The Problem of Predicting the Data Transmitting Delay in the Network with the Self-Similar Nature of Traffic, for the Purpose of Improving the Real-Time Conferencing

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Abstract- **This publication gives new results in applying the theoretical knowledge based on the Laplace-Stiltes transform. The main purpose is to predict packets transmitting delay, in a network based on the Internet technology. A new method for modeling the real-time networking process is designed.**

I. Introduction

The main ideas of the researched problem were presented at the Fraunhofer Institute Computer Architecture and Software Technology (Fraunhofer FIRST), Berlin, Germany and at the University of Applied Sciences Berlin, in the internal interviews.

Researching the Real-Time Network Applications, scientists defined the problem: how to predict the possibility of the Real-Time conference based on the Internet technology network? Even if the bandwidth of the connection is good, the packets transfer delays might prevent real-time conferences.

Nowadays, multimedia systems for the real-time conferencing become more and more challenging – ranging from the High-resolution conferences for the big halls, to the digital planetarium installations, and it is quite possibly that, in the nearest future, to the digital real-time planetarium which shows the video from the digital telescope which runs over the orbit around the Earth (or video from distributed space observatories). The other obvious alternative of the video source for these systems might be programs for 3 dimensional graphics (3D graphics). The example of a big installation for a video output is presented on Fig. 1.

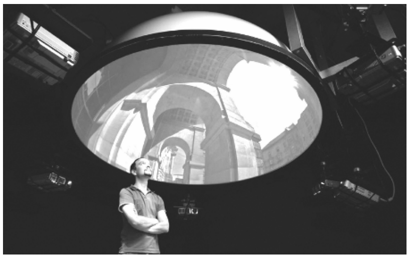

Fig. 1. The video installation

But, in most cases, for above-mentioned systems we need big and expensive computer clusters (to generate or compress the video flow). We want to use the video sources and the video outputs remotely with the best possible quality and we want to know if a real-time conference is possible with the Internet technology connection that we have. We want to know the most important set of the connection parameters and we want to get the simulation of presented problem.

Below, part II presents the problem-relative solutions. Part II gives information about the current solutions in the similar area to the presented research. Common information about prerequisites for research is presented in part III. This part presents the most important references, which are the base of current research. Part IV describes the main idea of the algorithm for predicting the packets transfer delay. The algorithm is based on the Laplace transform in

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the shape of Stiltes integral. Part IV presents the new effective algorithm formulated from wide mathematical theory about the Laplace transform properties. Part V is the key part as it presents the main results of the simulation. Part V shows to us opportunities for practical application of theoretical knowledge described in part IV. And, therefore, part V is the major contribution of this paper. Part VI provides the conclusions.

II. The Problem-Relative Solutions

At the present moment lot of products are produced under the title: "real-time conferencing". In this paper, real-time conferencing is the process with the defined properties:

- the conference is produced between two remote computer systems (between two participants);
- the systems produce data blocks;
- a time required for generating or obtaining the data block is available for measurement;
- a data block must be transmitted by the Internet-technology network;
- the amount of frames or packets which has the fixed length and required for transferring the data block (using the Internet technology) is available for counting;
- the amount of data blocks is available for counting;
- the timestamps of the packets arriving (to the receiver system) can be measured;
- the time required for processing the obtained data and the time required for the output device is also available for measurement.

The process with the abovementioned properties can be realized by the protocols:

- a transport protocol for real-time applications (RTP) [1];
- a real-time streaming protocol (RTSP) [2];
- a RTP control protocol (RTCP)[1];
- a Zimmermann RTP(ZRTP) [3];
- a resource reservation protocol (RSVP) [4];

and by the "specification of guaranteed quality of service" (QoS) [5].*

But, presented research has one more purpose: to produce as much control as possible (if and only if it is possible) over the network components for the purpose of obtaining the best possible quality of the real-time process.

It is not possible to use most of presented protocols for above-mentioned purpose. RTP, RTCP** and ZRTP gathers statistics on a media connection and information such as bytes sent, packets sent, lost packets, jitter, feedback and round trip delay. An application may use this information to increase the quality of service perhaps by limiting flow, or maybe using a low compression codec instead of a high compression codec [1]. So, the application should solve the frequency predicting problems and operate with the data and channel. RTSP takes advantage of streaming which breaks data into many packets sized according to the bandwith available between client and server [2]. But, it does not control the frequency of packets. The specification of guaranteed quality of service [5] says that the fixed delay is a property of the chosen path, which is determined not by guaranteed service but by the setup mechanism. Only queuing delay is determined by guaranteed service.

In this case, only RSVP is able to provide the required mechanisms. Presented research is focused on possible improving the functionality of RSVP for the rate-sensitive traffic. This paper is focused on the mathematical model for the real-time conferencing process, and on the predicting of the possibility of real-time conference with the fixed quality.

*It is possible to find more protocols that fit to the definition of conferencing process. This paper provides only the example for most famous protocols.

**RTP and RTCP exist as an Internet Standard.

III. Prerequisites For Research

A lot of recent scientific papers about the network traffic describes the self-similar nature of traffic. Basic ideas and results are presented in [6]. The research presented in [6] is verified by many authors, including [7-9].

This article presents the new mathematical model of the data blocks transmitting delay in the network with the self-similar nature of traffic.

For obtaining the expected value of the data block delay we use mathematical apparatus presented in [10]. The author of [10] applies the theoretical knowledge presented in chapter II of [10] to the distributed data base management systems (DBMS) algorithms, while our paper is focused on the networking and shows the new application of the theoretical knowledge.

In the presented article we use the definition of the process of data transmitting as the process described in details in [11].

Let's formulate an algorithm, using [10] in chapter IV of this paper.

IV. Algorithm For Determination Of The Time Of Receiving-transmitting A Data Block

To determine and predict the accurate time of receiving-transmitting a data block let's formulate the algorithm "A" from [10]

A1. Find the cumulative distribution function (CDF) *F(t)*; where *F(t)* is the CDF of random time of one packet transmitting.

A2. Find the Laplace transform in the shape of Stiltes integral (further Laplace-Stiltes transform) from *F(t)*.

$$\Psi(\omega) = \int_0^\infty e^{-\omega t} dF(t) \qquad (1)$$

A3. Experimentally determine the probabilities P_i that we need n packets for transmitting of the data block.

A4. Find the generating function of the packets count*.

$$\gamma(z) = \sum_{i=0}^{\infty} P_i \cdot z^i \qquad (2)$$

A5. Insert (1) into (2) to get the Laplas-Stiltes transform of the time to transfer all packets.

*In the strict sense the infinity limit in the sum at (2) makes (modifies) from algorithm a calculation method (in computer science an algorithm without finite steps property called - calculation method [16]).

$$T(\omega) = \gamma(\Psi(\omega)) \tag{3}$$

A6. Find the first derivative from (3).
A7. To use the property of Laplas-Stiltes transform [10]

$$T^{(n)}(0) = (-1)^n \cdot M(\xi^n) \tag{4}$$

where $T^{(n)}(0)$ – is the n derivatives $T(S)$, when $S=0$, $M(\xi^n)$ – the n^{th} initial moment of the random value ξ; $M(\xi^n)$ – expected value.

V. The Matematical Model Based On The Algorithm A

Obviously, the simulation based on algorithm A depends on the $F(t)$. We propose to use well-known approximations of network traffic according to the chapter III of this paper. The approximation ideas and models are presented in [12].

Undetermined time fractals have been widely used to model a number of phenomena in diverse fields from biology to finance and networking*.

In chapter 9 of [13] the set of undetermined time fractals, including the fractional Brownian motion (FBM), is presented. The existence of FBM was established by Benoit B. Mandelbrot and J. W. Van Ness [14].

FBM takes into consideration the Herst parameter, nowadays the Herst parameter is the main Self-Similarity metric. Description of Herst parameter mathematics runs out of this paper frames, but it is sufficiently represented in [6-9].

The base of FBM is the Gaussian Random Walk (GRW) [13]. Let's see how algorithm A functions with GRW apparatus.

The first derivative from CDF is the Probability density function (PDF). It is easier to use (1) transformed to the shape (5), when here and down the paper $F(t)$ is the PDF.

In this case, (1) transforms to

$$L(S) = \int_0^\infty e^{-St} F(t)\, dt \tag{5}$$

and we get the Laplace transform.

The PDF of GRW is the normal distribution**

$$F(x) = \frac{1}{\sqrt{2\pi} \cdot \sigma} \cdot e^{\left(-\frac{1}{2}\left(\frac{x-\mu}{\sigma}\right)^2\right)} \tag{6}$$

where μ is the expected value, σ is the variance and x is the variable of PDF.

*Not all undetermined time fractals fit perfectly to build the simulation using the algorithm A; nowadays we did not find solution for undetermined Wierstrass-Mandelbrot time fractal [15].

**The normal distribution, also called the Gaussian distribution, is an important family of continuous probability distribution, applicable in many fields.

Let's $\mu = 0$, according to [13].

Using (5) we get the Laplace transform of (6)

$$L(S) = \frac{\sqrt{\pi} \cdot \sigma \cdot e^{\left(\frac{\sigma^2 \cdot S^2}{2}\right)} \cdot \left(1 - erf\left(\frac{\sqrt{2} \cdot \sigma \cdot S}{2}\right)\right)}{\sqrt{2\pi} \cdot \sigma} \tag{7}$$

Then, we use (3) and (4) to get the expected time of transmitting the data block (First, we obtain the sum similar to (2) inserting $L(S)$ from (7) instead of z in (2). Then we get the first derivative from this sum, and afterward we assign $S=0;$ so we get $T^{(1)}(0)$. Finally we multiply the sum to $(-1)^1$. So, we get (8)).

$$M(\xi^1) = (-1)^1 \cdot \sum_{i=0}^{\infty} i \cdot P_i \cdot \left(\frac{1}{2}\right)^{i-1} \cdot \left(-\frac{\sigma}{\sqrt{2 \cdot \pi}}\right) \tag{8}$$

P_i – is the probability that for one data block we need i packets to transmit the data block.

$M(\xi^1)$ – is the expected time for transmitting one data block. σ – is the parameter of the packets transfer time variance. The longer time period for measuring σ, the more accurate is σ. We propose to calculate σ in the first seconds of data transferring and then to renew the σ value in the user-defined frequency.

If I is the quantity of probabilities P_i, then in the experiment with networking equipment, I is always a natural finite number. In this case, P_i when $i>I$ is 0 and all next members of sum (8) become 0. To transform the algorithm A to an algorithm in the strict sense, we give the finite steps property for A, changing the infinity symbol in the sum (8) to the finite natural number I.

The data measured in the real-time conference experiment is presented in table 1. The result of (8) with different σ is presented on Fig. 2. Fig. 2 shows that the GRW (non-fractal) network process depends only on σ and probabilities P_i. The model (8) provides the rule: the higher the process variance, the longer is the time of the data block transferring. Taking into account the results of [11] we can make an assumption about the future avoiding the self-similar models in preference to the random process models.

The model (8) does not take into account the possible self-similar nature of traffic. For this purpose we get (9) the PDF of FBM from [13].

TABLE I
The Data Fragment

Probability Number	Probability value	Packets count
1	0,011	1
2	0,012	2
3	0,303	3
4	0,510	4
5	0,163	5
6	0,001	6

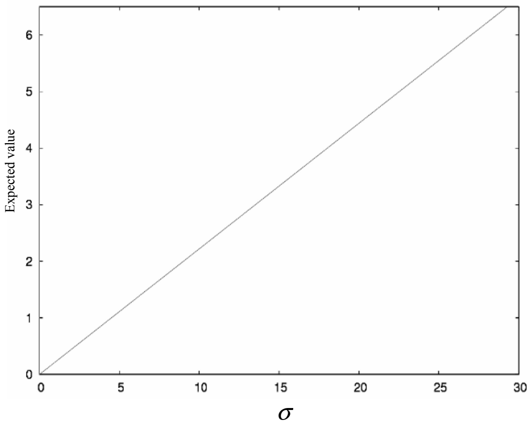

Fig. 2. The dependence of the data block transmission expected value to the process variance

Let's, $P(A/B)$ – is the conditional probability of event A, when the event B is established. Lets, $X(t_k)$ – is a random value in the moment of the time t_k, k – is a natural number. We are interested in a process such as $P(X(t_k) \le x_k / X(t_{k-1}) = x_{k-1})$, where: $t_1<t_2<\ldots<t_{k-1}<t_k$; here and down the paper, t_1 – is the initial time of the process, t_2 – is the time moment of obtaining the probability of the event A (this notation is equal to [13]). Lets $T_\Delta = t_2 - t_1$.

σ – is the packets transfer time variance, x is the variable of PDF, H – is the Herst parameter.

$$F(x) = \frac{1}{\sqrt{2\pi} \cdot \sigma \cdot {T_\Delta}^H} \cdot e^{\left(-\frac{x^2}{2 \cdot \sigma^2 \cdot {T_\Delta}^{2H}}\right)} \tag{9}$$

If $\sigma > 0$ and $0<H<1$, then the Laplace transform of (9) is

$$L(S) = \frac{e^{\left(\frac{\sigma^2 \cdot S^2 \cdot {T_\Delta}^{2H}}{2}\right)} \cdot \left[1 - erf\left(\frac{\sqrt{2} \cdot \sigma \cdot S \cdot \sqrt{{T_\Delta}^{2H}}}{2}\right)\right]}{2} \tag{10}$$

Using (3) and (4) we get the expected time of transmitting the data block for FBM.

$$M(\xi^1) = \sum_{i=0}^{I} i \cdot P_i \cdot \left(\frac{1}{2}\right)^{i-1} \cdot \left(\frac{\sigma \cdot {T_\Delta}^H}{\sqrt{2 \cdot \pi}}\right) \tag{11}$$

$M(\xi^1)$ – is the expected value of the data block transmission time. Fig. 3-5 presents the results of mathematical modeling.

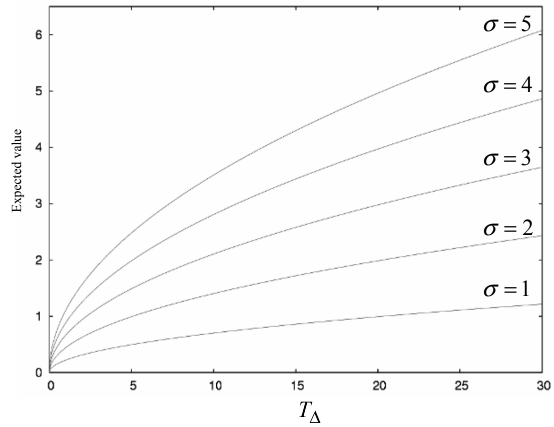

Fig. 3. The results of calculating the expected value of the data block transmission time, when H=0.3

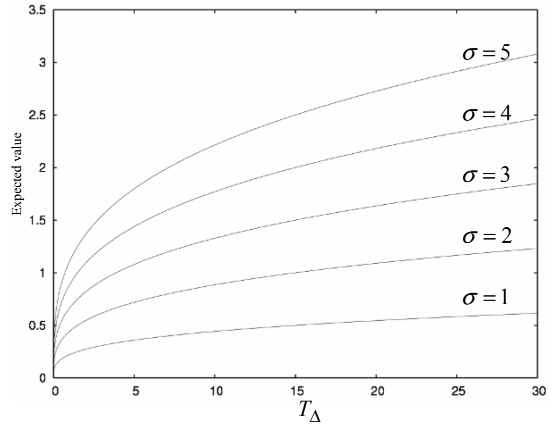

Fig. 4. The results of calculating the expected value of the data block transmission time, when H=0.5

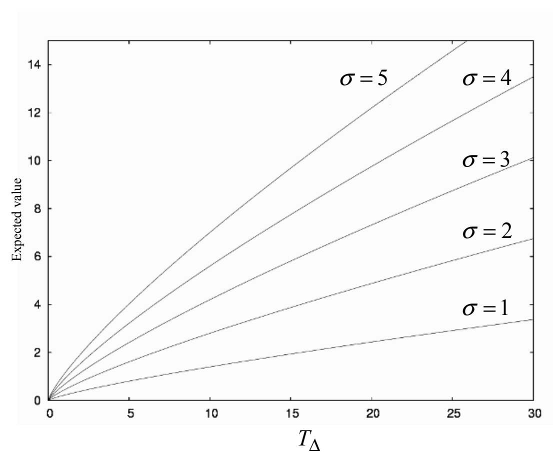

Fig. 5. The results of calculating the expected value of the data block transmission time, when H=0.8

Fig. 3-5 shows the results of the simulation, where the time of the data block transfer delay is obtained. Obviously, Fig. 3-5 shows that the dependence of the process on the previous events makes the forecast more problematic, and, in this case, the expected value of the block transmission time is increasing. If the variance σ is low, then it is possible to make the forecast to the longer T_Δ . Fig 2-5 shows the process with the variance up to 5; that is not typical for the modern networks. Fig 2-5 shows the process that is similar to a connection with a really bad quality, for the purpose of demonstrating the mathematical apparatus on the small values.

The software based on presented research might function in the following way: T_Δ is defined by the system user; P_i, I, σ are measured during the short-time test or during the conference. Then the software calculates the expected value. If the expected value is too high, than the system proposes to decrease the variance σ or to setup the lower T_Δ . In modern networks it is possible to get the different σ switching between the available routes in the routing map or upgrading the existed channel.

Presented model does not depend on the transport layer of the networking algorithm. If the packets loss exists and is controlled by the control protocol (or by the real-time protocol, like RTSP), then the variance σ grows. If the packets loss is not controlled, then it means that the data fragments loss is allowed by the system user.

VI. Conclusions.

Obtained knowledge can be effectively used for enhancing the quality of service of the network based on the Internet technology. The main application of the presented model is the real-time conferencing and a video flow transferring.

Conducted work gave the following outcomes:

– new method of the networking traffic modeling is designed;

– the new method takes into account all the basic properties of the modern Internet-technology network;

– this paper shows the importance in selecting the model;

– the opportunity to compare obtained data from the different models with the identical input values.

Presented research gives the opportunity to implement obtained knowledge to the software. The software tests and implementations is the main subject of the future research.

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