Paper

Statistical Analysis and Modeling of SIP Traffic for Parameter Estimation of Server Hysteretic Overload Control

Pavel Abaev[a], Rostislav Razumchik[b], and Ivan Uglov[c]

[a] *Department of Telecommunication Systems, Peoples' Friendship University of Russia, Moscow, Russia*
[b] *Institute of Informatics Problems of RAS, Moscow, Russia*
[c] *International Technology and Communication Center, Kolomna, Russia*

Abstract—The problem of overload control in Session Initiation Protocol (SIP) signaling networks gives rise to many questions which attract researchers from theoretical and practical point of view. Any mechanism that is claimed to settle this problem down demands estimation of local (control) parameters on which its performance is greatly dependent. In hysteretic mechanism these parameters are those which define hysteretic loops. In order to find appropriate values for parameters one needs adequate model of SIP traffic flow circulating in the network under consideration. In this paper the attempt is made to address this issue. Analysis of SIP traffic collected from telecommunication operator's network is presented. Traffic profile is built. It is shown that fitting with Markov Modulated Poisson Process with more than 2 phases is accurate. Estimated values of its parameters are given.

Keywords—Markov Modulated Poisson Process, overload control, SIP server, statistical analysis.

1. Introduction

The problem of overload control in SIP signaling networks gives rise to many questions which attract researchers from theoretical and practical point of view. Any mechanism that is claimed to settle this problem down demands estimation of local (control) parameters on which its performance is greatly dependent. This research is motivated by the idea that hysteretic control which is successfully deployed and used in SS7 networks may also be applicable and beneficial for overload control in next generation networks. Note that in hysteretic mechanism parameters that are subject to estimation are those which define hysteric loops. It is known that overload conditions that arise occasionally in signaling networks lead to severe loss of service quality which eventually affects network operators and/or service providers. There are many research papers that deal with analysis of systems with different overload control mechanisms, including hysteretic policy – [1]–[9] just to mention a few. In this paper system under consideration is SIP proxy server. The are two main approaches to analyze behavior of control mechanism in SIP networks: mathematical modeling and simulation. One of the drawbacks in mathematical modeling is the assumption concerning input flow and its parameters. Clearly if one chooses a simple model (say Poisson) then there is a risk that the nature of the real flow that is fed into the SIP proxy server is neglected too much and consequently the values of control parameters that are obtained may be inadequate. If one chooses more complex model, which reflects nature of real flow, it may lead to hardly tractable (with the growth of initial parameters values, i.e., system capacity, thresholds etc.) mathematical model. In this paper effort is made to address the issue of choosing input flow model and its parameters for SIP traffic, captured on SIP proxy server operating in telecommunication operator's network. We collect traffic circulating between two geographical regions, analyze it with well-known statistical methods and then try to fit Markov Modulated Poisson Process (MMPP) in SIP traffic data using different algorithms proposed in the literature [10]–[13]. In [14] there was build mathematical model of SIP server with MMPP input flow and two-level hysteretic policy and proposed optimization problem for choosing policy parameters. Using estimated in this study values of MMPP based on real SIP data one can calculate values of policy parameters for model in [14] and use them in practical implementations.

The paper is organized as follows. In the Section 2 description of traffic collection procedure and some insight into the traffic nature is given. Then SIP-I traffic model is being built in Section 3. Section 4 is devoted to statistical analysis of SIP-I traffic. In Section 5 the authors present the results of MMPP fitting. Conclusion contains short overview of obtained results and gives a glimpse of further research.

2. Traffic Collection and Data Description

Let's consider the fragment of the transit network of telecommunication operator depicted in Fig. 1. In this paper SIP-I traffic circulating between two different regions is analyzed [15]. Traffic aggregation happens on regional access

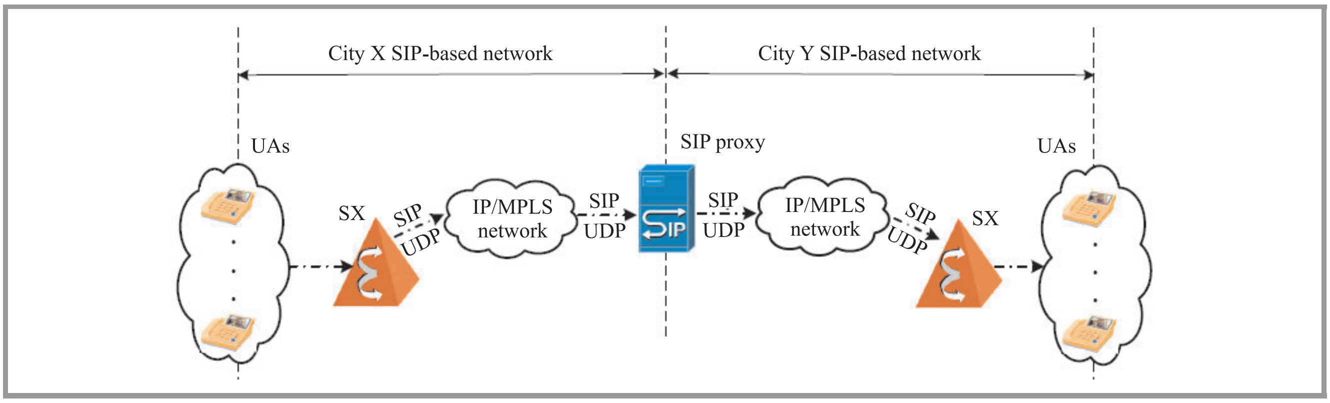

Fig. 1. SIP based transit network between two cities.

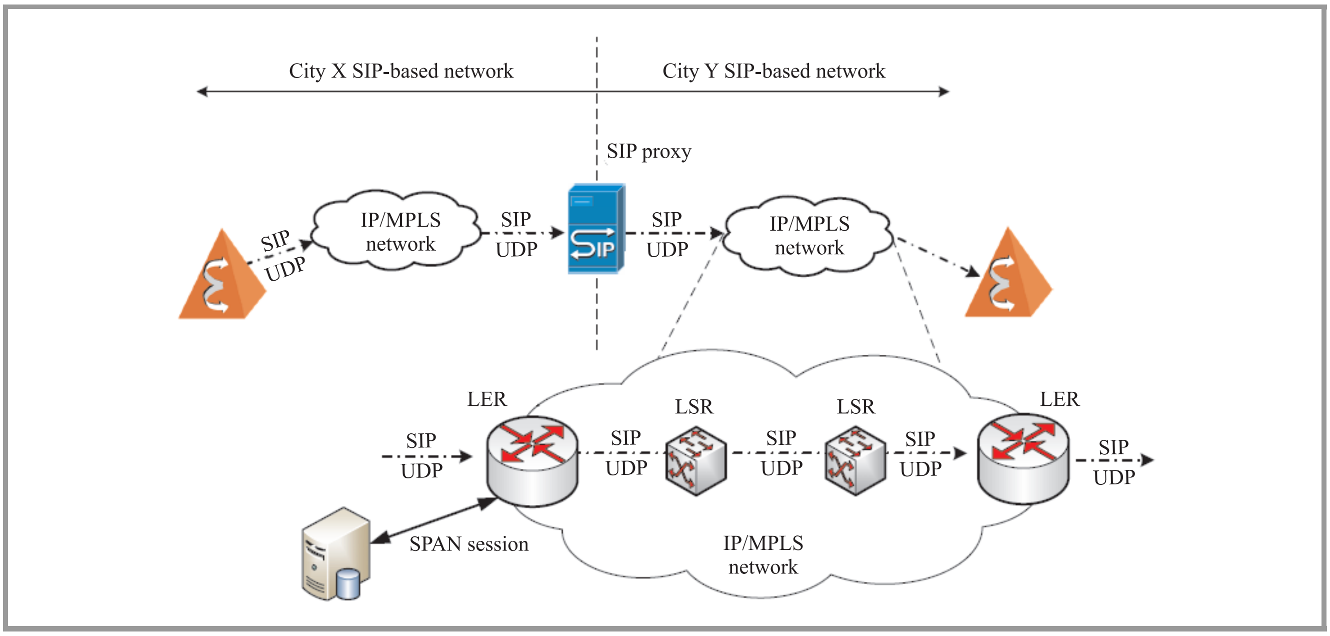

Fig. 2. Scheme of trace collection from transit segment.

network using Signaling System no. 7 (SS7). As the caller and call are located in different regions, the traffic goes through the two transit regional nodes – soft switches of the 5th class. Signalling exchange between them is organized by means of the SIP-I protocol. On the traffic route between soft switches SIP proxy server is set for the purpose of logical separation of regional networks. In the considered case this server performs the functions of the Interconnection Border Controller [16]. Aggregation networks are based on TDM technology with SS7 protocol stack. Besides traffic transit functions, soft switches may perform TMD-IP traffic conversion. Process of an exchange of signaling messages between the regions is completely described by the IETF RFC 3665 [17]. Number of messages that is necessary for call session initialization and termination in the majority of cases does not exceed 8.

Traffic data was captured on SIP proxy server's network interfaces. The organization of traffic measurement described in this paper is based on technical report [18] (Fig. 2). Passive recording system was chosen in order to guarantee the absence of significant influence of measurements on signaling flows. Analysis showed that for data recording it is possible to use standard network interface cards, installed on the data collection server [19]. During experiment L2/L3 switches were loaded up to 30–40% of its maximum throughput capacity. This guaranteed the absence of considerable influence of measurements on the values under test. Further analysis of the collected SIP-I traffic takes into account SIP protocol stack [15], [17].

Signaling information was collected from the network using mirroring technology with subsequent recording on L2/L3 network devices [20]. Traffic collection scheme is depicted in Fig. 2. All traffic circulation between two regions and entering one of L2/L3 switches was mirrored. Its recording was performed on the dedicated server for data recording and storage operation under Linux. Information was gathered with the standard utility (tcpdump) in the open libpcap format [21]. All network equipment which participated

in SIP-I traffic processing was synchronized using NTP protocol [22], [23].
The primary processing of libpcap files with SIP-I traces was carried out using standard functions of Wireshark software [24]. Automation of the processing was done using shell-script. As the result there were obtained data arrays, that contained all necessary information for further analysis of SIP traffic which was carried out in numerical computing environment Matlab [25].
All the SIP-I traces were captured during one week (7 consecutive days, starting from Saturday, 24 hours per day) by means of span session created on one L2/L3 switch. At this node the traffic is duplex. During one workday the maximum number of new call attempts per second (sum for two directions) does not exceed 95, on weekend – 70. Maximum number of concurrent active sessions during busy hour does not exceed 5000 and on weekend 7500. This information is summarized in Table 1.

Table 1
Basic characteristics of new call attempts flow

	Maximum new call attempts per second	Maximum number of concurrent active sessions (busy hour)
Workday	95	5000
Weekend	70	7500

The dynamics of new call arrivals (sum for two directions) during one week is depicted in Fig. 3. It can be seen that from 8 a.m. the load starts to increase and reaches its maximum level at 10 a.m. The sharp decline is observed at 8 p.m. and it continues up to minimum level at 11 p.m. The intensity of new call arrivals for workdays and weekend is somewhat similar. During workdays and weekend it alters in legible boundaries in the period between 10.30 a.m. – 4 p.m.

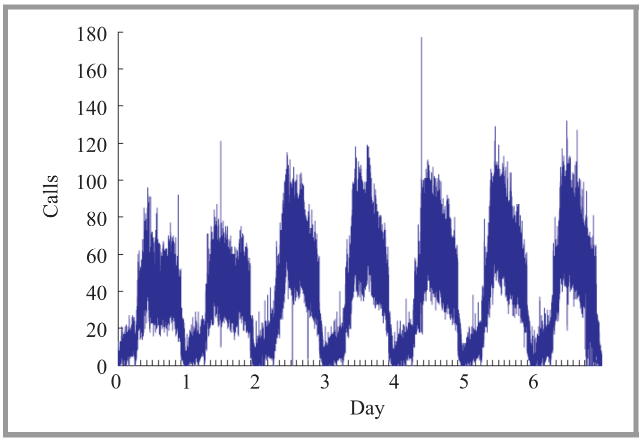

Fig. 3. Total number of new call attempts per second in both directions (one segment covers 2 hours).

The dynamics of new call arrivals (sum for two directions) during one workday (weekend) between 10.30 a.m. and 4 p.m. is depicted in Fig. 4, on weekend in Fig. 5. On

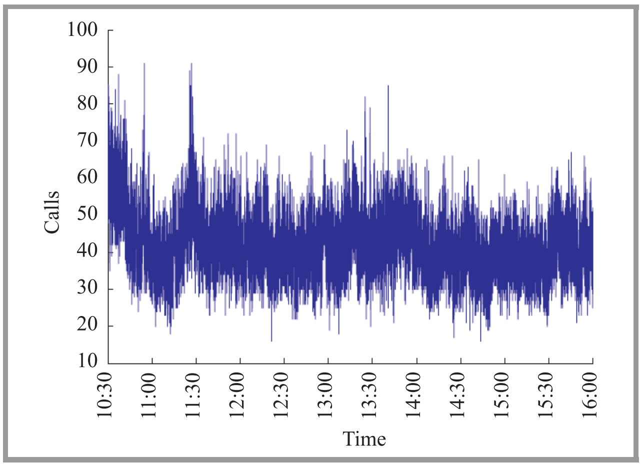

Fig. 4. Dynamic of new call attempts per second during one workday between 10.30 a.m. and 4 p.m.

the weekend one can see that there are not many intense changes. Herewith on a workday behavior is somewhat similar but one can see more abrupt jumps and absolute values of intensity are higher (Table 1).

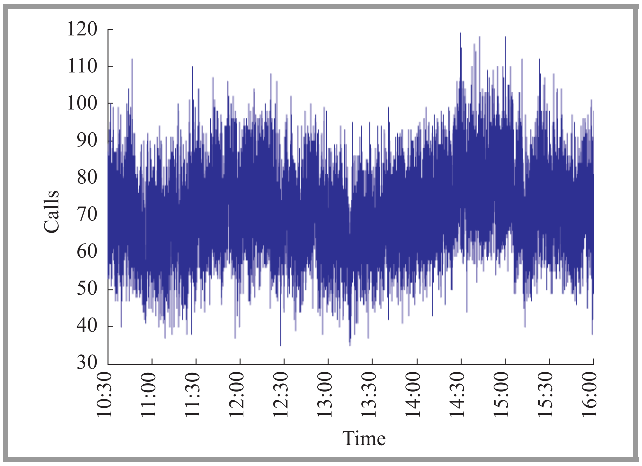

Fig. 5. Dynamic of new call attempts per second during one weekday between 10.30 a.m. and 4 p.m.

3. Building SIP-I Traffic Model

In the network under consideration the maximum size of SIP-I message is restricted by the maximum size of Ethernet frame (1542 bytes) [26]. During analysis the presence of fragmented packets in traffic traces was detected. Minor number of fragmented packets was found only for SIP-I messages that use INVITE, ACK, BYE, CANCEL methods. Among them the biggest number is for INVITE messages which is due to the big size of encapsulated IAM SS7 message. The presence of fragmented packets in the joint SIP-I flow may have impact on processors in network nodes but in the data set under study their number may considered to be insignificant in caparison to the its total volume.

Table 2
Statistical properties of SIP-I data flow

		ALL msg.	INVITE msg.	non-INVITE msg.
Mean	Workday	0.002022	0.028092	0.002359
	Weekend	0.003202	0.0231635	0.003716
Standard deviation	Workday	0.003001	0.0352475	0.003269
	Weekend	0.004909	0.028447	0.005280
Coefficient of variation	Workday	1.48386	2.01778	1.38626
	Weekend	1.5332	1.2283	1.4212
Skewness	Workday	3.16522	3.80822	2.76976
	Weekend	2.97935	2.3496	2.6567
Kurtosis	Workday	21.7262	18.497	17.0056
	Weekend	16.4755	12.255	13.968

Thorough analysis of collected traffic allowed us to estimate the frequency of each SIP-I message (Table 4). During traffic analysis one was interested in building traffic profile by identifying all possible session establishment scenarios. In order to do this all messages in the traces we grouped by Call-ID attribute. The depth of search was restricted by the maximum call length for the considered network which was 30 minutes. Clearly messages with identical Call-ID belong to the same call flow and form certain scenario. Going through the traces one can detect all unique scenarios and estimate their frequency observing that if two call flows consist of the same number and sequence of messages then they belong to one scenario. As the result traffic profile was obtained (Table 5). Very rare and very long scenarios (of 0.1% frequency and less) were left out of scope. Analyzing scenarios from Table 5 one can arrive at the following conclusions, based on assertions coming from RFC 3261 [15]:

- connection is established successfully with reply and correct session termination happens in 20.26% of all call cases (scenarios no. 3, 10, 13, 18, 23, 28, 30, 32, 37);
- attempting to call a busy callee and correct session termination happens in 10,12% of all call cases (scenarios no. 4, 12, 15, 20, 35, 40, 45);
- correct session termination without session establishment due to incorrect dialed number or unavailability of mobile subscriber happens in 25.92% of all call cases (scenarios no. 11, 14, 19, 26, 27, 29, 31, 34, 39, 44);
- connection termination on caller's side before session establishment, connection termination on callee's side, unavailability of signalling network (due to insufficient number of free time-slots etc.) happens in 36.3% of all call cases (scenarios no. 1, 2, 5–9, 16, 17, 21, 22, 24, 25);
- connection termination due to errors during negotiation of session establishment parameters, unexpected behavior of software etc. happens in 7.4% of all call cases (scenarios no. 33, 36, 38, 41–43, 46–48).

Notice that high number of call cases without reply is due to the fact that most of calls are long-distance/international, i.e., callers and callees are in different time zones.

4. Statistical Analysis of SIP-I Data Flow

Based on collected data during period 10.30 a.m. – 4 p.m. for each day of the week (weekend and 5 workdays) two types of samples for further analysis were created. One type of sample was composed only of SIP-I messages interarrival times. Three samples of this type were defined. The first sample (further referred to as ALL msg.) contained timestamps of consecutive message arrivals to SIP proxy server irrespective of their source (from A or from B). The second and third samples (further referred to as INVITE msg. and non-INVITE msg.) contained timestamps of consecutive INVITE (non-INVITE msg.) arrivals to SIP proxy server irrespective of their source. The second type of sample was formed based on the first type. Data in each of the three samples (ALL msg., INVITE msg., non-INVITE msg.) was aggregated over different timescales (i.e., representing number of SIP-I messages in respective time bin). Five time bins was used: 10 ms, 100 ms, 1 s, 10 s, 100 s. Note that in samples of this type SIP-I messages that arrive at SIP proxy server are counted irrespective of their source.

In Table 2 one can find basic statistical properties of the SIP-I data flow based on ALL msg., INVITE msg. and non-INVITE msg. interarrival samples for a workday and a weekend.

For workday (weekend) the value of each parameter is arithmetic average of the values, calculated for each workday (weekend) individually.

It can be seen that standard deviation is far above 1 in all three cases indicating that flows are not Poisson. The data is skewed to the right with sharp peak and long tail as indicated by the values of skewness and kurtosis respectively. In addition, the data is not normally distributed as indicated by the p-value of the Jarque-Bera statistic. There was validated the independence of messages interarrival times and number of arrived messages in disjoint time intervals. It was done using various tests: autocorrelation function (ACF), Box-Ljung statistic and visual inspection of consecutive arrivals (scatter plots). They all rejected independence assumptions.

In Fig. 6 the autocorrelation function for 1000 lags for ALL msg. sample (approx. 10 million consecutive interarrival times) is shown. At small as well as at higher time lags interarrivals are correlated. It is decaying slowly thus suggesting the presence of long-range dependency. In Fig. 7 the autocorrelation function for 1000 lags for ALL msg. sample aggregated over 1 s. time bin is presented. Again significant correlation is observed. Increasing the number of lags or the aggregation level sample (size of time bin) does not change significantly the behavior of ACF.

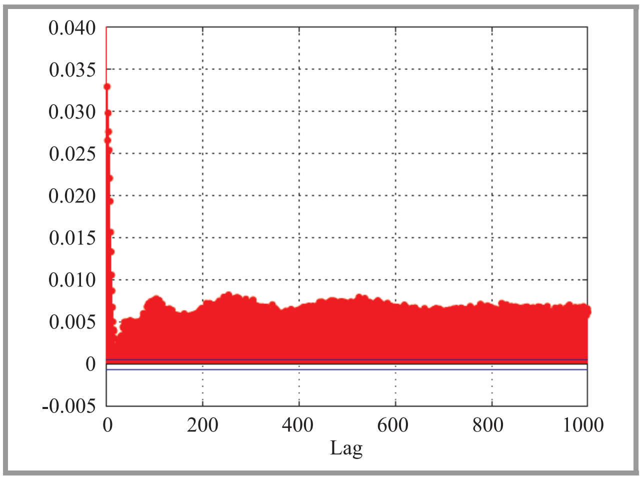

Fig. 6. Autocorrelation function for ALL msg. sample containing consecutive interarrival times and 95% confidence intervals.

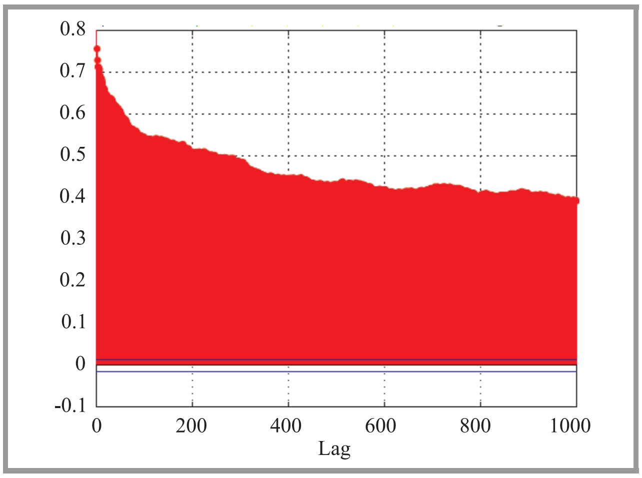

Fig. 7. Autocorrelation function for ALL msg. sample aggregated over 1 s time bin and 95% confidence intervals.

The Box-Ljung statistic shows that interarrival times and number of messages per respective time bin cannot be considered as independent and identically distributed with 95% confidence.

In order to reveal the presence or absence of dependencies in the dataset interarrival times in INVITE msg. sample, ALL msg. sample and its aggregation over different time bins were visually examined using scatter plots (Figs. 8–10). In Fig. 8 the X axis shows the time of i-th packet arrival, and the Y axis shows the time of (i+1)-th packet arrival. The plot is not symmetric and thus dependencies exist. For higher values (above 0.05) weak negative relationship is suggested. For aggregated sample (in Fig. 10) there seems to by weak positive (linear) relationship.

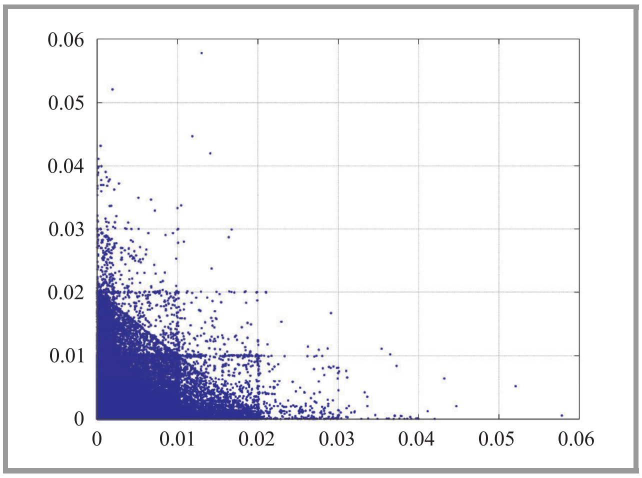

Fig. 8. Scatter plot for ALL msg. interarrival sample.

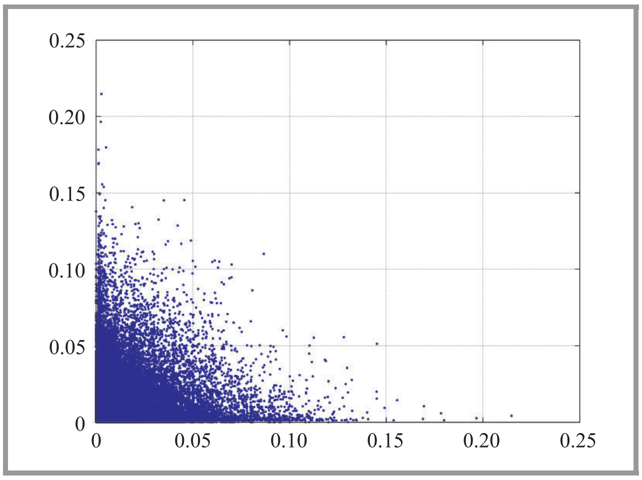

Fig. 9. Scatter plot for INVITE msg. interarrival sample.

Note that the presented results concern data flow consisting of consecutive message arrivals to SIP proxy server irrespective of their source (from A or from B). But one may be interested in nature of INVITE or non-INVITE flows only. Thus it is important to notice that test similar to presented above indicate that in this study one cannot closely approximate flow of SIP-I messages that use INVITE and non-INVITE methods with Poisson distribution.

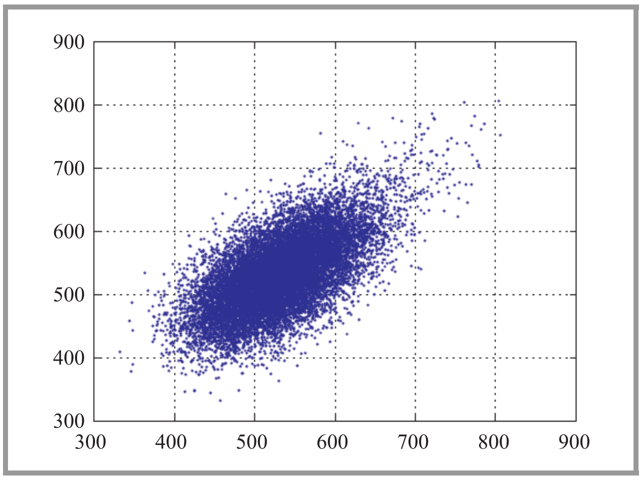

Fig. 10. Scatter plot for ALL msg. sample aggregated over 1 s time bin.

Notice that INVITE msg. sample includes not only INVITE messages that initiate new sessions but any re-INVITE messages that may appear during SIP dialogs. Statistical test show that if one sifts INVITE msg. sample, leaving only INVITE messages that initiate new sessions, such flow is also not Poisson but can be modeled by simple on-off source. Corresponding estimated values of fitted MMPP are given at the end of Section 5.

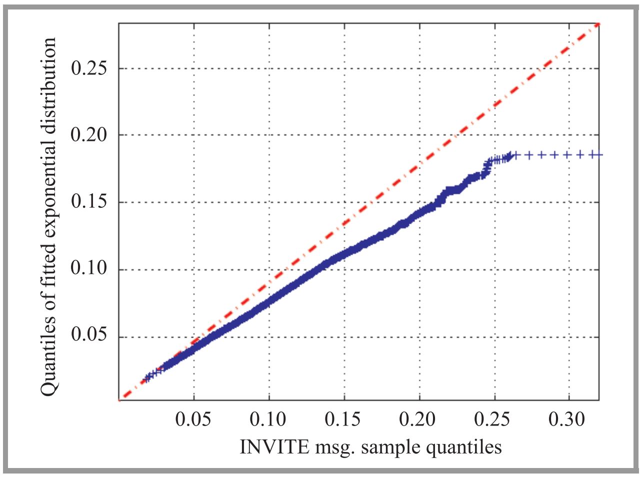

Fig. 11. Q-Q Plot for INVITE msg. sample and simulated exponentially distributed data.

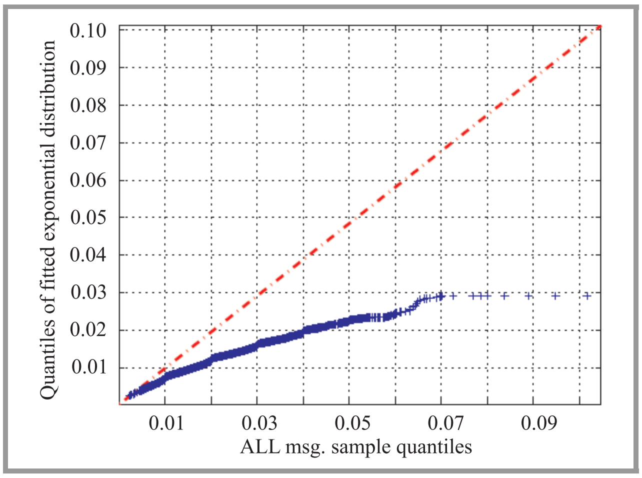

Fig. 12. Q-Q Plot for ALL msg. sample and simulated exponentially distributed data.

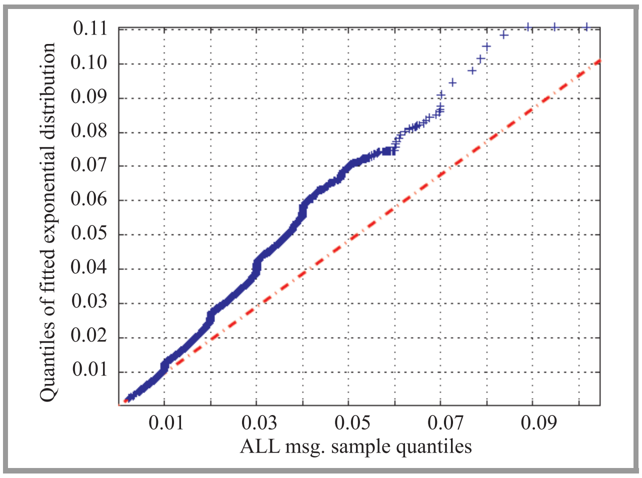

Fig. 13. Q-Q Plot for ALL msg. sample and simulated Weibull distributed data.

For each of the interarrival samples (ALL msg., INVITE msg. and non-INVITE msg.) its components were tested for exponential, Weibull and Pareto distributions using the Kolmogorov-Smirnov, Anderson-Darling and Chi-Squared tests. All tests reject null hypothesis with significance level of 5%. In Figs. 11–13 Quantile-Quantile (Q-Q) plots of different collected samples and data simulated from different distributions is shown. Necessary parameters for simulated distributions were estimated from samples.

Table 3
Estimated values of Hurst parameter from ALL msg. sample

Time bin	Absval	Aggvar	Diffvar	R/S
10 ms	0.8556	0.8583	0.5411	0.9001
100 ms	0.9435	0.9488	0.7056	0.9754
1 s	0.9578	0.9648	0.9272	0.9751
10 s	0.9485	0.9563	1.1138	0.9000
100 s	0.9414	0.9384	0.8509	0.6932

Finally for each of the aggregated samples Hurst parameter was calculated using the following methods: absolute moment, aggregate variance, difference variance, and R/S [27]. The estimated values for ALL msg. sample are given in Table 3.

Table 3 shows that basically Hurst exponent value does not vary significantly for bigger time scales (around 0.9)[1].

[1] Yet there exist several outliers depending on the estimation method.

This suggests that the arriving SIP-I messages to SIP proxy server (irrespective of source) is a persistent process with long-term memory.

5. Fitting SIP-I Trace to MMPP

In Fig. 14 ALL msg. sample aggregated over 1 s time bin is depicted. The plots for 10 s and 100 s time bins are similar. The arrival rates appear to vary among several distinct values, which is consistent with the path behavior of MMPP.

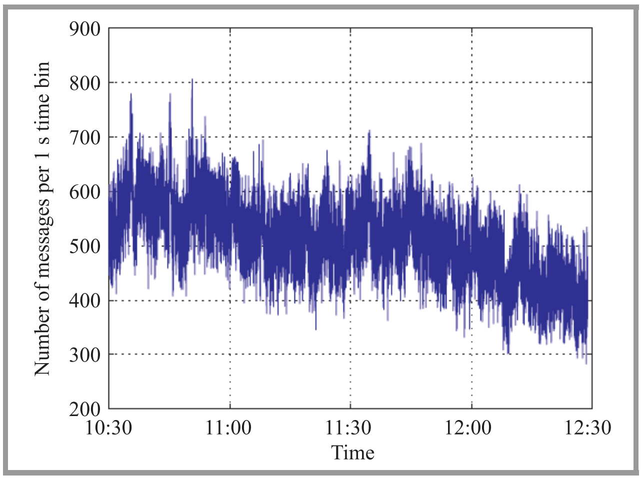

Fig. 14. ALL msg. sample for one workday aggregated over 1 s time bin.

Algorithm Lambda was used to fit ALL msg. sample aggregated over different time scales to a MMPP [28]. It was found that in this study MMPP model was able to adequately model collected data. For aggregation over 10 ms, 100 ms, 1 s, 10 s, 100 s time bins the number phases in fitted MMPP varies from 3 to 31. Here we restrict ourselves to the Q-Q plot as the measure of the goodness of fit of the model. In Fig. 15 the quantiles of the ALL msg. sample aggregated over 1 s time bin and of a simulation of the fitted process are shown. If both sets of data are drawn from the same distribution the plot is expected to be linear. It can bee seen, that the fit appears to be fairly good.

Phase transitions of fitted MMPP from Fig. 15 are governed by the infinitesimal matrix (generator) Q_6 of the form

$$Q_6 = \begin{pmatrix} -0.5275 & 0.4835 & 0.0440 & 0 & 0 & 0 \\ 0.0218 & -0.5679 & 0.5218 & 0.0253 & 0 & 0 \\ 0.004 & 0.1165 & -0.3634 & 0.2399 & 0.0066 & 0 \\ 0 & 0.0095 & 0.3075 & -0.4294 & 0.1099 & 0.0025 \\ 0 & 0 & 0.0274 & 0.4327 & -0.4874 & 0.0274 \\ 0 & 0 & 0 & 0.1739 & 0.7826 & -0.9565 \end{pmatrix}.$$

The vector of estimated arrival rates equals $\vec{\lambda}_6 =$(751.1845, 645.5535, 547.9225, 458.2916, 376.6606, 303.0296).

There was made an attempt to fit collected data to MMPP with two phases using interarrival statistic instead of frequency one. The goal was to check whether the simplest

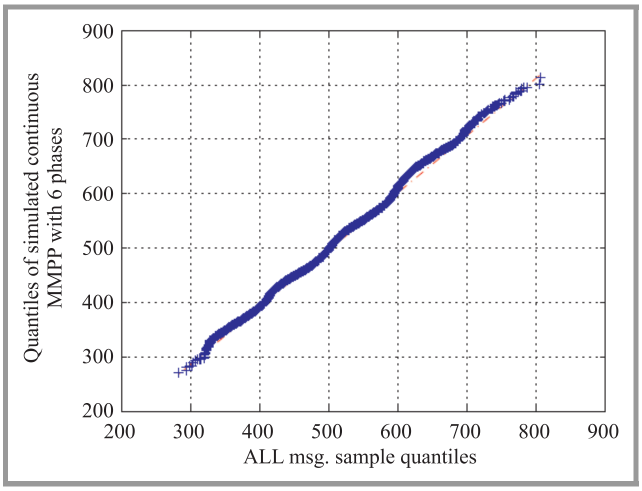

Fig. 15. Q-Q plot of the ALL msg. sample aggregated over 1 s time bin and simulated data, using fitted (with Lambda algorithm) MMPP with 6 phases.

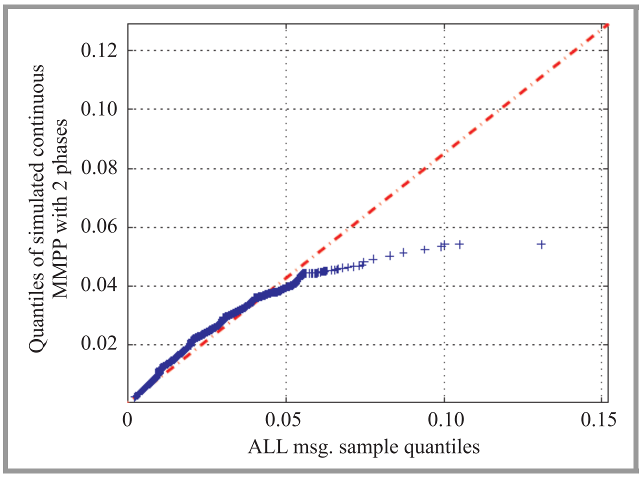

Fig. 16. Q-Q plot of the ALL msg. sample and simulated data, using fitted continuous two-phase MMPP.

MMPP model may be fairly accurate as well. In order to do this parameters of two-phase MMPP flow based on ALL msg. sample were estimated using KPC Toolbox [29]. Then synthetic data was generated using built in KPC Toolbox MAP random sample generator. Quantiles of the ALL msg. sample and of simulated fitted two-phase MMPP process are shown in Fig. 16.

Here phase transitions are governed by the infinitesimal matrix (generator) Q_2 of the form

$$Q_2 = \begin{pmatrix} -276.3723 & 276.3723 \\ 300.6946 & -300.6949 \end{pmatrix},$$

and vector of estimated arrival rates equals $\vec{\lambda}_2$=(63.7806, 978.6974).

One may observe too many outliers at high quantiles and fitting is not so accurate as with MMPP with more than 2 phases. Thus MMPP with two phases cannot adequately capture the behavior of real flow of SIP messages in the considered scenario (Fig. 1) and its use may underestimate impact of input flow on SIP proxy server performance.

It is worth mentioning that two-phase MMPP model (on-off source) turned out to be better than Poission for modeling of INVITE flow, consisting only of messages that initiate sessions. In Figs. 17–18 one can see Q-Q plots of the sifted INVITE msg. sample and simulated data, using fitted exponential distribution and continuous two-phase MMPP correspondingly. The estimated infinitesimal generator Q_{INVITE} of corresponding two-phase MMPP has the form

$$Q_{INVITE} = \begin{pmatrix} -182.9091 & 182.9091 \\ 129.2100 & -129.2100 \end{pmatrix},$$

and vector of estimated arrival rates equals $\vec{\lambda}_{INVITE}$=(165.4, 0).

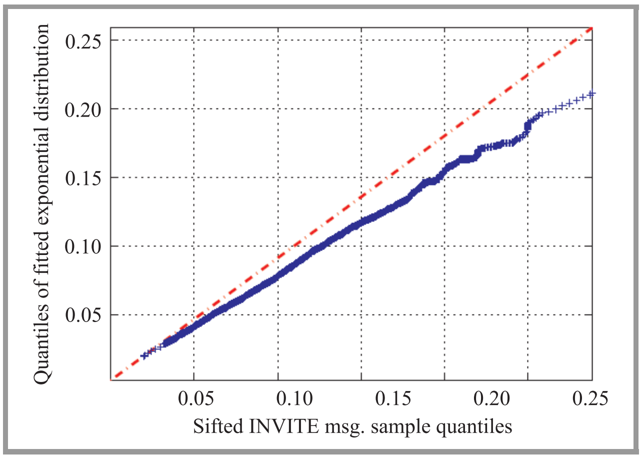

Fig. 17. Q-Q plot of the sifted INVITE msg. sample and simulated exponentially distributed data.

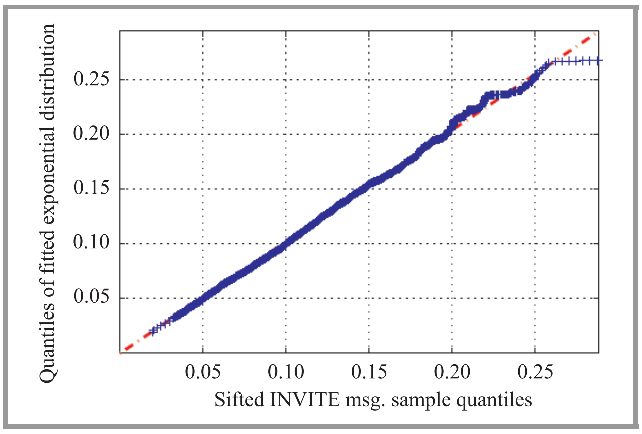

Fig. 18. Q-Q plot of the sifted INVITE msg. sample and simulated data, using fitted continuous two-phase MMPP.

Following [28] we tried to test the ability of the fitted on-off source to capture the features of flow of INVITE messages that initiate new sessions. In order to do this we simulated infinite buffer queue with constant service time using discrete-event simulation approach and fed this queue with sifted INVITE msg. sample, which contained interarrival times between consecutive arrivals of INVITE messages that initiate new sessions. The traffic intensity was varied by changing the service time, i.e., as mean time between consecutive arrivals in the considered sample equals ≈ 0.0146 sec., service time was varied between 0.009 and 0.0131. From simulation model we computed the mean number of customers in the system. We also computed mean number of customers in $MMPP|D|1$ using analytical expressions with infinitesimal generator and arrival rates given by Q_{INVITE} and $\vec{\lambda}_{INVITE}$. The results are shown in Fig. 19.

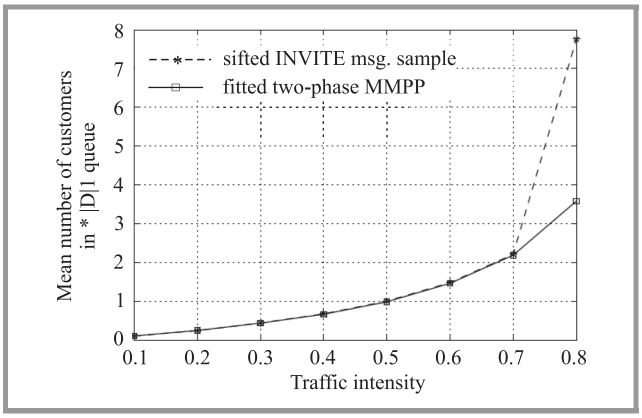

Fig. 19. Comparison of mean number of customers in $\cdot|D|1$ queue with sifted INVITE msg. sample as input flow and fitted continuous two-phase MMPP as input flow.

Mean delays computed from simulation and analytic models almost coincide for the traffic intensity less than 0.7 and in this region MMPP captures the relevant features of the sifted INVITE msg. sample. But at traffic intensities above 0.75 mean delay from simulation model is significantly larger than the one obtained from analytic expressions, which is consistent with observations in other studies, e.g. [28], [30].

6. Conclusion

In this paper the analysis of interregional SIP-I signaling traffic which circulates in telecommunication operator's network is presented. Traffic profile is built and typical call scenarios are identified. Statistical tests show that considered joint SIP-I traffic flow (and its INVITE component) is not Poisson and exhibits long range dependence. Traffic may be accurately fitted with MMPP with 3–31 phases. Based on obtained results further research will be devoted to estimating control parameters in mathematical models that represent SIP servers with embedded hysteretic load control mechanism and real network simulation.

Acknowledgements

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APPENDIX

Table 4
Size and frequencies of joint SIP-I messages flow

Message type	Total	Mean size (bytes)	Percent
ACK	2232516	476	14.03
BYE	5636610	606	3.54
INVITE	22341251	1314	14.04
CANCEL	7624459	451	4.79
INFO	212417	597	0.13
UPDATE	1	550	0
100	22063983	382	13.87
180	9810796	1329	6.16
183	33221739	937	20.88
200	19329687	713	12.15
400	18865	495	0.01
403	67135	593	0.04
404	544597	592	0.34
408	27792	277	0.01
410	16040	588	0
415	21	614	0
480	5090740	606	3.20
481	4046	422	0
483	944	944	0
484	23771	598	0.01
486	1965894	592	1.23
487	7666305	518	4.82
488	13153	520	0
500	343735	604	0.21
501	833	833	0
502	435713	595	0.27
503	244849	552	0.15
504	9124	634	0
604	455	578	0

Table 5
Typical session establishment scenarios
and their frequencies

No.	Sequence of messages	Percent
1	INVITE, 100, CANCEL, 200, 487, ACK	1.14
2	INVITE, 100, 180, CANCEL, 200, 487, ACK	2.28
3	INVITE, 100, 180, 200, ACK, BYE, 200	0.97
4	INVITE, 100, 180, 486, ACK	0.42
5	INVITE, 100, 183, CANCEL, 200, 487, ACK	4.62
6	INVITE, 100, 183, 180, CANCEL, 200, 487, ACK	14.84
7	INVITE, 100, 183, 180, 183, CANCEL, 200, 487, ACK	0.85
8	INVITE, 100, 183, 180, 183, 183, CANCEL, 200, 487, ACK	0.32
9	INVITE, 100, 183, 180, 183, 183, 183, CANCEL, 200, 487, ACK	0.26
10	INVITE, 100, 183, 180, 183, 200, ACK, BYE, 200	0.11
11	INVITE, 100, 183, 180, 183, 480, ACK	0.33
12	INVITE, 100, 183, 180, 183, 486, ACK	0.16
13	INVITE, 100, 183, 180, 200, ACK, BYE, 200	9.72
14	INVITE, 100, 183, 180, 480, ACK	0.99
15	INVITE, 100, 183, 180, 486, ACK	3.37
16	INVITE, 100, 183, 183, CANCEL,200, 487, ACK	3.14
17	INVITE, 100, 183, 183, 180, CANCEL, 200, 487, ACK	4.96
18	INVITE, 100, 183, 183, 180, 200, ACK, BYE, 200	1.95
19	INVITE, 100, 183, 183, 180, 480, ACK	0.20
20	INVITE, 100, 183, 183, 180, 486, ACK	1.02
21	INVITE, 100, 183, 183, 183, CANCEL, 200, 487, ACK	2.36
22	INVITE, 100, 183, 183, 183, 180, CANCEL, 200, 487, ACK	0.11
23	INVITE, 100, 183, 183, 183, 180, 200, ACK, BYE, 200	0.42
24	INVITE, 100, 183, 183, 183, 183, CANCEL, 200, 487, ACK	1.15
25	INVITE, 100, 183, 183, 183, 183, 183, CANCEL, 200, 487, ACK	0.27
26	INVITE, 100, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 480, ACK	0.19
27	INVITE, 100, 183, 183, 183, 183, 183, 480, ACK	0.33
28	INVITE, 100, 183, 183, 183, 183, 200, ACK, BYE, 200	0.70
29	INVITE, 100, 183, 183, 183, 183,480, ACK	1.61
30	INVITE, 100, 183, 183, 183, 200, ACK, BYE, 200	0.09
31	INVITE, 100, 183, 183, 183, 480, ACK	5.09
32	INVITE, 100, 183, 183, 200, ACK, BYE, 200	2.57
33	INVITE, 100, 183, 183, 404, ACK	0.22
34	INVITE, 100, 183, 183, 480, ACK	6.86
35	INVITE, 100, 183, 183, 486, ACK	0.69
36	INVITE, 100, 183, 183, 502, ACK	0.47
37	INVITE, 100, 183, 200, ACK, BYE, 200	3.73
38	INVITE, 100, 183, 404, ACK	0.65
39	INVITE, 100, 183, 480, ACK	6.72
40	INVITE, 100, 183, 486, ACK	1.73
41	INVITE, 100, 183, 500, ACK	0.83
42	INVITE, 100, 183, 502, ACK	1.51
43	INVITE, 100, 404, ACK	2.04
44	INVITE, 100, 480, ACK	3.60
45	INVITE, 100, 486, ACK	2.73
46	INVITE, 100, 500, ACK	0.53
47	INVITE, 100, 502, ACK	0.10
48	INVITE, 100, 503, ACK	1.05

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Pavel Abaev received his Ph.D. in Computer Science from the Peoples Friendship University of Russia in 2011. He is Associate Professor in the Telecommunication Systems department at Peoples' Friendship University of Russia since 2013. His current research focus is on NGN signalling, QoS analysis of SIP, and mathematical modeling of communication networks.

E-mail: pabaev@sci.pfu.edu.ru
Department of Telecommunication Systems
Peoples' Friendship University of Russia
Miklukho-Maklaya st 6
117198 Moscow, Russia

Rostislav Razumchik received his Ph.D. in Physics and Mathematics in 2011. Since then, he has worked as a senior research fellow at the Institute of Informatics Problems of the Russian Academy of Sciences. His current research activities focus on stochastic processes and queuing theory.

E-mail: rrazumchik@ieee.org
Institute of Informatics Problems of RAS
Vavilova st 44-1
119333 Moscow, Russia

Ivan Uglov is director of International Technology and Communication Center. His research interests include telecommunication network modeling, simulation and quality of service provisioning.

E-mail: uglov_ivan@mail.ru
International Technology and Communication Center
Devichie Pole st 21-88
140404 Kolomna, Russia