RATIONING THE MAIN PARAMETERS OF THE QUALITY OF NETWORK SERVICES

DOI: 10.36724/2072-8735-2020-14-11-72-76

Manuscript received 08 September 2020;
Accepted 26 October 2020

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Keywords: *Quality of Service, Communication, Latency, Jitter, Packet loss, Voice over IP technology, Packet-switched network, High Definition Television*

The European Telecommunications Standards Institute (ETSI) recommends that IP-telephony networks be divided into four classes according to the quality of service (quality of network services) QoS, the main indicator of which is packet delay (Y.1541). ITU-T Recommendation G.114 for public switched telephone networks provides delays close to ETSI gradations that correspond to different types of communication. The quality of the service from the user's point of view can be expressed by a set of indicators. These metrics are described in terms that are understandable to both the user and the service, and are independent of the network structure. Quality of service indicators are focused primarily on the effect perceived by the user, must be guaranteed to the user by the service and be objectively measured at the point of access to the service (ITU-T Recommendation I.350). ITU-T Recommendation E.862 provides possible approaches to accounting for the economic losses of the operator (in planning, design, operation and telecommunication network) and the user associated with technical failures. Operators of networks, working in market conditions, are interested in assessing possible losses due to failures and in comparing them with the costs of increasing the reliability of their funds.

Для цитирования:
Заин-аалабдаин аль-Намер. Нормирование основных параметров качества сетевых услуг // T-Comm: Телекоммуникации и транспорт. 2020. Том 14. №11. С. 72-76.

For citation:
Zain-aalabdain Al-namer. (2020) Rationing the main parameters of the Quality of Network services. *T-Comm*, vol. 14, no.11, pp. 72-76. (*in Russian*)

The quality of network services (QoS – Quality of Service) on the recommendation of ITU-T E800-user of a telecommunication network is not interested in the structural network and how it provides the necessary service. The user evaluates the quality of this service, comparing it with the quality of similar services. ITU-T Recommendation E800 defines the quality of service of network services as the total effect of the characteristics of the service, which determines the degree of user satisfaction with the service. ”

The quality of network services on the Internet of Things (QoS – Quality of Service) according to ITU-T recommendations –Y2066 and Y2068- It is recommended to maintain based, periodic and automatic communication modes between devices or IoT users while maintaining network performance. Communication device support is required to meet automatic communication requirements.

The European Telecommunications Standards Institute (ETSI) recommends that IP-telephony networks be divided into four classes according to the quality of service (quality of network services) QoS, the main indicator of which is packet delay (Y.1541). ITU-T Recommendation G.114 for public switched telephone networks provides delays close to ETSI gradations that correspond to different types of communication.

Table 1

Network classes for quality of service ITU-T latency recommendations are mapped to ETSI QoS classes.

Classes of Networks by Quality of Service		**Best**	**H**	**Medium**	**Lower**
Latency	ETSI	< 150 ms	< 250 *ms*	< 350 *ms*	< 400 *ms*
	ITU-T	< 150 *ms*	< 260 *ms*	< 400 ms	< 400*ms*

For each class of quality networks, ETSI provides the following subjective assessment of speech quality:

Best – equivalent or better than the public switched telephone network (PSTN) with the G.711 codec.

High – equivalent to a PSTN with a G.726 codec (32 kbit / s) or good quality GSM cellular communication with an EFR codec of increased reliability (12.2 kbit / s).

Medium – equivalent to GSM cellular communications with RF codec (13 kbit / s).

Lower- involves the use of VoIP on the Internet.

Table 2 shows the estimates of the MOS method for the quality of perception.

Table 2

Estimates of the MOS method

Assessment	Q lity	Level distortions
5	Beautiful	Not felt
4	Good	Perceptible but not irritating
3	Satisfactory	Perceptible and somewhat annoying
2	Insufficient	Annoying but not protesting
1	Bad	Very annoying and provoking

In the NGN concept, recommendations on the quality of service for various services were further developed. Recommendation Y .1541 gives the upper bounds for the network characteristics of the QoS classes to which NGN services are assigned (Table 3).

Table 3

Upper bounds for QoS network performance

Network specifications	Classes QoS					
	Classes 0	Класс 1	Класс 2	Класс 3	Класс 4	Класс 5
IPTD (Average Packet Delivery Delay)	100мс	400мс	100мс	400мс	1 с	-
IPDV (packet delivery delay variation)	50 мс	50 мс	-	-	-	-
IPLR (Packet Loss Ratio)	$1x10^{-3}$					-
IPER (Packet Error Rate)	$1x10^{-4}$					-

Recommendation Y .1541 establishes the following correspondence between quality of service classes and applications:

Class 0 – real-time applications that are sensitive to jitter , characterized by a high level of interactivity (VoIP , video conferencing);

Class 1 – real-time, interactive, jitter-sensitive applications (VoIP, video conferencing);

Class 2 – data transactions characterized by a high level of interactivity (for example, signaling);

Class 3 – data transactions, interactive;

Class 4 – applications that are sensitive to the loss of information during its transmission over the network (short transactions, data arrays, streaming video);

Class 5 – Traditional use of IP networks with default transmission characteristics

Designations: IPTD – IP Packet Transfer Delay;
IPDV – IP Packet Delay Variation;
IPLR – IP Packet Loss Ratio;
IPER – IP Packet Error Rate.

Recommendation Y.1541 provides specifications for a set of parameters that are related to measuring the real values of network characteristics – observation period, test packet length, number of packets, etc. In particular, when assessing the quality of transmission of voice packets in IP telephony, the minimum observation interval should be on the order of 1-20 with a typical transmission rate of 50 packets per second. The recommended measurement interval for delay, jitter and loss should be at least 60 s.

Assessment of the quality of service of message flows on the MCC link

Real-time traffic in the MSS is close to the traffic of circuit-switched networks, and classical methods of teletraffic theory are well suited for analyzing the quality indicators of its service.

As the main characteristic of the quality of service, the probability of blocking the establishment of a connection is taken when, due to the lack of free resources of the network link, the access control mechanism does not allow starting the transmission of messages.

Let the MCC link receive n real-time streams. Let us denote the speeds of the incoming flows through v1, v2, ..., vn, and the speed of the communication line at the network link through G. Let us call the unit of the channel resource the greatest common divisor (GCD) of these speeds

$$e = GCD\,(v1, v2, \dots, vn, G).$$

As a result, the integer representation of the speed of the communication line on the network link has the form $v = G\,/\,e$ channel units, and the integer requirement for the service rate for the k-th stream will have the form $b_\kappa = v_\kappa\,/\,e$, k = 1, 2, ..., n, channel units.

Example.

Let the line with a speed of C = 10 Mbit/s receive a stream of voice messages at a rate of v1 = 25 kbit/s (G.723.1 codec) and a stream of videoconferencing messages at a rate of v2 = 100 kbit/s.

$$e = gcd\,(25, 100, 10{,}000) = 25.$$

GCD- Greatest Common Diviso

$v = (10{,}000)\,/\,25 = 400\,units,$

$b_1 = \frac{25}{25} = 1\,channel\,unit,\ \ b_2 = \frac{25}{25} = 4$ channel units. Thus, out of 400 channel units of a communication line, each voice message occupies 1 channel unit for service, and each videoconference message occupies 4 channel units.

The quality of telephony systems is influenced by three main parameters of the communication channel:

1) atency. When transmitting voice or video, there are certain requirements for the maximum allowable delay. Various studies show that for a normal dialogue it is necessary that the "double delay" in voice transmission does not exceed 250-300 ms (delay budget). When this threshold is exceeded, participants begin to experience discomfort and seek to end the conversation. Thus, for conducting a comfortable conversation, the one-way delay should not exceed 150 ms (channel delay + codec algorithmic delay), which coincides with the ITU-T G.114 recommendation. To reduce the delay introduced by the network, you must use QoS (QualityofService).

2) Jitter . Ethernet is a packet-switched network. In general, this means that packets may not be received by the client in the order in which they were sent to it (different routes may be used to deliver packets). To solve this problem, a special "jitterbuffer" (smoothing buffer) is used. The purpose of this buffer is to pre-accumulate packets before transmitting them to the decoder. The jitter buffer also introduces some delay in the voice transmission process, so it is desirable to use a jitter buffer of such a size that, on the one hand, ensures acceptable speech quality, and on the other hand, minimizes the total budget of the two-way delay to 300 ms.

3) P ket loss. As you know, Ethernet networks allow packet loss. The effect of packet loss on speech quality is determined by the size of the packet, as well as the speech compression algorithm used. Voice information is more resistant to the loss of single packets. In any case, according to the ITU-T recommendation, no more than 1% of packets are allowed for the normal operation of IP-telephony systems, otherwise the degradation of speech quality will be noticeable. To improve the quality in the conditions of busy networks, you can use QoS or, if the packets are lost due to the nature of the network itself, then to improve the quality, you can use a more noise-resistant codec or reduce the size of the encoded frame. Consider separately the codecs for voice, video and data. Let's start by looking at the types of codecs for voice [5].

Data transfer rates

Table 4

Data transfer rates

Services		Coding technology	Transmission speed
Speech	Telephony (very high quality).	ITU-T G.711	64 kbit/s
	Telephony (high quality).	ITU-T G.729	8 kbit/s
	Telephony (quality from high to satisfactory).	ITU-T G.723.1	5.3/6.4 kbit/s
Audio Data	Streaming audio.	MP3	32-320 kbit/s
	Music CD	CD-DA	1441 kbit/s
	Music CD	MPEG **FFT**	192-256 kbit/s
	High quality audio recording.	MPEG **FFT**	384 kbit/s
Video data	Video conferencing.	ITU-T H.261	100 kbit/s
	Video streaming	MPEG-4	to 10 mbit/s
	Traditional tv	MPEG-2	2-4 mbit/s
	Digital television Professional quality	ITU-T BT.601	166 mbit/s
	DVD format for DVDs	MPEG-2	3-6 mbit/s
	High Definition HDTV	MPEG-2	25-34 Mbit/s

FFT is the fast Fourier transform.

The table shows that the most resource-intensive services are the transfer of video data. So, when providing telephony services to the MCC, the G.709 codec is usually used with a speed of 8 kbit/s, the provision of on-demand audio services is usually provided from 256 to 384 kbit/s, and for high-definition television when transmitting a compressed signal, it is necessary speed of the order of 30 Mbit/s.

The provision of these services varies significantly in the duration of the occupation of the transport resource. So, a telephone conversation on average can last from 2 to 25 minutes, while the duration of watching a TV channel lies in the range from 10 minutes to 3 hours.

The growth in the volume of television traffic is facilitated by the explosive growth in the speed of cellular mobile communication network technologies. According to forecasts in 2020, it is expected to begin commercial use of 5G (speed up to 1 Gbit/s).

The growth of video data traffic is facilitated by the mass introduction of multicast technology on MCC networks with the ability to transfer dozens of video streams corresponding to various television channels to large groups of subscribers at a speed of 2-4 Mbit / s for traditional television and 25-34 Mbit/s for high-definition television (English High Definitio Television, HDTV).

The need to control traffic flows in a multiservice communication network has led to the emergence of a new direction in the theory of teletraffic, which is called Traffic Engineering (TE). Traffic engineering is a complex of interconnected methods and mechanisms that allows the operator to optimize the throughput of communication networks. Briefly outline the goals of traffic engineering, this is the ability to direct information flows to where there is a resource for their maintenance.

This solves three groups of problems:

1. The first group of tasks is measuring the characteristics of information flows, processing measurement results, distributing traffic by directions, analyzing the parameters of the most popular communication applications; assessment of the speed of receipt and processing of information, etc.
2. The second group of tasks is the execution of actions for the formalized description of information flows.
3. The third group of tasks is modeling the process of receiving and servicing information flows. Based on the data of statistical observations, dependencies are derived that allow finding the characteristics of the user service quality using the values of flow parameters and information about the network topology.

Recommendations for the use of traffic engineering functions are formulated in RFC-2702 and RFC-3209.Ethernet framing sequence for transmitting voice messages over a packet-switched network.

Let's consider the principles of converting voice messages into data packets using the example of VoIP –Voice over IP technology. Conversion of analog signals to digital and their restoration from continuous digital stream to analog signals is performed using codecs either in digital telephones or in access gateways to which analog lines are connected.

This problem was solved even during the creation of the first digital transmission systems using codecs built according to ITU-T Rec. G.711. The method of pulse-code modulation (PCM) is used for coding. This codec is distinguished by its simplicity of implementation and high quality of speech transmission, which leads to its widespread use in VoIP technology. The disadvantage of G.711 codecs is the high bit rate (64 kbps).

When using PCM, the analog signal values are sampled every 125 µs, i.e. after 125 µs, one data byte is written to the packet. When transmitting voice messages, 80 or 160 bytes are usually placed in one packet. At 80 samples, the packetization delay will be 125 µs X 80 = 10 ms, and at 160 samples - 20 ms.

When forming an audio frame, 54 bytes of the address information of the protocols (Table 3) of the transport, network and data link layers of the Reference Model for Open Systems Interconnection (EMVOS) are added to 80 (160) bytes. The desire to reduce the bit rate and ensure high quality messaging has led to the development of a whole series of other codecs. The parameters of some of them are given in table 5.

Table 5

Protocols	Layers	Amount of data, bytes
	7. Cust	speech
G.711	6. Re esentations	80 (160)
H.323 / SIP	5. Sessi	80 (160)
RTP / UDP	4. ansport	12 + 8 + 80 (160)
IP	3. Netwo	20 + 12 + 8 + 80 (160)
Ethernet	2. Cha el	14 + 20 + 12 + 8 + 80 (160)
Stream bit	1. ysical	134 (214)

Legend: RTP – Real-time Transport Protocol; UDP – User Datagram Protocol; IP – Internet Protocol.

Table 6

Parameters of some codecs

Codec type	Codec speed, kbps,V kk	Speech frame size, bytes	Ethernet frame length, bytes	Packet speed, kbps Vkp	Quality Assessment
G.711	64	80 (160)	134 (214)	107,2 (85,6)	4,4
G.729*a*	8	20	74	29.6	4,0
G.726		40	94	75.2	4,3
G.723.1	8	25	79	25.3	3,9

Redundancy factors of the G.711 codec (the ratio of the Ethernet frame length to the length of the speech frame):

К rf Redundancy factors Vkp Packet speed Vkk Codec speed

To the G.711 $К_{rf\,G.711}^{(10)} = \frac{134}{80} = 1{,}676;$ (1)

To the G.711 $К_{rf\ G.711}^{(20)} = \frac{214}{160} = 1{,}338.$ (2)

Required speeds in a packet network, taking into account the transfer of address information:

$V_{kpG.711}^{(10)} = v_{kk\ G.711}\ К_{rf\,G.711}^{(10)} = 64*1{,}676 = 107{,}2$ kbit/s; (3)

$V_{kpG.711}^{(20)} = v_{kk\ G.711}\ К_{rf\,G.711}^{(20)} = 64*1{,}338 = 85{,}6$ kbit/s. (4)

In the implementation of packet switching technology, a delay is incorporated, the value of which is the sum of delays during encoding, during packetization and during transmission over the network.

Due to the headers of the EMVOC levels, the frame length increases, and, consequently, the required transmission rate.

Statement of the problem

Let the MCC link receive n streams of requests for the allocation of a channel resource for servicing real-time traffic streams. The arrival of claims of the kth flow obeys Poisson's law with intensityλ_κ, where $k = 1,2,\dots,n$. Let v be the speed of the communication link of the MCC link, expressed in units of the channel resource, b_κ – the number of units of the channel resource of the line required to service one claim of the kth flow, $(\frac{1}{\mu\times k})$ – the average duration of the occupation of the channel resource for its service.

We will assume that the durations of occupying the channel resource for servicing claims have an exponential distribution and do not depend on each other and on input flows. The MCC link model is shown in Fig. 1.

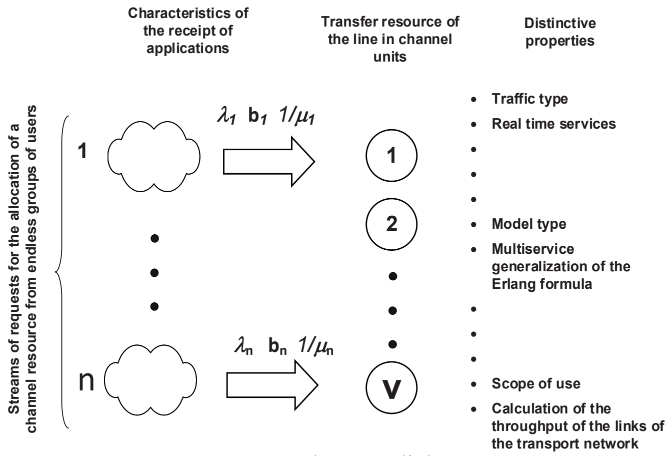

Figure 1. The MCC link

For a Poisson flow, λ_k is the intensity of the incoming call flow, and $1 / \mu_k$ is the average duration of servicing one call. In this case $\lambda_k / \mu_k = a_k$, where a_k is the intensity of the incoming load of the k-th flow.

Conclusion

For each of the n streams, the problem of estimating the proportion of lost claims will be solved, and we will also find the average value of the number of channel units of the line resource busy in servicing. The means for calculating the Qos indicators will be effective. All calculations will discuss in more detail the ways of solving the formulated problem on the next chapter.

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НОРМИРОВАНИЕ ОСНОВНЫХ ПАРАМЕТРОВ КАЧЕСТВА СЕТЕВЫХ УСЛУГ

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Аннотация
Качество услуги с точки зрения пользователя может быть выражено совокупностью показателей. Эти показатели описываются в терминах, понятных как пользователю, так и службе, и не зависят от структуры сети. Показатели качества услуги ориентированы по преимуществу на эффект, воспринимаемый пользователем, должны быть гарантированы пользователю службой и поддаваться объективному измерению в точке доступа к услуге (Рекомендация ITU-T I.350). В Рекомендации ITU-T E.862 приведены возможные подходы к учету экономических потерь оператора (при планировании, проектировании, эксплуатации и сети электросвязи) и пользователя, связанных с отказами технических средств. Операторы сетей, работая в условиях рынка, заинтересованы в оценке возможных потерь из-за отказов и в сопоставлении их с затратами на повышение надежности своих средств.

Ключевые слова: *качество обслуживания, связь, задержка, джиттер, потеря пакетов, технология передачи голоса по IP, сеть с коммутацией пакетов, телевидение высокой четкости.*

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