VoIP Call Simulation on UiTM Data Network

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***Abstract*-Voice over Internet Protocol (VoIP) becomes popular communication solution. A great evolution on Internet technology has brought up this technology to emerge and expand since it uses data network to transmit voice from sender to receiver which means voice and data travel using a single medium. By upgrading existing PBX to IP PBX, it makes VoIP to easily blend into existing communication technology. VoIP have been implemented and widely used in United States, European countries, and Asian countries. Informal observation and interviews discovered several organizations in Malaysia have already used VoIP. Therefore, it is a matter of time to UiTM to use this technology as a major communication medium among its 36 campuses nationwide. Majority campuses still use PSTN while the newest campuses already use VoIP. However, UiTM continues upgrading its Internet bandwidth to facilitate its daily operation. Therefore there are opportunities VoIP will replace PSTN as its major communication medium or at least allowing these two technologies to work together. The reason VoIP can be a great option is because it promotes call cost deduction in communication especially organization such as UiTM that have many branches. Next Generation Network (NGN) technology is believed and able to support VoIP to be implemented in UiTM. The good thing about NGN is amount of bandwidth can be allocated to allow voice transmission and it is flexible, it can be increased or decreased depending on network utilization. However, call cost deduction is not enough, the most important is number of VoIP calls that can be sustained by existing network while satisfying QoS requirements and leaving adequate capacity for future growth.**

Keywords—VoIP, next generation network, data network

I. INTRODUCTION

Universiti Teknologi MARA (UiTM) is not an ordinary public university in Malaysia. As of 2016, it has 45 active campuses spread all over Malaysia. The uniqueness of this university has made communication among branches are crucial. In the meantime, most of the campuses are having internal Public Switch Telephone Network (PSTN) and Metro-E data network to facilitate communication. However, there are several new campuses have using VoIP as its main communication technology. UiTM pay huge amount of many for bills and maintenance cost for both communication technologies. According to Telecommunication Unit, average traditional phone call in every branches is 187 phone calls in a single day and 478 phone calls in UiTM Shah Alam. The number will increase especially during examination period and convocation whereby the average phone call is 304 in every branches and 687 phone calls in UiTM Shah Alam. These phone calls include inter-branch call and outside call. In 2007, traditional phone call cost was about RM100,000.00 for all campuses. In 2015, UiTM paid around RM175,000.00 for traditional phone call cost to telecommunication provider.

In the meantime, according to InfoTech of UiTM, current bandwidth almost in all campuses has been increased to support current needs. Current cost of Internet bandwidth in 2015 is around RM25 millions for all campuses. It is going to increase from time to time especially to support teaching and learning processes where e-learning is getting attention by the university's top management.

InfoTech of UiTM allocated bandwidth according to services as depicted in Table 1. It shows multimedia application including voice service have 30% from the actual bandwidth. Usually, bandwidth for voice and multimedia is not shared with data traffic. Therefore, performance and quality of the VoIP call will not jeopardize if the network congested. VoIP has other issues which will not be discussed here as this study is focusing on determining number of VoIP call can be made according to 30% of bandwidth allocation.

TABLE 1. BANDWIDTH ALLOCATION FOR SERVICES

Portion in %	**Services**
30	Multimedia application and services including VoIP
30	Mission critical information systems
20	Standard data
20	Economy

UiTM do not have any plan to replace existing internal PSTN with VoIP. Existing PSTN will continue to serve because majority campuses already invested on this technology. Those campuses will apply toll-bypass whereby every phone calls do not have to go through external PSTN anymore [13]. It uses data network subscribed by UiTM. Only new campuses will use fully VoIP as its main communication medium. In order to support this technology, UiTM have upgraded existing Private Branch Exchange (PBX) to IP-PBX in all campuses. The old and new technologies have merged together at this point [7, 13]. The merge of these two technologies allow UiTM to apply toll-bypass as mentioned

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earlier. It gives huge benefits for inter-branch call where there is no longer charges apply to the phone call, video conferencing between campuses can be simply initiated and cost of traditional phone call can be reduced from time to time. Therefore, UiTM does not need to allocate more budgets to maintain existing internal PSTN and pay to the telecommunication provider for traditional phone call bills [2, 13].Current bandwidths have been upgraded as depicted in Table 2.

TABLE 2. BANDWIDTH ALLOCATION FOR EACH CAMPUSES

Campuses	**Bandwidth (Mbps)**	**30%**
UiTM Shah Alam	400 (200 x 2)	120
UiTM Perlis (Arau), UiTM Perak (Sri Iskandar)	150	45
UiTM Puncak Perdana, UiTM Selayang, UiTM Sg Buloh, UiTM Pulau Pinang (Permatang Pauh), UiTM Kelantan (Machang), UiTM Terengganu (Dungun), UiTM Pahang (Jengka), UiTM Negeri Sembilan (Kuala Pilah & Seremban 3), UiTM Melaka (Lendu & Jasin), UiTM Johor (Segamat), UiTM Sabah (Kota Kinabalu), UiTM Sarawak (Samarahan 1)	120	36
UiTM Perak (Tapah), UiTM Johor (Pasir Gudang)	110	33
UiTM Sarawak (Samarahan 2)	106	31.8
UiTM Seksyen 17 (INTEC), UiTM Terengganu (Bukit Besi), UiTM Pahang (Raub), UiTM Negeri Sembilan (Rembau), UiTM Melaka (Graha)	100	30
UiTM Puncak Alam, UiTM Dengkil, UiTM Terengganu (Chendering), UiTM Sarawak (Mukah 2)	80	24
UiTM Pulau Pinang (Bertam)	60	18
UiTM Intekma Resort, UiTM Kelantan (Kota Bharu), UiTM Perak (Training Center), UiTM Negeri Sembilan (Nilai), UiTM Sabah (Tawau)	50	15
UiTM Meranti College, UiTM Perak (Teluk Intan)	10	3
UiTM Cendana College, Akasia College, Cemara College, UiTM Jalan Othman	6	1.8

Those campuses in Table 2 are active campuses whereby there are operations in the campuses. Campuses that have been shut down and still in development are excluded from the list.

II. METHODOLOGY

In order to determine number of VoIP call can be initiated on UiTM data network, simulation methodology is applied [4, 5, 6]. There are many tools available, but we chose *VoIP Analytical Simulator*. More sophisticated tool such as OPNET can be used as well. In order to conduct the simulation, we applied eight crucial steps in Fig. 1 as suggested by [4].

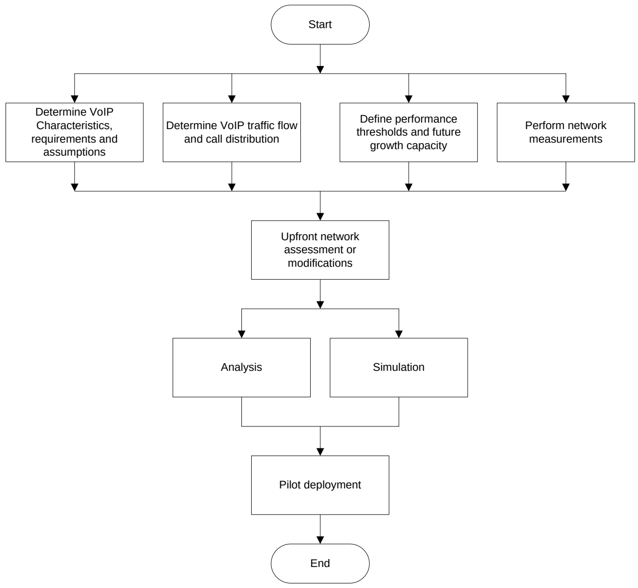

Fig. 1. VoIP deployment steps.

The eight steps above have covered every single aspects in VoIP deployment. Starting from the hardware elements, current traffic flow and distribution until the deployment step.

III. ARCHITECTURE

Since there are campuses still using PSTN, and it requires huge cost to total replace the technology, UiTM has upgraded the existing PBX to IP-PBX to handle VoIP call. Upgrading to IP-PBX allows analog voice from traditional phone being converted to digital and then rides through data network until it reaches the recipient IP phone or traditional phone. Therefore, the existing PSTN still can continue its services and only be integrated with data network [3, 9, 13]. Besides, it can be used as backup communication whenever data network having major issues such as hardware failure, broken network cable, and few others issues that caused data network cannot be used for certain period of time. In fact, there is no doubt regarding voice quality offered by PSTN as described by [2].

Fig. 2 depicts the convergence of existing PSTN and data network in UiTM [13]. This is the common setup for having the two technologies work together [11, 13]. Assume left and right side are two different campuses but connected to similar Metro-E subscribed by UiTM.

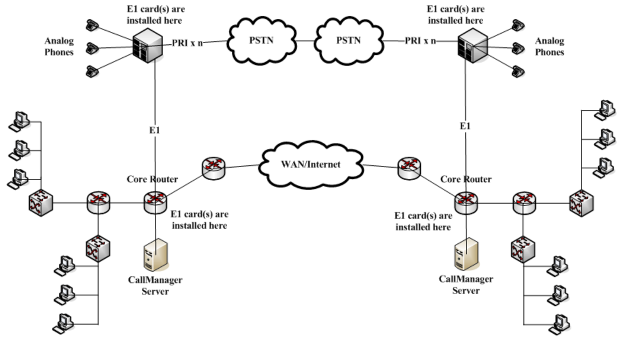

Fig. 2. Convergence of technologies in UiTM.

All hardware elements basically are similar in every campuses because controlled by InfoTech. Therfore, it can be conluded that all campuses have same setup and configuration.

IV. Simulation Procedure

VoIP Analytical Simulator tool developed by [6] is used to determine number of VoIP call can be made between campuses to achieve objective of this research. We used [9] standard configuration for VoIP call. Total packet size required is 18 bytes for header + 40 bytes for Internet Protocol (IP) + 20 bytes of voice payload = 78 bytes, which means equivalent to 624 bits. Then, to determine packet per second (pps), 8Kb codec bit rate / (20 bytes of voice payload * 8 bits) = 50 pps for one direction. Therefore, it requires 100 pps for both directions. Initial delay of G.729 standard is 20 ms of packet delay + 5 ms of initial look-ahead = 25 ms for first speech frame from sender [1, 10]. Terminals latency or network latency should be maximum 80 ms as recommended by [8, 12]. Maximum latency for acceptable one-way VoIP call is 150 ms as recommended by [8]. Therefore, receiver latency is 150 ms – (25 ms + 80 ms) = 45 ms. Those values are plugged into the simulation tool as depicted in Fig. 3 below.

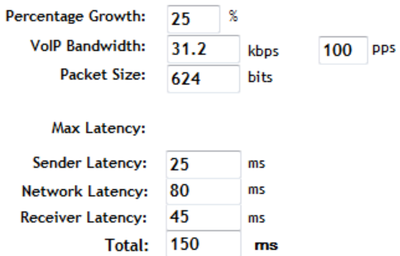

Fig. 3. VoIP settings.

G.729 standard has 3.92 Mean Opinion Score (MOS) as described by [8, 12] which mean quality of the voice is close to PSTN. Besides, it requires only 8Kb for VoIP call compare to PSTN which is 64Kb. There are other standards but not recommended for VoIP call as it may create lots of problems later on.

V. Result

Two reports generated by the simulation tools. Fig. 4 shows number of VoIP call based on bandwidth analysis. A total of 337 calls can be supported for the whole network. The report also shows there is a bottleneck at the router, therefore, it may require router replacement to support more calls. Fig. 5 shows number of VoIP call based on network analysis. Only 325 calls can be supported on the network as long as VoIP flows does not exceed that 80 ms. That 325 calls introduced a network delay of 11.92 ms. These number of calls can be initiated at the same time compare to average number of traditional phone call in every branches as mentioned earlier which made in a single day. Therefore, VoIP could handle more phone call compare to traditional phone call. Besides, these 325 calls are free for inter-branch call compare to traditional phone call which definitely will be charged by telecommunication provider. Thus, VoIP is the best technology that will benefit UiTM in reducing phone call cost every year.

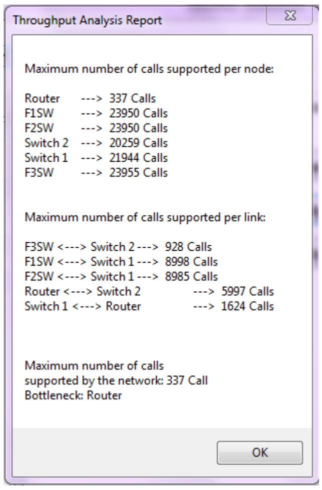

Fig. 4. Throughput Analysis Report.

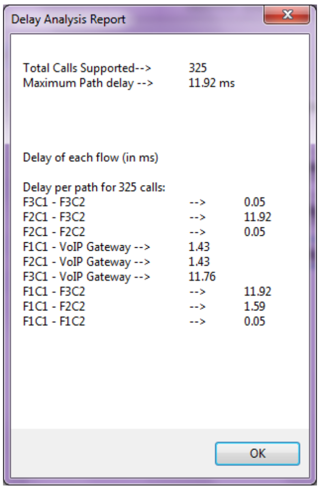

Fig. 5. Delay Analysis Report.

Based on the two reports, it can be concluded that the network can support up to 325 VoIP calls with end-to-end (mouth-to-ear) delay of 11.92 ms. Even though 30% of bandwidth has been allocated for multimedia traffic including voice traffic, 325 calls are accepted considering any other multimedia service for example video conferencing that share the bandwidth with voice traffic. Unless, InfoTech decides that 30% of bandwidth solely meant for voice only. Then, more number of VoIP call can be supported.

According to future plan of InfoTech, more bandwidth upgrade will come in the future as this public university to support teaching and learning process. The upgrade gives benefits to voice traffic as more VoIP call can be supported based on 30% bandwidth allocation for multimedia traffic including voice. In the meantime, UiTM able to reduce the cost of traditional phone paid to the telecommunication provider.

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