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The Influence of Transport Layer to Ethernet Services Quality

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Introduction

The developing of telecommunications technologies predetermines creation of multifunctional telecommunication products and various service presentations for users. All these innovations require high values of network availability and performance parameters and they ensure end-users and service providers that their network is capable to transport data [1–3]. But it is unclear what level of performance they should expect from their mission-critical applications. It is relevant for service providers to ensure end-users that most important applications can use full bandwidth of the up to date Ethernet services [4–6].

Transport layer is one of the most important because it is responsible for the end-to-end connections. This layer guaranties that the data segments are transferred from the network toward the application.

Two main types of transport protocols are used by applications to communicate between all of their locations: User Datagram Protocol (UDP) and Transport Control Protocol (TCP). They are the most popular open system protocols because of their successful application in different telecommunication networks. These transport protocols carry out a connection, but the main difference between them is that TCP protocol is connection oriented protocol with logical connection, while UDP – connectionless and establishing a connection before sending data is not necessary.

Data transferring control mechanisms are applied in user‘s premises and acts in fourth level of OSI model (Fig. 1). The other layers underneath the transport layer are the network, data link and physical layers. Separation between transmission and services divisions are caused by Ethernet over SDH. SDH offerings are positioned by sellers as capable to support cost-effective data transport [7–9].

Questions may arise for telecommunication transport service provider about what the configuration of SDH network nodes should be selected and which transport protocol is the most suitable for Ethernet data transferring through the SDH transport network?

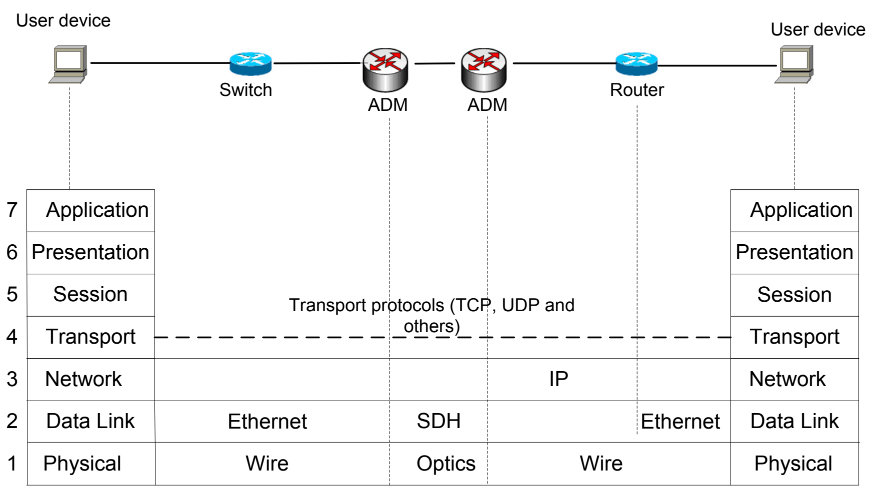

Fig. 1. Theoretical structure (OSI reference model): Ethernet over SDH

Ethernet over SDH

First of all it is necessary to discuss briefly the specifics related with data multiplexing in SDH network node, what allows network provider to transfer multiple data flows through one fibre. This procedure consists of data encapsulation and virtual container group (VCG) forming. STM-1 level (Synchronous Transport Module

Level–1) is the basic unit in SDH network, which allows 155.52 Mbps transfer rate. It supports SDH section layer connections (Fig. 1).

STM-1 frame consists of 2430 octets. A virtual container (VC) supports the path layer connections in the SDH. The Tributary Unit (TU) provides the adaptation between low order path layer and a high order path layer. One or more TUs, occupying fixed, defined positions in a high order VC-n payload are defined as a Tributary Unit Group (TUG). TU-11 and TU-12 can be multiplexed into a TUG-2. VC together with H1, H2, H3 byte pointers is called AU (Administrative Unit). AU assures adaptation between the high order path layer and the multiplex section layer. Two sorts of AUs are defined: AU-3 and AU-4. One or more AUs of fixed and defined positions in a higher order VC-n payload are defined as Administrative Unit Group (AUG). The AU-4 is placed directly in the AUG-1 [9]. The arrangement of section overhead (SOH) bytes is RSOH (Regenerator Section Overhead) and MSOH (Multiplex Section Overhead). These segments' information helps to do the monitoring of transmission quality for error detection and alarm managing in data channels. Section overhead SOH (RSOH and MSOH) is added to AUG, and finally STM-1 module is formed (Fig. 2).

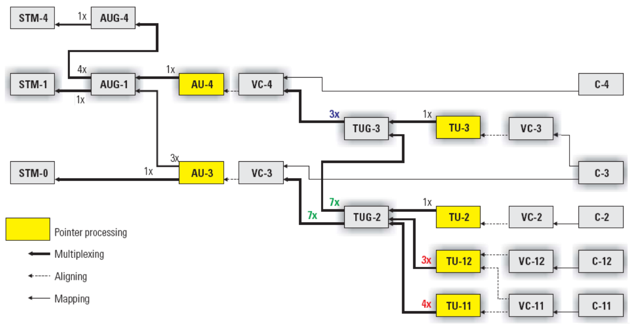

Fig. 2. SDH multiplexing structure [10]

Data encapsulation procedures should be analysed in more detail. Mostly for multiplexing GFP (Generic Framing Procedure) is used [11, 12], because it allows various lengths and higher level signals to be transferred over SDH networks. GFP better utilizes the information channel and assures user with flexible throughput between Ethernet and SDH data traffics. GFP frames may be with information or empty. Empty frames are inserted between frames with information, if asynchronous data flow is introduced into synchronous one.

In GFP a fixed length overhead is used for all packets irrespectively of their content. It consists of two segments: the first indicates the lengths of payload in Bytes, and the second – CRC-16 - indicates errors checking code for the first segment. Besides that, one bite error correction in the overhead is possible and this determines better reliability. GFP doesn't require any special means for frames processing as compared with HDLC (High-Level Data Link Control) protocols; the latter overhead depends on data transferred. Customer's signal data managing procedures are described in payload overhead.

At first it would be worthwhile to ascertain the influence of SDH network node configuration to the Ethernet data transferring quality.

Network node performance analysis

The influence of SDH network node (ADM - Add Drop Multiplexer) for the Ethernet transfer quality should be revealed. Therefore the determination is needed, how VC size affects packet loss. The model proposed is intended only for UDP protocol.

Data encapsulation procedure is performed, when user data gets into the network node. After the GFP frames are formed, the data is placed in to VC's. VC type determines the sort of VC, which is used to carry the data stream. Data placing into VC is performed at 125 μs intervals. Empty GFP frames are formed and placed in to VC, if there is no data. In such a way a synchronous data stream is obtained. At first the quality of optical lines is checked; data stream is transferred through the predetermined optical line, if the quality of the latter is good enough. An alternative transmission path is selected in the opposite case, when line quality characteristics are poor.

The asynchronous packets' flow coming to ADM is converted into synchronous one by means of empty VC, i.e. the empty VCs are formed, when there is not any data.

Some different node states are possible depending on different input rate (or intensity λ) and data flow processing (Fig. 3).

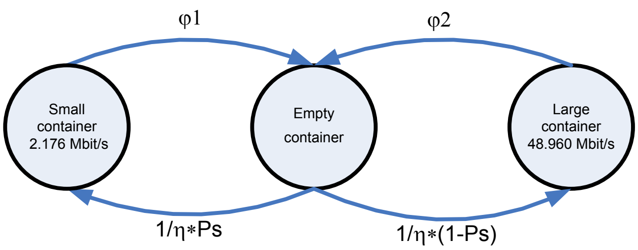

Fig. 3. Different VC forming probabilities

Supposing an 2,176 Mbps rate VC is formed (it indicates the probability of $1/\eta \cdot P_s$), if there is a small amount of data at the moment; empty VC is formed, if there is no data, and the big one, i.e. 48,960 Mbps VC is formed (this state probability is equal to $1/\eta \cdot (1 - P_s)$), when there is a large data flow at the input; here $1/\eta$ is an average duration of empty VC. If, in latter case, data stream intensity is more than three requests at once, losses appear. φ_1, φ_2 are a durations of 2,176 Mbps and 48,960 Mbps VCs formed (packet service time) correspondingly.

Characteristics of X, Y, Z should be taking into account, when creating the graph of possible node states. X indicates the number of requests coming to input; Y indicates 2,176 Mbps rate VC formed, and Z – 48,960 Mbps VC, correspondingly (Fig. 4).

The system presented (Fig. 4) can be described with Markov equations (1).

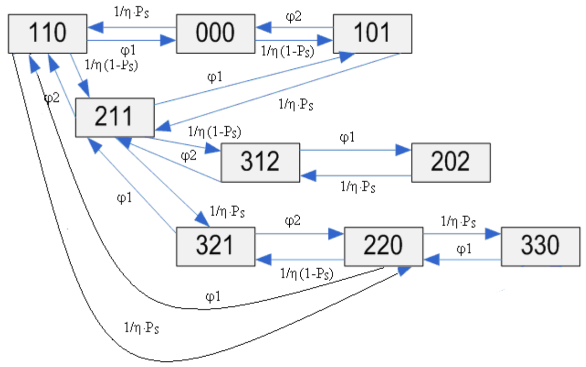

Fig. 4. Node states probabilities graph

$$
\begin{cases}
P_{000}+P_{101}+P_{110}+P_{211}+P_{312}+P_{202}+P_{321}+P_{220}+\\
+P_{330}=1;\\
P_{000}\cdot\left(\frac{1}{\eta}\cdot P_S+\frac{1}{\eta}\cdot(1-P_S)\right)-P_{110}\cdot\varphi_1-P_{101}\cdot\varphi_2=0;\\
P_{101}\cdot\left(\frac{1}{\eta}\cdot P_S+\varphi_2\right)-P_{000}\cdot\left(\frac{1}{\eta}\cdot(1-P_S)\right)-P_{211}\cdot\varphi_1=0;\\
P_{110}\cdot\left(\varphi_1+\frac{1}{\eta}\cdot(1-P_S)+\frac{1}{\eta}\cdot P_S\right)-P_{220}\cdot\varphi_1-P_{211}\cdot\varphi_2-\\
-P_{000}\cdot\left(\frac{1}{\eta}\cdot P_S\right)=0;\\
P_{211}\cdot\left(\varphi_1+\varphi_2+\frac{1}{\eta}\cdot(1-P_S)+\frac{1}{\eta}\cdot P_S\right)-P_{101}\cdot\left(\frac{1}{\eta}\cdot P_S\right)-\\
-P_{110}\cdot\left(\frac{1}{\eta}\cdot(1-P_S)\right)P_{312}\cdot\varphi_2-P_{321}\cdot\varphi_1=0;\\
P_{312}\cdot(\varphi_1+\varphi_2)-P_{110}\cdot\left(\frac{1}{\eta}\cdot(1-P_S)\right)-P_{202}\cdot\left(\frac{1}{\eta}\cdot P_S\right)=0;\\
P_{202}\cdot\left(\frac{1}{\eta}\cdot P_S\right)-P_{312}\cdot\varphi_1=0;\\
P_{321}\cdot(\varphi_1+\varphi_2)-P_{211}\cdot\left(\frac{1}{\eta}\cdot P_S\right)-P_{220}\left(\frac{1}{\eta}\cdot(1-P_S)\right)=0;\\
P_{220}\cdot\left(\varphi_1+\frac{1}{\eta}\cdot(1-P_S)+\frac{1}{\eta}\cdot P_S\right)-P_{321}\cdot\varphi_2-P_{330}\cdot\varphi_1-\\
-P_{110}\cdot\left(\frac{1}{\eta}\cdot P_S\right)=0;\\
P_{330}\cdot\varphi_1-P_{220}\cdot\left(\frac{1}{\eta}\cdot P_S\right)=0.
\end{cases}
\quad (1)
$$

Losses are received at these system states: 312; 321 and 330. Service time relates to requests intensity (2), when mean duration of VC empty is 1 ms.

$$\lambda=\varphi/VC\,. \quad (2)$$

Packet loss dependence regarding system service time is presented in Fig. 5 and Fig. 6 (2,176 Mbps VC is formed and 48,960 Mbps VC is formed, correspondingly). Mean duration of VC empty is 1 ms, and VC forming probability is equal to P_s=0.5.

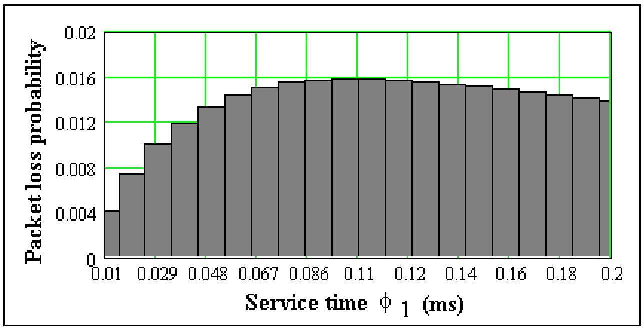

Fig. 5. Packets loss probability dependence from service time φ_1 (or number of requests) in the system

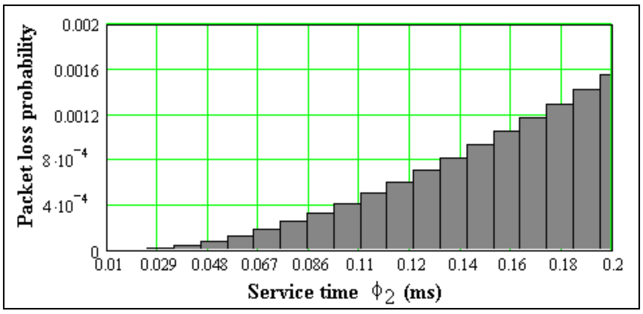

Fig. 6. Packets loss probability dependence from service time φ_2 (or number of requests) in the system

One can see that packet service time increases directly, when the number of requests (or packets) increases. This will result in losses being increased as well (Fig. 5). The ratio of losses proportionally increases up to 0.1 ms and further it has the tendency to decrease, when the number of requests becomes higher (Fig. 6). As it is seen in both cases, packet losses increase, when the virtual container formed is smaller.

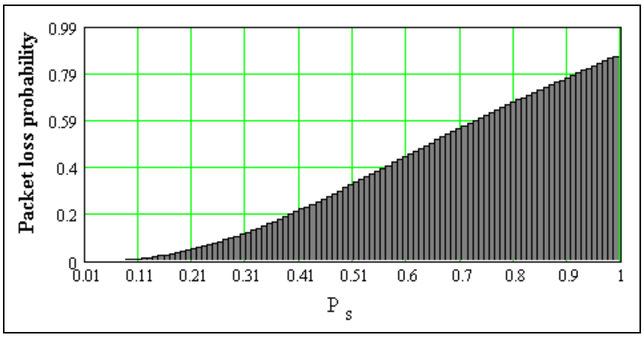

Fig. 7. Packet loss dependence from probability, that virtual container is small

The dependence of packet loss regarding VC size is obtained, when taking into account a variable probability of 2,176 Mbps VC forming; φ_1, φ_2 measures are constant and mean duration of VC empty is 1 ms (Fig. 7). As it seen the packet loss probability increased, if a little VC is formed.

Ethernet packet's service time proportionally increases depending on the growth of packet number arrived at the node and packet loss rate grew up as well.

Experimental analysis

As the influence of SDH network node configuration was estimated, it would be worthwhile to ascertain the influence of TCP and UDP transport protocol specifics to the Ethernet data transferring quality, i.e. to determine, how big the impact of datagram size and transport protocol window size is for Ethernet packet loss.

STM-1 level ring optical network with 100 Mbps Ethernet input line is chosen as the object of measurements (Fig. 8). The Iperf Client program packet was installed in PC1 and Iperf Server – in PC2. PC1 generates datagram flow of STM – 1 level order and transmits it through the switch to one of the STM-1 level ring network node (ADM), i.e. to 100 Mbps Ethernet port of ADM. Then flow is transmitted to PC2 from ADM Iperf Client datagram. Network quality characteristics should be evaluated from the latter.

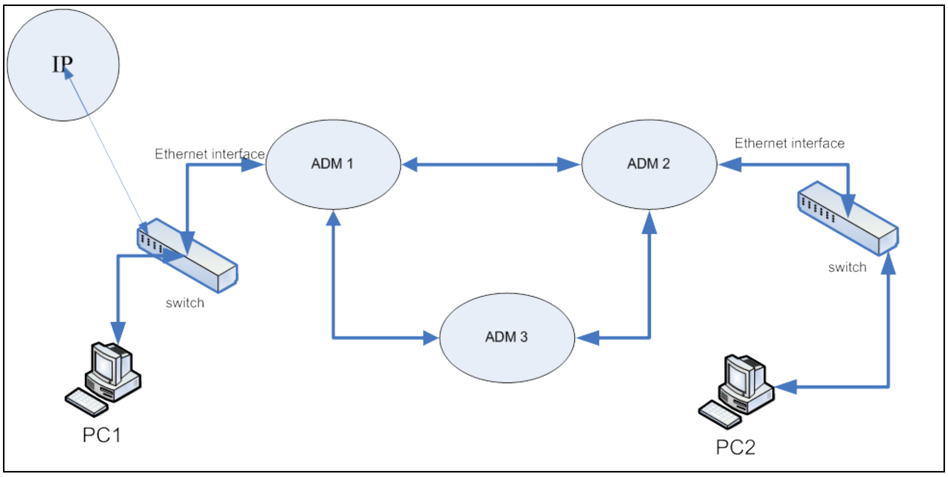

Fig. 8. The structure of network investigated

Investigation of TCP and UDP protocol influence to the transferring quality

The question is how Ethernet packets' transferring quality in STM-1 level network is affected with TCP protocol windows size and VCG formed at the node? It can be determined by transmitting various size TCP datagrams. The difference between columns of diagrams shows the different datagrams transmitted, when measurement time interval was equal to 10s for all datagrams (Fig. 9).

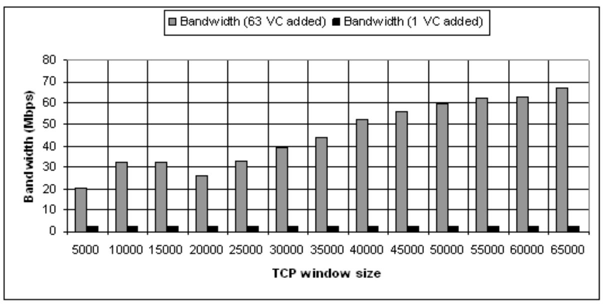

Fig. 9. Influence of different TCP windows applied onto network throughput

It is obvious, that network throughput obtained primary depends from TCP window size applied (Fig. 9). Throughput is growing directly increasing TCP windows or VC group; when using 65000 Bytes window size 65 - 68 Mbps transmission rate was achieved, comparing with 20 Mbps – in case of 5000 Bytes window size is applied. This can be explained in such a way, that if the bigger transmission window is applied, the bigger amount of information is transmitted at once.

The influence of UDP transport protocol for the packet transmission quality is analysed, using various datagrams and different virtual channel throughputs (i.e. different VCG at the node formed). It is worth to mention, that Iperf generates data flow of STM – 1 level order.

The results obtained are discrepant enough, when measuring packet loss dependence from datagram size (Fig. 10). The part of useful information increases in packet, when UDP datagram increases. Packet loss decreases in such a case. This tendency is perceptible in different rates. It can be explained in two ways: little data packets carry small enough amount of useful information comparing with overhead information in a packet. The big amount of overhead information deteriorates the performance of network equipment and the packet loss is increasing consequently. Another way to explain the dependence of packet loss from UDP datagram size could be related with Iperf performance characteristics. The number of packets transferred decreases almost in a half, when increasing datagram size. For example, 2967 packets of 200 Bytes were transferred in 2.2s and 1471 packets of 400 Bytes were transferred at the same time interval already. This reason defined the relatively smaller packet loss possibility, when transferring smaller flow of Ethernet packets.

As size of optimal datagram transferred varies from 800 to 1000 Bytes, the throughput decreases, when datagram size increases some more (Fig. 11).

An experimental jitter values obtained are the same character as theoretical one i.e. increases if datagram size increases. This tendency is obvious with a little variation in different rates (Fig. 12).

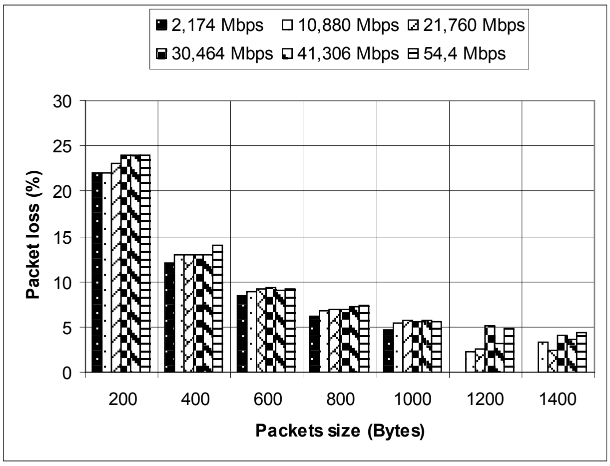

Fig. 10. Dependence of packet loss from different UDP datagrams

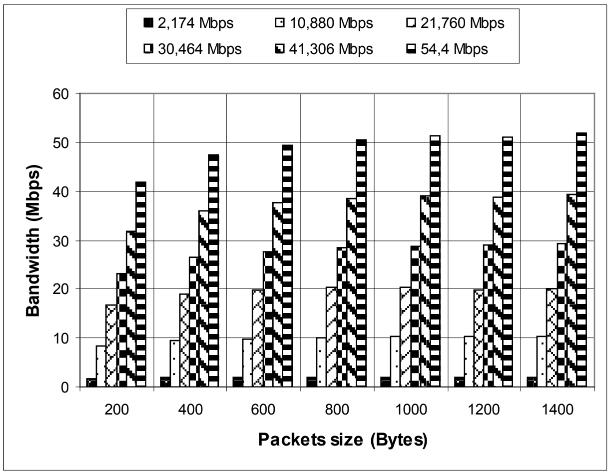

Fig. 11. Virtual channel throughput dependence from datagram size

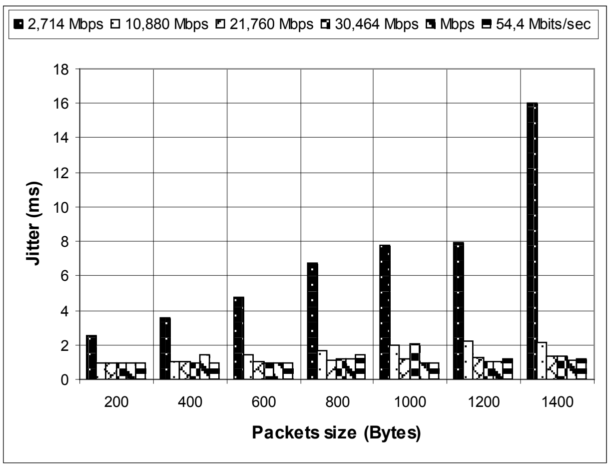

Fig. 12. Dependence of jitter from datagram size

Conclusions

The question about selection of proper ADM configuration in STM-1 ring network is answered with ADM node mathematical model created using Markov chain methods. The overall transfer rate is strictly limited with parameters of ADM, because of the number of VCs assigned to one input port. Configuring larger VC containers in ADM predetermines smaller values of Ethernet packet loss.

The analysis of influence of TCP and UDP transport protocols to Ethernet data transfer quality was made. It is determine, that the larger TCP windows assure the bigger amount of data flow transmitted at once. This would meet the requirements of Ethernet technology very effectively at high rates. The only shortcoming is that, if network line throughput is not sufficient, the loss probability of larger size packets and data delay value had increased because of additional packet transfer. The analysis of UDP transport protocol has showed, that big amount of small voice packets increased packet loss rate in a little, and it will have an insignificant influence to final quality of real time services. However, the results showed that, in spite of this, larger voice packets assured better network throughput, smaller packet loss and jitter values.

The analysis of encapsulation to Ethernet transferring quality should be done, as well as an investigation of the higher STM levels, using more flow generator with more functions. Also would be purposeful to analyse SCTP protocol characteristics. It can be the direction for future investigations of this area.

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Questions were analysed regarding the influence of SDH network nodes configuration specifics and TCP and UDP transport protocols on to Ethernet data transferring quality. The influence of virtual container (VC) size on ADM performance was revealed by means of mathematical model created, and it was determined, that the better choice was choosing the larger size VC, because of the smaller packets loss values obtained. The real experiments in STM-1 network helped in solving which transport protocol would be more suitable for data transferring. In case of TCP investigation, TCP windows or VC group formed in ADM size have showed the high and direct influence on to network throughput obtained, because the larger TCP windows applied assured the bigger amount of data transmitted at once. This would be meeting the requirements of Ethernet technology very effectively in high rates. The analysis of UDP showed that the most optimal UDP datagram size for Ethernet packets transferring would be of 800 – 1000 Bytes order. Investigation of UDP suitability for voice packets transferring showed that even if the smaller packet loss will have an insignificant influence on to final quality of real time services, the larger voice packets (800 – 1000 Bytes order) will assure better network throughput, smaller packets loss and Jitter values. Ill. 12, bibl. 12 (in English; abstracts in English and Lithuanian).

R. Jankūnienė, J. Prikšaitis. Transporto lygmens įtaka eterneto paslaugų kokybei // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2010. – Nr. 9(105). – P. 13–18.

Analizuojama transporto protokolų ir SDH tinklo mazgų konfigūravimo specifikos įtaka eterneto (Ethernet) duomenų perdavimo spartai ir jų praradimui STM-1 lygmens tinkle. Virtualiojo konteinerio (VC) dydžio įtaka ADM mazgo kokybinėms charakteristikoms buvo nustatyta naudojantis sukurtu ADM mazgo matematiniu modeliu. Pasirodo mažiau paketų prarandama, kai VC yra didesnis. Atlikti realūs eksperimentai STM-1 tinkle padėjo nustatyti transporto protokolų tinkamumą eterneto duomenims perduoti. Nustatyta, jog taikant TCP didelę ir tiesioginę įtaką tinklo pralaidumui turi TCP lango dydis, nes kai TCP langas didesnis vienu metu perduodama daugiau duomenų. Tai atitinka eterneto technologijai keliamus reikalavimus, esant didelei perdavimo spartai. UDP protokolo taikymo analizė parodė, kad optimalus UDP paketo dydis eterneto duomenims perduoti yra 800–1000 baitų. Tiriant UDP tinkamumą balso paketams perduoti, nustatyta, kad nors mažesnių paketų praradimai turi mažiau įtakos realių paslaugų kokybei, didesni balso duomenų paketai (800–1000 baitų eilės) užtikrina didesnį tinklo pralaidumą, mažesnius paketų praradimus bei vėlinimo fliuktuacijas. Il. 12, bibl. 12 (anglų kalba; santraukos anglų ir lietuvių k.).