Journal of Theoretical and Applied Information Technology
30th June 2021. Vol.99. No 12

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

THE IP CHANNEL BANDWIDTH DURING TRANSMISSION OF THE VIDEO AND TOMOGRAPHY SIGNALS

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ABSTRACT

The aim of the study is to develop a confident algorithm for assessing the bandwidth of an IP channel affecting image quality during transmission of the video signal and signals for tomography. The research method is based on the Monte Carlo method. The development of an algorithm for assessing the bandwidth based on information identifier (IP packet delay time and delay variation) is discussed in this paper. The experiments have been carried out taking into account the size of the information packet of the video signal and the signals for tomography. The outcomes of this study embrace the developed model of an IP channel, its software implementation, which allows assessing the dynamic bandwidth. The developed model enables to perform comparison of the efficiency of a compressor, a router, and a server with digital signal processing by the criterion of confidence of packet servicing time in the IP channel. Analytical calculations have been carried out for the purpose of quantitative calculation of the obtained confidence to confirm the results of modelling. The results of modelling allow taking into account the main interrelated factors that appear in the form of distortions in the image during the transmission of video signals and signals for tomography. The article discusses the results of the “Analysis of the functioning features of multi-service IP networks” project. The present paper reveals new knowledge about the development of confident algorithms that allow assessing the probability of delivery, packet loss, and IP channel bandwidth with a given probability. In prospective, these studies are necessary and relevant for performing remote surgical operations and decoding tomography images remotely.

Keywords: *Video signal, Signal for Tomography, IP Channel, Image Quality, Bandwidth, Algorithm, Confidence*

1. INTRODUCTION

In the near future, the possibility of remote surgical operations or their support, as well as remote consultation by medical specialists upon the results of magnetic resonance tomography, becomes more and more urgent. To conduct the mentioned works, the connection of terminal equipment with the help of an IP channel or through the Internet using IP Protocol is required.

Medical equipment uses IP networks to transmit tomography images. In this regard, it is particularly important to process the obtained quality of digital tomography images of human organs.

Due to increase in the number of data streams, consumer and enterprise services, development of the bandwidth information and communication network to be able to cope with increased loads, is becoming an urgent one. Currently, it is possible to observe deterioration in the quality of the video signal during a remote surgical operation or in the quality of tomographic images during remote interpreting of magnetic resonance images via the Internet or an IP channel.

Under these conditions, one of the effective methods for determining the bandwidth of an IP channel is the development of appropriate algorithms and procedures, involving the providing of access to the telecommunications network. The communication channel is presented as a block diagram indicating the main stations of the channel. In the case of several stations or devices in the channel, the task of monitoring their functioning as an IP packet passes through them becomes relevant.

This paper is devoted to the development and selection of an algorithm of confidence of the assessment of the bandwidth of IP channel based on the time of IP-packet servicing by transport infrastructure elements. The use of delay time of an IP packet, IPTD (Internet Protocol Packet Transfer Delay) as the information identifier, IPDV (Internet Protocol Packet Delay Variation) in the algorithm under development, as well as modelling as per the Monte Carlo, allowed creating the software implementation to determine the packet loss when exceeding TTL (Time to Live for a packet in the IP Protocol).

The Internet and IP channel is working under TCP/IP (Transmission Control Protocol/Internet Protocol) protocol stack and are based on two main fundamentals, which are: network offers only one class of service named best effort service, and network resources are overprovisioned as possible in order to minimize packet losses and packet delays. The Internet providers, telecommunications services providers aim to provide packet transfer to users as fast as possible, however they are far from guaranteeing, so called, rigid quality of service QoS (Quality of Service) and QoE (Quality of Experience) that are measured by the maximum allowable values of such parameters as IPTD and IPDV.

Network abilities of packet transfer determine the range of applications with which channel user receives the digital video signals and digital image signal.

The main research in the development of the method of determining the bandwidth of the IP channel during the transmission of tomographic signals was carried out in accordance with the given list of literature used. The theoretical analysis revealed a number of shortcomings, which were eliminated in the present study.

This article will provide a deeper understanding of the problems encountered when transmitting a digital tomographic signal, as well as simulate the process of its transmission over the IP channel. The following data are necessary for telecom operators organizing the connection channel to obtain high-quality digital tomographic images and video broadcasting of surgical operations at the receiving end.

2. PROBLEM STATEMENT AND THE WAY OF MODELLING

As it is known, compressors, routers and servers with digital signal processing, which introduce major delays into the communication channel, are important elements of the transport infrastructure of the IP channel. The approximate input delay time is as follows: 15...50 ms by compressors, 10...20 ms by routers, and 80...150 ms by servers with digital signal processing [1-3].

The main information identifier is the delay when determining the bandwidth. The delay is made up of the time of packetizing of traffic elements and the transmitting packets over communication channels between the nodes of the telecommunications IP network, as well as the waiting time of these blocks in the queues of intermediate routers and switches of the network [4-5].

The total network delay of 150...250 ms is considered acceptable. A jitter of 50...150 ms is considered acceptable; a more strict upper limit of 40 ms is sometimes established. It should be noted that the delay time input by the hardware is approximately in inverse proportion to the processor performance, which doubles every 1.5...2 years. Therefore, we can expect that in the coming years, the maximum delay in the IP network will approach 150 ms [5].

Video signals to a great extent depend on the network bandwidth, as well as on possible delays and variations in delays in the data transmission network. The fundamental difference is in the requirements for bandwidth on the part of Video on Demand (VoD) and IP broadcasting services. The first ones involve the transfer of information in the Unicast mode when an individual communication session is established between the video server and each client device. Ten clients mean ten video streams in an IP network, 100 clients mean 100 video streams, and 1,000 clients mean 1,000 video streams. It is clear that the IP network in this case should have a larger bandwidth with the possibility of its quick expansion. Such scaling is now possible only in IP networks with the use of the fastest Ethernet technologies and spectral multiplexing of fiber-optic lines (WDM). To transmit high-quality video over the IP network, it is required to use Differentiated Services (DS).

Different types of differentiated services are provided in IP networks:

1. Traffic offered at the Class A service level will be delivered with low delay.

2. Traffic offered at the Class B service level will be delivered with low losses.

3. Delays for 90% of traffic offered at the Class C service level will not exceed 50 ms.

4. 95% of traffic offered at the Class D service level will be delivered to the recipient.

5. Traffic offered at the Class E service level will be given twice as much bandwidth as traffic offered at the Class F service level.

In order to receive a particular service, data packets are marked with the DS field. The DS code is a tag applied to categorize data packets. A 6-bit code may describe 64 types of traffic class [3-4].

The DS technology provides prioritization of video packets, ensuring that exactly all video frames are delivered to users.

The introduction of digital TV broadcasting and related services in multi-service IP networks leads to certain difficulties, especially for the ISPs (Internet Service Providers). Video, multi-channel TV and HDTV broadcasting require more network resources than voice and data.

Over-IP video has clear parameters for packet loss, within the range of ten zeros, which actually exist in practice. IPTV packets can only be dropped due to invalid bits of information and network load.

There are two main types of video traffic: interactive video (video conferences) and streaming video (IPTV, multicast transmission).

There are many problems that affect the quality of the received signal, when transmitting data over the communication networks [6].

The unsatisfactory transmission service such as the voice, the data trans mittion or HD-video underestimated the so-called instrument of assured quality QoE and QoS.

QoE is a software product that gathers statistics, assessing the quality of perception of the services rendered. Gathered indicators are compared with the specified metrics, which allows you to understand how high-quality the services rendered for an individual client are. Based on the information received, the module initiates the execution of the necessary steps to improve the quality of the services rendered.

QoS is a collection of technologies that allow applications to request and receive a predictable level of services in terms of bandwidth, response delay time spread, and overall packet transfer delay.

The QoS analysis for certain types of traffic showed that when building and operating an IP network for rendering services, the following requirements are mandatory:

- delays of over 150 ms are not allowed;
- it is recommended to monitor delays within 30-50 ms for high-quality video data transmission;
- to broadcast video data, IP packets should not lose more than 1% of the total data, taking into account buffer caching.
- the lower limit of the HDTV video stream speed when using MPEG-4/H.264 compression for adequate service quality should be at least 15 Mb/s.

The influence of various factors during the transmission negatively affects the quality of the received signal, namely the quality of the digital image [7-10]. The unsatisfactory digitized tomographic images interfere with accurate diagnosis.

Analysis of any real processes is performed using mathematical modelling methods. It can be simpler and cheaper than designing appropriate devices. The model may include elements of randomness, taking into account the probabilities of possible actions to process the real parameters of the system in question. In the study of complex systems and phenomena, the process of simulation is used.

In simulation, the model implementing the algorithm reproduces the process of functioning of the system in time. The elementary phenomena that make up the process are imitated, preserving their logical structure and sequence of flow over time.

The results of simulation modelling of a stochastic system are realizations of random variables or processes. Therefore, to find the characteristics of the system requires repeated repetition and subsequent data processing. Most often in this case, a variety of simulation modelling is used, that is statistical modelling. One of the varieties of modelling is numerical modelling, which presupposes obtainment of the necessary quantitative data on the behaviour of systems or devices using certain suitable numerical method. The proposed simulation method should provide objective assessment of the quality of a digital tomographic image.

3. EXPERIMENTS

It is possible to use the Monte Carlo method to develop a packet servicing time model and perform the assessment of the IP bandwidth, considering the probabilistic nature of packet loss and packet delay [12].

The scheme of the Monte Carlo method is as follows. It is necessary to calculate some quantity *Z*. It is implied that it is possible to build a

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

random x variable with a Mx mathematical expectation equal to Z and with a Dx finite variance, whereupon, sample values, x_i of the x random variable are simply implemented by the random number generator. Let us find the approximation of the desired quantity, having built a large number of n sample values $x_1,\ldots,x_n$, based on the law of large numbers [12]:

$$Z = Mx \approx \bar{x}_n = \frac{x_1 + \ldots + x_n}{n}. \quad (1)$$

The main factor in using the approximation (1) is the possibility of effective implementation of sample values of random variables.

The essence of the method of mathematical modelling, is in that the process of functioning of the system is simulated by arithmetic and logical operations in the sequence of elementary acts, which is characteristic of the simulated process [13-14]. The simulation of random factors is performed using random numbers generated by a computer. Thus, an algorithm that allows, as per the specified parameters of delays, packet loss, and initial conditions, determining the probability of confidence of non-exceeding the delay time in accordance with the requirements of QoS mechanisms serves as a mathematical model of the process of delays, packet loss in the IP channel, and the bandwidth.

The statistical algorithm of the study includes the following main blocks (Figure 1):

- a block for obtaining random numbers in a programmable way;
- a block for the implementation of the formalized scheme of the process under study;
- a block for the analysis of the results and printing.

Quantitative assessment of the confidence of the information obtained on the bandwidth is of a certain interest. Such the assessment may be carried out analytically and with the help of probabilistic and statistical modelling. Calculations have been carried out using the machinery of probability theory. The following quantities were taken into account: introduced delay time by the compressor, router, and server with digital signal processing; compressor failure probability $P(\bar{C})$; server failure probability $P(\bar{S})$; router failure probability $P(\bar{R})$. Assuming that the above events are independent and shared, let us determine the probability of failure of all three devices $P(m)$:

$$P(m) = P(\bar{C}) + P(\bar{S}) + P(\bar{R}) - P(\bar{S})P(\bar{R}) - P(\bar{S})P(\bar{C}) - P(\bar{R})P(\bar{C}) - P(\bar{C})P(\bar{S})P(\bar{R}). \quad (2)$$

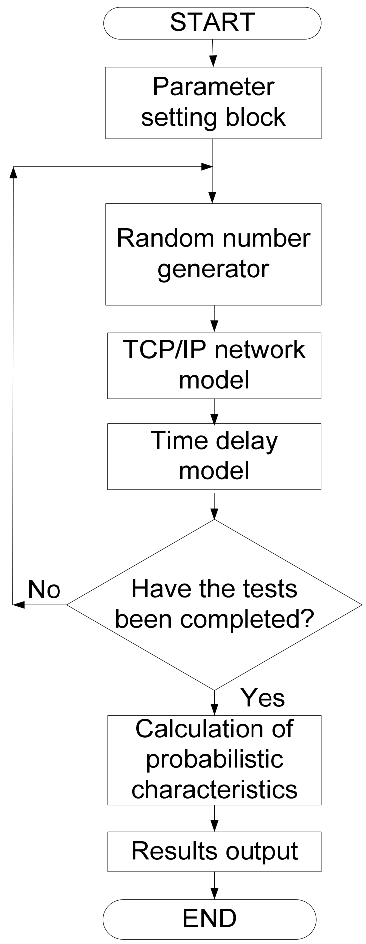

Figure 1: The statistical algorithm for studying packet service time in an IP network

According to (Avizienis, Laprie and Randell, 2001, Bagdonavicius, Nikulin, 2002), the following probabilities are obtained:

$$P(C) = 0.95;\; P(S) = 0.95;\; P(R) = 0.95\,. \quad (3)$$

Obviously, the product of three or more multipliers are negligible with the given values of the quantities (3) in the Equation (2) in comparison with the given quantities. Thus they may not be considered in the calculation of $R(m)$; in this regard $P(m) = 0.14$. Therefore, the true confidence, $R(r)$, of packet transmission through three of these devices under the given probabilities is determined by the following equation:

$$P(r) = 1 - P(m) = 0.86\,. \quad (4)$$

It is essential to determine the required number of tests when using the statistical modelling method. Increase in the number of tests increases the accuracy of statistical characteristics, however

Journal of Theoretical and Applied Information Technology
30th June 2021. Vol.99. No 12

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

at the same time the time spent on modeling is also increased. The number of experiments (N_x), which is required to expect, with a given probability (Q), that the investigated frequency of the *"r"* event will deviate from its probability *P(r)* by a quantity ε is determined by the following equation:

$$N_x = \frac{P(r)-(1-P(r))}{\varepsilon^2}\left[F^{-1}\left(\frac{1}{2}Q\right)\right]^2, \quad (5)$$

where F is the Laplace function.

Based on the fact that the error ε may be equal to the unit of the lowest position, we obtain ε=0.01. The confidence probability Q is assumed to be equal to Q=0.95 [15]. The probability *P(r)* is assumed to be respectively calculated according to the expression (3). The value *Nx*, which affects the accuracy of the model with the number of implementations, will be linked to the accuracy of the initial information and the practical task of studying the confidence of the information obtained during modelling.

3.1 The model of functioning of the IP channel based on a single identifier

To verify the calculation result, the probabilistic and statistical model of functioning of the IP network channel, with the help of which the confidence of the model readings has also been assessed, has been developed.

Modelling of the IP network channel is carried out as follows. With the help of the pseudo-random number generator, we obtain *C2, S2, R2*, random numbers that allow assessing whether the relevant devices failed or not: compressor, server, or router. The sequence of operation of IP channel devices is reflected in text algorithm description and corresponds to the actual sequence of functioning in the transmission channel.

The probabilistic characteristics of functioning of the IP network channel based on the modelling results, and the bandwidth were calculated as follows.

The probability of obtaining confident information about the bandwidth of the IP channel in the case of failure-free operation of the server, compressor, or router:

$$P(Z) = \frac{P_1}{O}, \quad (6)$$

where P_1 is the number of packets processed in the data stream to the subscriber; O is the total number of packets sent to the subscriber in the data stream.

The probability of obtaining confident information taking into account failures of a server, compressor, or router:

$$P(L) = \frac{S+C+R}{O}, \quad (7)$$

where S is the number of server failures; C is the number of compressor failures; R is the number of router failures.

The bandwidth of the IP channel is:

$$B = 8 \times P_1 \times P(n), \quad (8)$$

where P_1 is the number of processed packets in the data stream to the subscriber; *P(n)* is the packet size.

To obtain quantitative characteristics of changes in probabilities *P(Z)*, *P(L)* in the function of quantities *P(S)*, *P(C)*, *P(R)*, calculations have been performed on the model for the following cases $P(R)=0.95$: $P(Z)=f[P(S),P(C)]$, $P(L)=f[P(S),P(C)]$; $P(C)=0.95$: $P(Z)=f[P(S),P(R)]$, $P(L)=f[P(S),P(R)]$; $P(S)=0.95$: $P(Z)=f[P(C),P(R)]$, $P(L)=f[P(C),P(R)]$.

The text algorithm description of functioning of the IP channel based on a single information identifier:

```
var PC1, PS1, PR1, Pn, Nx, C, S, R, D, O, P1, I, PZ, PL, B: integer;
C1, S1, R1, dC, dS, dR: array;
begin
input data (PC1, PS1, PR1, Pn, Nx);
dC=[15..50];
dS=[80..150];
dR=[10..20];
C=0; S=0; R=0; D=0; O=0; P1=0;
for i=1 to (Nx-1) do
C1[i]=random(1000);
S1[i]=random(1000);
R1[i]=random(1000);
if C1[i] < min(dC) and C1[i] > max(dC)
then C=C+1;
D=D+1;
O=O+1;
end;
else if S1[i] < min(dS) and S1[i] > max(dS)
then S=S+1;
D=D+1;
O=O+1;
end;
else if R1[i] < min(dR) and R1[i] > max(dR)
then R=R+1;
D=D+1;
O=O+1;
end;
else T[i]=C1[i]+S1[i]+R1[i];
```

```
P1=P1+1;
O=P1+D;
end;
end;
end;
end;
PZ=P1 / O;
PL=T / O;
B=8*P1*Pn
end.
```

The results of the analytical calculation according to the Equation (1) and the modelling results at the values of probabilities (2) coincide with the accuracy of the third decimal place.

3.2 The model of functioning of the IP channel based on two identifiers

The analysis of the algorithm of the IP channel operation based on a single identifier showed that its main disadvantage was the lack of any information about the quantity of the delay variation IPDV (Internet Protocol packet Delay Variation).

The introduction of an additional information identifier of IPDV allows increasing the confidence of the obtained information about the bandwidth of the IP channel, which is confirmed by the calculations. To implement this method, the probabilistic and statistical model, which allows taking into account an additional identifier of IPDV, has been developed. The initial data for the calculation is the same as in the model of functioning of the IP channel based on a single information identifier.

Modelling of the IP channel operation to study the delay time and the calculation of the bandwidth associated with these factors is carried out as follows. With the help of the pseudo-random number generator, we obtain random numbers $C2$, $S2$, $R2$ that allow assessing whether the relevant devices failed or not: compressor, server, or router.

Let us present the statistical algorithm for modelling the IP channel for the study of packet delays and the bandwidth in the form of the sequence of the program steps:

1. An array of input information is introduced for the IP channel. The delay time of IP packets, the probabilities $P(C)1$,$P(S)1$, $P(R)1$, the packet size $P(n)$ and jitter are set.

2. Counters C (compressors failures), S (server failures), R (router failures), D (packet losses), $P1$ (number of processed packets), O (total number of packets) are reset.

3. Whether the compressor is sound or not is determined, to do this, the random number, $C2$, generated by the random number sensor is compared with the probability of compressor failure $P(\overline{C})1$. If the condition of $C2 < P(\overline{C})1$ is satisfied, the compressor failure is recorded, the counter C is increased by one, the counter of losses D is increased by one, and step 2 should be executed first. At $C2 > P(\overline{C})1$, the router actuation is detected and the next step is performed.

4. The packet delay time in the compressor is determined, to do this, the random number $C3$ generated by the random number sensor from the packet delay range of the compressor is taken as the delay time. If the condition IPDV≤50 ms is not satisfied, at each subsequent actuation of the random number sensor, step 2 should be executed first.

5. Whether the server is sound or not is determined, to do this, the random number, $S2$, generated by the random number sensor is compared with the probability of server failure $P(\overline{S})1$. If the condition of $S2 < P(\overline{S})1$ is satisfied, the server failure is recorded, the counter S is increased by one, the counter of losses D is increased by one, and step 2 should be executed first. At $S2 > P(\overline{S})1$, the server actuation is detected and the next step is performed.

6. The packet delay time in the server is determined, to do this, the random number $S3$ generated by the random number sensor from the packet delay range of the server is taken as the delay time. If the condition IPDV≤50 ms is not satisfied, at each subsequent actuation of the random number sensor, step 2 should be executed first.

7. Whether the router is sound or not is determined, to do this, the random number, $R2$, generated by the random number sensor is compared with the probability of router failure $P(\overline{R})1$. If the condition of $R2 < P(\overline{R})1$ is satisfied, the router failure is recorded, the counter R is increased by one, the counter of losses D is increased by one, and step 2 should be executed first. At $R2 > P(\overline{R})1$, the router actuation is detected and the next step is performed.

The text algorithm description of functioning of the IP channel based on two information identifiers:

```
var PC1, PS1, PR1, Pn, Nx, C, S, R, D, O, P1, I, PZ, PL, B: integer;
C1, S1, R1, dC, dS, dR: array;
begin
input data (PC1, PS1, PR1, Pn, Nx);
dC=[15..50];
```

```
dS=[80..150];
dR=[10..20];
C=0; S=0; R=0; D=0; O=0; P1=0; C1[0]=0;
for i=1 to (Nx-1) do
C1[i]=random(1000);
S1[i]=random(1000);
R1[i]=random(1000);
if C1[i] - C1[i-1] <= 50
if C1[i] < min(dC) and C1[i] > max(dC)
then C=C+1;
D=D+1;
O=O+1;
end;
else if S1[i] - C1[i] <=50
then if S1[i] < min(dS) and S1[i] > max(dS)
then S=S+1;
D=D+1;
O=O+1;
end;
else if R1[i] - S1[i] <=50
then if R1[i] < min(dR) and R1[i] > max(dR)
then R=R+1;
D=D+1;
O=O+1;
end;
else T[i]=C1[i]+S1[i]+R1[i];
P1=P1+1;
O=P1+D;
end;
end;
end;
end;
end;
end;
end;
PZ=P1 / O;
PL=T / O;
B=8*P1*Pn
end.
```

8. The packet delay time in the router is determined, to do this, the random number R3 generated by the random number sensor from the packet delay range of the router is taken as the delay time. If the condition IPDV≤50 ms is not satisfied, at each subsequent actuation of the random number sensor, step 2 should be executed first.

9. The total delay time and the bandwidth of the channel are calculated.

10. The condition of the implementation of the given number of tests is checked. If tests are completed, the calculation of the probabilistic and statistical characteristics of the model under study is carried out and the transition to the next step is made, otherwise, the jump to step 2 is executed.

11. The results of the calculations are displayed and the program is completed.

The data obtained from the program operation:

- *C* – number of compressor failures;
- *S* – number of server failures;
- *R* – number of router failures;
- *D* – number of lost packets in the data flow;
- *P*1 – number of processed packets in the data flow;
- *O* – total number of packets in the data flow;
- *P*(*Z*) - probability of delivery of information packets;
- *P*(*L*) - probability of loss of packets that pass through the information channel;
- *B* – bandwidth.

Probabilistic characteristics, bandwidth, and the probability of obtaining confident information about the bandwidth of the algorithm of functioning of the IP channel with an additional identifier of IPDV were calculated similarly to the Equations (6-8).

4. RESULTS AND DISCUSSION

Modelling of confidence of the assessment of the bandwidth of the IP channel and accordingly, the bandwidth calculation were carried out in the developed DelayProg software [16].

This piece of software is written in Delphi and consists of two windows. The first window of the program shows a Start button to start the work and three sectors to display all the simulation results. After clicking the File menu, a settings window is displayed for entering an array of input data.

The possibility to set the size of the IP information packet is implemented in DelayProg.

The algorithm developed and its software implementation allows taking into account the main interrelated factors that are manifested in the form of distortions in the digital image.

The principle of operation of the software is described in clause 2.2. By changing the values of the confidence of failure-free operation as a result of modelling using the software, we obtain the data on the change in the channel bandwidth (Table 1).

Table 1. The results of modelling the bandwidth of the IP channel based on one and two parameters

One parameter		Two parameters	
P(C), P(S), P(R)	***B*, Mb/s**	***P(C), P(S), P(R)***	***B*, Mb/s**
0.8	0.77	0.8	0.97
0.85	5.4	0.85	6.9
0.9	12.5	0.9	16.2
0.95	22.9	0.95	29.6
1	37.3	1	48.3

Let us present the data of Table 1 graphically (Figure 2).

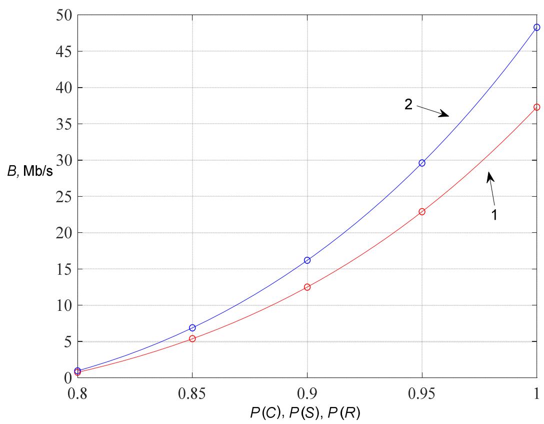

Figure 2: The bandwidth of the IP channel as a result of modelling based on one and two parameters

Figure 1 shows the graphical dependence of the IP channel bandwidth on the probability of failure-free operation of devices, which organize the communication channel. According to the modelling results, the bandwidth of the developed model based on two identifiers has higher rate.

The developed software implementation of the algorithm allows comparing the efficiency of the above models of functioning of the IP channel and obtainment of the data on the confidence of the bandwidth assessment (Table 2).

Table 2. The results of modelling changes in the confidence of the obtained information about the bandwidth of the IP channel based on one and two parameters

One parameter		
P(S)	***P(C)***	***P(Z)***
0.8	0.8	0.523
0.85	0.8	0.569
0.9	0.8	0.594
0.95	0.8	0.625
1	0.8	0.656
0.8	0.85	0.567
0.85	0.85	0.596
0.9	0.85	0.632
0.95	0.85	0.657
1	0.85	0.695
0.8	0.9	0.589
0.85	0.9	0.627
0.9	0.9	0.675
0.95	0.9	0.707
1	0.9	0.737
0.8	0.95	0.627
0.85	0.95	0.667
0.9	0.95	0.702
0.95	0.95	0.748
1	0.95	0.778
0.8	1	0.653
0.85	1	0.696
0.9	1	0.735
0.95	1	0.772
1	1	0.806
Two parameters		
P(S)	***P(C)***	***P(Z)***
0.8	0.8	0.620
0.85	0.8	0.672
0.9	0.8	0.702
0.95	0.8	0.744
1	0.8	0.776
0.8	0.85	0.662
0.85	0.85	0.710
0.9	0.85	0.748
0.95	0.85	0.793
1	0.85	0.821
0.8	0.9	0.703
0.85	0.9	0.743
0.9	0.9	0.795
0.95	0.9	0.838
1	0.9	0.869
0.8	0.95	0.744
0.85	0.95	0.788
0.9	0.95	0.833
0.95	0.95	0.882

1	0.95	0.921
0.8	1	0.770
0.85	1	0.821
0.9	1	0.871
0.95	1	0.921
1	1	0.959

Change of the confidence of the obtained information is received at probabilities of failure-free operation from 0.8 to 1 (Figure 3).

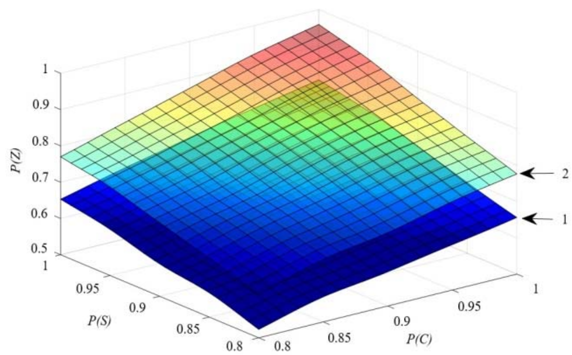

Figure 3: Change of the confidence of the obtained information about the bandwidth of the IP channel based on one and two parameters

The above diagram shows that the algorithm of the model of functioning of the IP channel based on two information identifiers has greater confidence and, therefore, it may more accurately characterize the bandwidth of the IP channel.

Probabilistic and statistical models of functioning of the IP channel are written in high-level Delphi language.

Practical experimental studies were carried out on the platform of the current service provider of the Republic of Kazakhstan according to the functional diagram on the Figure 4.

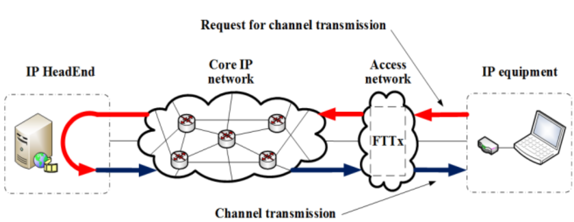

Figure 4: Functional diagram of practical studies

The delay of IP packet and IPDV were controlled with the help of the monitoring systems the quality of network Proviso.

The practical results of research the bandwidth of the IP channel for one parameter are shown in Figure 5 and for two parameters in Figure 6.

Figure 5: Computed tomography of abdominal organs while research an IP channel based on one parameter

Figure 6: Computed tomography of abdominal organs while research an IP channel based on two parameters

The tomographic image of the abdominal cavity (Figure 5-6) transmitted in the practical studying on the IP channel (Figure 4). During control only the delay time of IP packet arose such distortions on the transmitted image (Figure 5) as block distortion and the effect “dirty window” to occur with digital images [17]. These distortions are unacceptable during the image’s decrypting of tomography, especially during remote surgical procedures, where the video signal is transmitted, even more critical to delays. Control of the IP channel on two parameters allowed to obtain more reliable information about the capacity (Figure 3) and to achieve the image without distortion on the output of the channel (Figure 6).

5. CONCLUSION

Currently, based on cooperation of the centers of magnetic resonance imaging performed in the remote mode, there is increase in telesurgery, joint remote surgical operations, remote interpretation of tomographic examinations, etc. [18-20]. Remote mode of operations and image interpretation is implemented by means of dedicated fiber-optic lines or via the Internet [21-22]. The hardest part of these operations is to ensure minimum delay in the communication channel [23]. This requirement confirms the relevance of the presented studies.

A review of the literature showed significant gaps in the reliable determination of the IP channel capacity and its compliance with the quality of the transmitted image (Figure 5-6).

In connection with the above problem of reliable estimation of the IP channel capacity and transmission of high-quality tomographic images, the Monte Carlo method was proposed.

In the course of the study, the algorithm of the confidence model of the assessment of the IP channel bandwidth based on one and two information identifiers has been developed.

Given the delays of the transport infrastructure of the IP channel and the specified probabilities of failure-free operation of the equipment from 0.8 to 1, it has been found that when using one information parameter, the bandwidth varies from 0.77 to 37.3 Mb/s, when using two parameters, the bandwidth varies from 0.97 to 48.3 Mb/s.

Modelling of changes in the confidence of the obtained information about the IP channel bandwidth based on one and two parameters revealed greater adequacy of the model based on two information identifiers with the probability of 0.959, which is confirmed by the modelling results.

The transmission of the signal over the communication channel with higher bandwidth has minimal delays, which in turn affects the quality of medical images being transmitted (Figure 5-6).

Development of methods for calculating the capacity of IP-channel and evaluating the quality of transmitted tomographic images, taking into account the influencing factors, mathematical and simulation modeling are currently ongoing research, and will be presented by the authors in future works.

6. ACKNOWLEDGEMENTS

The research is carried out at Siberian State University of Telecommunications and Informatics within the framework of Siberian State University of Telecommunications and Informatics Competitiveness Enhancement Program grantand of International Cooperation project with S. Seifullin Kazakh Agrotechnical University “Analysis of the functioning features of multi-service IP networks”.

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ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

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