ISSN 1392-8619 print/ISSN 1822-3613 online
ŪKIO TECHNOLOGINIS IR EKONOMINIS VYSTYMAS
TECHNOLOGICAL AND ECONOMIC DEVELOPMENT OF ECONOMY
http://www.tede.vgtu.lt
2006, Vol XII, No 4, 327–333

INVESTIGATION OF BATCH DATA FLOW TRANSMISSION PARAMETERS

Paulius Tervydis[1], Ramutis Rindzevičius[2]

*Dept of Telecommunications and Electronics, Kaunas University of Technology,
Studentų g. 50-450, 51368 Kaunas, Lithuania
E-mail: [1]Paulius.Tervydis@takas.lt; [2]Ramutis.Rindzevicius@ktu.lt*

Received 15 June 2006; accepted 20 November 2006

Abstract. Rapid growth and increasing requirements for service quality, reliability, and efficiency have made traffic evaluation an essential consideration in design and operation of data transmission networks. In this paper we analyse performance measures of data transmission network servicing batch data flows. The article provides analysis of the one channel queuing system transmitting data packet flow general distributed. In this article we introduce queuing system with finite and infinite buffer capacity. In this article we analyse such data packets transmission performance parameters: average time spent by data packet in system, data packet delay variation, data packet loss. The data packet arrival rate and channel data packet transmission rate are independent of system state. A key mechanism of the proposed traffic model is that packet time spent in the system depends on channel utilization level, packet flow type and size of transmitted data. The proposed simulation model can be used when packet interarrival time follows such distributions: Pareto, Poisson, uniform, normal distribution. Simulation results of performance measures are defined by mathematical models. Considering simulation results recommendations were proposed to improve performance measures for batch data flow transmission over data networks.

Keywords: network, data, flow, transmission, parameters.

1. Introduction

A network operator must decide what services the network should deliver to the end user and the level of service quality that the user should experience. This is true for any telecommunications network, whether it is circuit or packet-switched, wired or wireless, optical or copper-based, and it is independent of the transmission technology applied. Further decisions to be made may include the type and layout of the network infrastructure for supporting the services, and the choice of techniques to be used for handling the information transport.

The goal when planning a telecommunication system is to adjust the amount of equipment so that variations in the subscriber demand for calls or data transmission parameters can be satisfied without noticeable inconvenience while the cost of the installations is as small as possible.

To ensure particular data transmission parameters it is necessary to investigate:

- user requirements for data transmission parameters;
- parameters and peculiarities of real traffic generated by a user;
- dynamics of user activity;
- number of users and its growth possibility.

Knowing the information it is possible to select appropriate infrastructure for data transmission: equipment, technology, topology, number of channels and their capacity.

For that purpose data network modelling and calculations should be applied. Role for the modelling is to expose properties of measurements that are important for particular engineering problems. This can happen when parameters are interpretable. Interpretable here means that that the parameter value has a relationship to an underlying cause or engineering question. As interpretable parameters change, we understand how something in real system changes – like system performance [1].

In simulations one often needs the ability to generate random but 'realistic' data as input. The starting point is usually measurements and a data model. Using a data model, one can generate large amounts of new data that agree in key properties with empirical measurements. In cases where parameters are interpretable, data or simulation models allow asking 'what if' questions. One can generate simulation inputs that span a range of realistic values and explore the resulting consequences.

Our proposed simulation model can be used to analyse the main data packet flow transmission performance parameters.

2. Problem of batch data flow transmission

Evaluation of data transmission parameters is complicated due to random nature of user's activity and intensity of data traffic generation [2]. Therefore, methods of statistical analysis should be used to analyse and describe the real traffic generated by a user. Usually more than one user is served by the system or its component, and that is another reason that gives randomness to the data transmission parameters.

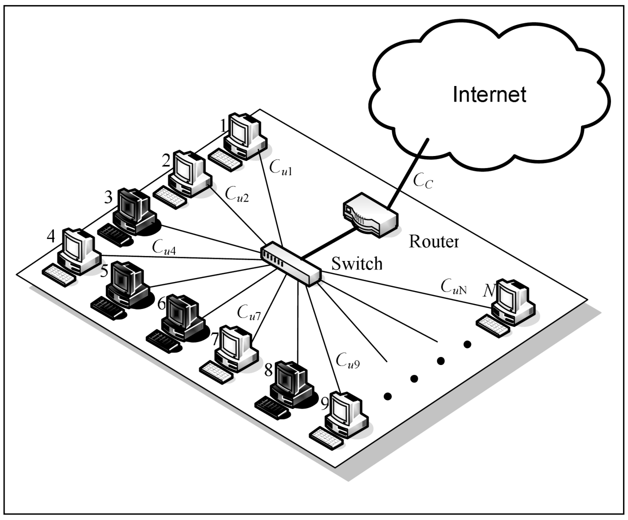

Fig 1. Scheme of a typical local area network

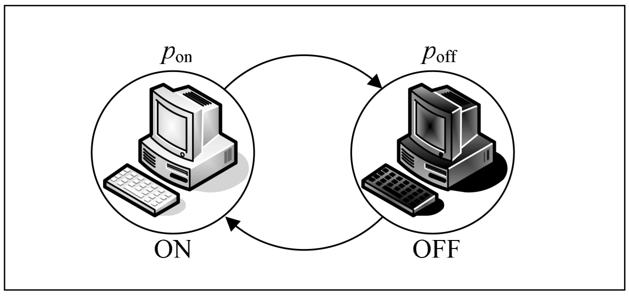

Fig 2. Two states of a network user

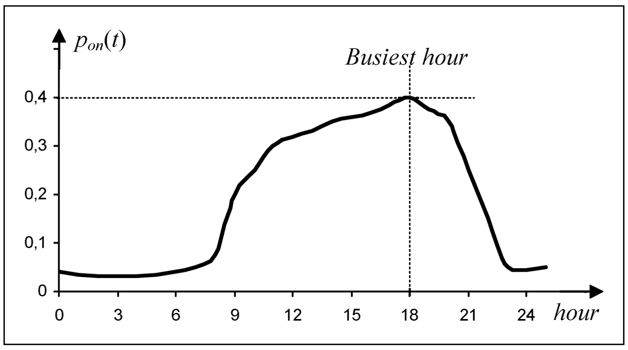

Fig 3. User's activity over daytime

Further, in this section the problem is illustrated by analysis of batch data flow transmission from a typical local area network (LAN) to Internet over single data channel (Fig 1).

Usually, LAN servicing a number (N) of users is connected to Internet network over single channel with particular (C_C) capacity [3]. All data generated by LAN users to Internet network is transmitted through the edge router. In present routers it is possible to configure a particular data rate (C_u) for a user, which cannot be exceeded. The rate may be different for different users or user classes, but in this case all users have the same C_u value.

Network user can be in two states: ON and OFF. In ON state user transmits his data, in OFF state the user is idle. User's state can be described by state probabilities: p_{on} and p_{off} (Fig 2).

Network administrator using traffic monitoring tools can collect statistics of user activity or amount of generated traffic. The graph of a typical user's activity over daytime is presented in Fig 3. The hour when the p_{on} has a maximum value is called the "*busiest hour*".

In this case, the probability for a particular number of active users (n_{on}) in the network can be calculated by binomial distribution formula [4]:

$$P(N, n_{on}) = \binom{N}{n_{on}} p_{on}{}^{n_{on}} (1 - p_{on})^{N - n_{on}} . \quad (1)$$

Then the mean and standard deviation of a number of active users are given by

$$\overline{n_{on}} = N \cdot p_{on} \quad (2)$$

and

$$\sigma_{n_{on}} = \sqrt{N \cdot p_{on} \cdot (1 - p_{on})} . \quad (3)$$

The probability distribution for n_{on} during *"busiest hour"* (Fig 3), when $p_{on} = 0.4$ and $N = 100$, is shown in Fig 4.

It is obvious, that in case when $n_{on} \cdot C_u > C_C$, a user, due to the level of channel's capacity utilization, will not be

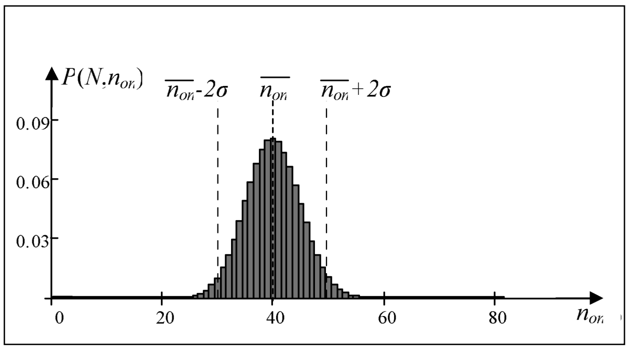

Fig 4. Probability distribution of n_{on}, when $p_{on} = 0.4$ and $N = 100$

able to transmit his data at the C_u rate. When N number and p_{on} results to:

$$n_{on} \le \frac{C_C}{C_u} \tag{4}$$

the actual data transmission rate C_{ur} is not lower or equal to C_u, and when

$$n_{on} > \frac{C_C}{C_u} \tag{5}$$

the actual data transmission rate C_{ur} will be lower than C_u. Therefore, the actual or real data transmission rate as a function of n_{on} is given by:

$$C_{ur}(n_{on}) = \begin{cases} \dfrac{C_C}{n_{on}}, if \dfrac{C_C}{n_{on}} < C_u; \\ C_u, if \dfrac{C_C}{n_{on}} \ge C_u. \end{cases} \tag{6}$$

Fig 5 presents how the actual user's data transmission rate depends on the total number of network users (N) during "busiest hour". The C_{ur} values are calculated using (6) formula. The mean and the confidence level values of C_{ur} are calculated using (2), (3) and (6) formulae (Fig 4).

The same calculations can be used to evaluate how the p_{on} dynamics (Fig 3) over time affects the actual user's data transmission rate. The dependence is illustrated in Fig 6.

The actual data transmission rate is critical to the data transmission parameters. If d is data packet length, then data packet transmission delay is given by [5]:

$$t_{out} = \frac{d}{C_{ur}}. \tag{7}$$

If C_{ur} is random, then t_{out} will be random too. Variation of data packet transmission delay, also called jitter, is often more important parameter than mean transmission delay. It is very important to the quality of real-time voice, video or multimedia transmissions over data networks.

In general case, C_{ur} is affected by the actual load of the main channel. The load is caused by the number of active users and also changes in time (Fig 3). How it affects a value of data packet transmission delay is illustrated in Fig 7, where the three slices of Fig 6 are presented.

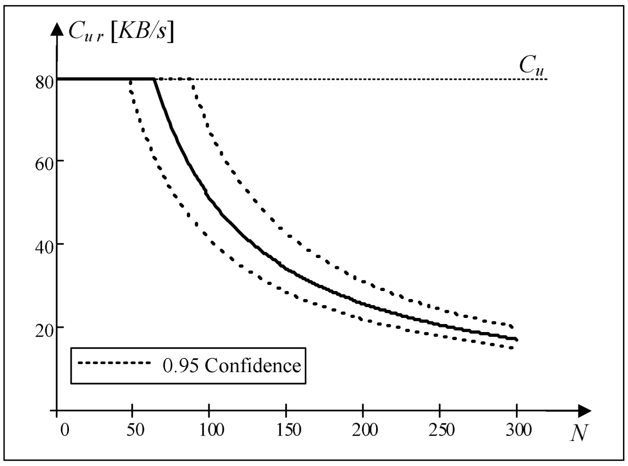

Fig 5. C_{ur} dependence on N, when p_{on} = 0.4, C_u = 80 [KB/s] and C_C = 2048 [KB/s]

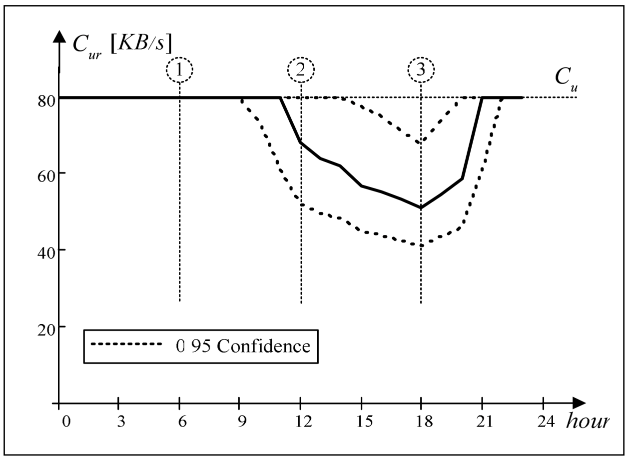

Fig 6. C_{ur} values over daytime, when N = 100, C_u = 80 [KB/s] and C_C = 2048 [KB/s]

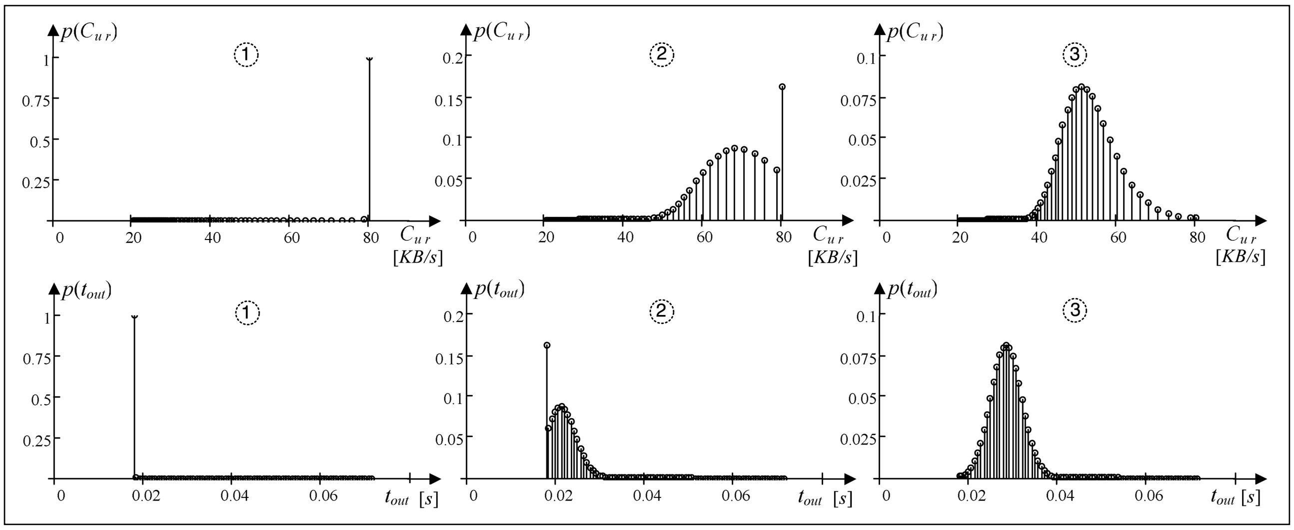

Fig 7. Distributions of C_{ur} and t_{out}, when N = 100, p_{on} = 0.4, C_u = 80 [KB/s], C_C = 2048 [KB/s], d = 1500 [B]

Therefore, there is a need for methods or models that can be used for data transmission parameter analysis, when data flow parameters are random and similar to one present in real networks.

In next sections we present our simulation model that can be used for that.

3. The proposed simulation model

There are many simulation and analytical models, that can be used for analysis of data transmission parameters [6]. But many of them are good only for particular conditions, for example, classical Markov models are simple and give precise results only when incoming data flow is Poisson distributed and data flow process time is exponentially distributed. There are models and computer simulation software that can be used to model and to analyse data flow parameters for different network topologies, technologies, etc. [7]. Unfortunately, the models have many hidden peculiarities, which make it difficult to understand how the results are calculated, and often there is a need to change some components of modelled system or to change the nature of initial parameters to achieve results similar to measured in real networks.

The main strength of our proposed model is in its simplicity. It is very clear and can be easily adjusted and programmed. To illustrate the main idea Fig 8 is presented. First of all, a set or array $\{t_{in}\}$ of time moment values of incoming data packets is collected from real data flow measurements or generated according to particular distribution. In the next step, a set or array $\{t_{out}\}$ of data packet transmission delay (or time) values is prepared; it also can be based upon collected statistics of real (or actual) data transmission rate or delay, or can be generated according to particular distribution (Fig 7)[6]. Here the data packet arrival rate and data packet transmission rate over channel are independent of system state. Having the values, it is cyclically compared if a packet is transmitted before next packet arrives, it allows determining whether a packet will be placed in a buffer or transmitted immediately.

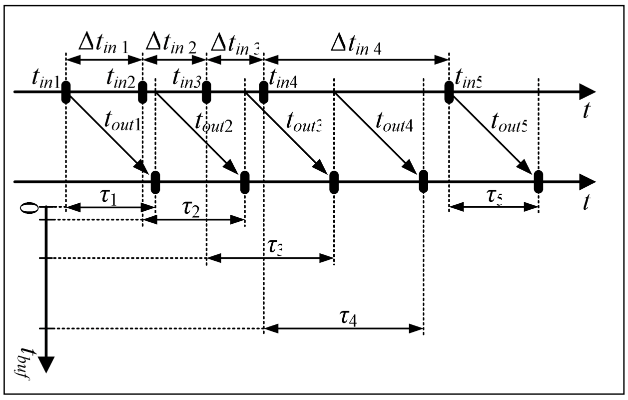

Fig 8. The main idea of the proposed simulation model

For example, as shown in Fig 8, the second packet arrives to the system before the end of the first packet transmission ($t_{in2} > t_{in1} + t_{out1}$), so the second packet is placed in a buffer for particular duration:

$$t_{buf\,2} = t_{in1} + t_{out1} - t_{in2} = t_{out1} - \Delta t_{in1} \tag{8}$$

and the total time spent in system for the second packet is given by:

$$\tau_2 = t_{buf\,2} + t_{out2}\,. \tag{9}$$

It is shown in the figure that, if time difference between packet arrival moments is smaller than packet transmission time, then packets are aggregated in the buffer – time spent in buffer and in the system also is aggregated (increases). And if time difference between packet arrival moments is bigger than packet transmission time, the buffer is emptied – time spent in buffer and in the system decreases. The situation is shown in the Fig 8, the time difference between 4-*th* and 5-*th* packets is enough to empty the buffer (transmission of 3-rd and 4-th packet are finished), and the time spent in system for 5-th packet is equal to t_{out5}.

3.1. The simulation model with infinite buffer

The algorithm of the model with infinite buffer (or infinite waiting time in buffer) is presented in Fig 9.

The system will be in stationary state if intensity of data packet arrival process is less or equal to systems ability to process the incoming data flow. When the requirements are not fulfilled, the system (network, data channel, etc.) is congested – data packets are placed in buffers or blocked

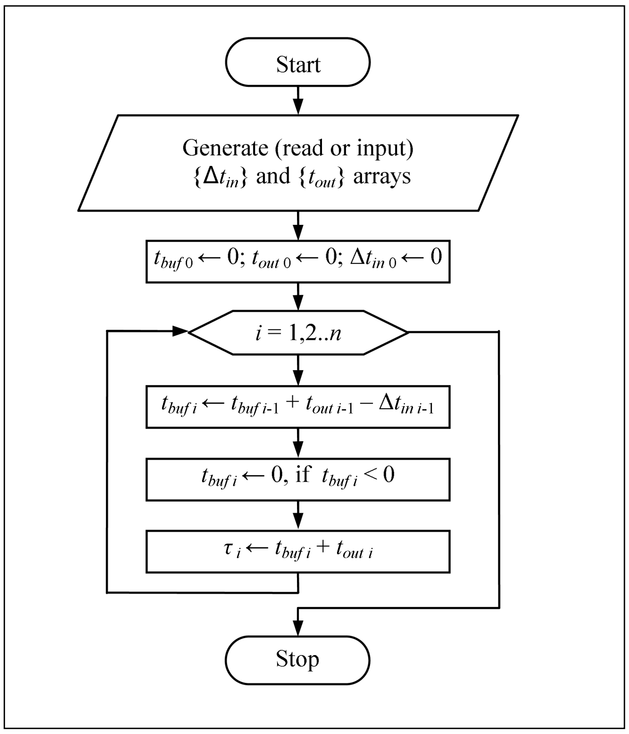

Fig 9. Algorithm of the simulation model with infinite buffer

on send phases, the data packet waiting time in buffer and total time spent in the system dramatically increases.

In real networks the problem is solved by feedback and time-out mechanisms between end stations.

Therefore, the proposed simulation model can be used to analyse data transmission parameters when stationary state requirements are met. Due to infinite buffer, there are no data packet losses.

In the next section, we present simulation model with finite waiting time in buffer (or with finite buffer), which can be used for the evaluation of data packet losses.

3.2. The simulation model with finite waiting time in buffer

In real networks or telecommunication systems buffers or memory registers are limited. Therefore, the waiting time (t_B) in buffer is also limited. In such case, if intensity of data packet arrival process is bigger than systems ability to process the incoming data flow, the buffers are overflowed and packets, which do not have room to fit in the buffers at the arrival moment, are discarded from the system.

The algorithm of the model with finite waiting time in buffer (or finite buffer) is presented in Fig 10.

Using the model it is possible to evaluate data loss ratio, data packet spent time in system and in buffer. Simulated results can be analysed using statistics methods. The proposed algorithm can be adjusted to model systems with finite buffer capacity expressed in bytes. Also, it can be modified to model data transmissions where data packet is discarded if data packet spent time in system exceeds particular value.

3.3. Evaluation of the proposed simulation model

To test the proposed models we compared simulation results with known analytical models results.

Proposed simulation models were programmed and tested with Mathcad application. To compare the simulation results with M/M/1 [8] model the following conditions were taken:

- Δt_{in} ~ Exponential (λ) – time difference between arriving data packets is distributed according to exponential distribution, with mean $1/\lambda$ value (where λ – data packet arrival intensity);
- t_{out} ~ Exponential (μ) – data packet process time in the system is distributed according to exponential distribution, with mean $1/\mu$ value (where μ – data packet process intensity);
- number of generated values – n = 1000000.

The comparison of results is presented in Table 1.

The similar precision is achieved when simulation results are compared to results calculated by M/D/1 and M/G/1 models [8] (Table 2).

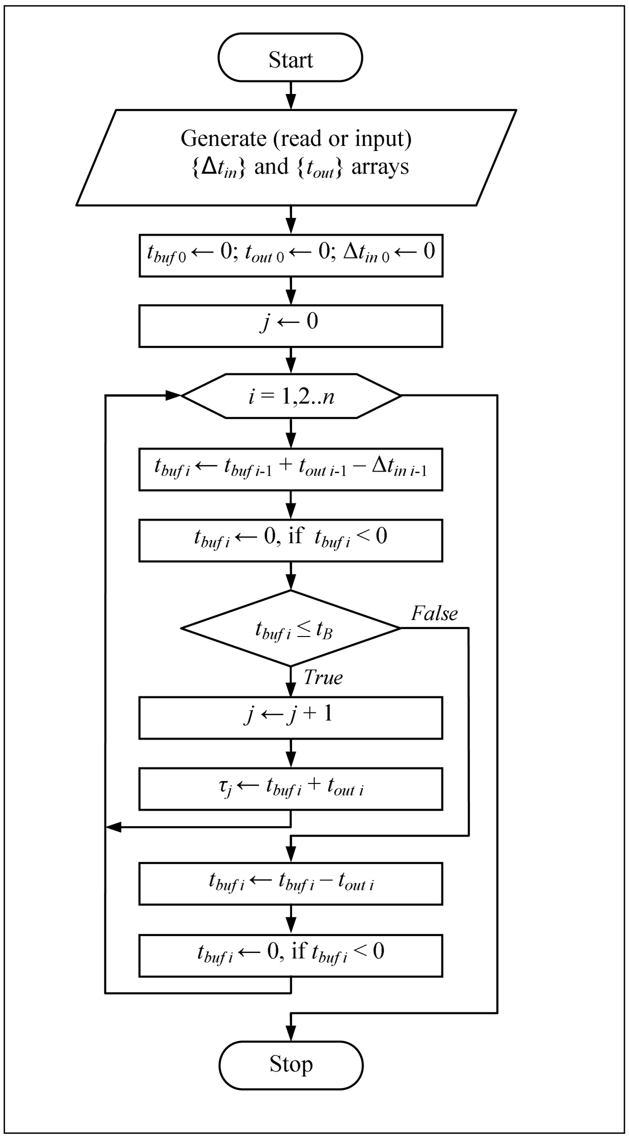

Fig 10. Algorithm of the simulation model with finite waiting time (t_B) in buffer

Table 1. Comparison of results between M/M/1 and the simulation model, when μ = 10 [pct/s]

Mean time spent in the system [s]		
λ [pct/s]	M/M/1	Sim. model
1	0.11111	0.11103
2	0.12500	0.12502
3	0.14285	0.14307
4	0.16666	0.16701
5	0.20000	0.20011
6	0.25000	0.24999
7	0.33333	0.33149
8	0.50000	0.50079
9	1.00000	1.02360
Corr(M/M/1, Model) = 0.999928		

Table 2. Comparison of results between M/D/1, M/G/1 and the simulation model, when μ = 10 [pct/s] (for M/D/1 and M/G/1) and t_{out} st. dev. σ = 0.025 [s] (for M/G/1)

Mean time spent in the system [s]				
λ [pct/s]	M/D/1	Sim. model	M/G/1	Sim. model
1	0.10556	0.10557	0.10590	0.10594
3	0.12143	0.12156	0.12277	0.12282
5	0.15000	0.14994	0.15313	0.15320
7	0.21667	0.21628	0.22396	0.22440
9	0.55000	0.54506	0.57813	0.57672

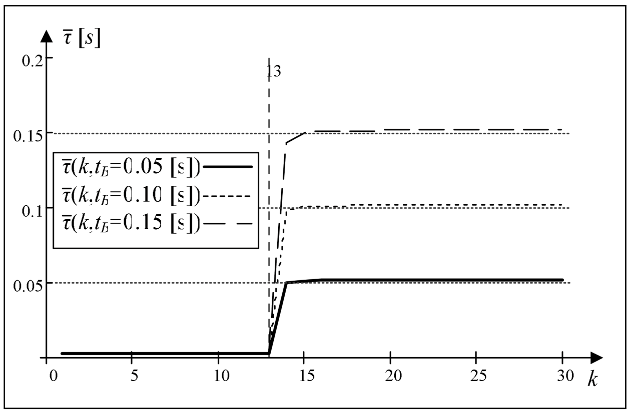

Fig 11. Dependence of mean time τ spent in system of a voice data packet on the number k of batch conversations

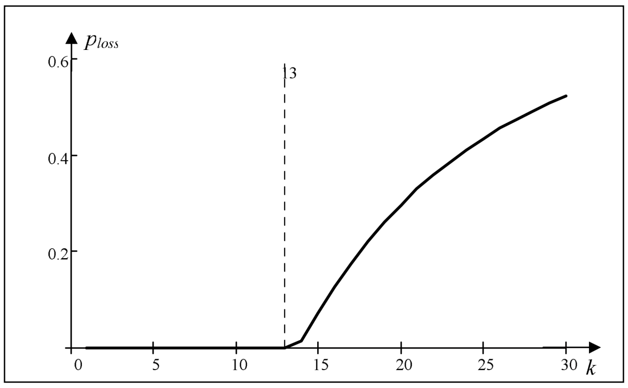

Fig 12. Dependence of loss probability p_{loss} of a voice data packet on the number k of batch conversations

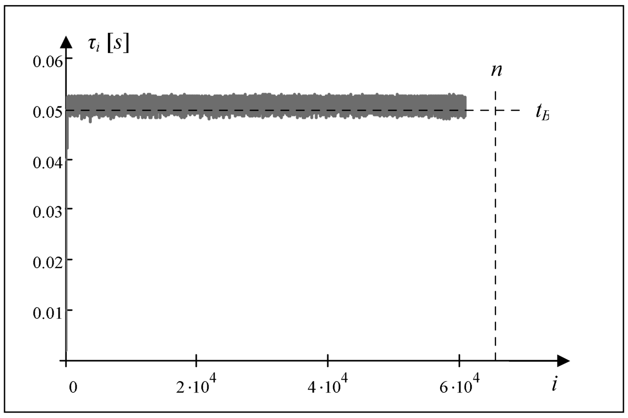

Fig 13. Simulation results of time spent in system values of a voice data packet, when n = 65536, k = 15, t_B = 0.05 [s], i = 1,2, ... n

4. Simulation results

In this section we present some simulation results, which were calculated using the proposed model with limited waiting time in buffer. The calculations were made to analyse how many voice (VoIP) conversations can be made at the same time moment during *"busiest hour"*, if one particular PC is replaced by VoIP gateway in Fig 1.

The distribution of actual data rate C_{ur} during *"busiest hour"* is presented in Fig 7, 3. If G.729 codec with 30 ms voice payload size is used, then the length (d) of voice data packet, including Ethernet, IP, UDP and RTP headers, is equal to 114 bytes [9]. Voice data packet process time can be calculated using (7) formula. And the mean value of the packets interarrival time is given by:

$$\Delta t_{in} = \frac{0.030}{k} \quad [\text{s}], \qquad (10)$$

where k – number of simultaneous voice conversations.

How the mean time spent in the system of a voice data packet depends on the number of simultaneous conversations is presented in Fig 11.

When $k \geq 13$ the $\bar{\tau}$ rapidly increases to particular t_B value. In the result some of voice data packets are lost (Fig 12).

An example of simulation results of time spent in system values of a voice data packet is shown in Fig 13.

It illustrates the situation when data packets arrive to the system with intensity bigger than systems ability to process the incoming data flow, data packet spent time in the system values is dispersed around t_B value and the number of processed packets is lower than n (some of them are lost due to buffer overflow).

The simulation results show, that during "*busiest hour*" the actual data transmission rate is enough only for 12 simultaneous (batch) voice data transmissions.

5. Conclusions

The load dynamics and utilization level of data transmission channel are crucial to the batch data flow parameters. Our calculations show that delay, delay variation and data packet loss parameters can be unacceptable when the channel's throughput is fully utilized. This is an important consideration for traffic shaping and control schemes, which also can be evaluated using the proposed simulation model for any type of $\{\Delta t_{in}\}$ and $\{t_{out}\}$ value distributions.

References

1. Crovella, M.; Krishnamurthy, B. Internet Measurement. Infrastructure, Traffic, and Applications. John Wiley & Sons, 2006. 495 p.

2. Choi, B. Y.; Moon, S.; Zhang, Z. L.; Papagiannaki, K.; Diot, C. Analysis of point-to-point delay in an operational network. In: Proceedings of IEEE INFOCOM Conference, Hong Kong, 2004, p. 1–11.
3. Duffield, N.; Lund, C. Predicting resource usage and estimation accuracy in an IP flow measurement collection infrastructure. In: Proceedings of the ACM SIGCOMM Internet Measurement Conference, 2003, p. 1–13.
4. Pióro, M.; Medhi, D. Routing, Flow, and Capacity Design in Communication and Computer Networks. Publisher: Morgan Kaufmann, 2004. 800 p.
5. Tanenbaum, A. S. Computer Networks, Fourth Edition. Prentice Hall, 2003. 912 p.
6. Duffied, N.; Lund, C.; Thorup, M. Estimation flow distributions from sampled flow statistics. In: Proceedings of ACM SIGCOMM, 2003, p. 325–336.
7. Feldmann, A.; Greenberg, A.; Lund, C.; Reingold, N.; Rexford, J.; True, F. Deriving traffic demands for operational IP networks: Methodology and experience. In: Proceedings of ACM SIGCOOM, 2000, p. 257–270.
8. Iversen, V. B. Teletraffic Engineering and Network Planning. http://www.com.dtu.dk/education/34340/material/telenookpdf.pdf, [revised 24 03 2006].
9. Voice Over IP – Per Call Bandwidth Consumption. http://www.cisco.com/warp/public/788/pkt-voice-general/ bwidth_consume.html, [revised 21 02 2006].

PLIŪPSNINIO DUOMENŲ SRAUTO PERDAVIMO PARAMETRŲ TYRIMAS

P. Tervydis, R. Rindzevičius

Santrauka

Tiriama pliūpsninio duomenų srauto perdavimo duomenų perdavimo tinklais problema. Analizuojami pliūpsninio duomenų srauto, kurį generuoja atsitiktinis vartotojų skaičius, perdavimo vienu kanalu ypatumai. Atlikta analizė parodo, kad dėl vartotojų skaičiaus aktyvumo dinamikos laike jų sukuriamas atsitiktinis duomenų srautas yra pliūpsninis. Dėl tokio duomenų srauto pobūdžio atitinkamai priklauso ir pagrindiniai duomenų perdavimo parametrai: duomenų paketų perdavimo trukmė (vėlinimas), aritmetinis vėlinimo vidurkis, jo sklaida, paketų praradimas. Nustatyta, kad laikotarpiai tarp pliūpsninio duomenų srauto paketų ir jų perdavimo kanalu trukmės, atsižvelgiant į kanalo apkrovą, gali būti atsitiktinai pasiskirstę pagal eksponentinį, gama, Pareto, normalųjį ar kt. skirstinį. Pasiūlytas imitacinis modelis, kuriuo remiantis galima įvertinti tokio duomenų srauto perdavimo parametrų priklausomybes.

Reikšminiai žodžiai: tinklas, duomenys, srautas, perdavimas, parametrai.

Paulius TERVYDIS. Received his MSc degree from Kaunas University of Technology in 2003. Currently he is working for his PhD degree in the Department of Electronics Engineering of Kaunas University of Technology. His key areas of interest include the research of voice and data transmission efficiency over IP based networks. He is author of 12 papers in technical journals and conference proceedings.

Ramutis RINDZEVIČIUS. Associated Professor in the Department of Telecommunications of Kaunas University of Technology, Doctor of Electrical and Electronics engineering of Kaunas University of Technology. The field of research – analysis of telecommunication systems by means of simulation and analytical models. The internship took place at Kopenhagen Ericsson training courses in 1994 and at Moscow Electrotechnics Institute of Telecommunications in 1974, 1984. He published about 70 papers and three handbooks.