internet.

To check the architecture of the developed network, the use of the OPNET Modeler module, the so-called NetDoctor, was carried out, which shows the reliability of the constructed network.

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