Measurement of V2oIP over Wide Area Network between Countries Using Soft Phone and USB Phone

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Abstract: *In this research, we propose an architectural solution to integrate the video voice over IP services in campus environment network and tested over LAN and WAN. Voice over IP technology has become a discussion issue for this time being. Today, the deployment of this technology on an organization truly can give a great financial benefit over traditional telephony. Therefore, this study is to analyze the V2oIP and investigate the performance areas evolved with the quality of service delivered by video phone system over LAN and WAN. This study focuses on quality of voice and video, packet loss, delay time and jitter through the USB video phone and soft phone over LAN and WAN. In this study, network management system is used to monitor and capture the performance of V2oIP between LAN and WAN. In addition, the most apparent of implementing soft phone and USB video phone platform is to define the best achievement between LAN and WAN that can be used in operational environment. Based on the finding result, it shows that users may experience V2oIP quality service degradations due to delays and losses in WAN compare to LAN.*

Keywords: *V2oIP, jitter, delay, packet loss, CPU, LAN, WAN, and NMS.*

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1. Introduction

As with most new technologies, Voice over Internet Protocol (VoIP) brings new challenges along with the benefits. The main challenge is VoIP's extreme sensitivity to delay and packet loss compared with other network applications such as web and e-mail services. A basic understanding of VoIP traffic and of the quality metrics provided by VoIP monitoring tools will help to keep VoIP network running smoothly. Existing hardware video phones have integrated video camera and microphone and can be used independently of any other equipment [1]. Traditionally, set top boxes are video broadcast receivers and are equipped with video/audio decoder (MPEG2) and usually based on H.263 codec for video and G.711, G.723 or G.729 codecs for voice [1]. The actual multimedia content (voice and video) is usually transmitted by means of the Real-time Transport Protocol (RTP) [2].

The objective of this study is to analyze and compare the performance of the V2oIP over USB Video Phone and soft phone between LAN and WAN. In addition, it is to develop V2oIP services environment in campus. This study focuses on quality of voice and video such as packet loss, delay time, CPU usage and jitter between LAN and WAN. Today, many people are talked and communicated face to face using USB Video Phone system. When VoIP is implemented using the public Internet, users may experience quality degradations due to dynamic delays and losses in the LAN and WAN. Packets may be lost, either in isolation or in batches, and may experience sudden delay increases [8]. Figures 1 and 2 show USB video Phone system and soft phone application will use in real network environment in LAN and WAN for the experimental. Based on previous work, there is no such study has been conducted on comparison of V2oIP performance over WAN between international countries using soft phone and USB phone.

Figure 1. USB Video phone system.

Figure 2. Soft phone application.

2. Related Works

Recently, VoIP is rapidly growing and becoming a mainstream telecommunication services, not only because of the lower cost compared with traditional PSTN (Public Switched Telephone Network), but also its convergence technologies of data and voice communication [6]. VoIP applications like Skype have also achieved great success [9]. However, due to the complexity of the Internet, it is unpractical to calculate VoIP performance metrics only through the mathematical modeling, as what was done in the telephone networks, so the performance evaluation of VoIP requires actual measurement activities. There have been numerous studies on VoIP measurement. For examples, most work focused on monitoring and analyzing performance of actual applications, like MSN and Skype [4, 5, 7]. Skype using more to video codec like H.263 and H.261. Skype also using the audio codec likes G.711, G.729, G.723, G.728 and GSM.

3. Methodologies

Figure 3 shows the overall framework of the V2oIP services in campus environment. There are five phases development process such as:

- Planning and research.
- Development.
- Implementation.
- Testing.
- Documentations.

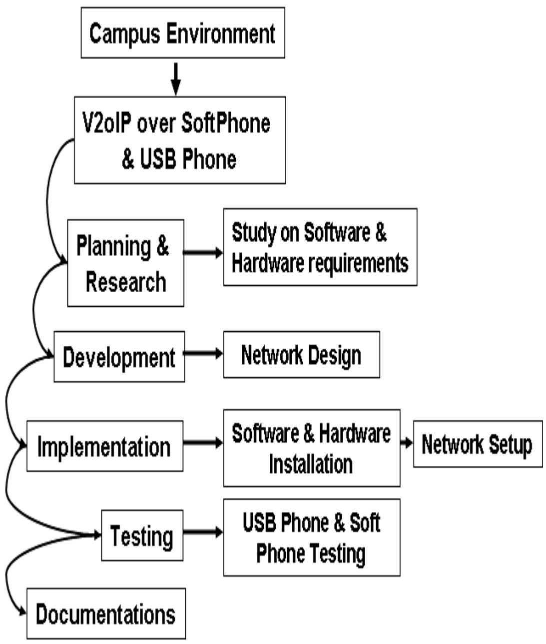

Figure 3. Framework of V2oIP services in campus environment development.

Figure 4 shows the overall framework of the V2oIP performance analysis. In the experiment, the performance analysis will focus on jitter, delay, packet loss and CPU usage over LAN. Network management system such as VQ manager and Colasoft Capsa are used to analyze V2oIP services in campus environment.

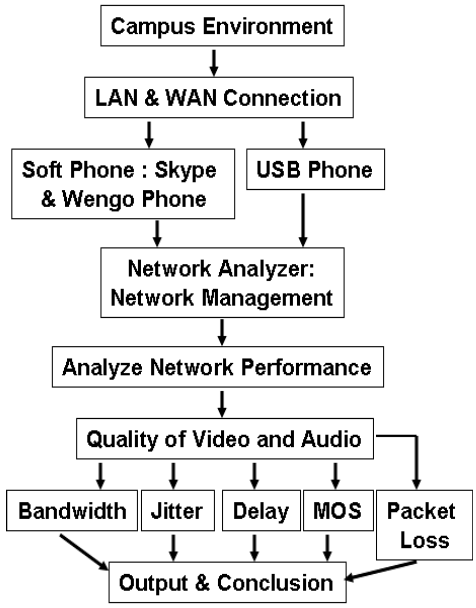

Figure 4. Framework of V2oIP performance analysis.

Figure 5 shows the framework of V2oIP conversation between originate and destination users. This framework shows the V2oIP conversation occurs between two different ISPs. There are six countries have selected to test V2oIP performance as follows: i) Malaysia; ii) Canada; iii) Turkey; iv) Japan; v) Lubnan; and vi) Norway. This experiment is to measure end-to-end performance between USB video Phone and Soft phone over LAN and WAN.

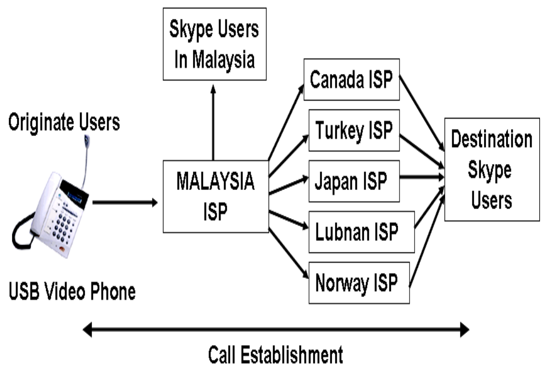

Figure 5. Framework of V2oIP over WAN between countries.

4. Proposed V2oIP Network Architecture in Campus Environment

We have setup a real network environment (LAN and WAN) to analyze and measure implementation of V2oIP services at University of Kuala Lumpur (UniKL) in Malaysia. This study posits several research questions: what is the performance level of the V2oIP over LAN and WAN; and is the analysis for evaluating and measuring V2oIP performance effective over LAN and WAN? Figure 6 shows the V2oIP architecture in real network over LAN and WAN. V2oIP quality can be monitored periodically through the measurement using VQnet (VoIP) management and colasoft capsa tools to gather quality variation information, avoiding the ignorance of unacceptable VoIP quality caused by the network failure or bandwidth bottleneck as shown in Figure 7.

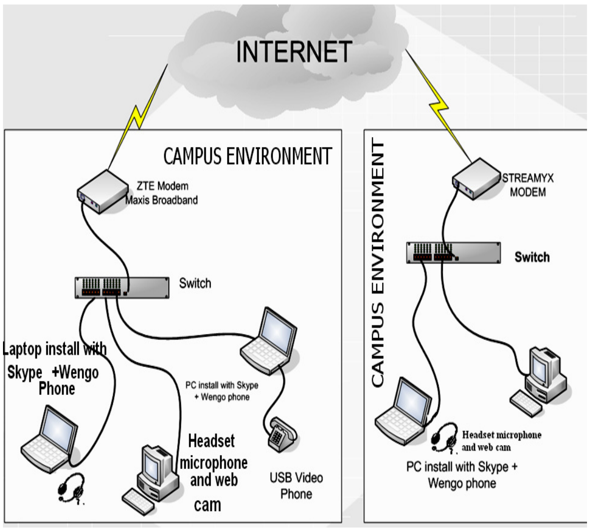

Figure 6. Development of V2oIP architecture in real network over LAN and WAN.

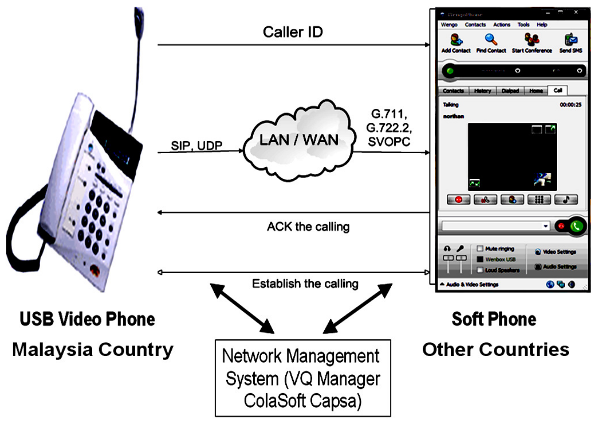

Figure 7. Installation of network monitoring system for V2oIP.

5. Experimental and Analysis Results

This section analyzes, measures and compares V2oIP performance over LAN and WAN. The experiment and traffic V2oIP in LAN are captured based on two situations such as peak hours and non peak hours. The V2oIP traffics are measured on every two minutes. VoIP software is available for use in the Internet, including Skype, Google-Talk, Windows Live Messenger, Yahoo Messenger, and Gizmo Project [3]. Only, Skype can support multi-party conferencing. Result in Figures 8 and 9 show that V2oIP transmission over WAN will contribute higher delay with more network jitter that can easily occur compared to LAN. While, result in Table 1 and Table 2 show the jitter statistics data comparison between countries over V2oIP during conversation between two parties.

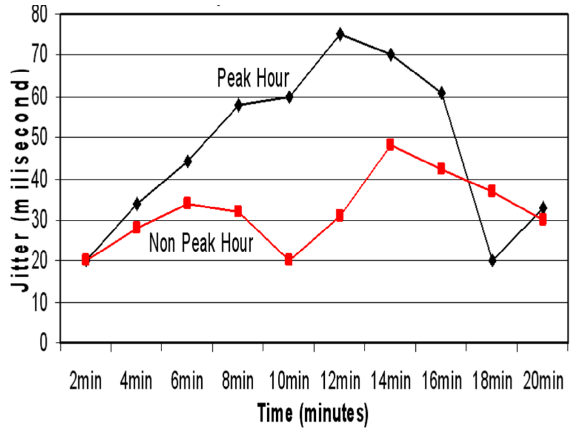

Figure 8. Comparison of jitter over LAN (peak/non- peak hours).

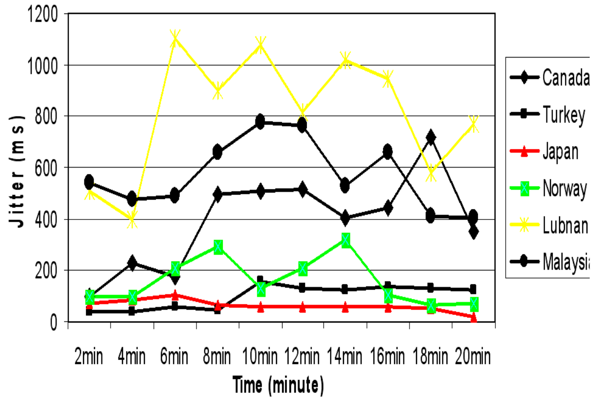

Figure 9. Comparison of jitter over WAN.

Table 1. Jitter occurs in Norway, Lubnan and Malaysia.

Time (minute)	Jitter (millisecond)		
	Norway	Lubnan	Malaysia
2	101	510	544
4	97	397	477
6	209	1103	486
8	291	903	658
10	133	1074	774
12	211	817	765
14	317	1017	526
16	104	946	657
18	67	583	410
20	69	772	407

Table 3 has shown the ITU standard recommendation for delay and jitter interval which are divided in three categories as follows:

- Acceptable.
- Acceptable and fair.
- Unacceptable.

Based on ITU standard measurement, Figure 9 shows that V2oIP over WAN has achieved ‘unacceptable’ and 'fair' level compare to LAN. For V2oIP, it has achieved 'acceptable' level on peak and non-peak hours as shown in Figure 8. The efficiency of V2oIP services can quickly deteriorate when the number of users increases. Jitter on V2oIP over WAN usually come from ISP and PSTN that uses packet switching technology. The reason is because processing delays over WAN is higher than LAN due to the fact that IP-to-PSTN is a fully operating packet and circuit switching.

Table 2. Jitter occurs in Canada, Turkey and Japan.

Time (minute)	Jitter (millisecond)		
	Canada	Turkey	Japan
2	99	40	69
4	227	36	86
6	175	60	102
8	493	45	67
10	506	156	58
12	518	128	60
14	403	125	59
16	443	135	58
18	720	130	55
20	353	127	20

Table 3. ITU Standard recommendation for delay and jitter.

Range in Milliseconds	Description
0-150	Acceptable for most user application
150 - 400	Acceptable and it's impact on the transmission quality of user application
Above 400	Unacceptable for general network planning purposes, however, it is recognized that in some exception cases this limit will be exceeded.

Again, as a result in Figures 10 and 11 have confirmed and proven that V2oIP achieve higher packet loss that can easily occur compared to LAN. The results end-to-end delay, jitter and packet loss increase higher the moment V2oIP traffic pass through ISP and PSTN network compare to LAN. Tables 4 and 5 show the packet loss statistics data comparison between countries over V2oIP during conversation between two parties.

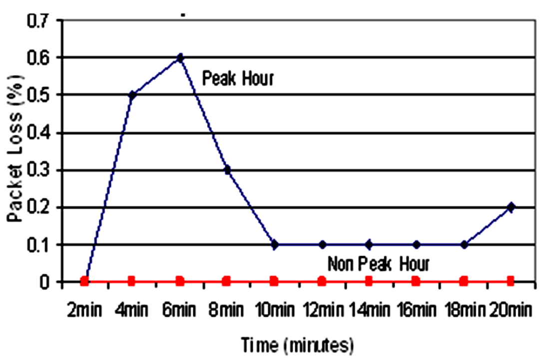

Figure 10. Comparison of packet loss over LAN for voice services.

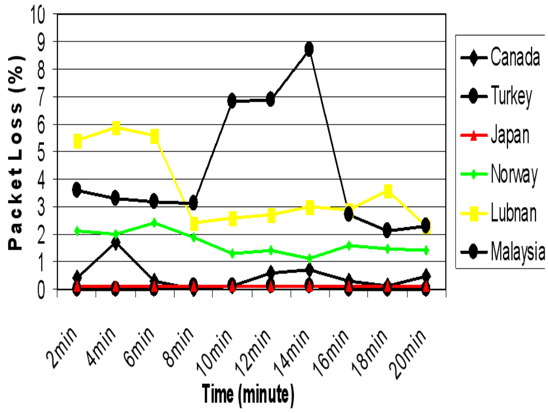

Figure 11. Comparison of packet loss over LAN for video services.

Table 4. Packet loss occurs in Norway, Lubnan and Malaysia countries.

Time (minute)	Packet Loss (%)		
	Norway	Lubnan	Malaysia
2min	2.1	5.4	3.6
4min	2	5.9	3.3
6min	2.4	5.6	3.2
8min	1.9	2.4	3.1
10min	1.3	2.6	6.8
12min	1.4	2.7	6.9
14min	1.1	3	8.7
16min	1.6	2.9	2.7
18min	1.5	3.6	2.1
20min	1.4	2.3	2.3

Table 5. Packet loss occurs in Canada, Turkey and Japan countries.

Time (minute)	Packet Loss (%)		
	Canada	Turkey	Japan
2min	0.4	0	0.1
4min	1.7	0	0.1
6min	0.3	0	0.1
8min	0	0.1	0.1
10min	0.1	0.1	0.1
12min	0.6	0.1	0.1
14min	0.7	0.1	0.1
16min	0.3	0	0.1
18min	0.1	0	0.1
20min	0.5	0	0.1

Figures 12 and 13 show the CPU usage performance that affects V2oIP services especially video transmission. Both medium (LAN and WAN) contribute different performance and characteristics. If the system load is very high, it can achieve higher CPU usage and make the throughput low that can cause inconsistent conversation between two parties. Therefore, it is important to understand how delay, jitter and CPU usage can affect speech quality. In addition, poor link quality network between two parties can also increase the delay and jitter for V2oIP services over WAN as shown in Tables 6 and 7. From the analysis result, it shows that V2oIP over LAN have a good network performance without implementation of any QoS level. The results of this study show that V2oIP is able to contribute and achieve reliability and scalability over LAN. Therefore, proper plan can mitigate most of the negative effects that are related to V2oIP characteristics.

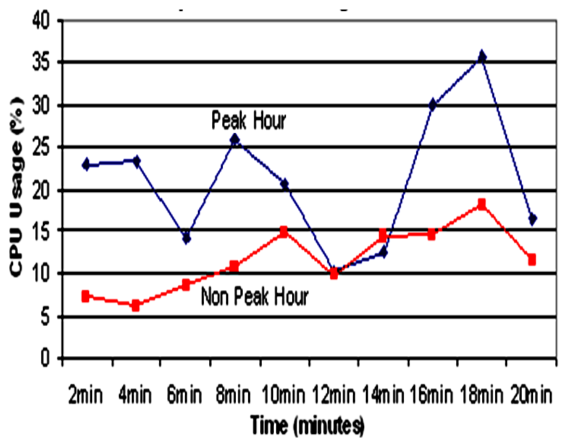

Figure 12. Comparison of CPU usage over LAN for V2oIP services.

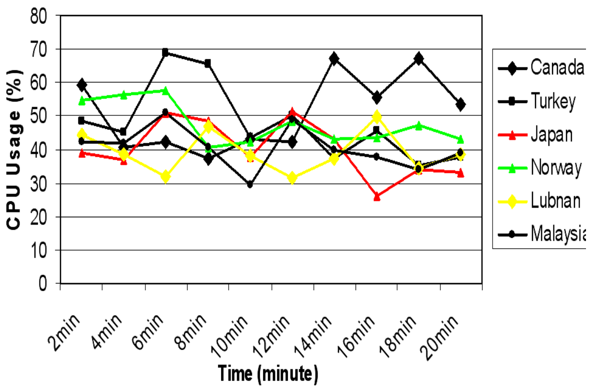

Figure 13. Comparison of CPU usage over WAN for v2oIP services.

We conclude that base on our findings, the result makes a comparison between V2oIP over LAN and WAN and its shows that WAN has achieved higher delay. In addition, packet losses caused by WAN connectivity such as propagation delay, codec delay, serialization delay and dijitter buffer.

Table 6. CPU Usage in Norway, Lubnan and Malaysia Countries

Time (minute)	CPU Usage (%)		
	Norway	Lubnan	Malaysia
2min	54.7	44.5	42.2
4min	56.3	38.5	41.7
6min	57.6	32.1	50.8
8min	40.5	46.9	40.6
10min	42.2	38.3	29.5
12min	48.4	31.3	48.9
14min	43.3	37.5	39.9
16min	43.5	49.7	37.7
18min	47.4	34.6	33.9
20min	43.3	38.6	39.1

Table 7. CPU usage in Canada, Turkey and Japan countries.

Time (minute)	CPU Usage (%)		
	Canada	Turkey	Japan
2min	59.1	48.3	39.1
4min	40.5	45.3	36.9
6min	42.2	68.7	50.8
8min	37.5	65.3	48.4
10min	43.3	43.4	37.9
12min	42.4	50.1	51.6
14min	67.3	37.3	43.3
16min	55.7	45.4	26.2
18min	67.1	35.4	33.8

6. Conclusions

This paper discussed which V2oIP services could produce good performance. We conclude that base on our findings, V2oIP over WAN can contribute higher delay, packet loss and CPU usage compare to LAN. If V2oIP want to implement in campus environment, it is recommended to enable QoS function in order that to achieve a good quality conversation between two parties. By measuring and analyzing jitter, end-to-end delay and packet loss before deployment of new V2oIP services, can aid in the correct redesign and configuration of traffic prioritization in V2oIP services. Mean Opinion Score (MOS) and bandwidth analysis over WAN will consider in future work. It should consider implementing techniques to improve quality of V2oIP over WAN. Therefore, efficient and effective QoS provisioning techniques are important to achieve robust services. There are several techniques should be studied and analyzed that can be used to increase performance of V2oIP in campus environment as follows:

- Dejitter buffer;
- Type of Service (ToS).
- Weighted Fair Queuing (WFQ).
- Random Early Detection (RED).

Implementing quality of service mechanisms on voice and video is a method to improve V2oIP network services performance in campus environment.

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