



Journal Website:
<https://theusajournals.com/index.php/ajast>

Copyright: Original
content from this work
may be used under the
terms of the creative
commons attributes
4.0 licence.

ABOUT QOS CLASSES OF MULTIMEDIA COMMUNICATION NETWORKS

Submission Date: December 17, 2022, **Accepted Date:** December 22, 2022,

Published Date: December 27, 2022

Crossref doi: <https://doi.org/10.37547/ajast/Volume02Issue12-05>

Bayjonova Lyudmila Egamberdievna

Senior Lecturer Of The Department Of Electronics And Radio Engineering, Tashkent University Of Information Technologies Named After Muhammad Al-Kharezmi, Uzbekistan

ABSTRACT

This article is devoted to the actual topic of our time.

The article describes the QoS classes of multimedia communication networks, the data transfer channel for QoS classes through the UNI-UNI interface, and the network QoS channel. The article analyzes the methods of decomposition of QoS indicators in an NGN class network, the principles of constructing an NGN network that are essential for further research, a set of QoS indicators in an NGN class network, and research results regarding the decomposition of QoS indicators.

KEYWORDS

Packet switching, multiservice networks, static distribution method, decomposition of quality indicators, QoS classes

INTRODUCTION

Analysis of the principles of decomposition of QoS indicators in an NGN class network. Principles of constructing an NGN class network. The evolution of communication networks is accompanied by the emergence and spread of new technologies. Their most massive penetration is observed in networks with

the largest number of users, i.e. in the cellular mobile network and in the public switched telephone network (PSTN). The process of upgrading these networks is aimed at creating the NGN network. This process involves a complete transition to packet switching technology and building a network that meets the

basic principles defined in international recommendations. Since this process requires the modernization of a significant amount of equipment of existing communication networks, it cannot be carried out in a short period of time. PSTN development programs at different levels of the hierarchy must be carefully coordinated so that the process of transition to packet switching technology is carried out while maintaining the functionality and quality of services in communication networks.

The construction of the next generation network is carried out on the basis of the so-called expanding core principle. At the initial stage of NGN formation, packet technologies are used only in the "core" of the network. First, a long-distance IP network must be created. Some countries have already implemented

such projects. The cost of such projects is determined by the size of the country.

With the modernization of urban and rural telephone networks, packet technologies are beginning to be used outside the long distance network. This process consistently involves networks of different levels, so it can be considered as an expanding core.

Figure 1. shows a model illustrating the options for establishing a connection between telephone sets of two local networks. Switching technologies can change at different points (they are indicated by letters of the Latin alphabet). These points are located on the borders with the NGN network. The classical approach to building NGN is based on the fact that these points are gradually shifting towards subscriber terminals.

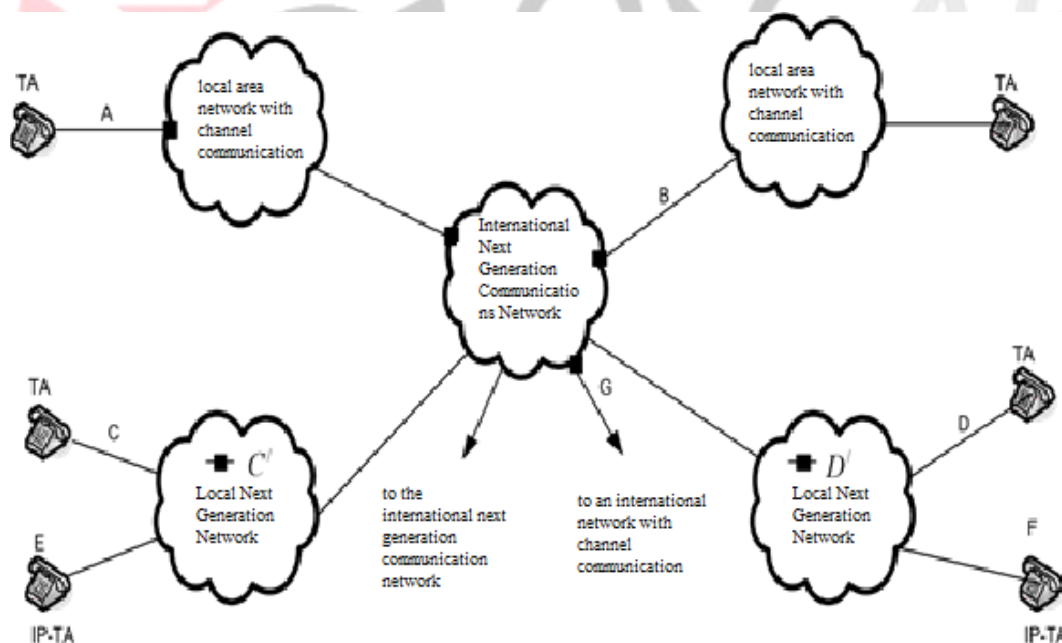


Figure 1. - Interaction of circuit-switched and packet-switched networks

The main criterion for evaluating the functioning of communication networks is the quality of services, since a communication service is the product for the production of which any communication networks are built. Therefore, one of the important criteria in choosing solutions, upgrading existing networks, building new communication networks and their operation is to ensure the quality of services.

The quality of the provision of most modern communication services is ensured by meeting the standards for the physical parameters of the functioning of communication networks, QoS (Quality of Service) parameters. The set of these parameters is defined in ITU-T recommendations and industry documents. It should be noted that the development of IT technologies and communication technologies leads to the development of new services and a change in perceptions of quality; therefore, it is possible that the regulations will also evolve.

This article discusses approaches and methods for decomposition of QoS indicators between UNIs (between user-network interfaces) for packet switching technology. Networks built on the basis of "packet switching" technology are significantly different from networks that have been implemented on the "circuit switching" technology. These differences can be viewed from several points of view. This article focuses on the differences in traffic QoS metrics. The main features of packet-switched networks can be summarized as follows:

- First, modern packet-switched networks are multiservice, that is, they serve the traffic of various services, such as voice, video, and data. Usually they are built on the basis of the NGN concept.
- Secondly, for packet-switched networks, performance indicators are introduced that

characterize the process of servicing packet traffic. First of all, these are probabilistic-temporal indicators and reliability indicators, such as the mathematical expectation of packet transfer time (IPTD), packet transfer time variation (IPDV) and packet loss probability (IPLR).

- In the tasks of planning and operating networks of the NGN class, determining the requirements for their resources (the values of the throughput of communication paths and the performance of routers), it is required to evaluate the numerical values of the parameters that must be provided in the designed or operated network section. In the general case, the considered network fragment (the network of the telecom operator) participates in the provision of communication services, i.e. only part of the data transport route passes through it. The quality standard must be ensured on the entire route, i.e. between user interfaces UNI-UNI, which in general can be in networks. To assess the target values of quality indicators on the designed or operated section of the network, it is necessary to decompose the main IPTD, IPDV and IPLR, normalized on the UNI-UNI section, for all the main network elements [26,1]. In ITU-T Recommendation Y.1542, there are two approaches for solving end-to-end IP packet delivery between two IPs, the distribution approach and the impairment accumulation approach.

Based on these two approaches, several methods are recommended for meeting the performance standards, which are given in recommendation Y.1541. Static methods are divided into the following types:

- static divider method;
- static allocation method based on the reference path;
- method of weighted elements.
- pseudo-static method (left for further study);

- signaling methods;
- interval distribution method;
- method of accumulation of distortions.

In addition, Recommendation Y.1542 lists methods left for further study.

It is contemplated that methods called "Costed Bids Method" and "Bid Discovery Using a Global Registry" may be proposed. These methods take into account technical and economic factors. This approach seems to be quite appropriate, taking into account the various possibilities of the telecommunications market participants in terms of ensuring the required level of quality of traffic service. They ensure compliance with quality of service requirements at a minimum cost for telecom operators and info communication service providers.

The body of Recommendation Y.1542 favors the static allocation method based on the reference path.

Recommendation Y.1542 outlines the key proposals for a method for decomposing all used traffic quality of service metrics. In addition, new methods of decomposition of IPTD, IPDV and IPLR indicators will solve the problem of concluding service level agreements - SLA, which are necessary for some part of customers and telecom operators.

Next, we will consider the decomposition of the most significant quality indicators IPTD, IPDV and IPLR on the example of the NGN network model, which provides for the participation of three telecom operators in the provision of services. Detailing implies consideration of one of the most important properties of the network - its ability to minimize the duration of IP packet delay by each telecom operator. It is proposed to take this possibility into account when distributing income from the type of service provided.

To ensure that the values of traffic quality of service indicators comply with the standards when exchanging IP packets between user-network interfaces, it is necessary to develop new methods for calculating them for the next generation communication network. Such a method is the decomposition of traffic service quality indicators. It should also be taken into account that the tendency for operators to enter into SLA agreements may lead to additional requirements for the decomposition procedure.

There are two main approaches to solving this problem. The first is to allocate quality metrics to a finite number of networks that participate in the organization of the IP packet exchange route between two UNIs, based on well-known techniques that have been developed and proposed by the International Telecommunication Union. The second method is based on the principles of concluding agreements between telecom operators, the essence of which is similar to concluding agreements on the level of SLA service quality.

The disadvantage of the second technique is that it does not take into account the state of the NGN network of each operator involved in the provision of the service. As a result, in some cases, the quality of services will be below the required level. The advantage of the methodology of the International Telecommunication Union lies in the simplicity of the calculations used.

The advantage of the technique, which implies the conclusion of agreements between all telecom operators whose networks are involved in organizing the connection, is that it allows you to guarantee the quality of traffic service. On the other hand, the method of decomposition of quality of service indicators that are normalized between a pair of UNIs

becomes noticeably more complicated. However, if appropriate methods for calculating the NGN network are developed, then the advantage of the second option of decomposition of quality indicators will be obvious.

General information about QoS classes.

This article discusses all the requirements of network quality indicators that are used to transfer user information when using packet technologies [5,67]. They are stated as IP layer specification parameters and are defined in ITU-T Recs Y.1540 and Y.1543. It can be noted here that from the user's point of view, these network QoS requirements are not defined for all characteristics, but only as part of the data transmission characteristics.

To determine the QoS classes, the boundaries of the values of network performance indicators between UNI-UNIs are defined. [15,1]. In the event that users (and individual network operators) do not reach the negotiated capacity exceeded, or the traffic and bearer contract is available (as defined in Recommendation ITU-T Y.1540), network service providers are required

to enforce these UNI interface limits together. -UNI for the duration of the session.

The practical network QoS that is assumed for a given flow will depend on the distance and "complexity" that the packet encounters along its path. In this case, the word "complexity" denotes all potential problems that are directly or indirectly related to the entire set of normalized indicators of the quality of service of multiservice traffic.

Often the quality will be higher than that defined by the value limits described in the QoS classes (Table 1.). The symbol "H" in Table 1 denotes an "undefined" or "unlimited" value. In the case where performance is related to a specific parameter whose value is defined as "H", ITU-T does not specify specific requirements for that parameter. In such a case, any requirements defined in Recommendation Y.1541 may be disregarded. Marking a parameter with the value "H" also means that the performance in relation to this parameter can be temporarily arbitrarily degraded over a wide range.

Table 1 – IP network QoS class definitions and network performance requirements

Network performance parameter	The essence of the network performance requirement	QoS classes					
		0	1	2	3	4	5
IP, IPTD packet delivery delay	(Example 1)	100 ms	400 ms	100 ms	400 ms	1 s	H

IP, IPDV Packet Delay Variation	(Example 2)	50 ms (Example 3)	50 ms (Example 3)	H	H	H	H
IP, IPLR Packet loss ratio	(Example 4)	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	H
IP, IPER Packet error rate	(Example 5)	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	H

Note:

1. Fairly long travel times eventually result in low end-to-end latency requirements not being met. In some cases, the IPTD requirements for classes 0 and 2 cannot be met. All service providers face this circumstance, so the range of IPTD requirements is divided into several alternative QoS classes. Delay value requirements for each of these classes are not prohibited by network service providers, but are proposed in agreements where they stipulate, for example, lower delay values. In accordance with the definition of the IPTD delay parameter in ITU-T Recommendation Y.1540, the packet delivery time is included in the IPTD requirement.

When evaluating these requirements, it was assumed that the maximum size of the packet information field is 1500 bytes.

2. The definition of the IPDV parameter requirement (given in ITU-T Y.1540) is a point-to-point variation of IP packet delay. See ITU-T Rec. Y.1540 and Appendix II for

more details on the nature of this requirement. For scheduling purposes, the IPTD delay value constraint can be derived from the upper limit of the IPTD minimum delay value; therefore, the 0.999 quantile limit can be obtained by adding the IPTD value and the IPDV value (for example, 150 ms for class 0).

3. The value of the delay depends on the bandwidth of the communication channels. A decrease in latency is possible if the channel throughput values are higher than the base rate (T_1 or E_1) or the packet size is less than 1500 bytes.

4. The requirements for class 0 and 1 traffic in terms of the IPLR parameter are based on studies showing that a setting of 10^{-3} will not have a significant impact on speech applications and speech codecs.

5. The value set assumes that the main cause of errors is packet loss and that the specified value of the indicator is valid when transmitting packets using IP protocols over a network with asynchronous delivery mode.

The requirements in Table 1 apply to public IP networks when analyzing network sections between two measurement points (reference points) that define the boundaries of an end-to-end IP network. In applying these guidelines, it is assumed that the required values can be provided in conventional implementations of IP networks.

The first column of Table 1 lists the statistical network performance requirements for which are listed in the following columns of the table.

The performance requirement for IP packet delivery delay is an upper acceptable bound for the IPTD value in a multi-packet flow.

Individual packets of a flow may be delivered with a delay greater than the limit given, but the average value of the IPTD delay over the lifetime (observation) of the flow (statistical estimate of the value) must be no more than the value defined in Table 1.

The performance requirements for changing the delay of an IP packet between two breakpoints are based on an upper bound of 0.999 quantiles of the delay value distribution (IPTD) for the packet stream. The use of the 0.999 quantile allows relatively short estimation intervals to be chosen (for example, the minimum allowable interval for this estimation is a time interval during which 1000 packets are transmitted). This allows for the flexibility of the network model when designing delay and queue length control buffers in routers, for which the requirement for an IPLR value of no more than 10⁻³ must be met. Using smaller quantile values can lead to an underestimation of the size of the dejitterization buffer. In this case, the actual packet loss may exceed the requirement for IPLR (for example, with a limit of 0.99 quantile, the packet loss ratio can be 1.1% with the norm IPLR = 10⁻³).

- The performance requirement for the IP packet loss ratio is an upper bound on the IP packet loss for a flow. Individual packets in a stream may be lost in transit; in a short time interval, the loss may exceed the limit, but the probability that any individual packet will be lost during transmission in the stream should be no more than the established limit given in Table 1.
- The assessment of the requirements given in table 1. is carried out as a result of measurements, i.e. not instantly, but over time. During evaluation intervals, subsets of the population of useful packets (defined in ITU-T Rec. Y.1540) are formed. It is desirable that the assessment (measurement) intervals be:
 - sufficiently long and allowed to obtain the required number of packets of the studied stream, taking into account the actual data transfer rate;
 - long enough to reflect the typical usage period (lifetime of the bit stream) for the given service.
 - short enough to provide a reasonable balance of applied performance over each interval (poor performance intervals should not be hidden in an overly long evaluation interval, they should be identified).
 - short enough to address the actual aspects of the measurement.

To perform telephony-related assessments, the minimum interval should be in the range of 10-20 seconds with a packet rate of 50 to 100 per second; the upper limit of the interval should be within a few minutes. The estimated (approximate) value is one minute. In any case, the measurement interval value used should be recorded along with the measured value, with any expected and confidence intervals. Any period of one minute must satisfy the requirements of IPTD, IPDV and IPLR, which are described in Table 1.

Static agreements for a QoS class can be developed by attaching marking packets (eg, high-order bits of the type of service or a differentiated services code pointer) to a particular service class. The protocols that are designated to support dynamic QoS requests between users and network service providers, as well as between network service providers themselves, are still subject, according to the ITU, to study [8,5]. By developing these protocols and support systems, users and network operators will be able to request and receive different QoS classes at the flow level. In this regard, certain technical characteristics required for different services and applications can be combined, evaluated and adopted to solve operational problems.

Data link for QoS classes over UNI-UN interface

The packet in each stream usually follows a specific path. Any stream that satisfies the relevant performance requirements can be considered fully compliant with the guidelines set out in ITU-T Recommendation Y.1541.

We will call the traffic served by the communication network between user interfaces with the network (user-network) "end-to-end" traffic[44,3]. The performance requirements between user-network

interfaces are defined for IP protocol metrics according to IP packet transmission reference events. These IP requirements for UNI-UNI interfaces apply from the end user-network interface to the user-network interface at the other end of the span (Figure 2). Let the route between these two user-network interfaces pass through several network segments (NS - Network Segment) and inter-network links that provide the transfer of IP packets from the UNI interface at the sender side (SRC) to the UNI interface at the other end of the destination (DST). The lower layer protocols, covering the IP layers from layer 1 to layer 3, according to the Open Systems Interconnection (OSI) model [45], are also considered part of the IP network. NS network segments (defined in ITU-T Rec. Y.1540) are equivalent to carrier domains and may include IP network architectures as described in ITU-T Rec. E.651 [94] and Y.1231. The data link shown in Figure 2 is an adaptive performance model from ITU-T Recommendation Y.1540.

The consumer equipment includes terminal equipment (TE), such as a host and a router or, if available, a local area network (LAN). It is assumed that there is only one user in several applications.

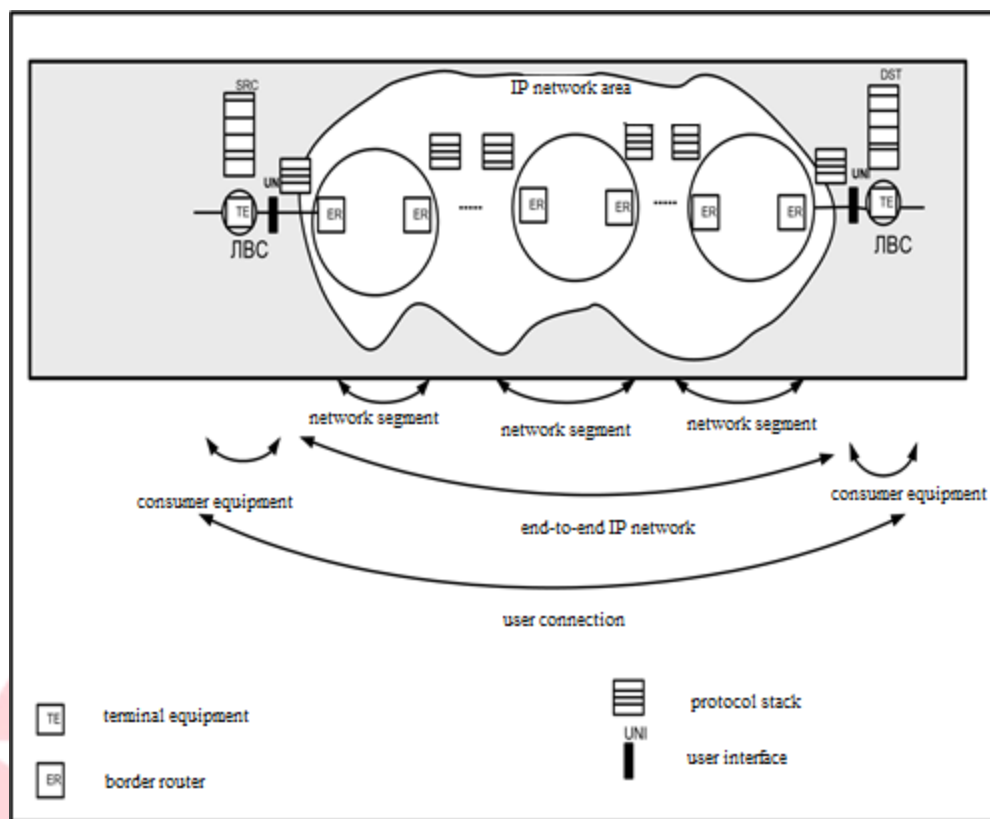


Figure 2 - UNI-UNI data link for network QoS requirements

The edge routers connected to the terminal equipment must also act as access gateways. Reference channels have the following characteristics (attributes):

- 1) An IP network can provide user-to-user, user-to-host, and other connection endpoint options.
- 2) Network segments can be represented as areas of the network between border routers and an indefinite number of internal routers that perform various functions.
- 3) The number of network segments in a given path may depend on the offered class of service along with the topology and geographic extent of each network segment.

- 4) The scope of Recommendation Y.1541 allows one or more network segments to be used on a route.
- 5) Network segments supporting the transmission of packets in a bitstream may change during its lifetime.
- 6) IP connectivity extends beyond international network boundaries, but does not follow identical circuit switching conventions (eg, there may be no identifiable gateways at the international boundary).

Network QoS classes. The article deals with the (currently defined) traffic QoS classes. Each such class has a certain combination of restrictions on performance values. 1. The table contains rules that

indicate the possibility of using each of the traffic service classes. These rules do not impose restrictions on the choice of any particular class of service in a predetermined environment[8,3].

Quality of Service requirements apply to public IP networks. It is assumed that the QoS requirements can be achievable on conventional IP network implementations. The network service provider's obligation to the user is to deliver packets in a manner that ensures that all applicable requirements are met.

The vast majority of IP circuits claimed to comply with ITU-T Y.1541 should satisfy these requirements. For some parameters, performance on short and/or easy routes can be significantly better than the specified values.

The evaluation of parameters in the communication network is carried out according to the data obtained as a result of measurements (observations). In this case, the accuracy of the estimates obtained depends on the number of observations and on the duration of the observation interval. For the parameters IPTD, IPDV and IPLR, the estimated time interval is assumed to be one minute. In any case, both the measured value and the duration of the observation interval must be recorded. Any period of time in one minute must satisfy the specified requirements. Individual Service Providers may offer agreements with better network performance values than those provided by the general requirements.

In the general case, a communication session can be represented as consisting of three phases - establishing a connection, transmitting information, and terminating the connection. ITU-T Y.1540 considers only the session phase, the IP packet delivery phase.

Recommendation ITU-T Y.1540 defines the following parameters that characterize the delivery of IP packets:

IP packet transfer delay (IPTD). The IPTD parameter is defined as the amount of time required to deliver a packet between source and destination for all packets, both successfully transmitted and packets delivered with errors.

IP packet delay variation (IPDV). IP packet delivery delay variation, or jitter, is manifested in the fact that successive packets arrive at the recipient at random times.

IP packet loss ratio (IPLR). The IPLR loss ratio is defined as the ratio of the number of packets lost to the total number of packets in a selected set of transmitted and received packets.

IP packet error ratio (IPER). The IPER ratio is defined as the number of packets received with errors divided by the sum of successfully received packets and packets received with errors. The IPER value determines the portion of the IPLR that is due to packet loss due to corruption. For this reason, there is no point in studying the value of IPER from the point of view of its subsequent decomposition.

Also, the ITU-T Y.1541 recommendation defines the numerical values of the parameters that must be performed in IP networks on international routes connecting user terminals [24,2]. Quality objectives are divided into different QoS classes, which are defined depending on the applications and network mechanisms used to provide them.

One of the basic services of communication networks is a telephone service, so let's consider the basic principles of its implementation. Among the basic principles for the construction and development of

telephone networks, it is advisable to single out three aspects that are directly or indirectly related to the issues discussed in the article:

- network structure at all levels of the hierarchy;
- indicators of the quality of service for voice traffic;
- services provided to subscribers.

The telephone network model proposed by the ITU in the Y series of recommendations is shown in Figure 3. This model is universal for communication networks of any kind, including NGN networks.

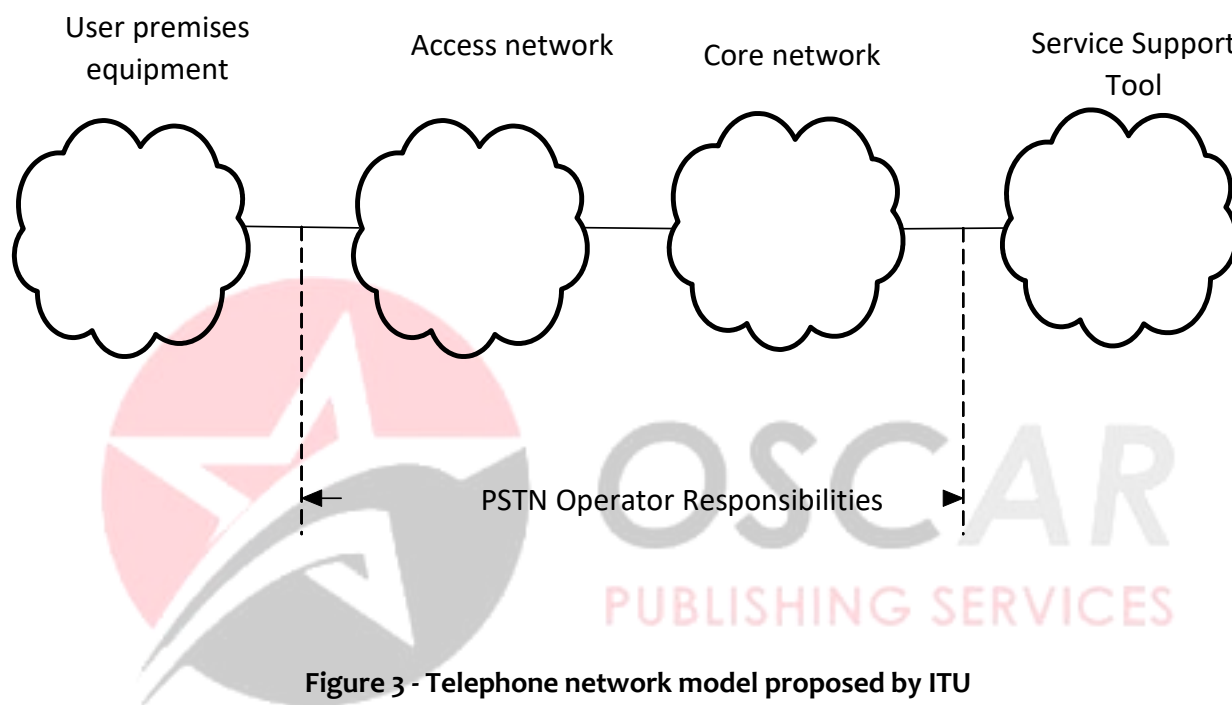


Figure 3 - Telephone network model proposed by ITU

In the basic network model (core network), it is customary to conditionally allocate four fragments - Figure 3. These fragments form the upper levels of the PSTN hierarchy. The bottom layer is the access network. Strictly speaking, the access network is an integral part of any local telephone network. Its allocation as a separate "cloud" is explained by a number of important properties of the access network.

CONCLUSION

1. One of the tasks of building communication networks of the NGN class is to support normalized quality indicators for the provision of communication

services. It can be provided by monitoring the quality of service parameters of QoS traffic in a packet-switched network.

2. The indicators of the quality of service for traffic in the NGN network are the parameters that have the greatest impact on the quality of service provision. These include parameters characterizing the speed and reliability of data (packet) delivery, defined by ITU-T recommendations as IPTD, IPDV and IPLR.

3. Ensuring the quality of service provision in the NGN network is achieved by normalizing these parameters on the data delivery route, which is defined as the



route between user interfaces (UNI-UNI). Normative values of QoS parameters are defined in ITU-T recommendations for various traffic service classes.

4. In general, the UNI-UNI traffic route passes through the networks of several telecom operators. Therefore, the task of ensuring the quality of the service is solved by the interaction of the networks of all operators involved in servicing traffic. To solve this problem, the decomposition of quality indicators (standard values) into sections of the UNI-UNI route in the areas of responsibility of each of the telecom operators is required.

5. Currently, only one of the options for solving the problem of decomposition of the IPTD indicator is described in the ITU-T materials. The task of decomposition of IPLR and IPDV indicators currently does not have a definite solution. It should also be noted that the described IPTD decomposition method does not take into account the peculiarities and interests of telecom operators.

6. Thus, the development of a method for the decomposition of quality indicators is relevant. Its

solution, taking into account the interests of telecom operators, will optimize the relationship between telecom operators and avoid unnecessary costs for re-equipment of networks, while ensuring normalized requirements for the quality of service for multiservice traffic.

REFERENCES

1. Бакланов И.Г. NGN: принципы построения и организации. /– М.: Эко-Трендз, 2008. – 400 с.
2. Балькин Г., Михайлов В., Москалец В., Хиленко В. Киевская городская сеть: переход на пакеты. – Сети и телекоммуникации, № 1 – 2, 2004.
3. Башарин Г.П. Лекции по математической теории телетрафика. – М.: Издательство Российского университета дружбы народов, 2007. – 268с.
4. Бертсекас Д., Галлагер Р. Сети передачи данных. – М. Мир. 1989. – 544 с.
5. Битнер В.И., Попов Г.Н. Нормирование качества телекоммуникационных услуг. – М.: "Горячая линия – Телеком", 2004. – 312 с.