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SIP Flow Time Interval between Messages of Statistical Analysis

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***Abstract*—SIP protocol is most often used for voice information transmitting over IP networks via communication session. Communication session process has been managed by SIP proxy server. In order to determine its operating characteristics, such as message service time, and message blocking probability, it is necessary to know the statistical features of the served traffic including the time interval of incoming message flow, the time interval distribution between arriving messages, mean value and their possible deviation. The statistic characteristics of time intervals of individual SIP messages and the total flow entering into SIP Proxy server have been experimentally determined in the paper. It has been found that the inter-arrival time of the sum total of SIP message flow does not correspond to the Poisson distribution thus General distribution should be used in the tests.**

***Index Terms*—Telecommunication services, IP networks, packet switching, internet telephony, signalling.**

I. INTRODUCTION

IP-based network resources are increasingly used in voice services. During the provision of voice services, network devices are loaded with signalling and user information. Using packet mode communication it is difficult to guarantee the required quality of service associated with the establishment of the connection session and voice communication. SIP (Session Initiation Protocol) [1]–[3] is the main signalling protocol for the transmission of voice information over IP networks. One of the main characteristics of voice quality related to the servicing of signalling messages at signalling nodes is the time needed to establish the connection session. It depends not only on the intensity of voice services usage, but also on the algorithm implementing the service.

Analytical simulation models are used [4]–[7] in order to perform the time analysis needed to establish connection for services delivered over IP network. While developing any mass servicing system analytic or simulation model, distribution of incoming messages should be determined. For evaluation of statistical characteristics in analytical models of IP telephony networks it is often assumed that incoming messages enter the servicing system according to Poisson distribution [7]. This is correct if characteristics such as the time interval between the calls or the call handling time statistical characteristics are analysed for voice transmission through circuit switching networks [8].

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In packet networks, during any connection session, signalling and voice information message flows are created, the statistical characteristics of which can differ from the usually assumed Poisson distribution. It is also necessary to take into account that voice and signalling information flows put different loads on the network devices. Signalling information messages are processed in the SIP communication servers and their performance is important while servicing the messages flows.

In Fig. 1 typical signalling network that is used to provide voice service is presented, when voice service connection is made using one voice services provider in IP network. The network is made of SIP proxy servers (SIP PS), application server (AS), data bases (DB), media resource functions (MRF) and IPv4 network of signalling information transmission. SIP PS is the core element of VoIP network when SIP signalling is being used. This functional element is responsible for the forwarding of signalling messages between end users, to other SIP PS or to AS. AS, DB, MRF that are optional elements of VoIP networks. AS has the advanced voice services logic. The voice information for IVR services is kept in MRF. The supplementary information needed for services (i. e. service execution time, users of service) is stored in DB.

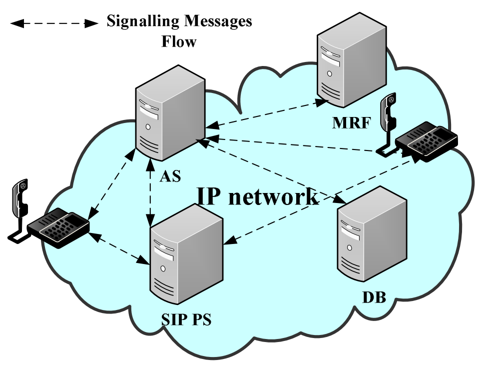

Fig. 1. Typical signalling network used to provide voice services when connection is made in IP network of one VoIP provider.

Statistical features of time intervals between SIP signalling messages arriving to SIP Proxy server (SIP PS) moments obtained during the experiment are presented in this work. The observation of message arriving times was established in SIP PS for one Lithuanian VoIP service provider. All observed calls are established using only one SIP PS. IPv4 is used for SIP messages transported for this service provider.

Signalling messages from the basic voice service connection algorithm entering SIP PS were recorded during the experiment (Fig. 2) [1], [8].

Upon initiating a call, INVITE message is created followed by announcements TRYING, RINGING, OK, ACK, BYE, which are related to the creation and termination of the connection session. These messages are interdependent and their sequence is determined by the service algorithm (Fig. 2). Some messages, for example, OK, are transmitted several times during one session.

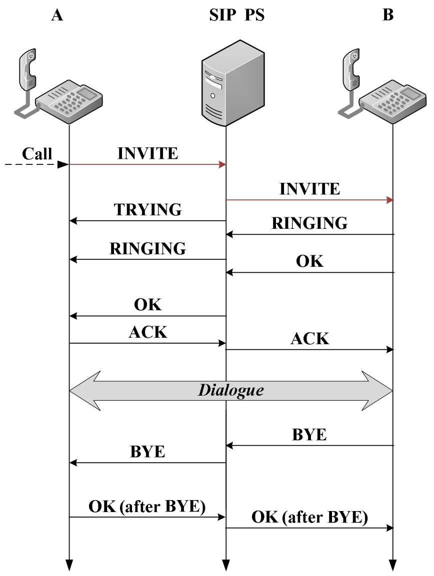

Fig 2. Basic call voice service algorithm.

II. Determination of the Distribution of Time Intervals between the Signalling Message Arrivals to the SIP PS

Times of signalling messages coming into the SIP PS servicing the basic voice calls are shown (Fig. 3).

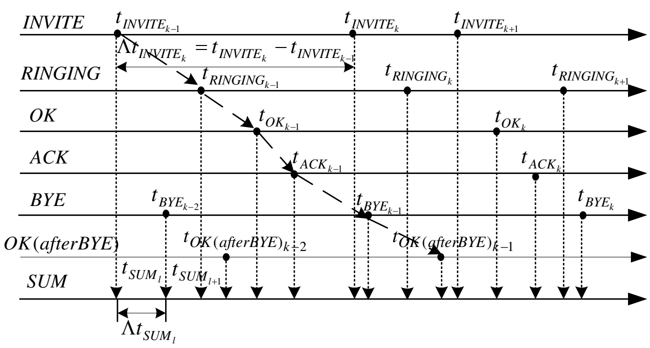

Fig. 3. The model of messages flow coming to SIP proxy.

During observation period 4954 basic voice service calls were serviced. Their arrival times in the SIP PS were captured. Traffic of each type of the signalling messages is characterized by the distributions of time intervals ($\Lambda t_{i_k} = t_{i_{k+1}} - t_{i_k}$) between two adjacent signalling messages of the same type, and the total flow – interval time between the arrival of two neighbouring ($\Lambda t_{SUM_l} = t_{SUM_{l+1}} - t_{SUM_l}$) signalling messages. The moments of appearance of the signalling messages for a single call servicing is strictly regulated voice service algorithm and the following inequality holds

$$t_{INVITE_k} < t_{RINGING_k} < t_{OK_k} < t_{ACK_k} < t_{BYE_k} < t_{OKafterBYE_k} \cdot \quad (1)$$

Basic voice service connection is initiated when the INVITE message is sent from the user equipment to the SIP PS server.

The averages of time intervals between the appearance of adjacent messages (separately for each type of messages and total flow) are found

$$\overline{\Lambda t}_i = \frac{\sum_{k=1}^{N_i} \Lambda t_{i_k}}{N_i}, \quad (2)$$

where N_i – the number of messages during i-th type of observation.

Time interval dispersion is evaluated using standard deviation

$$\sigma(\Lambda t_i) = \sqrt{\frac{\sum_{k=1}^{N_i} \left(\Lambda t_{i_k} - \overline{\Lambda t}_{i_k}\right)^2}{N_i - 1}}. \quad (3)$$

The absolute error of the duration of time intervals between the appearance of adjacent messages of type i is

$$\varepsilon_{p,\Lambda t_i} = z_p \cdot \frac{\sigma(\Lambda t_i)}{\sqrt{N_i}}, \quad (4)$$

where z_p – is the standard normal distribution p – th quintile, selected according to p value.

The value of relative error of average time interval between adjacent i-th type signalling messages arrival times with 99% confidence level ($p = 0.99$) is

$$\Delta_{\Lambda t_i} = \frac{\varepsilon_{p,\Lambda t_i}}{\overline{\Lambda t}_i} \cdot 100\% . \quad (5)$$

The parameters of time intervals between the moments of arrival of adjacent signalling messages are shown in Table I.

TABLE I. THE PARAMETERS OF TIME INTERVALS BETWEEN THE MOMENTS OF ARRIVAL OF ADJACENT SIGNALLING MESSAGES.

Message	$\overline{\Lambda t}_i$, s	$\sigma(\Lambda t_i)$, s	$\varepsilon_{p,\Lambda t_i}$	$\Delta_{\Lambda t_i}$, %
INVITE	1.079	1.05	0.038	3.57
RINGING	1.079	0.953	0.035	3.24
OK	1.084	1.007	0.037	3.41
ACK	1.084	1	0.037	3.38
BYE	1.084	1	0.037	3.38
OK (after BYE)	1.084	1	0.037	3.38
Total	0.18	0.881	0.013	7.32

The mean and variance of time intervals between adjacent messages of the same type differs from the observed values by no more than 3.57% and the total signalling messages flow – by no more than 7.32% with 99% confidence level.

Based on the experiment data, histograms of signalling messages servicing times are drawn. Bounds of the histogram grouping intervals are defined as follows

$$l_{\Delta t_i,\min} = 1.2\lg N_i\,,\; l_{\Delta t_i,\max} = 3.3\lg N_i\,. \qquad (6)$$

In this case the size of the individual signalling messages sample is $N_i = 4953$, so $l_{\Delta t_i,\min} = 4$ and $l_{\Delta t_i,\max} = 12$. The size of total signalling messages sample is $N_7 = 29723$, so $l_{\Delta t_i,\min} = 5$ and $l_{\Delta t_i,\max} = 15$. For the grouping of time intervals between the arrivals of individual signalling messages $l_i = 8$ is used, for the total flow - $l_i = 10$.

Sample width is determined for each group of experiment values

$$IP_{\Delta t_i} = \Delta t_{\Delta t_{i\max}} - \Delta t_{\Delta t_{i\min}}\,, \qquad (7)$$

where $\Delta t_{i_{\min}}$, $\Delta t_{i_{\max}}$ – the minimum and maximum interval time between the arrival of adjacent *i*-th type of signalling messages.

For each type messages histograms are step determined

$$d_{\Delta t_i} = \frac{IP_{\Delta t_i}}{l_{\Delta t_i}}\,. \qquad (8)$$

The probability $P_{Eksper,i,n}\left(a_{i,n} \le \Delta t_{i_k} < b_{i,n}\right)$ that the Δt_{i_k} falls in the range $\left[a_{i,n}, b_{i,n}\right)$ is calculated from experimental data using the following equation

$$P_{Eksper,i,n}\left(a_{i,n} \le \Delta t_{i_k} < b_{i,n}\right) = \frac{z_{i,n}}{N_i}\,, \qquad (9)$$

where $z_{i,n}$ – the number of values of Δt_{i_k} that fall in the range $\left[a_{i,n}, b_{i,n}\right)$, n – index of the interval.

The probability of exponential distribution $P_{Ekspo,i}\left(a_{i,n} \le \Delta t_{i_k} < b_{i,n}\right)$ that the Δt_{i_k} falls in the range $\left[a_{i,n}, b_{i,n}\right)$ is found by the equation

$$P_{Ekspo,i}\left(a_{i,n} \le \Delta t_{i_k} < b_{i,n}\right) = \frac{1}{\overline{\Delta t_i}} \int_{a_{i,n}}^{b_{i,n}} e^{-\frac{\Delta t_i}{\overline{\Delta t_i}}}\, d\Delta t_i\,. \qquad (10)$$

The calculation results are given in Fig. 4–Fig. 10.

The results indicate that the distributions of the time intervals between the arrival of two adjacent signalling messages provided by experimental data and theoretical calculations (using exponential distribution (10)) are the same (Fig. 4–Fig. 9). The mean and standard deviations of the distributions differ by no more than 3.75% (Table I). Thus the condition of the exponential distributions that the mean and standard deviation are equal is satisfied.

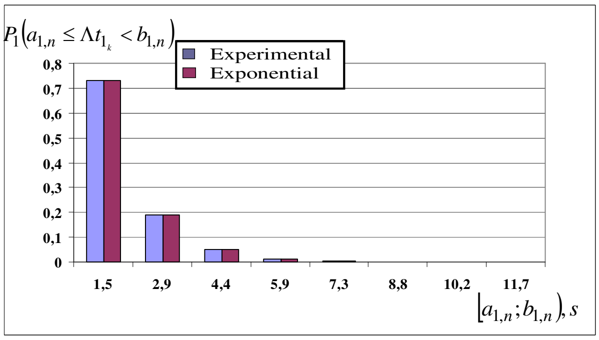

Fig. 4. Probability histogram of INVITE signalling messages inter-arrival time at SIP PS.

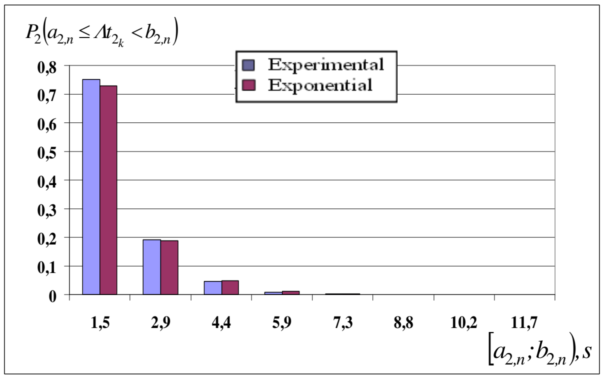

Fig. 5. Probability histogram of RINGING signalling messages inter-arrival time at SIP PS.

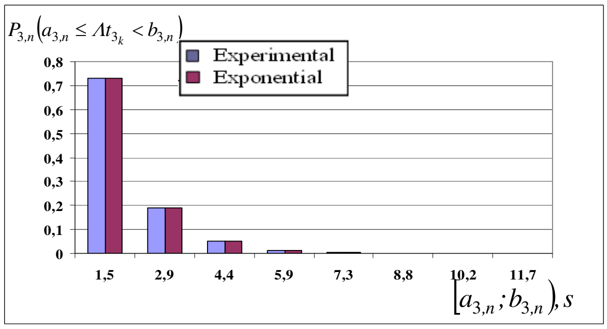

Fig. 6. Probability histogram of OK signalling messages inter-arrival time at SIP PS.

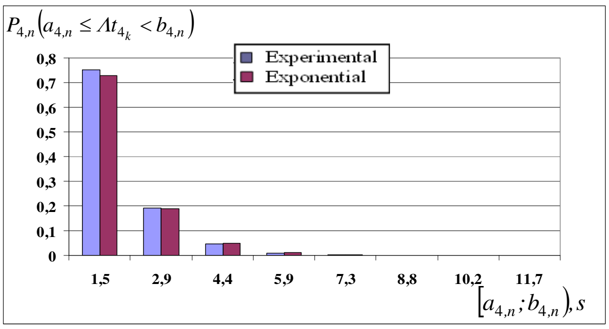

Fig. 7. Probability histogram of ACK signalling messages inter-arrival time at SIP PS.

Therefore, we can conclude that the values of time intervals between the arrivals of two adjacent signalling messages of the same type are distributed according to the exponential distribution. Mean while, the total signalling traffic entering into the SIP, PS, while servicing the basic call connection is not exponential (Fig. 10). This is confirmed by the fact that the mean and standard deviation of the distributions differs in value almost five times (Table

I).

As the signalling message sequence of the voice service connection/termination is strictly defined (by the connection establishment algorithm), time interval between the arrival of one type of signalling messages is dependent on the time interval between the arrivals of another type of signalling messages for the same service.

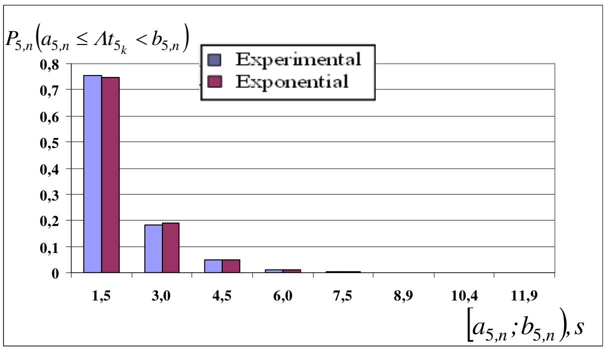

Fig. 8. Probability histogram of BYE signalling messages inter-arrival time at SIP PS.

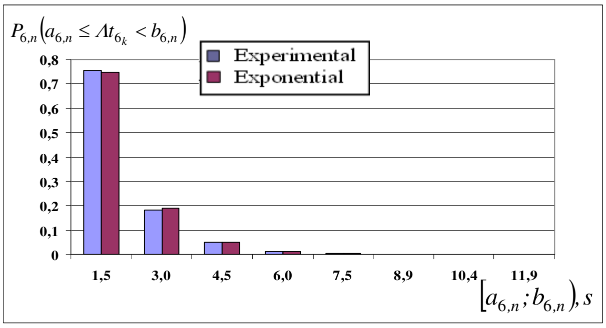

Fig. 9. Probability histogram of OK (after BYE) signalling messages inter-arrival time at SIP PS.

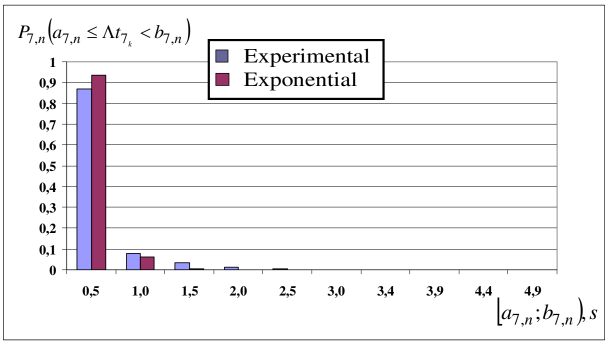

Fig. 10. Probability histogram of any type signalling messages inter-arrival time at SIP PS.

To check this assertion, correlation of time intervals between the arrivals of two consecutive messages of the same type is evaluated. The correlation coefficient between i-th and j-th type messages of inter-arrival time is

$$r_{i,j} = \frac{\sum_k \left(\left(\Lambda t_{i_k} - \bar{\Lambda} t_i \right) \left(\Lambda t_{j_k} - \bar{\Lambda} t_j \right) \right)}{N_i \cdot \sigma(\Lambda t_i) \cdot \sigma(\Lambda t_j)}, \tag{11}$$

where $i \neq j$ but $N_i = N_j$.

The results show that the RINGING, ACK, OK after BYE messages time interval is highly correlated ($r_{i,j} = 0.99$) with the INVITE, OK, BYE message interval. Interdependent messages are part of one call. INVITE and RINGING messages are part of the initiation of the call, ACK – connection establishment; BYE and OK after BYE are part of the connection termination transactions. Among There is practically no relationship ($r_{i,i+x} = 0.02$) between the signalling messages belonging to different transactions.

The absence of correlation between the messages of different transactions can be explained by random processes that occur between transactions, independent on the signalling network operation. Call pickup process intervenes between call initiation and connection establishment transactions. Meanwhile, voice message intervenes between connection establishment and call termination. Such processes are dependent on the user behaviour.

III. Conclusions

1. Distribution of the time intervals between signalling messages of the total flow entering SIP proxy server is not exponential, although it is created by servicing call flow, which time intervals between consecutive messages of the same type are distributed according to the exponential distribution.

2. Time intervals between signalling messages of the same type entering SIP PS are distributed according to exponential distribution.

3. After examining total signalling message flow coming to SIP PS it was determined that this flow is not exponential although it is created by the servicing voice service call flow that is exponential. It was also determined that part of signalling messages inter-arrival times in total messages flow are interdependent. Of your paper observes the textual arrangement on this page.

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