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Bandwidth Calculation for VoIP Networks Based on PSTN Statistical Model

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Abstract: This paper shows an analysis how to calculate proper bandwidth for VoIP calls after proper dimensioning of PSTN network. For this purpose, we use Erlang B and extended Erlang B formulae. Further, we have developed a software tool, named Bandwidth Calculator to calculate proper number of the circuits on the PSTN side and after that IP bandwidth. Traffic analysis is conducted for VoIP networks considering impact of many factors on the bandwidth such as: voice codecs, samples, VAD, RTP compression. The results obtained by bandwidth calculator are compared to simulation results and data obtained by measurements.

Keywords: VoIP, bandwidth calculation, tele-traffic analysis.

1 Introduction

BANDWIDTH is probably the most expensive component of the network. The network administrator has to know how to calculate proper bandwidth and how to reduce overall bandwidth consumption [1–3]. In this paper we develop a method to calculate proper bandwidth for VoIP calls after proper dimensioning PSTN network and determining optimal number of the circuits for the peak hour time.

In this paper, the two categories of Internet subscribers are considered: dial-up and ADSL. In the Literature ([4–11]), it is assumed that the Internet traffic has property of self-similarity. In order to express the degree of self-similarity numerically, we use so-called Hurst parameter. In [4] and [6] it has been shown that when value of Hurst parameter is small, i.e. $H < 0,7$, a classical model can

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be used, which is based on Erlang B and Extended Erlang B and their inversion functions. Thus, our first task is to check if classical method for calculation of number of circuits can be used. This can be done by calculating Hurst parameter for both type of traffic. Here, we use linear and square regression to show future trends of the ADSL and dial-up traffic behaviors and for the calculation of Hurst parameter.

The main task is to find method for proper dimensioning of a typical PSTN network. We consider a typical part of the network; however results and tools can be applied to any similar situation. We show that the value of Hurst parameter is in range $H < 0.7$ for dial-up traffic and for ADSL traffic is bigger than 0.7 . Erlang based model can be applied for the network with low load and rare loss and for the high speed and highly aggregated traffic on the backbone network (network core) because that traffic tends back to Poisson [12–16] . Because of its mathematical simplicity, well defined and well-known conditions for the simulations, it is still useful tool [13–19].

Further, obtained traffic analysis is applied to VoIP network, and proper bandwidth for VoIP calls is calculated. We have developed a software tool which operates as bandwidth calculator for such network [19]. First, we calculate proper number of circuits. After that, based on this calculation, the proper bandwidth is calculated for several VoIP codecs. The bandwidth calculation takes into consideration impact of many factors on the bandwidth such as: voice codecs, VAD, RTP compression. The results obtained by bandwidth calculator are compared to simulation results and data obtained by measurements.

The paper is organized as follows. In chapter 2, PSTN statistical model is described. The regression analysis is made in Chapter 3. Tele-traffic analysis is presented in Chapter 4. Finally, some case studies are discussed in Chapter 5. Chapter 6 describes traffic analysis of the VoIP network.

2 PSTN Statistical Model

Erlang B and Extended Erlang B models are used to determine proper number of circuits to carry voice traffic during the busy hour time (BHT), because it is period with maximum traffic load which one network must support [20]. However, our aim is to calculate proper number of circuits for VoIP calls. In determining busy hour traffic (BHT) intensity for certain day, we have to use simplification in which BHT is 17 percent of all traffic for that day [21]. The Erlang B traffic model [19–24] is based on the following initial assumptions: (i) an infinite number of sources, (ii) Poisson arrival process, (iii) block calls cleared, and (iv) holding times are exponentially distributed. But it can be shown that (iv) assumption is valid for arbitrary holding time distribution [20, 22]. In this paper is checked which distribution is valid in our case.

The following expression is used to derive the Erlang B model:

$$P_B = \frac{\frac{A_i^c}{c!}}{\sum\limits_{k=1}^{c} \frac{A^k}{k!}} \tag{1}$$

Here, P_B is the probability of blocking a call, c is number of circuits, and A_i is offered Internet traffic[1]. Calls with retrials are not used into the consideration in Erlang B model. For this purpose Extended Erlang B model is used which can be calculated on the base of already calculated Erlang B model. Based on (1), Erlang B, Extended Erlang B models and their inversion functions, we have developed the software tool named **Bandwidth Calculator** [19]. The Bandwidth Calculator can be used to determine proper number of circuits for VoIP calls. The user interface of the software tool is shown and explained later.

Erlang based model can be applied for the network with low load and rare loss and for the high speed and highly aggregated traffic on the backbone network (network core) because that traffic tends back to Poisson. Because of its mathematical simplicity, well defined and well-known conditions for the simulations it is still useful tool [13–19].

3 Regression Analysis

In order to analyze how increased number of subscribers influence on the traffic and to calculate Hurst parameter, we use regression model for the specific category of subscribers under consideration. The regression analysis is used to examine trends of the subscriber behavior in the future. We use regression analysis model which is based on the real values of the variables obtained by measurements. The regression analysis includes estimation of the unknown parameters, calculating dispersion characteristics and other statistical-analytical indices [25, 26].

The regression model is used when it is desired to express analytical relation between data appearance. The aim of this model is to find relationship between single variable Y and several variables X. The simplest method of the regression is linear regression [25]:

$$Y = a + bX + u \tag{2}$$

[1]Offered traffic is theoretical parameter, it can not be measured and it is used for calculation like in Erlang formula. Calculations on the rest of the paper is based on the real, measured traffic known as carried (serverd) traffic.

Here, X is independent variable, Y is dependent variable, u is error term known as residual, and a, b are parameters which have to be determined. Equation (2) can be written as:

$$y_i = \hat{y}_i + u_i, \text{ where } \hat{y}_i = a + bx_i, \text{ for } i = 1,2,\ldots,n. \tag{3}$$

where, $\hat{y}_i$ is a deterministic part of model. Residual u_i can be expressed as:

$$u_i = y_i - \hat{y}_i, \;\; u_i = y_i - a - bx_i, \;\; \text{for } i = 1,2,\ldots,n. \tag{4}$$

In order to achieve better accuracy, square regression and method of the least squares can be used: The square regression model is expressed as [24]:

$$y_i = a + bx_i + cx^2 + u_i, \text{ for } i = 1,2,\ldots,n. \tag{5}$$

Residual sum of squares can be written as:

$$SQ = \sum_{i=1}^{n} (y_i - \hat{y}_i)^2 = \sum_{i=1}^{n} (y_i - (a + bx_i + cx_i^2))^2. \tag{6}$$

Parameters a, b and c can be found by solving the following set of equations:

$$\frac{\partial SQ}{\partial a} = 0; \;\; \frac{\partial SQ}{\partial b} = 0; \;\; \frac{\partial SQ}{\partial c} = 0 \tag{7}$$

Mathematical calculations are performed by using Matlab and by operating with data obtained by measurements.

4 Tele-Traffic Analysis

When setting up a statistical model of a network, it is very important to choose proper mathematical traffic model. For connection-oriented circuit switched application, such as voice, relatively simple analytical model exists based on the Poisson process. This is classical model [20]. Connectionless packet-oriented data traffic (Internet traffic) is much more variable and thus less predictable. The main reason is in the fact that a typical multimedia application contains packets from the different sources, such as video, voice and data, which differ in statistical pattern. However, the traffic poses certain degree of correlation between arrivals and long range dependences on time, thus it can be defined as self-similar traffic [9], [10].

The fundamental questions related to the traffic model are:

1. Whether it is possible to use the classical model in the presence of self-similar traffic, and,

2. How self-similarity influences on the network performance in the situation under consideration.

In this paper, the property of self-similarity and Hurst parameter which is used to measure degree of self similarity is explained. After that, self-similarity of dial-up and ADSL traffic are analyzed, for the network segment under consideration. All analysis and other calculations are performed using traffic measurements in a real network.

4.1 Self-similar traffic

A self-similar phenomenon displays structural similarities across a wide range of timescales: milliseconds, seconds, minutes, hour, even days and weeks. It means that it is not matter what time scale is observed, similar pattern is seen (fractal-like behavior) [4]. The definition of self-similarity is given below [4, 6, 9, 10]:

X is defined to be a wide sense stationary random process with mean μ, variance σ and autocorrelation function $r(k)$. A stationary random process $X = (Xt; t = 1, 2, 3.)$ is statistically exact second-order self-similar if it has the same autocorrelation function $r(k) = E[(x_t - \mu)(x_{t+1} - \mu)]$ as the series X^m for all m, where X^m is the m-aggregated series $X^{(m)} = (X_k^{(m)}; k = 1, 2, 3, \ldots)$ obtained by summing the original series X over non-overlapping blocks of size m, $X_k^{(m)} = \frac{1}{m}(X_{km-m+1} + X_{km-m+2} + \cdots + X_{km})$.

The self-similar process has the following properties: i) slowly decaying variance, and ii) long range dependence. Hurst parameter H is used to determine the degree of the self-similarity. It expresses the speed of decay of the autocorrelation function. For a self-similar process, Hurst parameter is in range $\frac{1}{2} < H < 1$; for $H = \frac{1}{2}$ the time series is short range dependent; for $H \to 1$, the process becomes more and more self similar [4]. In [4] and [6], it is shown that for $H < 0{,}7$, the long-range dependence is unimportant, and self-similar traffic pattern can be neglected. In this case classical model for traffic engineering can be used. This type of traffic has short-range dependency.

For the larger values of Hurst parameter H, the property of self-similarity has bad influence on the network performance. In [5], it is shown that in this case, buffering can be extremely large. However, increased buffering leads to large queuing delays and overall delay which has impact on QoS for real-time applications, e.g. for VoIP. In the case under consideration, H parameter has to be calculated for the dial-up and ADSL traffic.

4.2 R/S method for the calculation of Hurst parameter

In [6], [8] and [9], several methods for calculation of Hurst parameter are explained. However, in this paper, we use R/S method.

For empirical time series X_t (for $t = 1,2,...,n$) with arithmetic mean $\overline{X(n)}$ and variance $S^2(n)$, we define R/S statistics, also called Rescaled Adjusted Range, with $R(n)/S(n)$ [9]. Here $R(n)$ is defined as:

$$R(n) = \max\left[\sum_{i=1}^{n}(X_i - \overline{X(n)}),\ 1 \leq k \leq n\right] - \min\left[\sum_{i=1}^{n}(X_i - \overline{X(n)}),\ 1 \leq k \leq n\right] \quad (8)$$

For almost all naturally occurring time series, the rescaled adjusted range statistic for set of samples of size n follows the following relationship:

$$E\left[\frac{R(n)}{S(n)}\right] \rightarrow cn^H \text{ as } n \rightarrow \infty,\ c \text{ is constant.} \quad (9)$$

If $\log[R(n)/S(n)]$ is plotted versus $\log(n)$, the slope of the curve is equal to Hurst parameter H.

5 Case Studies

In this part, we consider a typical part of the network; however results and tools can be applied to any similar situation.

5.1 Description of the network segment under consideration

Figure 1 shows situation under analyses. From Fig. 1, it is obvious that there are two types of the traffic which must be observed differently.

1. **Served (carried) dial-up traffic**: It can be seen from the Fig. 1. that subscribers from Tandem Office (TO in Fig. 1.) use same circuits for voice and data for transmission to the Main Exchange (ME in the Fig. 1.) in Zvornik for dial-up category of the subscribers. The switching part of EWSD exchange in Zvornik (ME in Fig. 1.) separates voice from data using dialed numbers. This is done due to the fact that anonymous dial-up subscribers dial 081-50-1433 and classical dial-up subscribers dial 081-59-1432 number, and thus all subscribers dialing those numbers are directed to Access Server. Access Server and switching part of EWSD exchange are connected via 90 transmission trunks. After that data is transferred to IP-MPLS bus across edge router. All traffic between main exchange and all exchanges which are connected to the main exchange is measured.

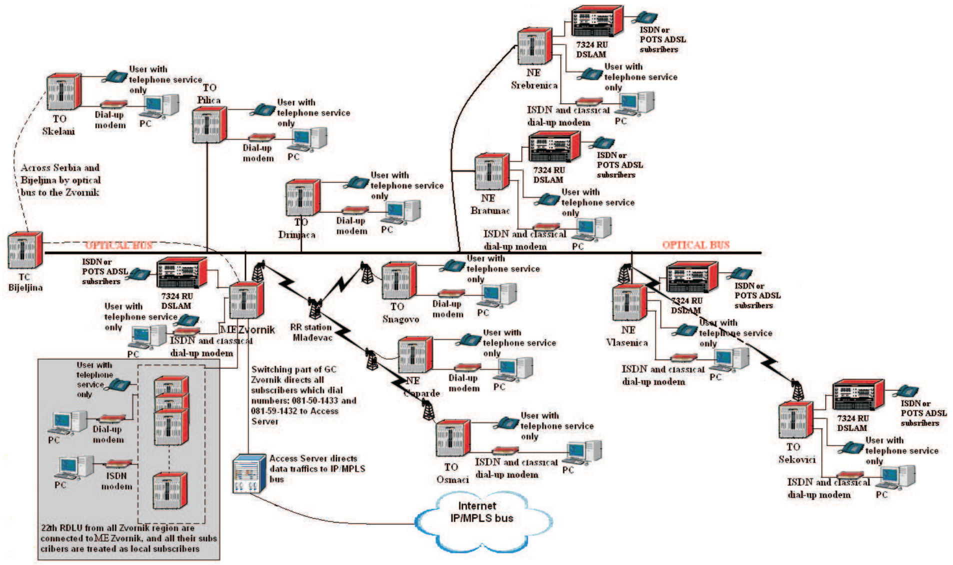


Fig. 1. Network segment under consideration

2. **Served (carried) ADSL traffic** (data part of ADSL traffic): From each Node Exchange (NE) and TO with ADSL to the ME (Main Exchange), see Fig. 1, data part of ADSL traffic is transmitted using separate optical transmission network. The voice part of the ADSL traffic from all exchanges is mixed with voice traffic, from ordinary PSTN subscribers [19]. Whole data part of ADSL traffic, from NE, TO and ME, is directed via edge router in Zvornik to IP/MPLS bus to core router in Banja Luka.

Overall served traffic is composed of served telephone and data traffic:

$$A_{mix} = A_t + A_i \tag{10}$$

Here, A_{mix} is overall served traffic, A_t is served telephone traffic and A_i is served Internet (data) traffic. In our case:

$$A_i = A_{dial\_up} + A_{ADSL}. \tag{11}$$

In (11), A_i is served data traffic directed from edge router in Zvornik to core router in Banja Luka by GEth (Zx) interface, and

$$A_{dial\_up} = A_{dial\_up\_56k} + A_{dial\_up\_ISDN} \tag{12}$$

where, $A_{dial\_up\_56k}$ is served traffic from subscribers with dial-up connections to Internet, and $A_{dial\_up\_ISDN}$ is served traffic from subscribers with ISDN connection to Internet and A_{ADSL} is served traffic from ADSL subscribers [19].

ADSL served traffic and dial-up served traffic are different, by their nature, because ADSL served traffic has bigger self-similarity than dial-up served traffic which can not be neglected. ADSL served traffic has different statistical properties. The third difference is the fact that data part of ADSL served traffic is directed permanently to IP/MPLS bus. The rest of traffic, i.e. dial-up served traffic and voice served traffic, which come to main exchange from all NEs, (Node Exchanges), TO and RDLU (Remote Digital Line Unit) is directed to the IP/MPLS bus after separation of voice and data, see Fig. 1.

Having in mind the self-similarity property of ADSL traffic and the fact that IP/MPLS bus has great capacity and DWDM equipment is ready to start with works, there is no sense to perform dimensioning of that part of the network. Statistical models used here are based on real data obtained by measurement of traffic between Access Server and edge router.

5.2 Traffic variables

There are two key traffic variables for dial-up traffic: holding time and interarrival time [27, 28]. In order to test initial assumption that holding time is exponentially distributed according to Erlang B model, we take only first variable in consideration. There are several methods to examine the underlying distribution [28]. In this paper, we use two: i) a histogram or density plot, and, ii) the squared coefficient of variance.

In order to test assumption of underlying distribution, we use data obtained by measurements during 6 months. Based on the collected data, it is very useful to plot frequency of occurrence of each score. This is known as frequency distribution, or histogram, which is simply a graph plotting values of service times (holding times), on x axis, and number of observations (frequency) on y axis. In ideal case, i.e. in the case of normal distribution, all service times will be distributed symetrically around the centre of all values (peak value at the mean). However, from Fig. 2, we see that this is not the case.

From Fig. 2, we can see that a positive skew is evident in the data. This implies that there are a many short sessions with short serving times. The large number of short sessions can be based on large number of aborted sessions. A histogram or density plot is more visual test and additional test has to be performed to examine distribution. The squared coefficient of variance C_x^2 is calculated to provide additional information about underlying distribution [27].

From Table 1, one can be read data about variance (column 8), mean duration of

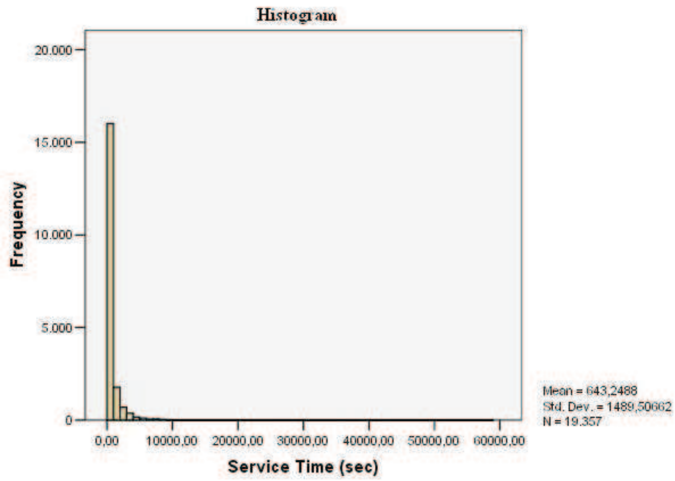


Fig. 2. Histogram of the aggregate holding times (based on data obtained by measurements from august 2008 to January 2009)

session (column 3) and square coefficient of variance (column 9). If squared coefficient of variance is equal to one, i.e. $C_x^2 = 1$, then X has exponential distribution. If $C_x^2 = 1/k$, then X has Erlang-k distribution. Finally, if $C_x^2 > 1$, than X has n-stage hyper-exponential distribution. The squared coefficient of variance (column 9) indicates that the services time follows hyper-exponentially distribution[2] [28].

Table 1. Daily Service Time Statistics

	Average Numbers of Sessions	Mean (sec)	Standard Deviation	25% Q3	50% Median	75% Q1	Variance	C_X^2
Monday	979,00	643,25	1.489,51	47	167	616	2.180.630	5,36
Tuesday	961,00	588,62	1.373,00	44	132	564	1.885.139	5,44
Wednesday	945,00	598,88	1.291,33	42	130	573	1.667.536	4,65
Thursday	971,00	621,78	1.498,38	40	146	592	2.245.164	5,81
Friday	945,00	614,40	1.467,29	47	138	584	2.152.931	5,70
Saturday	821,00	622,87	1.617,34	52	129	552	2.615.778	6,74
Sunday	808,00	637,38	1.425,07	54	130	590	2.030.814	5,00

From Table 1, we see that the mean holding times per day are slightly different, with the biggest value on Monday. In our case median holding time provides a little more insight into actual traffic [26, 28]. Based on the median value, we conclude

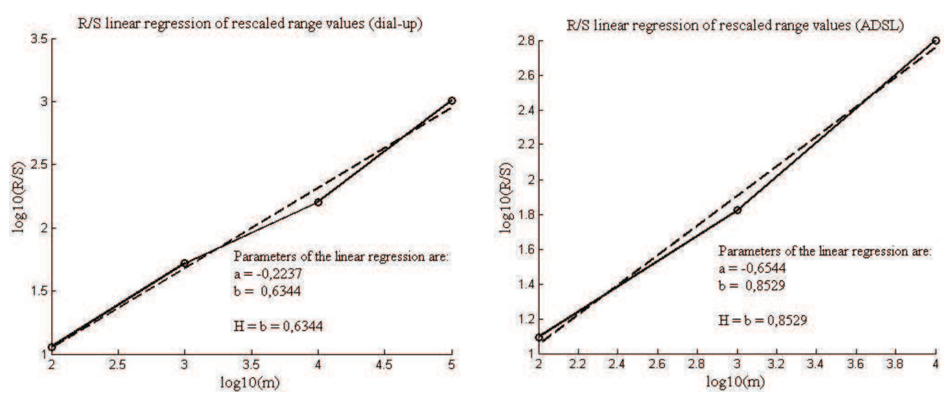
[2]Important properties of Erlang B formula is insensivity of underlaying distribution

that 50 percent of all sessions involve short holding times. Seventy-five percent of all sessions are less than 10 minutes in duration, and twenty-five percent of all session are between one half and one minutes. These results are in accordance with histogram in Fig. 2 and positive skew in it.

5.3 Self-similarity and Hurst parameter

In analysis of the dial-up traffic, we have used data obtained by traffic measurement during 17 months for the network segment under consideration, see Fig. 1. In analysis of ADSL traffic, we have used data obtained by traffic measurement during 8 months.

From Fig. 3.(a), we can see that the value of Hurst parameter for dial-up traffic is $H = 0,6344$. We can conclude that in the case of dial-up traffic we can use classical model for network dimensioning because $H < 0,7$. This fact is in accordance with results of [4] and [6].



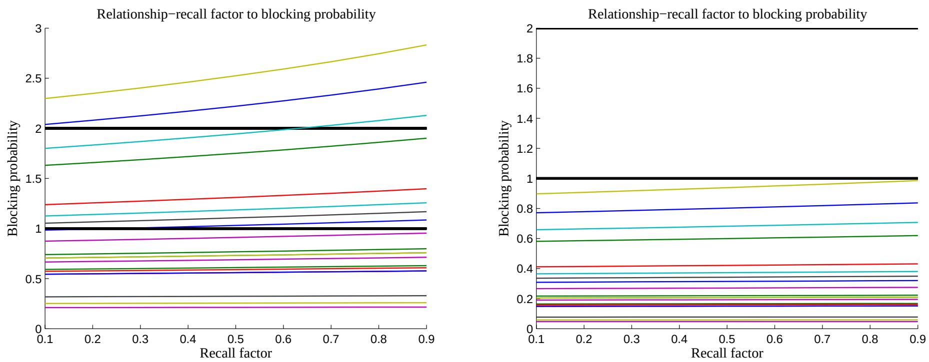
(a) (b)

Fig. 3. (a) R/S plot for dial-up traffic and, (b) ADSL traffic

From Fig. 3.(b), we can see that $H = 0,8529$, thus Hurst parameter is bigger than 0,7. We can conclude that in the case of ADSL traffic the property of self-similarity is significant, thus it is not possible to use classical model for network dimensioning.

5.4 Network dimensioning and Bandwidth calculator

Figures from 4(a) to 4(b) show relationship between blocking probability P_B and recall factor (repeated call attempt), which is varied in the range from 0.1 to 0.9, for each of 20 measured days, (table 2).



(a) (b)

Fig. 4. Relationship between blocking probability and recall factor for the constant number of trunks (from 39 in (a) to 42 in (d)).

Table 2. Served traffic for 20 days in ME Zvornik

Day/Year	Served Voice and Data Traffic BHT (in Erlang)	Served Data Traffic BHT (in Erlang)
15–Dec–08	74,30	26,70
16–Dec–08	70,10	25,19
17–Dec–08	74,80	26,88
18–Dec–08	69,20	24,87
19–Dec–08	74,30	26,70
22–Dec–08	75,90	27,27
23–Dec–08	76,20	27,38
24–Dec–08	74,60	26,81
25–Dec–08	75,90	27,27
26–Dec–08	76,20	27,38
19–Jan–09	66,10	27,77
20–Jan–09	71,60	30,08
21–Jan–09	67,60	28,40
22–Jan–09	67,20	28,23
23–Jan–09	68,20	28,65
26–Jan–09	72,50	30,46
27–Jan–09	70,00	29,41
28–Jan–09	64,60	27,14
29–Jan–09	66,80	28,06
30–Jan–09	70,70	29,70

The measurements are performed in ME Zvornik exchange. From Fig. 4(a), it can be seen that for the two days probability P_B is larger than 2%, therefore we must enlarge number of circuits. Hence, when the number of blocking the circuits is 42, as shown in Fig 4(b), blocking probability P_B is less then 1%. This result corresponds to the result which is obtained from the Bandwidth calculator. From Fig. 4(b), it is obvious that for the maximum served traffic of 30.46 Erlangs and blocking probability P_B of 1%, we obtain same result as from bandwidth calculator, which is shown in Fig. 7.

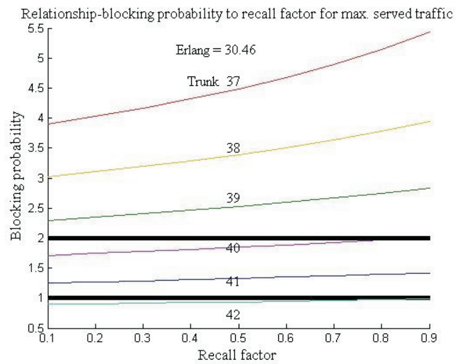


Fig. 5. Relationship between blocking probability and recall factor for the max. served traffic for 20 days in ME Zvornik (30,46 Erlangs)

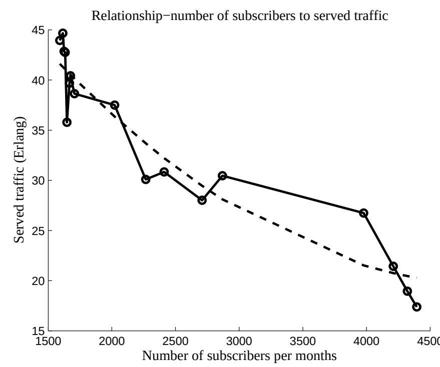


Fig. 6. Relationship between number of subscribers and measured traffic for dial-up subscribers

Figure 5 shows relationship between blocking probability P_B and recall factor, which is varied in the range from the 0.1-0.9, for the maximum served traffic of the 30.46 Erlangs. The results are tested versus different number of circuits ranging from 37 to 42. It is obvious from Fig. 4(b) and Fig. 5 that for the maximum served traffic of 30.46 Erlangs and blocking probability P_B of 1%, we obtain same result as from Bandwidth calculator, which is shown in Fig. 7.

Figure 6 shows relationship between calculated parameters of square regression, dashed curve in Fig. 6, and real data obtained from the billing system, solid curve in Fig. 6. The parameters of regression model are estimated using data obtained by measurements. Matlab is also used to check results obtained by Bandwidth Calculator and this is shown in Fig. 7.

6 Applaying Traffic Analysis to VoIP Networks

After proper dimensioning of PSTN network, the next task is to determine the portion of bandwidth on the link between switching part of the ME exchange and Access Server (see Fig. 1) that is used for VoIP. In this analysis, we consider influence of different Voice codecs, which are supported by Access Server, and Voice Activity Detection (VAD) on the bandwidth for VoIP call. The Access Server supports the following VoIP codecs: G.711, G.729, G.723 and G.726. By increasing

Bandwidth Calculator

BHT Traffic (Erlang)	30,46
Blocking Probability	0,01
Recall Factor	0,01
Number of Circuits:	42

Calculate

Algorithm	Codec Bit Rate (kbit/s)	Codec Sample Interval (ms)	Voice payload size (bytes)	Packet per second	IP/UDP/RTP	L2	Layer 2 header (bytes)	cRTP	Total packet size (bytes)	Bandwidth per connection no VAD (kb/s)	Bandwidth per connection with VAD (kb/s)	Bandwidth (no VAD) between Access Server and switching part of the exchange (Mb/s)	Bandwidth (with VAD) between Access Server and switching part of the exchange (Mb/s)
		10	80	100					158	126,400	82,160	5,309	3,451
G.711	64	20	160	50	40	Ethernet	38	N/A	238	95,200	61,880	3,998	2,599
		30	240	33					318	83,952	54,569	3,526	2,292
		10	80	100					128	102,400	66,560	4,301	2,796
G.711	64	20	160	50	40	PPP	8		208	83,200	54,080	3,494	2,271
		30	240	33					288	76,032	49,421	3,193	2,076
		10	80	100					92	73,600	47,840	3,091	2,009
G.711	64	20	160	50	40	PPP	8	4	172	68,800	44,720	2,890	1,878
		30	240	33					240	63,360	41,184	2,661	1,730
		10	10	100					88	70,400	45,760	2,957	1,922
G.729	8	20	20	50	40	Ethernet	38	N/A	98	39,200	25,480	1,646	1,070
		30	30	33					108	28,512	18,533	1,198	0,778
		40	40	25					118	23,600	15,340	0,991	0,644
		10	10	100					58	46,400	30,160	1,949	1,267
G.729	8	20	20	50	40	PPP	8		68	27,200	17,680	1,142	0,743
		30	30	33					78	20,592	13,385	0,865	0,562
		40	40	25					88	17,600	11,440	0,739	0,480
		10	10	100					22	17,600	11,440	0,739	0,480
G.729	8	20	20	50	40	PPP	8	4	32	12,800	8,320	0,538	0,349
		30	30	33					42	11,088	7,207	0,466	0,303
		40	40	25					52	10,400	6,760	0,437	0,284
G.723r63	63	30	24	33	40	Ethernet	38	N/A	102	26,928	17,503	1,131	0,735
		60	48	16					126	16,128	10,483	0,677	0,440
G.723r63	63	30	24	33	40	PPP	8		72	19,008	12,355	0,798	0,519
		60	48	16					96	12,288	7,987	0,516	0,335
G.723r63	63	30	24	33	40	PPP	8	4	36	9,504	6,178	0,399	0,259
		60	48	16					60	7,680	4,992	0,323	0,210
G.723r53	53	30	20	33	40	Ethernet	38	N/A	98	25,872	16,817	1,087	0,706
		60	40	17					118	16,048	10,431	0,674	0,438
G.723r53	53	30	20	33	40	PPP	8		68	17,952	11,669	0,754	0,490
		60	40	17					88	11,968	7,779	12,010	0,327
G.723r53	53	30	20	33	40	PPP	8'	4	32	8,448	5,491	0,355	0,231
		60	40	17					40	5,440	3,536	0,228	0,149

Fig. 7. Bandwidth calculator

number of voice samples per frame, the required bandwidth is decreased because overall amount of headers required is reduced for the call. But this can increase voice delay and decrease voice quality [1, 2]. The number of voice samples to be sent per frame depends on the type of codec that is used. It also depends on the ratio: high resolution and high bandwidth (20 ms sample duration) or lower resolution and lower bandwidth (50 ms sample duration) [3]. Default settings for all codecs are shown in Fig. 7. Typical voice conversation can contain from 35 up to 50 percent of silence. VAD sends RTP packets only when voice is detected. For VoIP bandwidth planning, it is assumed that VAD reduces bandwidth by 35 percent; this fact is used in the Bandwidth calculator [19]. The results obtained by Bandwidth calculator are displayed in Fig. 7.

The switching part of the ME exchange is connected to the Access Server by Ethernet network, and that is why Ethernet as L2 layer is considered in bandwidth calculator. PPP is also considered as L2 layer, thou procedure of the calculation for any L2 is same. The only difference is the packet size which corresponds to certain

L2 layer and consequently all other calculations follow this change. The protocol cRTP can be used with PPP to compress IP/UDP/RTP header to two or four bytes. This functionality is not supported in Ethernet [29]. Bandwidth per connection for a single VoIP call (with and without VAD) is calculated and it is shown in the 11th and 12th columns of Fig. 7. In Fig. 7, in the last two columns the overall bandwidth (with and without VAD) is shown. This is required bandwidth to support all calls during the peak hour. The overall bandwidth is product of the number of circuits calculated for the traffic in peak hour (BHT in Fig. 7) for the some blocking probability and bandwidth per connection. This represents full-duplex bandwidth or bandwidth required in both directions.

7 Conclusion

The bandwidth is probably the most expensive component of the network. This paper has shown a way how to calculate proper bandwidth for VoIP calls after proper dimensioning PSTN network and determining optimal number of the circuits. The model incorporated in software tool which works as bandwidth calculator is validated by simulation results in Matlab. We have used as an example a segment of Telekom Srpskes network for which we have enough data. However, calculation will be very alike for another provider or group of subscribers using leased lines for Internet access and who want to save resources with proper determining bandwidth for their purposes.

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