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# Method of Distribution Network Resources after Restoration, the Networks MPLS-TE Use of Various Telecommunications Technologies to Construct Backbone Networks

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## Abstract

The modern telecommunication system is characterized by the rapid improvement of information and communication technologies, such as the improvement in the data rate, power consumption and Quality of Service (QoS). Based on this development, the next generation network (NGN) is shaped. The NGN is represented by a packet transport network demarcated functions and services. To provide transport functions in NGN, using Multi-Protocol Label Switching (MPLS) technology, there are two main problems, which are multipath routing and traffic distribution, which are what this paper works with. Accordingly, this paper proposes a solution to these two problems using optimization procedures to find the shortest path Dijkstra and Bellman-Ford, which is characterized by a high-speed-oriented selection of a single shortest path between the source and the destination, based on several selected criteria related to the optimal use of the network resources. Solving the information flow distribution problems in large number of nodes network applying salesman procedure or having NP-completeness, causes significant time delay, which means that the proposed solution is not suitable for real time applications. Accordingly, for real time applications, there is a need to move to other methods based on the use of several independent shortest paths.

## Keywords

QoS, NGN, MPLS, Routing Protocols, Backbone Networks, BER, PEP

## 1. Introduction

Distribution protocols are regarded as the main criteria to improve the communication data distribution over the network, for example, in [1] the routing protocol performance was discussed for some selected networks resulting to the optimum way of utilizing the resources of the network. In [2] [3] and [4], the influence of improving the routing protocols over communications network was presented for broadcasting network to show how choosing the proper protocol algorithm improves the network performance, so, in [2], combining the transmitted packets resulted to much less transmission traffic over a lossy channel network and in [3] over the physical layer with introducing the Bet Error Rate (BER) for the combined network. In [4], improving the transmission protocols over the long term evaluation advanced communication system resulted to much better Packet Error Probability (PEP).

The information distributing in modern practical communication network is in either packets or in data streams forms as shown in [5] [6] and [7]. The distribution of the flow path involves selection of a specific unit for a couple of the recipient. Per packet transmission makes better use of available network resources. However, using this method may change packet sequence consistency, increased delays in the collection messages and, ultimately, the quality of service degradation. Practice shows that the method is a more rational distribution of traffic. This lowers the load on the device that serves the traffic, and, in general, there is no problem of packet reordering [6] [7]. This method is used in many modern technologies, including in Multi-Protocol Label Switching (MPLS) as a method for optimal load balancing across multiple independent paths for MPLS-TE networks, consisting of consistent application of the optimal shortest path selection procedures. As the selection criterion of the path integral criterion is used, parameters such as the combined capacity of the path and the maximum delay path are taken into account.

## 2. Backbone Networks Construction Telecommunication Technologies

Nowadays, the construction of the main areas of multiservice networks commonly used three-tier architecture, with the network layer functions performed by a single duct—IP. The proposed work in this paper with additional protocols participate to solve the problem of the dynamic routing, signaling, quality of service..., etc.

To create a transport network can be used by several techniques (**Figure 1**) that act as data link layer and the network layer.

The traditional networks, based on the SDH, are regarded as an outdated nowadays, however, ATM technology, in recent years actively developed alternative approaches to build high-performance networks with packet switching. Particular attention is paid to the use of Ethernet for building global networks: collectively, these solutions are known as Metro (Carrier) Ethernet. The Adoption Committee IEEE 802.3 [8] [18] series of 10 GBase-SW technologies, 10 GBase-LW, 10 GBase-EW, cemented Ethernet as promising for the construction

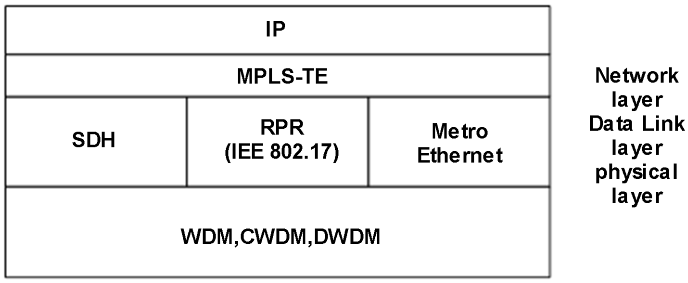



**Figure 1.** Structure of transport technologies.

of global networks. At the moment these technologies are still unresolved questions of management, alarm and monitoring network, but IEEE 802.3 committee and a forum of Carrier Ethernet equipment are actively working to implement the relevant standards.

An alternative approach is to build high-performance networks with packet switching committee proposed IEEE 802.17 standard [9] [10] and [11]. RPR (Resilient Packet Ring) technology uses two optical rings to transmit information in different ways. Structure and sub levels standard 802.17 technology is shown in **Figure 2**.

The structure in **Figure 2** is unlike FDDI technologies [9], which has traditionally been built using double fiber ring in the RPR for transmitting information involves both rings simultaneously. Although the standard 802.17 is still in the approval stage, many companies offer proprietary solutions based on this architecture.

Separately, it should be noted MPLS-TE technology, which is an intelligent tool for the maintenance of traffic and the implementation of new services based on almost any vehicle. Since MPLS-TE mechanisms combined link and network levels, administrators have new mechanisms that allow optimal distribution of traffic, while maintaining the performance of technologies such as ATM.

In addition to traditional office tasks undertaken within each technology, in recent years' special attention is paid to improving the reliability of the network and development of restoration techniques after a network failure. At the moment, many standards describe ways to solve this set of problems in a variety of technologies [8] [12] [13] and [14]. However, nowadays, it has become apparent that a simple switching to the backup path is not sufficient, in addition, must optimize traffic distribution and maximize the use of available network resources. Despite the fact that the solutions to this problem are offered in only one technology, MPLS-TE, should pay attention to the recovery process in various technologies.

## 3. Disaster Recovery in a Variety of Transport Technologies

To create a secure telecommunications network, within each technology should be defined mechanisms for backup and recovery. The different transport technologies, these procedures are implemented in different ways, but in each of

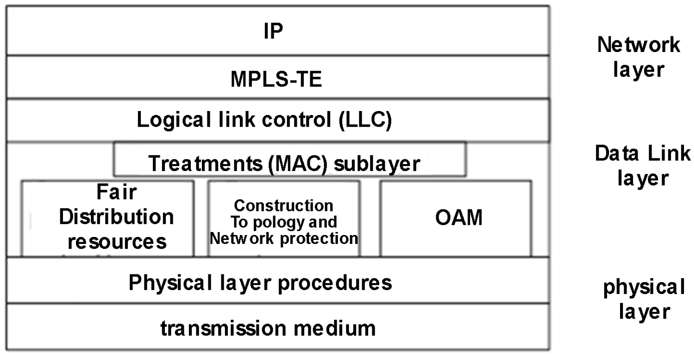



**Figure 2.** Structural levels of RPR technology.

them, the recovery problems are solved on the physical and data link layers. According to [15] and [16] network recovery should occur at the time of no more than 50 ms, *i.e.*

$$t_{\bar{a}} - t_{\overline{na}} < 50 \text{ ms}, \tag{1}$$

where $t_{\bar{a}}$ —the duration of the network failure.

In general terms, in the process of restoration of the network performance consists of three stages as shown in **Figure 3** and **Figure 4**. An early detection output is taken in the first step to confirm building the network, where the detection time and switching to the backup path is considered as a critical parameter and should not exceed tens of milliseconds.

In contrast, the detection of a failure at the network level may take from a few seconds, if the link state protocol OSPF and IS-IS, to a few minutes, in the case of routing protocols running on distance vector, such as RIP, BGP.

When using the MPLS-TE in combination with extensions such as fast routing a (Fast Reroute) [9] and [10], in IP networks an opportunity to achieve the restoration of the network after damage for tens to hundreds of milliseconds, regardless of used transport technology.

In the second stage, after the detection of a failure the fact traffic channel released from the system switches to standby. Normally a backup (or protective) channel pre-created and is idle. With the MPLS-TE technology, it is possible to dynamically create a tunnel (virtual channel) only after the exit from the main building. However, this approach is rarely used in practice, since the traffic will be lost when you create a virtual bypass channel.

The last stage—load balancing optimization of a characteristic of the network layer, and MPLS-TE technology. As a result of the exit of the channel system, or node occurs a structural change in the network, it is possible non-optimal distribution of traffic and incomplete use of available network resources. On traditional routing protocols, traffic distribution is based on the routing metrics and network convergence after recalculation traffic distribution takes a little time. However, the convergence of the network itself takes considerable time, which leads to a break between sessions to higher level applications. This problem is

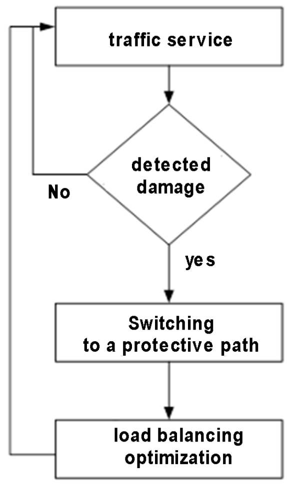



**Figure 3.** Generalized recovery algorithm after the failure in TN.

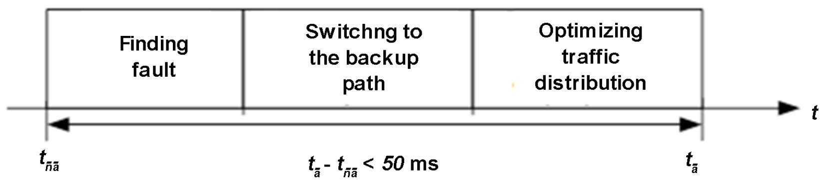



**Figure 4.** Stages of network failover.

solved in the MPLS-TE, which combines intelligent network-level mechanisms and speed channel. Table 1 provides a summary of the recovery time in various technologies.

## 4. Study of the Network Settings under Different Network Modes

### 4.1. Layout Study of Network Parameters

The circuit layout used to study the network setup shown in Figure 5. As a network of clients, we were using PCs running Windows 2000. Operating system to calculate the one-way delay both computers are synchronized using the method of time synchronization for IP-based networks, NTP protocol (Network time Protocol). The source and destination of traffic and routers connected to the switch with Fast Ethernet technology at the speed of 100 Mbit/s in full duplex mode routers Cisco 1760. Fast Ethernet router interfaces were used for modeling the transport network site, as well as computers that are connected to the Cisco Catalyst 2950. Switches as an external network connections serial router interfaces are configured to operate in asynchronous mode, with a maximum transmission rate that was used being 115,200 bit/s output queue servicing discipline as the packet arrival (FIFO). For the experiment, the data link layer to the series connection was implemented using PPP (Point-to-Point Protocol). To study the characteristics of the network traffic in the channel disconnection delay

Table 1. Mechanisms for disaster recovery in a variety of transport technologies.

| | The level of the OSI model | Convergence Time, c | Mechanisms for detecting damage |
|---|---|---|---|
| SDH/SONET | Physical, canal | $<5 \times 10^{-6}$ | Triggers the physical layer, APS |
| SRP (802.17) | Physical, canal | $<5 \times 10^{-6}$ | Posts physical layer |
| Ethernet (STP) | Physical, canal | 1 - 2 (802.1 w)<br>30 - 50 (802.1 d) | BPDU about the network topology changes, producing a timeout BPDU |
| MPLS-TE | The channel, Net | $<10^{-5}$ | SDH/SONET The alarm message, Timeouts RSVP-hello packets |
| IP (IGP) | Net | 2 - 5 (for LS)<br>>10 (for DV) | Timeout service messages |

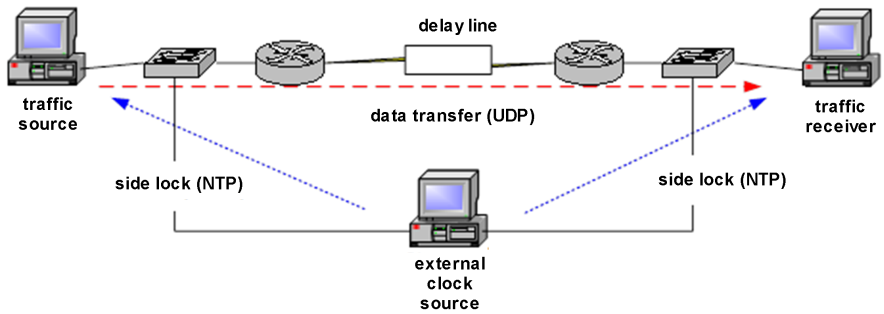



Figure 5. Scheme of the investigated area network.

line has been included with a variable delay of up 0 mks 4 mks.

Investigation of traffic characteristics was performed using the software package D-ITG. With ITGS end traffic generator in the direction from the source to the receiver created a flood of traffic. In an experiment using the power generated UDP packet stream with the normal intensity distribution packet departure. The UDP protocol has been chosen due to the need to study the characteristics of the aggregated flow, excluding certain indicators "micro flow". Using TCP, especially at a high value of the delay lines, would result in multiple retransmissions of lost packets, and distortions in the final results. Given that the experiment was conducted to study the parameters of the network of the main areas, issues related to the correction of errors, the collection of fragmented packets will not be considered and must be addressed end systems. For each delay value held two series of experiments for different size packages—512 and 1024 bytes. The study varied the intensity of traffic generation MO—10 p/s, 20 p/s, 30 p/s and 40 p/s. On the receiving side via ITGResv component recording and subsequent processing of the experimental results. Traffic time for each experiment is 120 second.

## 4.2. Experimental Results

During the experiment, the measurements were the main characteristics of the network—packet loss and delay. Figure 6(a), and Figure 6(b) show the absolute values of packet loss for each value of the delay introduced. As can be seen,

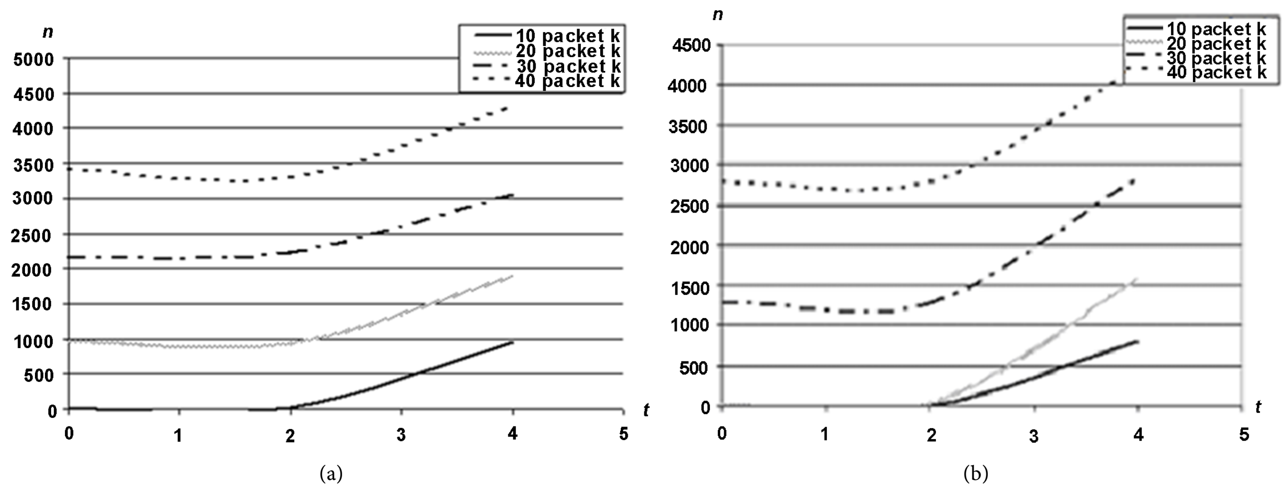
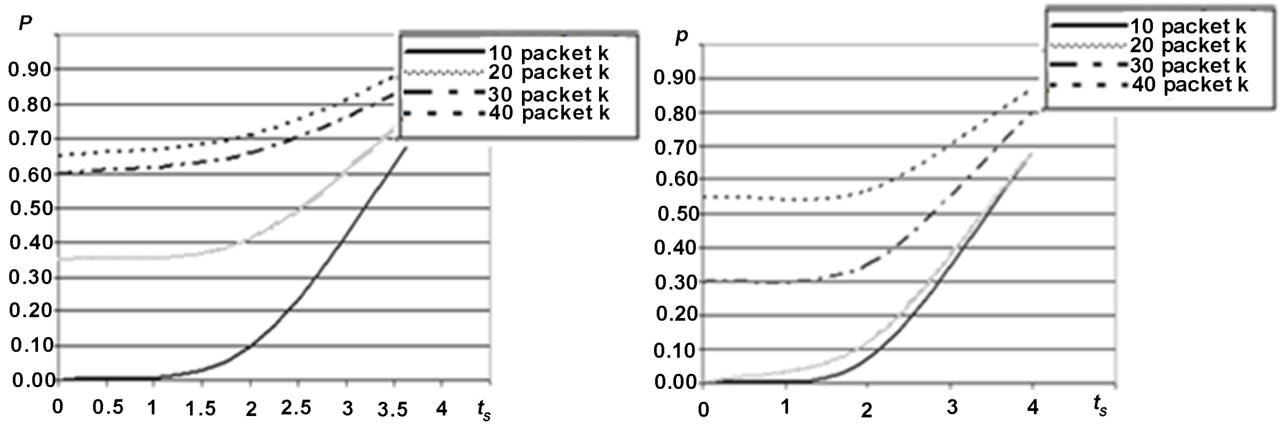



**Figure 6.** Absolute packet loss: the packet size to 1024 bytes (a); 512 bytes (b).

packet loss increases as the physical delays in the transmission line, a significant rostrum packet loss observed with increasing delays 2 mks to 4 mks. Losses at zero delay line caused by the overflow queue at the transmitting router, and depend only on the incoming traffic intensity, size, queue service discipline and the outgoing channel capacity. It is obvious that an increase in the intensity of packet arrival, the losses associated with the overflow queue, also will increase (Figure 6(a), Figure 6(b)) and form a "permanent" component, which does not affect the characteristics of the transmission medium.

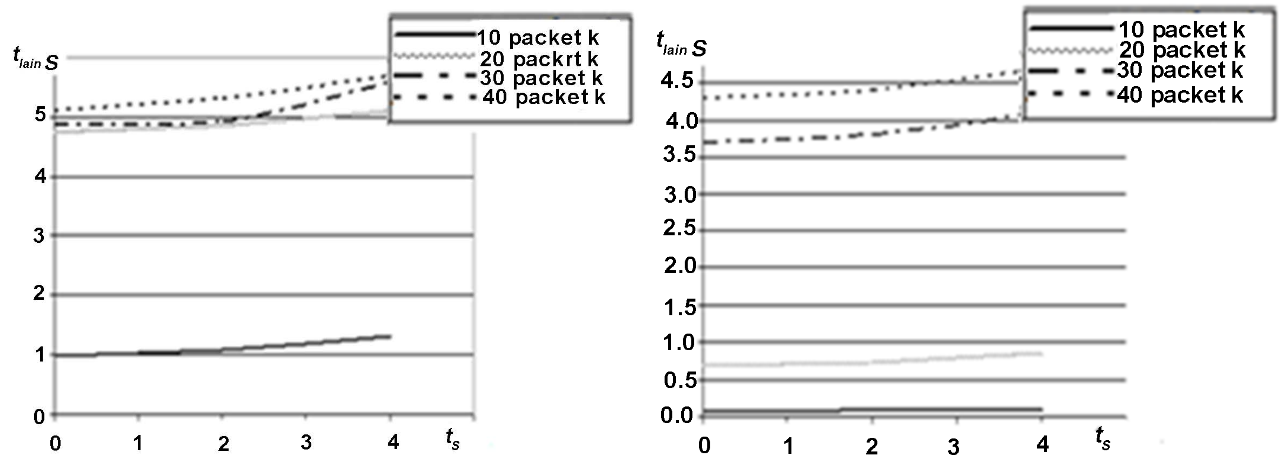
Figure 7 presents the probability characteristics of packet loss. A significant increase in the probability of losses plot 2 mks - 4 mks associated with the deterioration of the physical layer characteristics. From the viewpoint of the terminal equipment, the communication channel is workable, however, given the considerable losses in it (80% - 90% at 4 mks delay, taking into account losses in the queue), to transmit information thereon is impossible. According to [11], the transmission of speech over IP, packet loss allowable value depending on the codec type and packet loss recovery techniques without significant degradation in speech quality, may vary in the range of 2% - 3%. However, when transmitting data packets (*i.e.* TCP protocol segments) are required to transmit each packet is lost again. Therefore, the maximum permissible limit of loss must not exceed 3%, with an increase in the probability of packet loss, it is necessary to carry out the routing of flows on the bypass routes.

Since the main component of the losses of traffic over high-quality transmission medium, which is typical, for example, fiber optic, up losing in the queues of the transmitting device, a quantitative measurement of the losses on the practice may be carried out by monitoring the queue for outbound interfaces, and when thresholds are exceeded, perform a full or partial routing in roundabout ways. In the case of large losses in the communication channel, which greatly exceeds the allowable threshold, the monitoring should be performed on the receiving side by the sequence numbers of packets or the transport network and to




(a) (b)

**Figure 7.** The probability of packet loss: packet size of 1024 bytes (a); 512 bytes (b).




(a) (b)

**Figure 8.** The probability of packet loss: 1024 bytes' packet size (a); 512 bytes (b).

notify the sender using the signaling protocol.

According to [17] [18], for voice packets over IP-based networks, one-way delay, *i.e.*, speech packet transmission time from the source to the receiver, must not exceed 150 ms.

A further increase in the delay of voice packets leads to significant speech quality degradation. **Figure 8** shows the characteristics of packet delay in a simulated network. **Figure 8** shows that the change in the physical layer characteristics of virtually no effect on the value of packet delays. The main component is a delay while waiting for service in the transmitting device queue. In practice, the measurement of one-way packet delay is difficult and requires a centralized timing source and destination using the NTP protocol. Local packet delay on the main portions with high accuracy can be forecasted by monitoring the packet queue at the transmitting device. Moreover, one should consider a variety of service requirements for voice and data traffic. To minimize the delay of voice packets to the local node may be using a queue service discipline. In addition, at a constant load queue may perform routing separate streams for backup paths.

## 5. Conclusion

The task of network restoration in critical situations due to its non-stationarity can be solved based on the management of the corresponding resources using the procedure of recursive estimation of the traffic state. The control algorithm, in this case, can be constructed using the results of the separation theorem, that is, if the linear solution is possible, the Gaussian character of the estimated process, the change in traffic, the given algorithm is implemented in the form of a sequence of two separate procedures, optimal stochastic estimation of the network state and deterministic linear control of network resources, where the control parameter is the obtained estimate. Qualitative characteristics of the estimation and the structure of control algorithms are obtained, and recommendations are given on the choice of the parameters of these algorithms.

## References

[1] Alhihi, M. (2017) Practical Routing Protocol Models to Improve Network Performance and Adequacy. *Journal of Computer and Communications*, **5**, 114-124. https://doi.org/10.4236/jcc.2017.56007

[2] El-Hihi, M., Attar, H., Solyman, A.A.A. and Stankovic, L. (2016) Network Coding Cooperation Performance Analysis in Wireless Network over a Lossy Channel, M Users and a Destination Scenario. *Communications and Network*, **8**, 257-280. https://doi.org/10.4236/cn.2016.84023

[3] Attar, H. (2016) Physical Layer Deterministic Network Coding Using PUM Turbo Codes over AWGN Channel, N Nodes through a Base Station Scenario. *Communications and Network*, **8**, 241-256. https://doi.org/10.4236/cn.2016.84022

[4] Attar, H., Stankovic, L., Alhihi, M. and Ameen, A. (2014) Deterministic Network Coding over Long Term Evaluation Advance Communication System. 2014 *Fourth International Conference on Digital Information and Communication Technology and its Applications* (*DICTAP*), Bangkok, 6-8 May 2014, 56-61. https://doi.org/10.1109/DICTAP.2014.6821657

[5] Khosravi, M.R., Basri, H. and Rostami, H. (2016) A Solution for Scalable Routing in Depth Divisions-Based DUSNs via Adding a Scalable Parameter to Control Depth Clusters: Creating an Energy Efficient and Low Delay NI-Independent Communication Protocol. *Journal of Computer and Communications*, **4**, 55-61. https://doi.org/10.4236/jcc.2016.47008

[6] Attar, H., Stankovic, L. and Stankovic, V. (2012) Cooperative Network-Coding System for Wireless Sensor Networks. *IET Communications*, **6**, 344-352.

[7] Attar, H.H. and Stankovic, L. and Stankovic, V.M. (2009) Physical-Layer Network Coding Based on PUM Turbo Codes. 3*rd Mosharaka International Conference on Communications*, *Signals and Coding*, 19-21 November 2009.

[8] Carrier Sense Multiple Access with Collision Detection (CSMA/CD) (2005) Access Method and Physical Layer Specifications. Standard IEEE-802.3.

[9] Olifer, V.G. and Olifer, N.A. (2006) Computer Networks. Principles, Technologies, Protocols: Textbook for High Schools. 3rd Edition, Peter, St. Petersburg, 958c.: ill.

[10] Tsiang, D. and Suwala, G. (2000) The Cisco SRP MAC Layer Protocol. RFC-2892.

[11] Resilient Packet Ring (RPR) (2004) Access Method and Physical Layer Specifications. Standard IEEE-802.17.

[12] Thompson, K., Miller, G.J. and Wilder, R. (1997) Wide-Area Internet Traffic Patterns and Characteristics. *IEEE Networks*, **11**, 10-23. https://doi.org/10.1109/65.642356

[13] Types and Characteristics of SDH Network Protection Architectures. ITU-T Recommendation G.841, 1998.

[14] Lemeshko, A.V. and Belenkov, A.G. (2003) A Dynamic Model for the Integrated Solution of Routing and Subscriber Access Tasks in Territorially-Distributed Telecommunication Networks. *Open Information and Computer Integrated Technologies*, **18**, 134-139.

[15] Liu, G. and Ramakrishnan, K.G. (2001) A Prune: An Algorithm for Finding k Shortest Paths Subject to Multiple Constraints. *Proceedings of the INFOCOM* 2001 *Conference*, Anchorage, Alaska, April 2001, 743-749.

[16] Pan, P., Swallow, G. and Atlas, A. (2005) Fast Reroute Extensions to RSVP-TE for LSP Tunnels, RFC-4090.

[17] Olvein, V. (2004) Structure and Implementation of Modern MPLS Technology: Per. with English. M.: Williams Publishing House, 480 p.

[18] Alhihi, M. (2017) Setting an Optimization Problem of Distributing Traffic across Multiple Independent Paths. *Transactions on Networks and Communications*, **5**, 40.