VOIP PERFORMANCE MEASUREMENT USING QoS PARAMETERS

A.H.Muhamad Amin
IT/IS Department, Universiti Teknologi PETRONAS,
31750 Tronoh, Perak Darul Ridzuan, Malaysia
Correspondence Email: ananghudaya@petronas.com.my

ABSTRACT

This paper presents an analysis of the performance of VoIP. In our study, three different VoIP communication aspects have been considered. These comprise the call signaling protocols, networking environments in which the VoIP communications take place, and the Virtual Private Network (VPN) protocols for securing the data transmissions. The performance evaluation involves the identification of the QoS parameters, which would be relevance to the VoIP communications. These parameters have then been measured on the RTP packets transmission in VoIP communications. The results gathered from the analysis shows that these communication aspects have a significant impact on the QoS in the VoIP communications and the impact varies according to the parameters and the communication aspects selected for the analysis.

Keywords: Voice over IP (VoIP), Quality of Service (QoS), Queuing Mechanisms, Virtual Private Network (VPN).

1. INTRODUCTION

The performance of the Voice over IP (VoIP) protocol is of interest when planning for public access. The technologies, which work well for limited use often, fail to scale-up to the user requirements. According to [1], High-quality VoIP services are required as for the Internet communications to be an alternative towards Public Switched Telephone Network (PSTN). The deployment of VoIP in the Internet network does not promise a good Quality of Service (QoS), since Internet is a kind of best-effort networks [2]. The privacy consideration is also of importance when provisioning voice services on the Internet; particularly from the business use perspective. A full study, which comes up with a definitive set of recommendations would require considerable work over a substantial period of time, however some information on the performance may be obtained by re-enacting the most commonly occurring conditions in the lab to ascertain the sensitivity of the VoIP to its key QoS parameters.

The aim of this paper is to analyze the performance of VoIP communications by observing the QoS parameters variation with respect to the some of the pertinent communication aspects. The aspect chosen in this regard comprise the call signaling protocols, the networking environments, and VPN protocols. Some similar studies [1, 3 and 18] have also been used as a comparative measurement towards the results obtained from this research.

This paper is divided into five sections. The first section is basically the introduction towards the topic of the research. Section 2 provides a brief description on the research methodology that has been undertaken. Section 3 focuses on the communication aspects that have been taken into consideration. Section 4 entails the experimentation process conducted. Section 5 provides the results from the research that has been undertaken.

2. RESEARCH OVERVIEW

As been mentioned earlier, this study relates the VoIP Performance towards the three communication aspects: call signaling protocols, network environment, and security. Figure 1 illustrates the overview of the research framework that has been carried out.

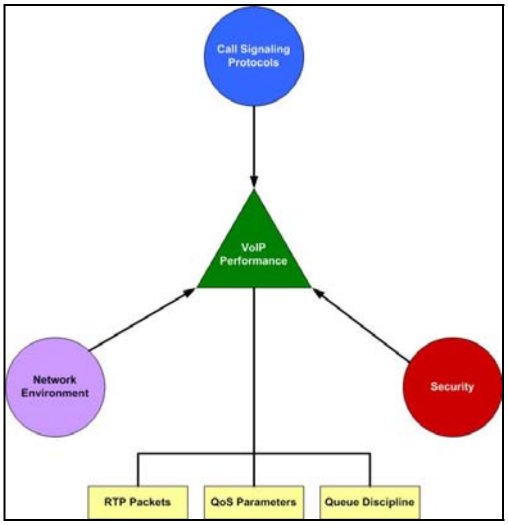

Figure 1: Overall Research Framework

It should be noted that the study is focused on Real-Time Transmission Protocol (RTP) packets transmission in order to observe the QoS for VoIP communications. The main reason for this is such that the study being undertaken is concerned with the network conditions for VoIP communication. Hence, the observation at network packet level would be considered as a better mean to achieve the objective of this study. It should also be noted that the limitation of this study is such that it does not consider the variable types of speech codec being used. For the purpose of this study, the codec used was ITU-T G.711. This codec uses 64kbps channel and based on the Pulse Code Modulation (PCM) scheme, operating at 8KHz sample rate, with 8 bits per sample.

2.1 QoS Parameters

In ensuring the QoS for VoIP, researchers have come out with some QoS parameters as to be observed. In general, the focus of this paper is directed to the Network QoS, based on the QoS Framework as suggested by [4]. This simply means that the analysis that has been taken out is to evaluate the network environment in which VoIP communication is being conducted. There are several QoS parameters that have been identified to be implemented in this study. Table 1 provides a list of Network QoS parameters available as derived from [7].

Table 1: Network QoS Parameters.

Category	*Parameters*
Timeliness	Delay Response time Jitter
Bandwidth	Systems-level data rate Application-level data rate Transaction time
Reliability	Mean time to failure (MTTF) Mean time to repair (MTTR) Mean time between failures (MTBF) Percentage of time available Packet loss rate Bit error rate

In our analysis, three QoS parameters have been selected, namely delay, jitter, and packet loss rate. The main justification for this selection is such that the study focuses on the performance of VoIP communication over different networking conditions and environment, and hence the time and reliability would be the major concerns for evaluation. The results of studies conducted by [1] also shown that high delay and high delay variability (jitter) has been experienced by a large number of Internet paths that resulting in poor VoIP performance. It should be noted that these QoS parameters have also been adopted in a number of previous studies [8], [9], and [10].

2.2 Related Works

There are several related works that have also being carried out by other researchers in this field including [1, 3, 11, 13, 14, 15, and 16]. Observation on the QoS for VoIP over the Internet has been done by [1], while [13] focuses on the delay patterns in VoIP network using simulation method. There are also some other initiatives in monitoring and measuring QoS of VoIP in wide area network using pattern methods as in [18]. These works have been a motivation for the authors. Generally, the authors have adopted lab experimentation as a mean to study the VoIP performance. This is simply due to the fact that it is more feasible to control the environment in the lab, as compared to real-life experimentation, as well as simulation. However, it should be noted that sometimes lab experimentation could not provide the real results, but just as approximations.

3. COMMUNICATION ASPECTS

In our analysis, three communication aspects have been considered. These include the VoIP call signaling protocols, the network environment, as well as the security for VoIP communication.

3.1 VoIP Call Signaling Protocols

For this research, the author has selected two of the most commonly used call signaling protocols, namely H.323 and SIP. H.323 is a call signaling protocol introduced by International Telecommunications Union (ITU) [7]. On the other hand, SIP was developed by Internet Engineering Task Force (IETF) as an alternative to H.323 [7].

3.2 Network Environment

Two different network environments have been adopted in this study. These include Ethernet network and wireless LAN (WLAN) network. Both networks provide the best-effort mechanism in which there are no differentiated services for specific types of data being transmitted. Therefore, it is best suited to observe the VoIP performance under this condition. However, queuing mechanism have been adopted in order to simulate the real environment in which RTP packets would undergo queuing process between gateways and routers.

3.3 Security for VoIP Communication

In ensuring the security of VoIP communication, the authors have adopted the usage of VPN as the security approach. VPN has arisen from the needs to ensure the privacy of communications among enterprises, using the existing Internet infrastructure as the medium of communication. Two different VPN protocols have been implemented in this regard, namely PPTP and IPSec.

4. EXPERIMENTATION

In analyzing the QoS performance of VoIP communications, the research has taken into account two distinct cases. These are the secure (with VPN) and the non-secure VoIP communications. The VoIP communications were implemented with both the call signaling protocols (i.e. H.323 and SIP) for the tests. A queuing mechanism was adopted for simulating the behavior of a real network. The results from the experiments undertaken included the values of delay, jitter, and packet loss obtained from each of the VoIP implementations. The following sections provide a brief description of the set ups used for the experiments.

4.1 Test Environment

Essentially, any research involved with a networking environment should be able to portray the 'real' traffic conditions. Usually, the real traffic conditions are simulated within the software or by using a test environment in the laboratory. Studies conducted by [1] and [11] use the real traffic conditions. On the other hand, this study makes use of the later option as a replacement for the real environment. A number of different applications usually share the same network bandwidth. However for this study, only a single VoIP application (SJLABS™ SJPhone™) was configured for utilizing the entire bandwidth. This was done in order to measure the QoS of VoIP with respect to the changes made to the underlying networking infrastructure and the transport protocol without any interference from other sources. A queue management mechanism was later introduced to simulate a real networking environment. In this paper we refer the former as the 'ideal network environment' and to the later as the 'non-ideal network' environment.

4.2 The Test Cases

The following cases were studied to evaluate the performance of VoIP with respect to the selected QoS parameters, the network environments, and the VPN protocols.

4.2.1 Network-to-Network

Two different network topologies were implemented for this purpose. Figure 2 provides an overview of the topology used for a wired-LAN environment.

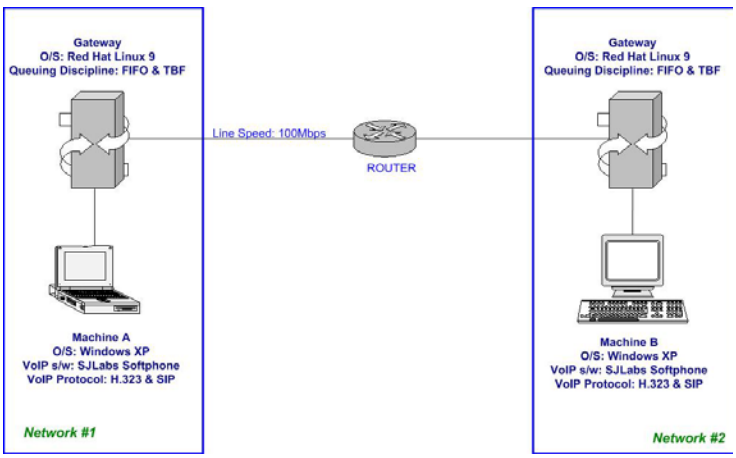

Figure 2: Network-to-Network VoIP communications using the Ethernet environment.

The second topology included a wireless-LAN as shown in Figure 3.

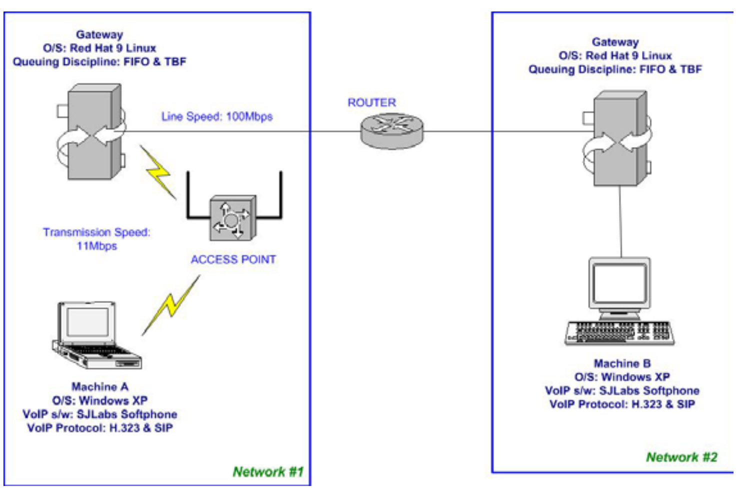

Figure 3: Network-to-Network VoIP communications using the Wireless-LAN.

4.2.2 Secure Network-to-Network

The security for the VoIP communications was implemented using two different VPN protocols. These protocols were implemented between the gateways as shown in Figure 3. The setup of the

VPN for the wireless-LAN was also very similar to the wired-LAN setup and hence it has not been reproduced in the paper.

5. RESULTS

In overall perspective, usually the assessment of VoIP is carried out using the subjective quality measure called Mean Opinion Score (MOS) [1]. MOS scales from 1 (lowest quality) to 5 (highest quality). However, this type of measurement focuses on the perceived quality provided by users. MOS is useful when considering the overall end-to-end quality of communications. In this research, the assessment using MOS was not being adopted, since the focus is towards measuring the performance of VoIP communications with regards to the network environment.

Based on the experiments, the delay and jitter values have been organized in the form of histogram charts, representing the number of packets versus the delay or jitter values. This shows the number of RTP packets experiencing the same jitter values or delay. It should be noted that these values do not represent any real progression. Data from this study (delay, jitter values, and packet loss rate) has been collected using ETHEREAL network monitoring software. The delay values are obtained by calculating the difference between the RTP packet actual arrival time and the estimated arrival time. On the other hand, the jitter values have been derived from the differences in the inter-arrival time of the RTP packets. The packet loss values are represented in the percentage form of the total RTP packets being transmitted.

5.1 Ideal Network Environment

For the ideal Ethernet LAN Environment, the default queue algorithm used was the First-in-First-Out (FIFO). Based on the results obtained, it demonstrates that Both PPTP and IPSec incur a higher delay as compared to normal VoIP communication within the ideal network. This is true for both H.323 and SIP based VoIP communications.

On the other hand, the jitter values for SIP and H.323 VoIP communications remained generally similar to each other. The packet loss was not considered as a factor within the ideal and ideal-secure environments owing to a very low bit error rate of the lab network.

5.2 Non-Ideal Ethernet Network Environment

The ideal network was then converted into a non-ideal one; with congestion and delays. In this regard an advanced queue discipline was implemented in the gateways. Token Bucket Filter (TBF) was used for the queue management. The advantage of using TBF queue discipline is such that it provides the condition where packets would be heavily queued at one end and will be burst out and some of the packets might be dropped due to the limitation of the bucket size. Table 2 shows the parameter values that have been implemented in the TBF queue.

Table 2: TBF Parameters that have been adopted.

Parameters	*Description*	*Values selected*
Bucket/Burst Size	This is the size of the bucket. Indirectly, it is also considered as the burst size, since the queue will be burst when the bucket is full.	*1024 Kbytes*
Latency	The amount of time in which a particular packet is allowed to reside in the TBF bucket.	*100 ms*
Rate	Rate of the arrival for tokens.	*50 kbps*

According to [17], traffic shaping (as being conducted in this research) could also be used as a mean to reduce the impacts of interfering bursts on network performance. As one would expect, both SIP and H.323 incurred higher delays in non-ideal secure network-to-network (more so under the IPSec) environment, with respect to the maximum RTP packets distribution. Figure 4 and 5 show the average delay values for both SIP and H.323 VoIP communications.

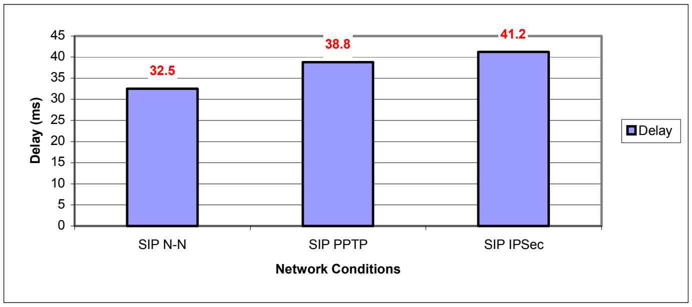

Figure 4: Comparison of Average Delays for RTP Packets transmission in SIP VoIP Communications (Non-Ideal Network-to-Network).

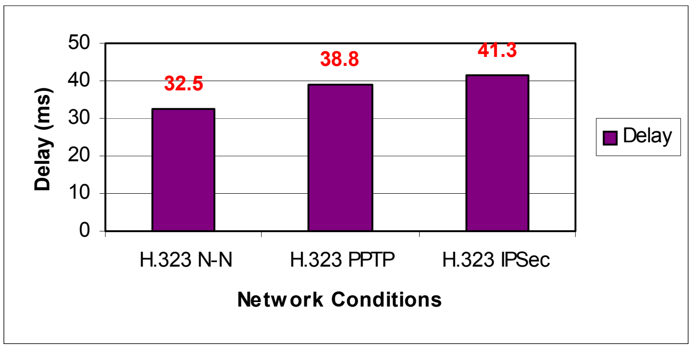

Figure 5: Comparison of Average Delays for RTP Packets transmission in H.323 VoIP Communications (Non-Ideal Network-to-Network).

It should be noted that the results from the delay analysis shows lower values as compared to the assessment done by [1] over the real Internet backbones. This is considered as acceptable, since the analysis that has been carried out was done in a controlled lab environment, with only one single VoIP communication being conducted. According to [14], the delay values between 100 – 150 ms and above are detectable by humans and can impair the interactivity of the conversations. The results obtained are far less than those values. However, the analysis has been done in the controlled environment, unlike the real environment where the impact of delay and jitter are more severe.

In analyzing the jitter values, the overall results obtained show that SIP exhibits higher jitter values as compared to H.323 VoIP communications in all the cases. Figure 6 illustrates the comparison made between H.323 and SIP based VoIP communications in a non-ideal networking environment.

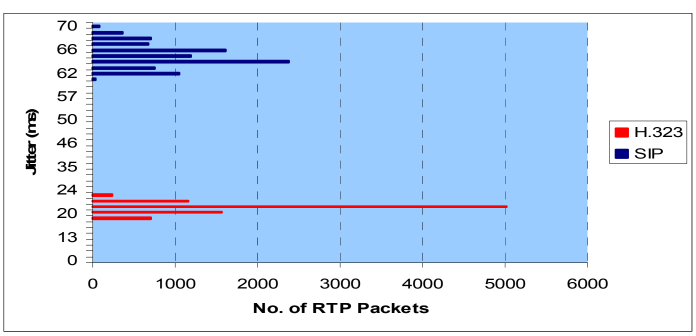

Figure 6: Comparison of Jitter Values for RTP Packets between H.323 and SIP VoIP Communications (Non-Ideal Network-to-Network).

The introduction of IPSec and PPTP increased the jitter values for both H.323 and SIP based VoIP communications. In this study, IPSec produced the highest jitter values for both H.323 and SIP communications. Figure 7 shows the average jitter values for both H.323 and SIP.

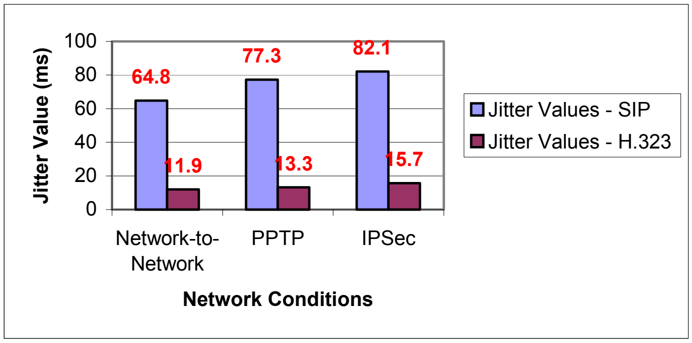

Figure 7: Comparison of Average Jitter Values for RTP Packets of SIP and H.323 VoIP Communications for Non-Ideal Secure Ethernet Environment.

In relation to the packet loss analysis, the non-ideal and non-ideal secure network-to-network environments consistently produced relatively high packet loss rates. Packet loss rate is calculated by determining the number of RTP packets that are lost (unreachable to the destination) over the number of RTP packets being transmitted. Table 3 provides the values of packet loss (in terms of percentages) for all the cases examined for the non-ideal environment. It is also best noted that the use of IPSec and PPTP further increased the packet loss rate in both Ethernet and WLAN environment.

Table 3: Comparisons of Packet Loss Rates for Non-Ideal Ethernet and WLANEnvironment.

Cases	Packet Loss Rate (% of Total Packet) Ethernet	Packet Loss Rate (% of Total Packet) WLAN
Non-Ideal SIP	38.4	38.4
Non-Ideal H.323	39.0	38.4
Non-Ideal Secure (PPTP) SIP	48.4	50.2
Non-Ideal Secure (PPTP) H.323	48.4	50.9
Non-Ideal Secure (IPSec) SIP	51.4	51.4
Non-Ideal Secure (IPSec) H.323	51.5	51.5

With regards to the results shown by [1], the percentages of packet loss rate shown in Table 2 correlates with low MOS score (around 2 -3). This simply means that the quality that will be perceived by the users is low. In addition, based on a comparative study done by [18], it shown almost equivalent results for non-ideal scenarios (both using SIP and H.323). The results for non-ideal secure scenarios show significant increases of about 10% of the total packets sent.

5.3 Non-Ideal Wireless LAN Environment

Both SIP and H.323 offered almost similar delay distributions for the wireless-LAN environment. However, SIP was noted to incur higher jitter values in all the cases. Figure 8 shows the jitter values distributions for both H.323 and SIP based VoIP communications in this regard.

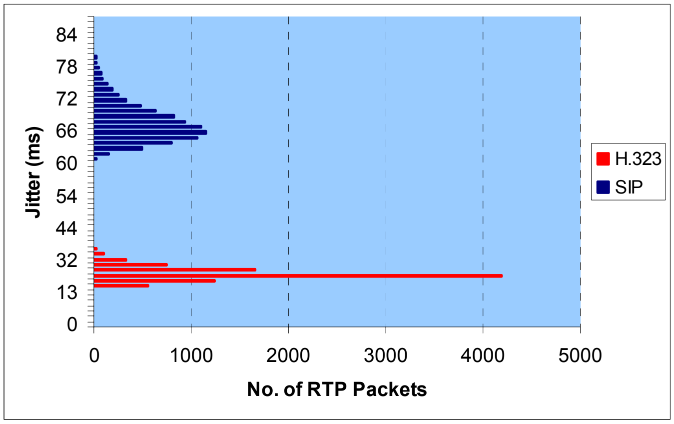

Figure 8: Comparison of Jitter Values for RTP Packets between H.323 and SIP VoIP Communications (Non-Ideal Wireless LAN Network-to-Network).

An overview of the jitter and delay variations is shown in Figures 9, 10, and 11 for SIP and H.323 using the VPN and the insecure networking environments. IPSec-based VoIP communications generally incurred the highest jitter values in all the cases.

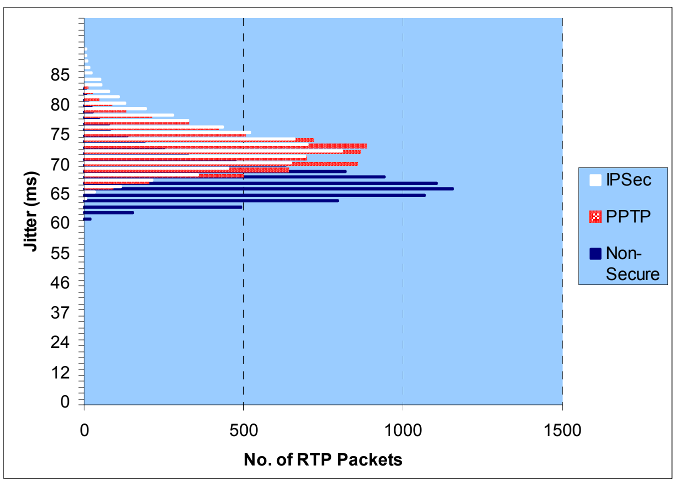

Figure 9: Comparison of Jitter Values for RTP Packets of SIP VoIP Communications for Non-Ideal Secure Wireless-AN Environment.

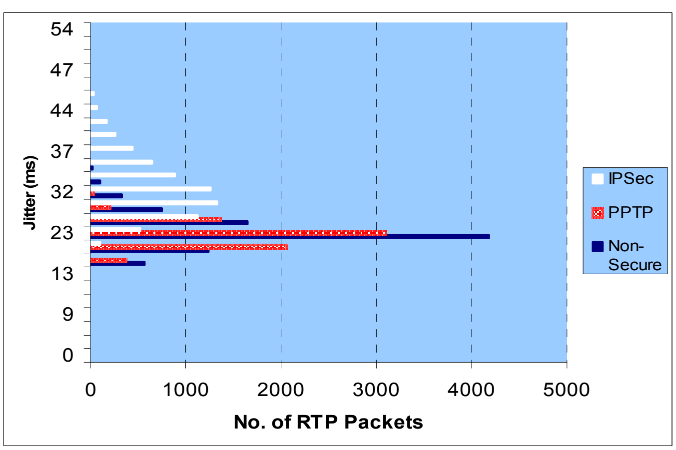

Figure 10: Comparison of Jitter Values for RTP Packets of H.323 VoIP Communications for Non-Ideal Secure Wireless-LAN Environment.

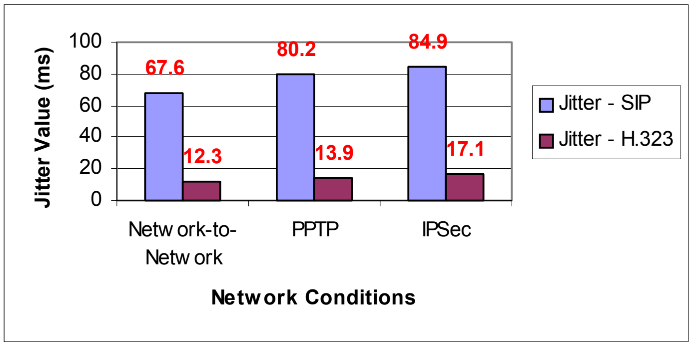

Figure 11: Comparison of Average Jitter Values for RTP Packets of SIP and H.323 VoIP Communications for Non-Ideal Secure Wireless-LAN Environment.

From an overall perspective, the results illustrate significant differences in QoS between H.323 and SIP based VoIP communications for different networking environments.

CONCLUSIONS

The study shows that the call signaling protocol used, the networking environment being implemented, and the VPN affect the performance of the QoS parameters such as the jitter and the delay. It's also noted that in general, the QoS of VoIP communications is decreased with the introduction of network congestion owing to the relatively higher packet loss. The VPN protocols also cause a tangible degradation in the QoS of the VoIP communications.

As one would expect, the best values for the QoS parameters were achieved within the network with no congestion. In the non-ideal environment with the queuing discipline, these parameters showed a significantly high rate of deterioration, especially in the case of the packet loss rate for the RTP. The results obtained for the non-ideal environment are of greater significance since the real-world deployment of VoIP is unlikely to be completely free of the network congestion.

It may be concluded from the results that the call signaling protocols including H.323 and SIP produce dissimilar jitters. In this context, H.323 offers significantly lower jitter values for RTP packets as compared to SIP. However, both SIP and H.323 are not significantly apart with respect to the delay and the packet losses.

The implementation of the VPN protocols significantly affects the QoS of VoIP communication. All the three QoS parameters examined shows deterioration for the RTP packets transmission; more so in the case of the non-ideal environment with an IPSec implementation.

Finally, it should also be noted that VoIP performance in both wired and wireless LAN experienced the increased in delay and jitter values when VPN is being implemented.

For future works, further analysis using other optimization methods proposed by [16] would be a good point of start, where the analysis should also consider the implementation of VPN over VoIP communications. In addition, analysis using Mobile IP, IPv6 and other security protocols in VoIP communications would be of interest. Also of interest would be the use of IEEE 802.11(a) QoS-based wireless-LAN standard for the VoIP communications.

REFERENCES

[1] Markopoulou A.P., Tobagi, F.A., and Karam, M.J., "Assessing the Quality of Voice communications over Internet Backbones", IEEE/ACM Transactions on Networking, Vol. 11, No. 5, October 2003.

[2] Kobayashi, K. and Katayama, T., "Analysis and Evaluation of Packet Delay Variance in the Internet", IEICE Transactions on Communications, Vol. E85-B, January 2002.

[3] Hole, D.P. and Tobagi F.A., Capacity of an IEEE802.11b Wireless LAN Supporting VoIP, Proceedings of the IEEE International Conference on Communications (ICC) 2004.

[4] Jha, S., and Hassan, M. "Engineering Internet QoS". Artech House. ISBN: 1580533418.2002.

[5] Das, S.K., Lee, E., Basu, K., and Sen, S.K. "Performance Optimization of VoIP Calls over Wireless Links Using H.323 Protocol". IEEE Transactions on Computers, Volume 52, No. 6. June 2003.

[6] Schulzrinne, H., Casner, S., Frederick, R., and Jacobson, V. "RTP – A Transport Protocol for Real-Time Applications". IETF RFC 1889. January 1996.

[7] Goode, B. "Voice Over Internet Protocol (VoIP)". Proceedings of The IEEE. Volume 90. No. 9. September 2002.

[8] Kos, A., Klepec, B., and Tomažič, S. "Techniques for Performance Improvement of VoIP Applications". IEEE MELECON 2002. Cairo, Egypt. May 2002.

[9] Feigin, J., Pahlavan, K., and Yliantilla, M. "Hardware-Fitted Modeling and Simulation of VoIP over a Wireless LAN". IEEE, VTC'2000, Boston, September. 2000. IEEE Publications. 2000.

[10] Shirdokar, R., Kabara, J., and Krishnamurthy, P. "A QoS-based Indoor Wireless Data Network Design for VoIP Applications". Proceedings of IEEE Vehicular Technology Conference. September 2001. IEEE Publications. 2001.

[11] Karam, M.J. and Tobagi, F.A. "Analysis of the Delay and Jitter of Voice Traffic over the Internet". Proceedings of Infocom 2001.

[12] Rosenberg, J., Schulzrinne, H., Camarillo, Johnston, A., Peterson, J., Sparks, R., Handley, M., and Schooler, E. "SIP: Session Initiation Protocol". IETF RFC 3261. June 2002.

[13] Han, J.S., Ahn, S.J., and Chung, J.W. "Study of Delay Patterns of Weighted Voice Traffic of End-to-End Users on the VoIP Network", Int. J. Network Management 2002.

[14] Amir, Y., Danilov, C., Hedqvist, D., and Terzis, A., "1-800 OVERLAYS: Using Overlay Networks to Improve VoIP Quality", Technical Report CNDS-2004-2, August 2004.

[15] Bearden, M., Denby, L., Karaçah, B., Meloche, J., and Stott, T., "Experiences with Evaluating Network QoS for IP Telephony", 10th International Workshop on Quality of Service (IWQoS 2002), USA, May 2002.

[16] Gardner M.T., Frost, V.S., and Petr, D.W., "Using Optimization to Achieve Efficient Quality of Service in Voice over IP Networks", IPCCC 2003 – The 22nd International Performance, Computing, and Communications Conference, Phoenix, Arizona, April 2003.

[17] Fidler, M., "On the Impacts of Traffic Shaping on End-to-End Delay Bounds in Aggregate Scheduling Networks", Springer, LNCS 2811, Proceedings of COST 263 QoFIS, pp. 1-10, October 2003.

[18] Miloucheva, I., Nassri, A., and Anzaloni, A., "Automated Analysis of Network QoS Parameters for Voice over IP Applications", D41 – 2nd Inter-Domain Performance and Simulation Workshop (IPS 2004).