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MODELING AND SIMULATION APPLIED TO CAPACITY PLANNING OF VOICE GATEWAYS: A CASE STUDY

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ABSTRACT

In this work, we applied modeling and simulation to plan and evaluate the capacity of a real enterprise voice gateway system. We modeled the analyzed voice gateway and assessed it under different workload scenarios. We evaluated the actual setup under the real current workload demand, as well as future expected demands in terms of voice long-distance calls using PSTN and VoIP providers. Also, the existing voice gateway capacity was tested facing calls to mobile phones through GSM *Sim* cards. Finally, we tested a proposal setup to reduce the number of simultaneous E1 channels used per voice call, and found that our approach reduced the E1 usage rate by 14%.

1 INTRODUCTION

Although communication through social networks, e-mail, and instant messaging has grown significantly in recent decades, the telephone remains essential for many enterprise environments. The integration of telephone with data networks, using VoIP (Voice over Internet) protocols (Goode 2002), allows one to make phone calls using data network infrastructures, which mainly enables low cost long-distance and international calls through the Internet.

In general, the implementation of VoIP starts with the merging of the legacy telephone system with the IP telephony technology. The replacement, at once, of the legacy telephone infrastructure requires high investment and does not occur in short term. In large organizations, it is common that the first step of the VoIP implementation is the deployment of a voice gateway system, between the legacy telephony and IP telephony networks, in order to provide VoIP services. Given the fast growth of the VoIP usage and the essential role played by the voice gateways in this transition process, it is mandatory to plan the capacity of these gateway systems in order to guarantee the required quality of service. This task should be performed proactively, so the services provided can meet the current and future demands in terms of service level agreements (SLA).

Aiming to contribute to the body of knowledge in this field, the main goal of this work is to analyze and plan the capacity of a real enterprise-class voice gateway system, which supports the voice communication of six university campuses. We propose the use of computer modeling and simulation for planning the critical resources of the investigated gateway system. In this work, different scenarios are simulated to assess the current and future capacity of the system under study. We also evaluate alternative gateway setups for optimization purposes.

The remainder of the paper is organized as follows. Section 2 presents related works on modeling and forecasting in call centers. The voice gateway system investigated is presented in Section 3, and Section 4 describes the methodology used to model and simulate it. Section 5 presents the model validation. The main findings of this study are presented in Section 6. Finally, Section 7 summarizes our conclusions.

2 RELATED WORKS

Given that we did not find previous specific studies on modeling and simulation of voice gateway systems, in this section we analyze similar works applied to call centers.

Ibrahim et al. (2012) compare the accuracy of point estimate and distributional forecasting models applied to call center data. The square root transformation is used to stabilize the data set variance and the mean squared error to compare the models' precision. The distributional models showed better accuracy for the lead-time of one day and fourteenth days.

Robbins and Medeiros (2010) assess the errors from Erlang C models when used with real call center data. The space filling design approach, based on LHS (Latin Hypercube Sampling), is used to define the experimental region used in the simulation of a real call center. They conclude that the error prediction from Erlang C models were strongly correlated to the call abandon rates.

Mehrotra and Fama (2003) show how computer simulation can be used for decision making in call centers. Several scenarios are simulated to estimate the best number of active and receptive agents to achieve a given SLA. They conclude that simulation is a powerful tool for helping call center analysts to make educated decisions for capacity planning purpose.

Avramidis, Deslauriers, and L'Ecuyer (2004) present two statistical models that capture the variability of traffic intensity of phone calls during the day, and the non-zero correlation between the numbers of call at different period of the day. Data from a real call center confirmed the simultaneous presence of these two patterns. They conclude that compared to traditional NHPP (non-homogeneous Poisson process) and doubly stochastic models, the proposed models achieve better forecasting accuracy.

In (Ibrahim and L'Ecuyer 2013), the forecasting accuracy of four models representing the call arrivals in call centers is analyzed. Lead times ranging from hours to weeks are considered. The authors conclude that the models accounting for different correlation structures in the data obtained better results.

In (Taylor 2008), five prediction methods applied to univariate series of incoming calls in real call centers are evaluated. The results show that seasonal ARIMA models and an extension of Holt-Winters (Taylor 2003) are the most suitable models for predictions of up to two or three days ahead. For forecasts beyond three days, the simple arithmetic mean is hard to overcome.

Steinmann and Freitas Filho (2013) use simulation to generate synthetic data to compare call prediction algorithms in call centers. They present a simulation model that includes random events that typically affect the call centers' performance in real world scenarios. They conclude that synthetic data through simulation is a good alternative to test models for call center analysis.

3 CASE STUDY

In this work, we model and simulate a telephone system of a university with multiple campuses. This system not only connects the internal phone extensions, but it also connects to three phone providers: wired (PSTN), wireless (GSM), and VoIP. Figure 1 shows the architecture of the system under study.

Specifically, the investigated voice gateway connects the university's legacy PABX (Private Automatic Branch Exchange) to the VoIP provider. This gateway translates the standard ISDN (Integrated Services Digital Network) protocols to VoIP protocols. In this enterprise environment, the VoIP provider is used to make cost efficient long-distance calls. The calls that cannot be completed via VoIP are then forwarded to the PSTN (Public Switched Telephone Network). This happens when the destination number cannot be reached through the VoIP provider.

The voice gateway and PABX are connected through an E1 trunk with ISDN signaling, allowing the use of 30 simultaneous voice channels. As both systems are physically located at separate buildings, a single-mode optical fiber is used to connect them. The SIP protocol (Rosenberg et al. 2002) is used for signaling between the voice gateway and VoIP provider, and the RTP protocol (Schulzrinne et al. 2003) for transporting the voice traffic.

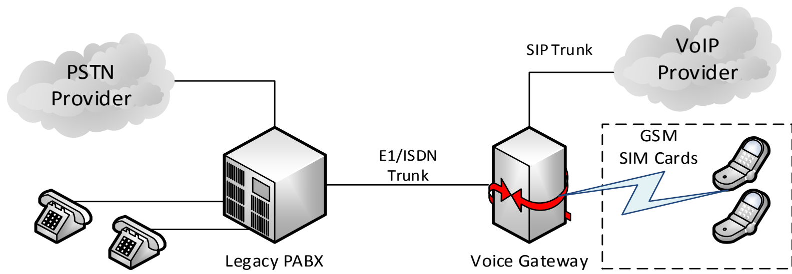


Figure 1: Telephone system investigated.

The voice gateway's call routing table contains the direct distance dialing (DDD) codes that can be completed via VoIP provider. This is a synchronized local read-only copy of the original database table stored into the VoIP provider's SQL database. Since the legacy PABX does not have the ability to query routes from SQL databases, its call forwarding logic is programmed to route all long-distance calls to the voice gateway. Upon receiving a call, the voice gateway checks its routing table and if it can be completed using the VoIP provider the call is forwarded to it through the Internet. Otherwise, the call is routed back to the PABX to be complete through the PSTN provider. Note that when a call firstly passes through the voice gateway and then goes back to the PABX, it uses two channels of the E1 trunk. In Section 6.5 we propose a solution to optimize the E1 channels usage for this situation.

In addition to the long-distance calls, the university is currently planning to also reduce costs of calls to mobile phones, using the existing voice gateway. The plan is to contract with a mobile phone carrier with competitive call rates and use GSM cards in the voice gateway to complete calls to mobile phones. Each GSM card supports multiple *Sim* cards, which will limit the number of possible simultaneous calls to mobiles. The square (dashed line) in Figure 1 illustrates the addition of GSM *Sim* cards to the current voice gateway setup in order to meet this new demand. In this new scenario, calls from PABX's extensions to mobile phones will be forwarded to the voice gateway via E1 trunk and completed using one of the new voice gateway's GSM *Sim* cards.

The PABX and voice gateway integration described above is commonly found in many organizations deploying VoIP along with their legacy telephone systems. As can be observed, the voice gateway turns to a critical component of the entire communication infrastructure. We model and simulate this system in order to analyze and plan its capacity to meet the current and near-future organization's demand for DDD via VoIP. Also, we evaluate the impact of serving calls to mobile phones through this voice gateway. For both purposes, we mainly focus on the capacity planning of E1 trunk and GSM *Sim* cards.

We look for answering the following research questions: Do the existing voice gateway resources (E1 channels) support the current DDD demand? Is it possible to optimize the use of existing voice gateway resources? How many GSM *Sim* cards are required to meet the university's demand for calls to mobile phones? Will the existing E1 channels be able to meet this aggregated demand? What is the percentage of lost calls per new demand scenarios?

4 METHOD & MATERIAL

4.1 Simulation Model

Figure 2 shows a high-level view of the simulation model created for the case study presented in Section 3. The calls are categorized into three types: (I) long-distance calls via VoIP, (II) long-distance calls via PSTN, and (III) calls to mobile phones. Given that in a near future the university wants to serve calls to mobiles via voice gateway, we also included this type of call in the simulation model.

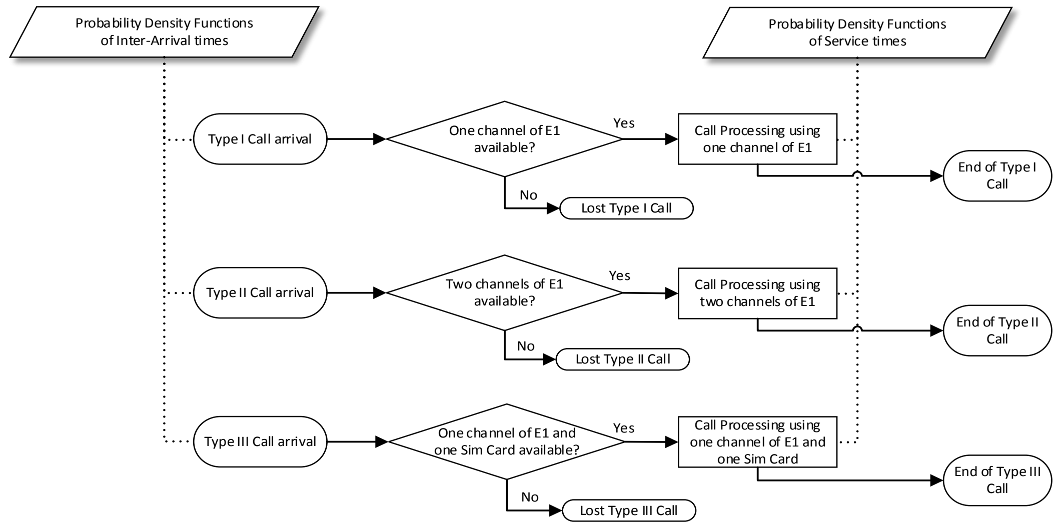


Figure 2: Simulation model overview.

When a call arrives at the voice gateway, it is checked whether there are sufficient resources to complete the call. If so, the allocated resources (E1 channels or GSM *Sim* cards, or both) are occupied during the call duration, and released in the end of call. Otherwise, the call is lost. To complete a call of type I, only one channel from E1 trunk is necessary. For type II, two E1 channels are needed, and for type III one E1 channel and one GSM *Sim* card are required.

Each evaluated scenario is simulated separately with the call data grouped in two 5-hour shifts: Morning (08:00am to 01:00pm) and Afternoon (01:00pm to 06:00pm). For each scenario, we perform 1,000 simulation replications and calculate the 95% confidence intervals for all response variables of interest. Our goal is to pinpoint bottlenecks caused by the congestion of E1 trunk and GSM *Sim* cards in different demand scenarios. Once we detect bottlenecks, we simulate the system with alternative amount of resources to find out the setup that meets the evaluated scenario's demand. Section 6 presents and discusses the simulation results.

Table 1 shows the resource used for each type of calls simulated in this study. Scenarios #1 and #2 simulate the current call demand and are used to validate the model. Scenario #3 also simulates the current demand, but with only 10 *Sim* cards to complete the mobile calls. Scenario #4 considers the demand increase of calls. Scenario #5 considers the demand increase of calls and an optimized setup we propose to reduce the number of E1 channels to complete calls of type II.

Table 1: Simulated call types and related system resources.

Scenario	Type of call	Resource
#1	I, and II.	30 channels (E1)
#2	I, II, and III.	Unlimited E1 + unlimited *Sim* cards
#3	I, II, and III.	30 channels (E1) + 10 *Sim* cards
#4	I, II and III with demand increase of 3%, 5% and 10% of type I and III, respectively.	30 channels (E1) + 16 *Sim* cards
#5	I, II, and III with demand increase of 3%, 5%, and 10%, respectively.	30 channels (E1) + 16 *Sim* cards + optimized setup

Table 2 shows the estimated response variables for each simulated scenario. In addition to their point estimate (average), we also compute the half-width of the 95% confidence interval ($\pm HW_{CI}$) for each average.

Table 2: Simulated response variables.

Response Variable	Description
E1_max	Average maximum number of E1 channels simultaneously used
E1_avg	Average number of E1 channels simultaneously used
E1_rate	E1 channels usage rate
GSM_max	Average maximum number of *Sim* cards simultaneously used
GSM_avg	Average number of *Sim* cards simultaneously used
GSM_rate	*Sim* cards usage rate
VoIP_total	Average total number of long-distance calls using VoIP (type I)
PSTN_total	Average total number of long-distance calls using PSTN (type II)
Mobile_total	Average total number of calls to mobiles (type III)
LDC_lost	Average number of lost calls of type I and II
Mobile_lost	Average number of lost calls of type III

4.2 Sampling Design

The sampling strategy adopted is based on call detail records (CDR) collected from the real voice gateway and PABX systems. We considered the outgoing calls during business hours (08:00am to 06:00pm), in one-year interval (Jan/2012 to Dec/2012). The CDR of long-distance calls (types I and II) were collected from the voice gateway, and to mobile calls (type III) from the PABX. Given that the system bottleneck manifests when the rate of resource utilization is high, we selected the 30 days with the highest number of calls from the annual sampling. This selected data set is available in (Alves, Matias Jr., and Freitas Filho 2013). It would not be worth to simulate the system with low number of calls, since no performance problems would manifest. Table 3 shows the descriptive statistics for the selected 30-day sample. All values (except for the "Total") are related to the daily calls.

Table 3: Descriptive statistics for number of calls.

	Type I (VoIP)	Type II (PSTN)	Type III (GSM)
Total	11921	19112	46557
Average	397.4	637.1	1551.9
Median	378	627	1552
Minimum	232	539	1418
Maximum	540	780	1748
Std. deviation	74.8	56.5	76.98

For types I and II, we considered the completed and uncompleted calls. The call duration is also taken into account, because during this time resource consumption occurs. Due to setting limitations of the legacy PABX, we only used CDR of type III calls that were completed.

All random variables are modelled through probability distributions (Goldsman 2007). We use three statistical tests, Kolmogorov-Smirnov (KS) (Massey 1951), Anderson-Darling (AD) (Anderson and Darling 1954), and Chi-squared ($\chi 2$) (Boero, Smith, and Wallis 2005), to check the goodness of fit (GoF) of different theoretical distributions to the respective variables' collected data.

For the variable "inter-arrival time of calls", we did not find a theoretical distribution with demonstrated good fit. It occurred due to the variability in the number of calls during different periods of the day. Hence, we follow the guideline in (Law and Kelton 1991) and adopt the strategy of grouping the calls in shorter periods; we used 15-minute periods. Based on this new data set, the goodness-of-fit tests indicated (p-value > .05) that the data follow an exponential distribution for all periods of the day. As expected, the average value varied from period to period, i.e., the arrival process is Poisson but the call arrival rate varies over time (see Figure 3). This pattern suggests a non-homogeneous Poisson process, thus we model this variable through a non-stationary exponential distribution (Law and Kelton 1991). The numerical data plotted in Figure 3 are listed in Table 11 (see Appendix A).

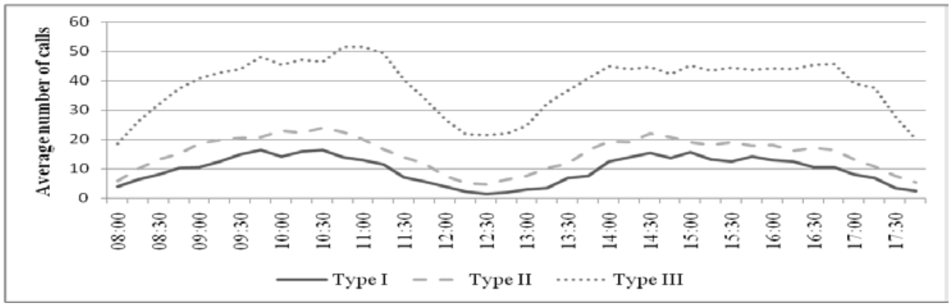
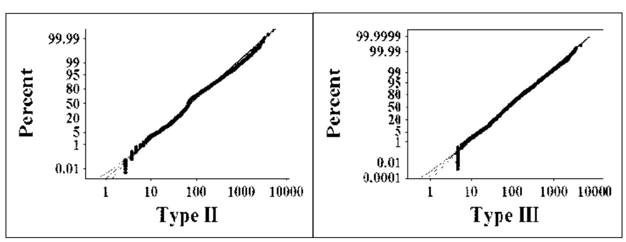


Figure 3: Average call arrival rate per 15-minute periods.

Next, we model the "call duration" for the three types of calls, separately. Table 4 shows the descriptive statistics of these three data sets. For calls of type I, the Chi-squared test presented a *p*-value of 0.0589 for the lognormal distribution, logn(119, 218). However, the Chi-squared and AD tests presented *p*-values less than 0.05 for the samples of type II and III, thus rejecting the hypothesis that these samples follow one of the tested theoretical distributions. Since we are using large number of samples (>11,000), it is not uncommon for these GoF tests to incorrectly reject the null hypothesis (Law and Kelton 1991). Therefore, we also use graphical analyses to support the model fitting procedure. Figure 4 shows the lognormal probability plots for calls of type II and III, respectively.

Table 4: Descriptive statistics for call duration (in seconds).

	Type I (VoIP)	Type II (PSTN)	Type III (GSM)
Average	118.9	130	126.4
Median	52	69	78
Minimum	4	4	6
Maximum	5162	3843	4461
Std. deviation	200.7	195.2	164.8



(a) (b)

Figure 4: Probability plots of "call duration" for calls of (a) type II, and (b) type III.

The graphical analysis suggests the good fit of the two data sets to lognormal distributions, whose parameters (LogMean and LogStd) are listed in Table 5.

Table 5: Lognormal distribution parameters (call duration).

Type of Call	LogMean	LogStd
I (VoIP)	119	218
II (PSTN)	123	155
III (GSM)	123	144

5 MODEL VALIDATION

To validate the model described in Section 4, it was necessary to verify whether it accurately represents the system under study (Law 2009). We firstly interviewed the engineers in charge of the university's telephony department, presenting all details of the simulation model and elicited their feedbacks to improve the model. Besides the interviews, we also performed a quantitative validation comparing the output of the simulation model with the results from the real system, which is described next.

Initially, we simulate the model using the resources and type of calls of scenarios #1 and #2 (see Table 1) to calculate estimates for the response variables of interest (see Table 2). Next, we obtain the observed values of the response variables using the real raw data (CDR). Lastly, we compare the observed and the simulated values, and based on their difference we assess the quality of the simulation model. As mentioned in Section 4, the data from the 30 days with the highest number of calls in 2012 were used as input for the simulation model. So, the CDR from these 30 days were used in the model validation. Figure 5 shows the validation process. First, we perform the model validation by analyzing the values of the response variables *VoIP_total*, *PSTN_total*, and *Mobile_total*, in order to verify if the amount of calls generated by the simulation model is comparable to the amount of calls observed in the real system. For the real observed values, we calculated the response variables using the average number of outgoing calls of the selected 30 days. Next we compare the values of the response variables *E1_max* and *GSM_max* to verify if the maximum number of E1 channels and GSM *Sim* cards used in the real system is comparable to the values produced by the simulation model.

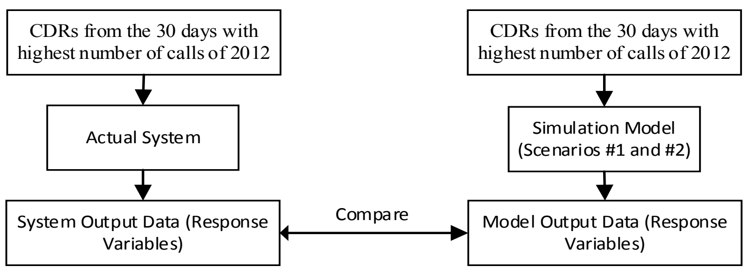


Figure 5: Overview of validation process.

We also compare the observed and simulated values for the average number of E1 channels simultaneously used (*E1_avg*), and the average number of *Sim* cards simultaneously used (*GSM_avg*). Equation (1) is used to calculate *E1_avg* and *GSM_avg* directly from the CDR data set,

$$Avg\_usage = T^{-1} \times \int_0^T B(t)dt\,, \qquad (1)$$

where T is the period of time analyzed (in seconds), and $B(t)$ is the amount of the evaluated *resource* allocated at time t. The validation results are described next.

5.1 Model validation based on scenarios #1 and #2

Scenario #1 represents the existing demand and voice gateway setup. Calls of types I and II are simulated and the PABX connects to the voice gateway through one E1 trunk (30 channels). Table 6 shows the simulated values for the response variables, as well as the observed values obtained from the real raw CDR data.

As explained in Section 4, the simulation is executed for two shifts (morning and afternoon). For each response variable, we calculated the 95% confidence interval ($\pm HW_{CI}$). Comparing the simulated values against the observed values, we note that the simulation outputs are comparable with the real observed values, indicating the good accuracy of the simulation model. In the second scenario, calls of types I, II, and III are simulated. We use unlimited E1 channels and unlimited GSM *Sim* cards to allow the voice gateway to route the calls. Hence, it is possible to analyze the system behavior also serving calls to mobile phones. Table 7 summarizes the simulation results, which are accurately comparable with the real observed values.

Table 6: Simulated and observed outputs for scenario #1.

Response Variable	Morning			Afternoon		
	Simulated	$\pm HW_{CI}$	Observed	Simulated	$\pm HW_{CI}$	Observed
E1_max	20.51	0.15	20.56	19.45	0.14	19.6
E1_avg	5.36	0.02	5.57	5.46	0.02	5.64
E1_rate	17.88%	0.88%	18.56%	18.22%	0.88%	18.8%
VoIP_total	189.4	0.87	188.9	198.14	0.87	197.76
PSTN_total	302.25	1.11	301.7	306.20	1.08	305.66
LDC_lost	0	0	0	0	0	0

Table 7: Simulated and observed outputs for scenario #2.

Response Variable	Morning			Afternoon		
	Simulated	$\pm HW_{CI}$	Observed	Simulated	$\pm HW_{CI}$	Observed
E1_max	29.2	0.17	29.43	27.82	0.15	27.33
E1_avg	10.51	0.03	10.41	10.94	0.03	10.84
E1_rate	N/A	N/A	N/A	N/A	N/A	N/A
VoIP_total	189.12	0.86	188.9	197.74	0.87	197.76
PSTN_total	301.73	1.06	301.7	305.97	1.06	305.66
LDC_lost	0	0	0	0	0	0
GSM_max	15.22	0.10	14.23	14.92	0.09	13.7
GSM_avg	5.18	0.01	4.84	5.50	0.01	5.19
GSM_rate	N/A	N/A	N/A	N/A	N/A	N/A
Mobile_total	700.22	1.60	700.06	742.69	1.68	742.8
Mobile_lost	0	0	0	0	0	0

6 RESULT ANALYSIS

6.1 Scenario #1

The results for this scenario (see Table 6) show that the maximum number of E1 channels used simultaneously is slightly higher in the morning. The difference between the average and maximum number of channels used is due to the call rate variation within the shift, i.e., there are several peaks of calls during a shift that raise the number of busy channels. Considering the current total demand for long-distance calls (*VoIP_total* and *PSTN_total*), we observe that 38.6% (morning) and 39.1% (afternoon) of the calls were completed via VoIP provider, which means a reduction of the call costs. Also, the average rate of E1 usage is less than 20% of its total capacity, and the E1 maximal usage is around 21 channels. Thus, we conclude that the 30 existing E1 channels are adequate to meet the current demand for calls of types I and II, given that no call is lost due to the lack of channels in the two shifts analyzed.

6.2 Scenario #2

Based on the results for this scenario (see Table 7), we observe that the average maximum number of E1 channels used simultaneously increased 42% in the morning and 43% in the afternoon, when compared to the first scenario. Importantly, we found that to meet the current demand for calls to mobile phones, an approximate number of 16 *Sim* cards would suffice to peak times. The maximum number of *Sim* cards used simultaneously in the morning is slightly higher than in the afternoon. This is noted in Figure 3, where the highest call arrival rate from type III occurs between 10:30am and 11:30am. The results also show that, in peak times, the number of E1 channels used simultaneous is close to the current capacity (30 channels).

6.3 Scenario #3

In this scenario, we simulate the current demand of type I, II and III calls using one E1 trunk (30 channels) and 10 GSM *Sim* cards. In scenario #2, we found that 16 GSM *Sim* cards are needed to meet the current demand for mobile calls. However, we want to evaluate the percentage of lost calls that may occur if only 10 GSM *Sim* cards are available to complete mobile calls. Table 8 shows the simulation results. As observed, the amount of lost calls to mobiles was approximately 31 (morning) and 29 (afternoon). This represents, respectively, 4.4% e 3.9% of lost calls to mobile phones.

Table 8: Simulation output of scenario #3.

Response Variable	**Morning**		**Afternoon**	
	Estimate	**$\pm HW_{CI}$**	**Estimate**	**$\pm HW_{CI}$**
E1_max	27.65	0.12	26.81	0.12
E1_avg	10.31	0.03	10.75	0.03
E1_rate	34.38%	0.01%	35.83%	0.01%
VoIP_total	188.84	0.85	197.60	0.86
PSTN_total	302.35	1.09	306.33	1.07
LDC_lost	0.23	0.04	0.10	0.02
GSM_max	10	0	10	0
GSM_avg	4.96	0.01	5.27	0.01
GSM_rate	49.68%	0.01%	52.75	0.01%
Mobile_total	701.26	1.67	742.45	1.70
Mobile_lost	30.72	0.72	28.78	0.65

6.4 Scenario #4

In this scenario we want to evaluate the current system capacity considering the following demand increases per call types: type I (3%), type II (5%), and type III (10%). We consider the system setup composed of one E1 trunk (30 channels) and 16 GSM *Sim* cards. Table 9 shows the simulation results.

The maximum number of E1 channels used simultaneously are 29.14 (morning) and 28.45 (afternoon), respectively. Also due to the increased demand simulated, comparing with scenario #2, the utilization (*E1_avg*) rose from 10.51 to 11.28 in the morning, and from 10.95 to 11.73 in the afternoon.

Table 9: Simulation output of scenario #4.

Response Variable	**Morning**		**Afternoon**	
	Estimate	**$\pm HW_{CI}$**	**Estimate**	**$\pm HW_{CI}$**
E1_max	29.14	0.07	28.45	0.09
E1_avg	11.28	0.03	11.73	0.03
E1_rate	37.60%	0.01%	39.11%	0.01%
VoIP_total	194.38	0.82	203.69	0.88
PSTN_total	317.40	1.09	321.50	1.16
LDC_lost	0.88	0.08	0.39	0.05
GSM_max	15.50	0.04	15.35	0.05
GSM_avg	5.70	0.02	6.03	0.02
GSM_rate	35.62%	0.01%	37.69%	0.01%
Mobile_total	771.10	1.78	816.84	1.79
Mobile_lost	1.68	0.14	0.86	0.09

The lost calls to mobiles were approximately 1.68 (morning) and 0.86 (morning), and the lost DDD calls were 0.88 (morning) and 0.39 (afternoon).

6.5 Scenario #5

As explained in Section 3, in the current voice gateway setup to complete a call of type II two E1 channels are used. This happens because this type of call passes through the voice gateway and goes back to the PABX before it leaves to the PSTN service provider.

In this simulation scenario, we evaluate an optimization proposal that will use only one E1 channel per call of type II. We know that the PABX device can forward a call to a secondary route if the primary route cannot complete the call. It happens when the network through which the call has been primarily routed does not serve the called number. In (ITU-T 1998), this situation is referred to as termination code number 3. Based on this feature, our proposal is to program the voice gateway to return the termination code 3 to the PABX, avoiding routing the call back to the PABX. Thus, calls of type II only use one E1 channel for 2 seconds, time needed to query the routing table and terminate the call with code 3. So, in this scenario we evaluate the system behavior considering the procedure of terminating calls of type II above described. We simulate the increased demand considered in scenario #4 using one E1 trunk (30 channels) and 16 GSM *Sim* cards. Table 10 shows the results.

Table 10: Simulation output of scenario #5.

Response Variable	**Morning**		**Afternoon**	
	Estimate	**$\pm HW_{CI}$**	**Estimate**	**$\pm HW_{CI}$**
E1_max	19.09	0.09	18.62	0.09
E1_avg	7.00	0.02	7.42	0.02
E1_rate	23.36%	0.07%	24.76%	0.08%
VoIP_total	194.48	0.85	203.76	0.88
PSTN_total	317.38	1.08	321.44	1.10
LDC_lost	0	0	0	0
GSM_max	15.50	0.05	15.39	0.05
GSM_avg	5.69	0.01	6.04	0.02
GSM_rate	35.59%	0.01%	37.77%	0.01%
Mobile_total	770.58	1.77	817.72	1.79
Mobile_lost	1.02	0.10	0.65	0.07

We observe that for the voice gateway optimized setup, the usage rate of E1 channels is about 23% in the morning and 24% in the afternoon. In scenario #4, which did not consider the E1 optimization, the usage rate was 37% in the morning and 39% in the afternoon. Thus, we conclude that our optimization proposal reduced the E1 trunk utilization rate in more than 14% when compared to the current setup. It allows the voice gateway to support an increase of demand for all three types of calls investigated.

7 CONCLUSION

In this work, we use computer simulation for capacity planning of a real enterprise voice gateway system. Due to the numerous details involved in the telecommunication system here studied, many analytical approaches could not successfully deal with this complexity. Indeed, using simulation it was possible to incorporate all necessary aspects of the real system required to study different scenarios of interest. Also, the visualization and animation of the simulation model were valuable features during the engineers' interviews for validating the model, which allowed us identifying the voice gateway setup optimization discussed in scenario #5; this optimization showed effective to reduce in 14% the E1 trunk usage rate.

We presented the results of five scenarios of interest that allowed answering all research questions raised at the end of Section 3. However, the approach used can be easily adapted to answer new questions such as the maximal call demand supported by the current setup; the cost increase per service provider contract for different demand scenarios of calls; the required bandwidth of the Internet link to provide VoIP calls to a new campus; and many others.

As future work, we will improve the presented simulation model in several directions. We will broaden the study including new response variables, especially related to resource costs. Further explorations will consider as input the forecasts of call demands estimated through statistical techniques applied to time-series derived from the CDR data sets. Also we will improve the model by adding the redials when the caller tries to make a call and there are no resources available to complete the call.

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A NUMERICAL DATA

Table 11: Average call arrival rate per 15-minute periods for the 30 days with the highest number of calls.

Period	Value			Period	Value			Period	Value		
	I	II	III		I	II	III		I	II	III
08:00-08:15	3.9	6.03	18.53	11:30-11:45	7.13	13.93	40.26	15:00-15:15	15.66	18.96	45.1
08:15-08:30	6.33	10.06	26.16	11:45-12:00	5.56	11.7	33.93	15:15-15:30	13.1	18	43.53
08:30-08:45	8.13	13.2	31.96	12:00-12:15	3.86	7.63	26.9	15:30-15:45	12.4	19	44.53
08:45-09:00	10.3	15.13	37.26	12:15-12:30	2.13	5.13	21.63	15:45-16:00	14.23	17.93	43.6
09:00-09:15	10.46	18.5	40.66	12:30-12:45	1.43	4.76	21.5	16:00-16:15	12.83	18	44.26
09:15-09:30	12.36	19.73	42.7	12:45-13:00	1.83	6.5	22	16:15-16:30	12.3	16.2	43.9
09:30-09:45	14.86	20.46	43.86	13:00-13:15	2.9	7.63	24.76	16:30-16:45	10.36	17.2	45.5
09:45-10:00	16.23	20.7	48	13:15-13:30	3.5	10.43	32.26	16:45-17:00	10.43	16.33	45.6
10:00-10:15	14.23	22.9	45.5	13:30-13:45	6.73	11.73	36.73	17:00-17:15	8.06	13.16	39.03
10:15-10:30	15.76	22.33	47.16	13:45-14:00	7.56	16.3	40.83	17:15-17:30	6.86	10.93	37.56
10:30-10:45	16.33	23.93	46.43	14:00-14:15	12.33	19.3	44.93	17:30-17:45	3.3	7.73	27.93
10:45-11:00	13.8	22.36	51.46	14:15-14:30	13.96	19.1	43.9	17:45-18:00	2.5	5.4	20.1
11:00-11:15	12.9	20.16	51.53	14:30-14:45	15.43	21.93	44.7				
11:15-11:30	11.53	16.76	49.33	14:45-15:00	13.6	20.73	42.26				

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