Design and Implementation of a VoIP System for Campus Usage: A Case Study at NPRU

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***Abstract*—Voice over Internet Protocol (VoIP) technology has continually gained popularity and has been widely adopted for personal and enterprise usage over the past decade. This results from its capability of reducing the cost of a pricy phone bill while being able to increase the organization's productivity as a whole.**

The contribution of this work is to discuss the guidelines in designing and the implementation issues in deploying a VoIP system for campus usage. Specifically, this paper illustrates how VoIP is designed to be integrated with the existing analog phone system, in order to provide a solution to the problems of internal phone line shortage and frequently damage of an analog telephone card, at Nakhon Pathom Rajabhat University (NPRU). The system's performance is then evaluated in terms of voice quality and the sufficiency of the number of trunks required, which reveals that the designed system operates with satisfactory performance.

I. Introduction

With the expansion of the Internet service area and high speed mobile data services made available via mobile networks, the continually and rapidly growing use of the Voice over Internet Protocol (VoIP) technology in Thailand has been witnessed over the last decade. This can be evidenced by the rise of many new VoIP providers in Thailand, e.g., TOT NetCall, True NetTalk, Call Cafe, CAT2Call, EasySIP, MounthMun and many more. From the user's perspective, VoIP provides a lower cost and more efficient way for voice communication than what the traditional telephone system can offer. This cost efficiency, together with other advance call features such as video call, voice mail to email and call conference, certainly leads to the improvement of the organization's productivity as a whole.

Although VoIP is highly seductive due to its advantages over traditional phone systems, there still are some drawbacks that needs awareness before making a decision to implement the technology. Understanding these drawbacks clearly can help the system developers to foreseeing what problem is to be expected and what precaution should be made during the design stage. To specifically mention, these drawbacks include voice quality, bandwidth dependency and security [1].

In VoIP, voice is packetized, compressed and then transmitted, in a form of data packets, over IP-based networks. Thus, voice quality of a call depends largely on the quality of the network, which varies due to numerous factors such as the hardware, the provider, the current internet connection status, the available bandwidth, etc. This leads to the difficulties in maintaining the good quality of VoIP call. In general, the quality of VoIP calls is expected to be worse than that of PSTN (Public Switched Telephone Network) calls due to the difference in the technology used for voice transmission (packet switching for VoIP vs circuit switching for PSTN). Moreover, VoIP call tends to experience longer delay which can be caused by packetization, compression and transmission processes. If the delay from these processes is accumulatively higher than a certain threshold (e.g., 150 ms [2]), this gives a rise to an echo which can be very annoying among the call parties. In addition to the call quality issue, by utilizing the IP-based networks instead of using the dedicated phone lines to transverse voice, VoIP is highly prone to network security threats (e.g., identity and service theft, viruses and malware, denial of service (DoS), call tampering, etc.) [3]. Security issue is often regarded as one of the most important issue in networking since any security compromise can lead to enormous losses and damages. As the matter of fact, many enterprises decide to delay the implementation of VoIP mainly because of the security concern.

By taking all these drawbacks in mind, we illustrates in this paper how a VoIP system can be designed to be integrated with the existing analog phone system to obtain satisfactory performance. The goal of the design is to provide a solution to the problem of internal phone line shortage and frequently damage of an analog telephone card at Nakhon Pathom Rajabhat University (NPRU). Unlike the work in [4] that also presents the use of VoIP over a campus environment, it does not provide the details of how a VoIP system is designed but mainly focusing on the performance illustration of the designed system. Also, while [5], [6] focus their studies to answer the question of whether the campus should switch from existing traditional phone system to VoIP, we focus our work on how the VoIP should be designed and implemented.

The rest of this paper is organized as follows. Section II describes the motivation of this work. Specifically, the section discusses the current technology that is used for voice communication at NPRU and the reason why it needs new technology. Section III then describes how NPRU-VoIP is designed to solve such problem. Next, in Section IV, the E-model technique is discussed in details and is used to perform a voice quality evaluation of the NPRU-VoIP system. We then conclude our work in Section V.

II. Current NPRU's telephone system

The telephone system of NPRU is an analog-based PBX that currently provides call services for a total of 300 users. The system supports the following call functions: auto attendant, call transfer, call waiting as well as caller ID. It is also connected with a PSTN service provider (e.g., TOT), in order to allow incoming and outgoing calls from and to users in PSTN networks. Let us define the term "internal call" as a call originated and destined within the campus while the term "external call" is denoted for a call originated within the campus and is destined to the users in PSTN networks. The PBX allows only the external calls that are destined to the phone numbers within Thailand. There are a total of 16 available connections for supporting the traffic flow (inbound and outbound calls) between the PBX and the PSTN network.

In spite of the existing traditional telephone system, there are two major reasons making NPRU become interested in implementing a VoIP system. These include (1) the frequent damage of analog telephone cards, used in the PBX, which is due to electrical surges and (2) the expansion in the size of the campus, leading to the problem of internal phone line shortage. Although the surge problem can be alleviated by installing good grounding system and surge protectors, it is not working very well for NPRU due to the fact that some of the phone lines are wired along with the power line, making it highly prone to be sensitive to the surge. Reinstalling the wiring of the phone lines is also not an option since this would involve in a costly investment in both time and money.

Alternatively, VoIP telephony is quite promising in solving these problems. This is because, in VoIP telephony, voice is converted from analog to digital signal and then transmitted from a caller to a called party in a form of data packets over a computer network. Since NPRU's computer network is formed by optical fibers wiring underground, this makes it quite immune to electrical surge.

To address the problem of insufficient phone lines due to the expansion of the campus size, the choice of adding more analog cards so as to increase the number of users the analog PBX can support is impossible because the maximum number of card the PBX can support have already been reached. Either having a new analog phone system that can support a larger number of users installed or integrating VoIP technology to the existing phone system is a possible choice. When cost is a primary concern which is the case for NPRU, the latter choice seems to be more attractive since installing a whole new analog PBX system that support a large number of users can be very costly. These lead us to the decision to integrate VoIP technology with the existing analog PBX system.

Specifically, the system to be designed and implemented for NPRU (NPRU-VoIP) must have the following characteristics:

- minimize the change in existing analog PBX so that there is the least impact on users in analog system.
- minimize the share of the campus's Internet bandwidth
- support the total of 45 VoIP users distributed throughout the campus
- support basic call functions such as call transfer, call waiting, caller ID, call log as well as conference call. Additional features are preferred but not mandatory.
- be able to interconnect smoothly with the existing analog PBX so that a call between the two systems is possible
- be able to interconnect with users in PSTN network so as to allow VoIP users to make outbound PSTN calls. The inbound calls will be routed through the analog PBX.

TABLE I
Call Traffic Collected during Oct.–Dec., 2011, from NPRU's Analog PBX with the total of 300 users

Type of Calls	No. of Calls/hour	Duration (s)/call
Internal	183	58.73
External	69	180

III. The Design of NPRU-VoIP

A. Analyzing the Traffic

Since there are going to be two systems (analog and VoIP) providing voice service, interconnecting users between the two systems is mandatory. When interconnecting two PBXs (e.g., analog PBX and IP-PBX for VoIP system), the number of channels or trunks required to allow the voice traffic flow between these PBXs must be determined.

To do so, the statistical data of call traffic must be examined. Since the VoIP system does not yet exist at the time, we assume that the traffic to be generated from 45 VoIP users would exhibit similar traffic distribution to what the 300 users in the analog system currently perform. We thus collect the information of the average number of call/hour and the average call duration (in seconds) for both internal and external calls. For internal calls, these information can be obtained from the call log made available by the analog PBX. Note that for NPRU's analog PBX, only successful calls are logged and it is logged for the length of the call in seconds. On the other hand, for external calls, we obtain the average number of call/hour and the average call duration from the phone bills generated by the PSTN service provider (in NPRU case, it is TOT). In detail, the calls are billed in one-minute rounding and since there is no charge for inbound calls (e.g., calls from PSTN network to NPRU), the service provider does not provide any detail for inbound calls. As a result, the information collected from the PSTN provider only reflect the traffic load for outbound external calls. Table I shows the statistical data obtained from the analog PBX and the PSTN provider.

To calculate the number of channels needed to interconnecting VoIP and analog users, we analyze the traffic flow (A) according to the given expression [7]:

$$A = C \times T, \tag{1}$$

where C and T are the number of calls originated during a period of one hour and the average holding time of the call (hour), respectively. While the number of calls originated during a period of one hour is quite self-explained, the holding time of the call may need some clarification. Specifically, the holding time is defined as the total time the trunk is in use. This

TABLE II
ESTIMATED CALL TRAFFIC DERIVED FROM TABLE I FOR THE FUTURE 45 VOIP USERS

Type of Calls	C	T (hour)	A (Erlangs)	No. of trunks[1]
Internal	27.45	0.018	0.494	4
External	10.35	0.055	0.569	4

TABLE III
RELATIONSHIP OF VOICE CODEC, VOICE QUALITY AND BANDWIDTH REQUIREMENT [7]

Codec Type	Coding Algorithm	MOS score	Bandwidth (kbps) [2]
G.711	PCM	4.1	80
G.726	ADPCM	3.85	48
G.728	LD-CELP	3.61	32
G.729	CS-ACELP	3.92	24

includes the time at which the the trunk is in use for dialing, call setup, ringing, conversation and releasing the trunk. Since call logs collected by PBXs generally report only the time during which the conversation takes place or what we refer earlier to as a "successful call", they need to be adjusted in order to obtain T.

As recommended by [7], for a system that uses one-minute increments round upward, calls on average will have 30 s. of extra holding time per call. To adjust the average duration of each call to reflect the actual holding time, an extra of 10% to 16% can be added on top of what is reported from the log of the PBX, in order to account for overhead. By taken this into account, the data from Table I is adjusted with 10% of call overhead for both internal and external calls. The adjusted number of call/hour and call duration are then normalized into the C and T for 45 users. The results are shown in Table II. By using Eq. (1), it is possible to compute the estimated traffic load (A) for 45 VoIP users. Next, the number of trunks needed to support the traffic load A with the probability of blocking p can be obtained using Erlang B table. From the above calculations with $p = 1\%$, the number of trunks for both internal and external calls are 4.

B. IP Bandwidth Requirement

Once the number of trunks has been specified, the next step is to compute the bandwidth requirement needed to support the traffic flow. By implementing the VoIP system, voice traffic is now allowed to transverse over the campus computer networks. We thus need to determine the impact of the voice traffic on the campus network's bandwidth. To compute the bandwidth requirement to handle a voice call, the algorithm used for voice codec must be specified first. Table III [2], [7] shows the relationship between the type of codec, voice quality (in terms of Mean Opinion Score: MOS) and bandwidth requirement.

The data shown in Table III indicates that tradeoff between the quality of a call and the bandwidth requirement must be considered. For internal calls that are routed within the campus, G.711 is preferred since the Intranet bandwidth of NPRU is at the rate of 1 Gbps which can support as many as 6250 simultaneous G.711 VoIP calls[3]. Thus, for internal calls, we prioritizes the call quality over the bandwidth consumption.

For external calls, there are two types of trunk connections available for connecting VoIP users with PSTN networks: by using (1) VoIP gateways and (2) VoIP service providers. The choice of choosing one over another depends on multiple factors such as the space requirement, the extra cost of hardware, the ease of maintenance, the difference in tariff charge etc. By using a VoIP gateway, the administrator has a complete control on how the trunk interacts with the IP-PBX (e.g., type of codec used for the trunk, caller ID transmitted with a call). Moreover, since the traffic is routed through PSTN network via analog phone lines or GSM channels, the sharing of the campus Internet bandwidth is minimized. However, additional hardware, such as FXO card, GSM gateway or ATA, is required. On the other hand, external calls can be routed over the Internet to VoIP service providers (e.g., CAT2Call, MountMun, True Nettalk, etc.), which have interfaces to PSTN networks. By doing so, there is no need to install any hardware when connecting the VoIP system to the VoIP service providers; however, VoIP administrator may not be able to have a full control on some call features. Furthermore, since connection between VoIP system to the VoIP service provider is achieved through the Internet connection, the more the number of simultaneous calls flow between the two PBXs, the larger the bandwidth is taken to support VoIP system. In addition to the bandwidth consumption issue, connecting a IP-PBX with a service provider requires a number of IP-PBX's ports to be opened to the Internet, in order to enable communication between the two servers. This consequently leads to security vulnerability.

Since our objectives are to minimize the Internet bandwidth sharing as well as to tighten the system's security, we decide to use the GSM gateways (Yeastar Neogate TG200) that support 4 GSM channels to support external calls from VoIP system. To interconnect analog and VoIP users, an FXO card with 4 FXO ports (Yeastar TDM400) is used. Moreover, this FXO card also handles inbound calls from PSTN to VoIP users. Fig. 1 shows the design of NPRU-VoIP that supports 45 VoIP users, 4 GSM channels for outbound external calls and FXO card with 4 FXO ports for inbound PSTN calls. Table IV summarizes the flows of calls for NPRU.

C. IP-PBX Design

An IP-PBX is a software-based PBX that provides the core functionality of a VoIP system. These include connecting a call between users, providing call features such as voicemail, call transfer, caller ID, call conference etc. In this project, the

[1] According to Erlang B table with the probability of blocking at 1%.

[2] The bandwidth required for one leg of a call.

[3] In this calculation, we assume that there is no other type of traffic sharing the IP bandwidth.

[4] If different codec is used for different trunk, then the IP-PBX needs to perform codec transcoding. The amount of CPU and RAM used during this process is dependent on the type of codec being transcoded and the number of calls being transcoded simultaneously.

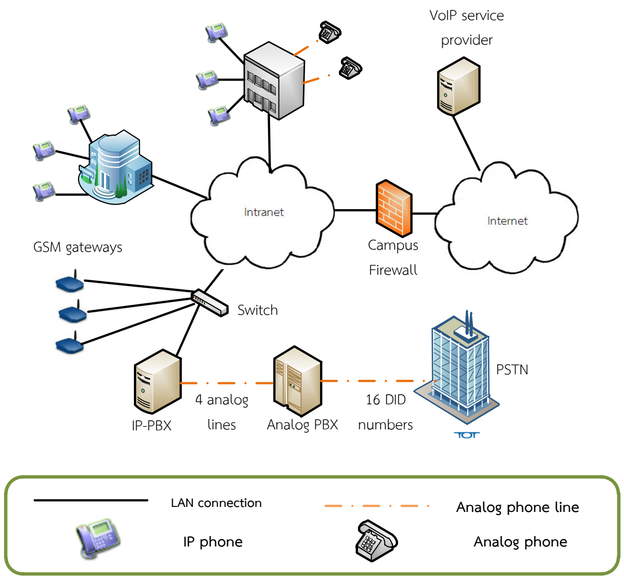

Fig. 1. The Design of NPRU-VoIP System.

TABLE IV
FLOWS OF A VOICE CALL AFTER THE INTEGRATION OF A VOIP SYSTEM WITH THE ANALOG PBX

Call Type	Flow of Traffic
VoIP → VoIP	A → IP-PBX → B
VoIP → analog	A → IP-PBX → PBX → B
VoIP → PSTN	A → IP-PBX → GSM gateway → B
analog → VoIP	A → PBX → IP-PBX → B
analog → analog	A → PBX → B
analog → PSTN	A → PBX → PSTN → B
PSTN → VoIP	A → PSTN → PBX → IP-PBX → B
PSTN → analog	A → PSTN → PBS → B

IP-PBX is developed using an open-source software, called Elastix [8]. Specifically, Elastix is a unified communication platform that combines many popular sets of software available for Asterisk-based PBXs into a single interface. As a result, Elastix is highly user-friendly, especially for those users who are not familiar with Linux.

In terms of hardware design, it is important to clearly specify all the parameters consuming the IP-PBX's resources (e.g., CPU, RAM and hard disk). These include the number of concurrent calls, type of codec used for each trunks, call features (e.g., voice mail, music on hold, call conference etc.) to be supported by the IP-PBX, how the system and the calls should be logged and how often the backup should be performed. Although it is very difficult to exactly specify how much each of these parameters consume the IP-PBX resource individually, this information can help us to roughly determine

TABLE V
IP-PBX FUNCTIONALITY AND ITS RESOURCE CONSUMPTION

Call Functionality	CPU	RAM	Hard disk
No. of concurrent calls	√	√	
No. of simultaneous transcoding[4]	√	√	
Space for Voice mail			√
Call log and Backup			√

the specification of the hardware for developing the IP-PBX.

For NPRU-VoIP project, although there are only 45 users in the initial phase, the total number of users can be as high as 400 users in the near future. Also, although there is no transcoding at the present, transcoding will be needed later when we activate the service for VoIP users connecting from out of campus network via Internet connection. For such scenario, the codec used for these users will be G.729 so as to minimize the Internet bandwidth sharing. Thus, extra RAM should be spared for transcoding process. We consequently use a tower-type server with Intel XEON processor–3.2 GHz, RAM–8 GB, Hard disk–1 TB and hardware raid–1 as the IP-PBX for NPRU-VoIP.

D. Numbering System

1) Number of digits: Since there are going to be two phone systems (analog and VoIP) working together, numbering becomes an important issue and need to be designed carefully. This is because a good numbering system can reduce call processing delay, provide user location and zoning as well as lead to easier management. To determine the number of digits to be used to form a phone number, it is intuitively that these number of digits must be able to support current and expected number of users in the near future. The digit used to form an extension number should be composed of at least four digits since lesser digits not only limits number of extensions in the system, but also may be prone to security vulnerability as well as overlap with the set of pre-configured feature codes used by the IP-PBX. Although there will be only 45 VoIP users for our initial design, it is expected that the number of users will grow to approximately of 400 users in a next few years. By taking all these facts into consideration, an extension number for NPRU-VoIP is then designed to be a 4-digit number.

2) Leading Digit: Next, we need to determine what set of the 4-digit numbers should be used for VoIP extension numbers. To avoid any confusion that may occur, the set of 4-digit numbers used for VoIP extension numbers should not overlap with the numbers that have already been assigned to extensions in analog system. Keeping this in mind, we are next going to select the possible leading digits for our VoIP system. Generally, leading digit is used for specifying the trunk to which a call should be routed. As the IP-PBX can be connected to other systems such as PSTN, VoIP service providers, or GSM gateway, routing a call based on leading digit can be one of the techniques. Since the 1, 2, 6, 7, have been given as the leading digit for NPRU's analog phone system, we then choose 3 to be leading digit for our VoIP extension numbers. Thus, the extension number of VoIP users will be 3XXX where X is any possible number ranging between 0-9, resulting in the total of 729 possible VoIP extension numbers. This is more than enough for supporting any increment in number of users for the next several years.

E. Endpoint Terminal

Generally, there are 2 types of phone for VoIP system: (1) Softphone and (2) Hardphone. While both share the common

TABLE VI
COMPARISON ON POWER CONSUMPTION OF AT-610 AND LINKSYS PAP2T

Phone state	Power consumption (W)	
	AT-610	PAP2T[5]
Idle	1.98	2.62
Ringing	2.31	4.02
Speaking	2.20	3.90

purpose of making and receiving phone calls that are routed over the Internet, the two are quite different in many aspects. Softphone refers to a piece of software that can be installed in a computer (e.g., PC, notebook, smartphone, tablet, etc.) and utilizes the host media resources for echo cancellation. Nowadays, there are numerous softphones available for free to download, for example, X-lite, 3CX, Zoiper and many more. Moreover, the call features provided by softphones typically are broader when comparing with what hardphones can offer. As being able to be inexpensively obtained and easily installed on to a computer, softphone offers a low cost and fast installation, which is suitable for users with mobility.

On the other hand, hardphone refers to a physical equipment that is dedicated for telephony services. Hardphone is usually implemented with DSP chip dedicated for echo cancellation, resulting in a better voice quality. Since hardphones come in a form of a physical phone, when a user wants to make or receive a call, he/she just lifts the handset and start dialing or speaking. Comparing with the use of microphone and headset for a call made with softphone, hardphone is considered to be more user-friendly. Despite providing better voice quality and being more user-friendly, hardphone is more expensive (the price is dependent on the features of the phone) and typically involves with costly installation.

Although minimizing the cost of the system is one of our main objective, voice quality and user-friendliness are also main priority concern, in this case, we thus choose to deploy hardphones for NPRU-VoIP. Specifically, there are two types of hardphone that can be used with VoIP system: (1) IP phone and (2) analog phone with Analog Terminal Adapter (ATA). As suggested in [5], [6] that the power consumption of IP phone and ATA should be investigated so that the choice of implementation can be chosen based on the minimization of the long-run electrical charge. Table VI illustrates the comparison of power consumption between IP-phone (AT-610) and ATA (Linksys PAP2T).

Table VI indicates that the use of IP-phone leads to higher power consumption than that of the ATA. Thus, we decide to deploy ATA with analog phones for most VoIP users while the IP-phones are given to only those users who need advance call features (e.g., voice mail, call transfer, etc).

IV. PERFORMANCE EVALUATION

Once the VoIP system has been completely set up and run, it is important to ensure that the voice quality and the number of trunks determined in the design stage really meet the actual traffic requests.

[5]The power consumption is obtained from having two analog phones connecting with the PAP2T.

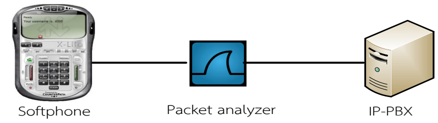

Fig. 2. Experiment setup for measuring network parameters for calculation of MOS score.

A. Examining the Voice Quality

To examine the voice quality, we use a subjective technique called the E-model [9] to obtain the overall transmission quality rating which is usually referred to as the R-factor. In particular, the E-model estimates how much impact the network impairments have on voice quality. Specifically, the network impairments used in E-model include the impairments caused by the delay, the echo and the packet loss. As suggested by [9], R-factor can be obtained from:

$$R = 93.2 - I_{\mathrm{d}} - I_{\mathrm{e}} - A, \qquad (2)$$

when I_{d} denotes the impairment factors due to delay and echo effect, I_{e} is the codec-dependent equipment impairment factor caused by jitter and packet loss and A is an advantage factor, where A = 0 for wireline transmission and A = 5 wireless transmission.

As given in [10], [11], I_{e} for G.711 can be computed using:

$$I_{\mathrm{e}} = 22\ln(1 + 0.2l), \qquad (3)$$

and for G.729, I_{e} is given by:

$$I_{\mathrm{e}} = 11 + 31\ln(1 + 0.15l), \qquad (4)$$

when l is the percentage of the packet loss. Next, I_{d} is given in the expression of [2]:

$$I_{\mathrm{d}} = 0.024d + 0.11(d - 177.3)H(d - 177.3), \qquad (5)$$

where d is the one-way packet delay (ms) and $H(x)$ is a unit step function.

In order to obtain the R-factor, as suggested in Eqs. (3) and (5) we need the information on the one-way packet delay and the packet loss. To do so, we setup an experiment using a softphone (X-lite) and a packet analyser (Wireshark) according to Fig. 2 and perform an internal VoIP-to-VoIP call test. Fig.3 shows the result of the test call from which the following information can be extracted: $l = 0$ and $d = 30.13$. These values are then substituted into Eqs. (3) and (5) to obtain $I_{\mathrm{e}} = 0$ and $I_{\mathrm{d}} = 0.723$, respectively. Next, according to Eq. (2), I_{e} and I_{d} are used to obtain $R = 92.477$. Similar test is also performed on outgoing calls, on which G.729 is applied. The results obtained on this particular case are l = 0 and $d = 70.43$, which consequently leads to $R = 80.51$.

Once the R-factor has been determined, it can be converted into a MOS (Mean Opinion Score) rating which is typically used to indicate the quality of voice. Table VII [10], [11] summarizes the relationship of R-factor, MOS rating and user satisfaction. Based on this table, the R-factors of NPRU-VoIP system with the codec of G.711 and G.729 are 92.477 and 80.51, respectively or are equivalent to the MOS rating of

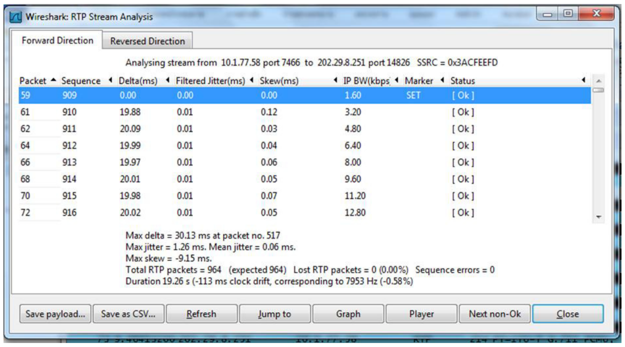

Fig. 3. Result of network parameters obtained by Wireshark.

TABLE VII
R-FACTOR, MOS SCORE AND USER SATISFACTION

R-factor	MOS	User satisfaction
$90 < R < 100$	$4.34 < MOS < 4.50$	Very satisfied
$80 < R < 90$	$4.03 < MOS < 4.34$	Satisfied
$70 < R < 80$	$3.60 < MOS < 4.03$	Some users dissatisfied
$60 < R < 70$	$3.10 < MOS < 3.60$	Many users dissatisfied
$50 < R < 60$	$2.58 < MOS < 3.10$	Nearly all users dissatisfied
$0 < R < 50$	$1.00 < MOS < 2.58$	Not recommended

4.34-4.50 and 4.03-4.34, respectively. These indicate that the voice quality of both the internal and the external calls are quite good.

B. Verifying the Sufficiency of Number of Trunks

Next, we attempt to verify whether the number of trunks interconnecting between the VoIP and the analog PBX are enough to handle the actual traffic requested by the users from both the VoIP and the analog PBX systems. This step is required because the traffic load used in designing the number of trunks is estimated based on only the traffic load generated by 300 users in the analog system. Fig. 4 illustrates the percentage of traffic (calculation based on the number of sec. the trunk is being occupied) serviced by each trunk. As can be seen from the figure that trunk no. 4 never has to serve any call yet. Thus, the number of blocked call is currently at 0 which is desirable. Due to space limit, we do not show the VoIP-PSTN trunk usage. However similar result (no block call) can be obtained for the GSM gateway usage.

V. CONCLUSIONS

In this paper, the design and implementation of a VoIP system for NPRU have been discussed in details. The main focus of the design is to solve the problem of the internal campus phone line shortage which is caused by (1) the frequent damage of an analog card required by analog PBX system due to electrical surge and (2) the expansion in the size of the campus. In order to minimize the cost of installation and hardware investment, the VoIP system has been designed to be integrated with the existing analog telephone system. As being a pilot phase, the scope of the work has been limited to support only 45 VoIP users, all of which must be able to smoothly interconnect with the other 300 users in analog telephone system. Moreover, according to the university's policy, the Internet bandwidth should be highly reserved for data usage, thus, the VoIP system is designed in such a way that it gains the least share of the campus Internet bandwidth.

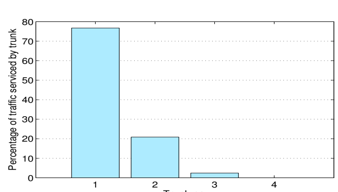

Fig. 4. VoIP-analog PBX trunk usage.

Having these requirements in mind, we have shown the step-by-step of how to achieve the design of such a VoIP system. This includes the traffic and bandwidth requirement analysis, the numbering system and the IP-PBX design, as well as the endpoint terminal selection. In addition, we provide the technique used in evaluating the quality of voice call which shows that the system offers a superior voice quality. The number of trunks has also been rechecked again after the deployment so as to ensure that the system can really handle the actual traffic requests.

It is important to mention that although the design and implementation issues reported in this paper are taken from a case study at NPRU, they can be useful as guidelines that help to fasten the VoIP development.

ACKNOWLEDGMENT

This project receives funding support from Nakhon Pathom Rajabhat University, 2012.

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