Performance Analysis on the Effect of G.729, Speex and GSM Speech Codec on 802.11g Wireless Local Area Network over VoIP using Packet Jitter

Imran Edzereiq Kamarudin[1], Syahrizal Azmir Md Sharif[1] and Tutut Herawan[2]

[1]Faculty of Computer System and Software Engineering
Universiti Malaysia Pahang
Lebuh Raya Tun Razak, Gambang 26300, Kuantan, Pahang, Malaysia
[2]Department of Mathematics Education
Universitas Ahmad Dahlan
Jalan Prof Dr Soepomo 55166, Yogyakarta, Indonesia

{edzereiq,syazmir}@ump.edu.my, tutut81@uad.ac.id

Abstract

In this paper, three speech codecs; G.729 (8 kbps), Speex (8kbps) and GSM (13kbps) were tested together with several predetermined SNR value ranging from 10dB to 45dB with a sample of 8 second speech. VoIP QoS such as packet jitter is analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11g environment. Result shows that at lower SNR, GSM achieve higher packet jitter than that G.729 and Speex. At higher SNR, GSM achieves lower packet jitter as compared than that G.729 and Speex.

Keywords: *Speech codec;G.729; Speex; GSM; VoIP; 802.11g; WLAN, Packet Jitter*

1. Introduction

Voice over IP or VoIP is a term used in IP telephony for a set of facilities that use the Internet Protocol (IP) to deliver voice information. It all started when in February of 1995 by a small company called Vocaltec Inc. [1]. One of the main reason why VoIP became so popular and slowly but surely replacing the traditional public switch telephone network (PSTN) is when PSTN line is being used, we typically pay for time used to a PSTN line company, in other words the more time we stay at phone and more we need to pay. During the "carry over" session, codec is being used. This is the method of how the "audio" data is placed within the UDP datagram. Information about what codec to use is between the systems and is negotiated during the call setup. Some codecs use compression, while others do not. Some standard voice codecs available are ADPCM (Adaptive Differential Pulse Code Modulation), G.711 (A-Law and μ-Law), G.723.1 (pass through), G.729, GSM, iLBC, Linear, LPC-10, and Speex. Today, the installed-based of Wi-Fi client devices exceeds 200 million worldwide in 2007 [2]. IEEE 802.11g offer adequate capacity for supporting Voice over WLAN (VoWLAN) applications. Research have shown that various speech codec gives different speech quality [4-8]. Speech codec which require higher bandwidth have a better quality voice compare to speech codec which require lower bandwidth. Hence, by increasing SNR and better data rate, it gives a better voice quality with respect to any speech codecs available. A key determinant in voice quality is the speech codec, but wireless network performance will have a substantial impact as well.

Degradation of voice quality caused by network performance or SNR value is still one of the technical barriers of VoIP over WLAN.

In this paper, three speech codecs; G.729 (8 kbps), Speex (8kbps) and GSM (13kbps) were tested together with several predetermined SNR value ranging from 10dB to 45dB with a sample of 8 second speech. VoIP QoS such as packet jitter is analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11g environment. Eight values of SNR have been chosen for the analysis on wireless performance. The SNR values are 10dB, 15dB, 20dB, 25dB, 30dB, 35dB, 40dB, and 45dB.

In summary, the contribution of this work is summarized as follow.

We evaluate the performance of speech quality for G.729, Speex and GSM speech codec by gathering data based on a VoIP QoS parameter i.e. packet jitter.

We evaluate and compare the performance of speech quality for G.729, Speex and GSM speech codec on various network performance based on pre define SNR values based on a VoIP QoS parameter, *i.e.*, packet jitter.

The rest of this paper is organized as follow. Chapter 2 explains the experimental design while Chapter 3 discusses all the findings, analysis and discussion. Finally, the conclusion of this work is described in Chapter 4.

2. Experimental Design

The experimental setup was made to emulate real VoIP over single wireless local area network 802.11g standard. Based on the methodology in design phase, it is necessary to ensure that the hardware and software used will bring significant value to the research. As in Figure 1, in our experimental setup we used an infrastructure based WLAN.

2.1. Experimental parameters and procedures

In our experimental setup, we used an infrastructure based WLAN where all nodes are able to communicate with each other by communication through the access point. Figure 1 shows the infrastructure configuration, the installed software and configured IP address. All the experiment and measurements were conducted in a close office environment to isolate and avoid any other 802.11g wireless signal being transmitted around the test area that can affect the test result.

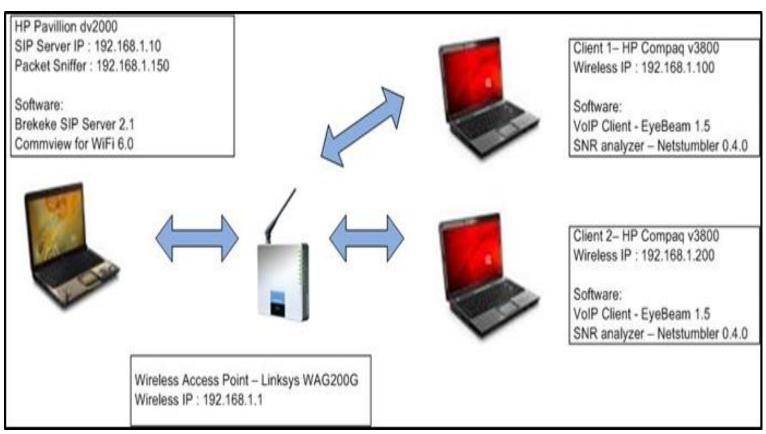

Figure 1. Overall Network Design

The wireless access point was configured with IP address 192.168.1.1. VoIP client 1, VoIP client 2 and SIP server wireless connection also be configured to be associated with the wireless access point. The encryption and handshaking, medium reservation, MAC, encryption for wireless interface, 802.1 x security modes, RADIUS MAC and all power-save features of the wireless cards and wireless access point were disabled during the experiment procedure [11]. Several precautions were taken to ensure the accuracy and precision of experiment results.

1. Measurements were obtained while no other device operating in the 2.4 GHz band was functioning
2. The experimental network was isolated from any other network and the Internet to prevent any application from automatically connecting to the Internet and skewing the measurements.
3. When the laptop was moved to a new location, measurements were not taken immediately; we waited one or two minutes before taking measurements to allows the laptop wireless adaptor to adjust to an appropriate transmission rate (data rate). This was especially important when the SNR at the new location is below 25 dB.
4. To avoid any interference from any wireless access point in which might affected the measurements an isolated channel have been selected.

For the SIP Server, IP address 192.168.10 is used to associate the SIP Server to the wireless network while the IP address 192.168.1.150 is being configured as passive mode dedicated for wireless network monitoring and analyzing. The packet analyzer requires a dedicated wireless network card to run in passive mode as an agent to capture wireless packets. This explains why we need to separate wireless card for the SIP Server. As in [11], all measurements were taken during weekend to minimize the effect of human into the wireless channel. During the experiment to avoid any application from automatically connecting to the Internet and skewing the measurements the experimental network was isolated from any other network [11]. For the purpose of VoIP analysis and testing purpose, we are using one speech files with duration of 8 seconds and containing segments from a male and female speaker were used [11, 12]. Based on ITU-T P.800, Series P: Methods for objective and subjective assessment of quality, the following words of conversation between a male and female speaker were recorded [13]. Figure 2 describes an example of speech material.

You will have to be very quiet.

There was nothing to be seen.

They worshipped wooden idols.

I want a minute with the inspector.

Did he need any money?

Figure 2. Example of Speech Material [13]

In testing the conversation between two clients, we are using the conversational speech model as specified in ITU-T P.59 recommendation [9, 11, 14]. The conversation between two clients is modeled as a four state of Markov:-

1. ‘Client 1’ talking and ‘Client 2’ silent – Markov 1

2. ‘Client 1’ silent and ‘Client 2’ talking – Markov 2
3. Both talking – Markov 3
4. Both silent

However, in our study we assume that voice packets are generated only when a user is in the talking state. In other words, we make the simplifying assumption of silence suppression where no voice packets are generated when a user is silent [11]. Thus, in our experiment procedure, we are only focus on three states excluding the ‘both silent’ state. One of the objectives of the research is to evaluate the speech quality of respective VoWLAN speech codec namely G.729, Speex and GSM by gathering data based on QoS parameter such as packet jitter. Thus, our analysis will only analyze the speech quality of three speech codec namely G.729, Speex and GSM. The respective speech codec will be selected or activated separately. For analysis of G.729 speech codec, in EyeBeam configuration we will only activated G.729 speech codec. Equally while analyzing Speex and GSM speech codec we will only trigger Speex or GSM speech codec. The QoS parameter packet jitter, packet loss, MOS score and R-factor can be captured by using the CommView for WiFi VoIP analyzer. The experimental procedure of this dissertation can be summarized to 3 main experiments which need to be repeated for each speech codec and for each predefined SNR value.

1. Experiment M1: “Client 1” talking to “Client 2” silent – Markov 1 (M1)
2. Experiment M2: “Client 1” silent to “Client 2” talking – Markov 2 (M2)
3. Experiment M3: “Client 1” talking to “Client 2” talking – Markov 3 (M3)

3. Experiment Results

In this chapter, the results of the experiment will be discussed. The speech quality of three speech codec namely G.729, Speex and GSM under various network performance based on pre-determined SNR values will be evaluated and compare against. Several tests are constructed to prove that it meets the interest of investigation. The experimental procedure of this dissertation can be summarized to 3 main experiments which need to be repeated for each speech codec and for each predefined SNR value. All the 3 experiments; Experiment M1, Experiment M2 and Experiment M3, need to be repeated for all three speech codec G.729, Speex and GSM with each respective SNR values; 10 dB, 15 dB, 20 dB, 25 dB, 30 dB, 35 dB, 40 dB and 45 dB. In determining the SNR value, Netstumbler is being used at both VoIP clients. For comparing all three speech codec, at each scenario the CommView for WiFi will capture and analyze the VoIP QoS namely packet jitter.

3.1 Experiment M1 scenario: “Client 1” talking and “Client 2” silent

In this experiment, G.729 and Speex has a lower bandwidth (31.2kbps) requirement compare to GSM (36.4 kbps). This result shows that GSM have a higher packet jitter compare to G.729 and Speex at SNR below 20 dB. As SNR decrease so does bit rates, it affects the performance of GSM speech codec due to the needs of higher bandwidth requirement GSM requires more bandwidth thus it is more sensitive to congestion or resource utilization. Increasing SNR will lead to higher bit rate so its reducing the effect of packet jitter and hence the improvement in performance of VoIP. As SNR increase, GSM records much lower packet jitter compare to G.729 and Speex. We can see that G.729 and Speex performance are almost at par at any given SNR. Packet jitter is lower than GSM for both speech codec at SNR below 20 dB. To obtain excellent quality calls, the receivable mobile stations needed a SNR of 20 dB or higher. Thus, all three speech codec shows at SNR 20 dB and above, the packet jitter value is almost negligible or at acceptable level.

QOS parameter	Speech Codec	SNR							
		10dB	15dB	20dB	25dB	30dB	35dB	40dB	45dB
Packet Jitter (ms)	G.729	47.76	42.18	18.42	12.60	9.34	9.59	9.01	6.56
	Speex	47.47	42.95	21.30	14.54	11.92	8.51	3.97	2.10
	GSM	92.68	77.75	10.95	15.12	8.88	4.26	2.04	1.22

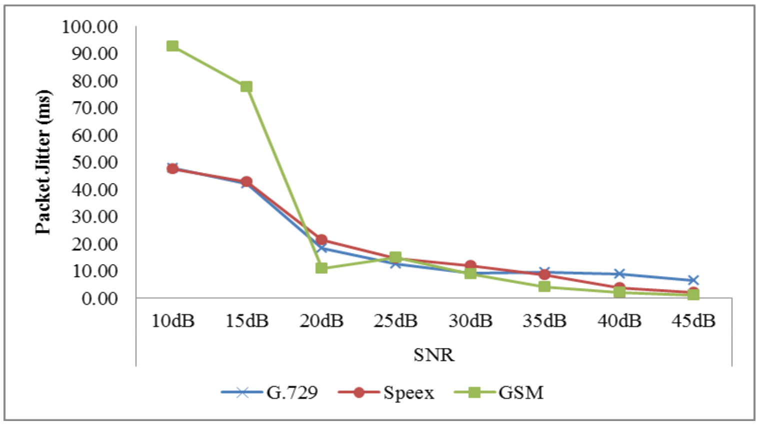

Figure 3. M1 for G.729, Speex and GSM on Packet Jitter Comparison

3.2 Experiment M2 scenario: "Client 1" silent and "Client 2" talking

In this experiment, G.729 and Speex has a lower bandwidth (31.2kbps) requirement compare to GSM (36.4 kbps). This result shows that GSM have a higher packet jitter compare to G.729 and Speex at SNR below 20 dB. As SNR decrease so does bit rates, it affects the performance of GSM speech codec due to the needs of higher bandwidth requirement. GSM requires more bandwidth thus it is more sensitive to congestion or resource utilization. Increasing SNR will lead to higher bit rate so its reducing the effect of packet jitter and hence the improvement in performance of VoIP. As SNR increase, GSM records much lower packet jitter compare to G.729 and Speex. We can see that G.729 and Speex performance are almost at par at any given SNR. Packet jitter is lower than GSM for both speech codec at SNR below 20 dB. To obtain excellent quality calls, the receivable mobile stations needed a SNR of 20 dB or higher. Thus, all three speech codec shows at SNR 30 dB and above, the packet jitter value is almost negligible or at acceptable level.

QOS parameter	Speech Codec	SNR							
		10dB	15dB	20dB	25dB	30dB	35dB	40dB	45dB
Packet Jitter (ms)	G.729	49.30	41.20	32.56	23.47	19.31	14.74	6.78	1.78
	Speex	48.17	39.94	25.40	20.67	17.31	9.35	3.58	1.78
	GSM	72.23	60.29	22.50	15.12	8.88	4.26	2.04	1.22

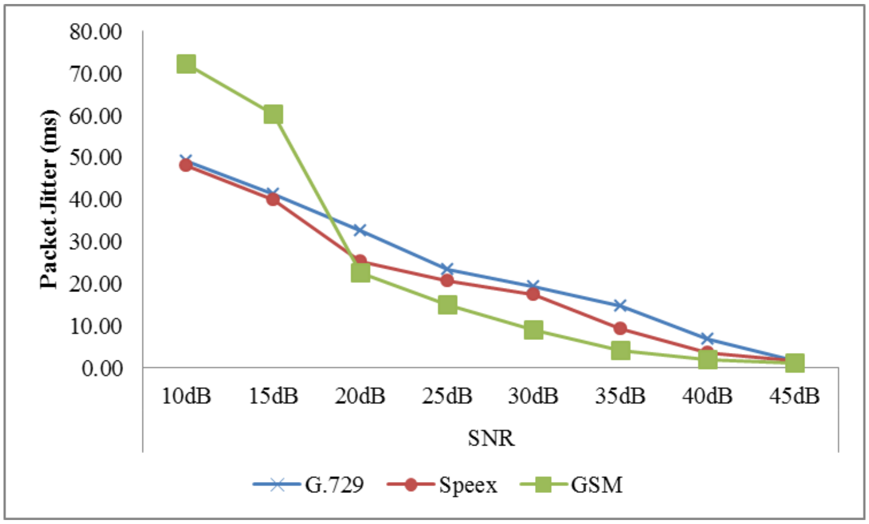

Figure 4. M2 for G.729, Speex and GSM on Packet Jitter Comparison

3.3 Experiment M3 scenario: "Client 1" talking and "Client 2" talking

In this experiment, G.729 and Speex has a lower bandwidth (31.2kbps) requirement compare to GSM (36.4 kbps). The result shows that GSM have a higher packet jitter compare to G.729 and Speex at SNR below 20 dB. However, the numbers of packet jitters recorded almost double than the ones recorded in M1 and M2 experiments. As SNR decrease so does bit rates, it affects the performance of GSM speech codec due to the needs of higher bandwidth requirement. GSM requires more bandwidth thus it is more sensitive to congestion or resource utilization. Increasing SNR will lead to higher bit rate so its reducing the effect of packet jitter and hence the improvement in performance of VoIP. As SNR increase, GSM records much lower packet jitter compare to G.729 and Speex. We can see that G.729 and Speex performance are almost at par at any given SNR. Packet jitter is lower than GSM for both speech codec at SNR below 20 dB. To obtain excellent quality calls, the receivable mobile stations needed a SNR of 20 dB or higher. Thus, all three speech codec shows at SNR 35 dB and above, the packet jitter value is almost negligible or at acceptable level.

QOS parameter	Speech Codec	SNR							
		10dB	15dB	20dB	25dB	30dB	35dB	40dB	45dB
Packet Jitter (ms)	G.729	95.81	84.41	52.61	41.19	33.25	24.66	11.75	3.88
	Speex	89.54	79.41	48.27	36.46	26.53	18.32	9.34	2.65
	GSM	164.91	138.04	30.93	23.58	12.42	6.68	3.95	2.48

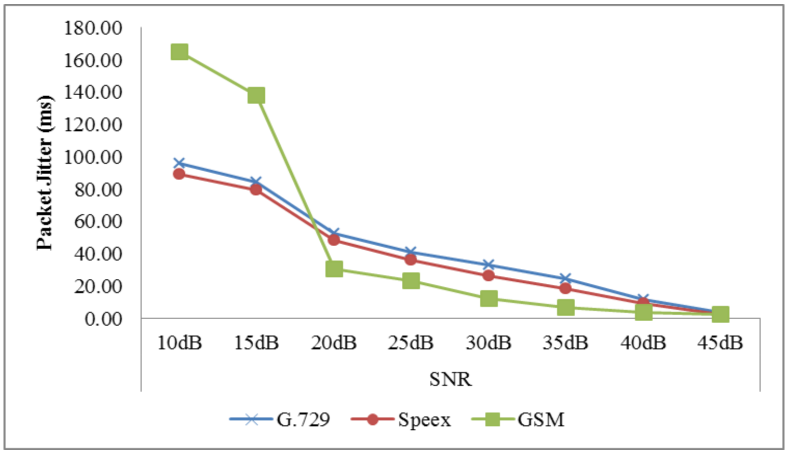

Figure 5. M3 for G.729, Speex and GSM on Packet Jitter Comparison

4. Conclusion

The popularity and widespread use of VoIP and its extension to wireless local area network has led to an increased interest in the study of voice over wireless LANs. Users clearly defined the decisive factor in selecting VoIP applications, which is the speech quality. The choice of speech codec has a major influence in the perceived quality. The combination factors of speech codec quality and WLAN signal strength by referring to SNR value will clearly give a substantial impact on voice quality. This paper focuses on three speech codecs; G.729, Speex and GSM. In the 802.11g wireless environment, with predetermined SNR value, the performance of selected speech codec was analyzed. VoIP QoS such as packet jitter became the benchmark in evaluating the effect of speech codec to VoIP performance in wireless 802.11g local area networks.

When analyzing the effect of speech codec to performance of VoWLAN, few factors of causes were identified clearly. Speech codec properties such as bit rate and bandwidth requirements in order to convey voice packet from one client to another client was taken into consideration. The relationship between SNR, 802.11g performance and capacity and data rate certainly gave an impact to perceived speech quality. Also, an association between packet jitter and speech quality was analyzed thoroughly. Theoretical knowledge of an acceptable packet loss rate for VoIP was needed for us to argue the impact of speech codec to VoIP. VoIP QoS such as packet jitter is used and analyzed in order to make a comparison of speech quality of those three speech codecs in wireless LAN 802.11g environment. Result shows that at lower SNR, GSM achieve higher packet jitter than that G.729 and Speex. At higher SNR, GSM achieves lower packet jitter as compared than that G.729 and Speex.

Acknowledgements

This research is supported by Universiti Malaysia Pahang Research Grant (RDU120388)

References

[1] "VoIP Primer: Introduction to Voice over IP", netIQ whitepaper, **(2006)** January.
[2] Federal Communications Commission, Consumer & Governmental Affairs Bureau, "Voice over Internet Protocol (VoIP) consumer facts", **(2006)** October.
[3] N. Shetty, S. Choudhury and J. D. Gibson, "Voice capacity under quality constraints for IEEE 802.1 la based WLANs", In Proceeding of IWCMC, **(2006)**.
[4] Rumin and E. Guy, "VoIP over WLAN 802.11b simulations for infrastructure and adhoc networks", http://www.it.uc3m.es/~acuevasr/publicaciones/LCS06.pdf.
[5] O. Awoniyi and F. A. Tobagi, "Effect of fading on the performance of VoIP in IEEE 802.11a WLANs", In Proceeding of ICC'04, **(2004)**, pp. 3712–3717.
[6] W. Wang, S. C. Liew and V. O. K. Li, "Solutions to performance problems in VoIP over 802.11 wireless LAN", IEEE Transactions on Vehicle Technology, **(2005)**, pp. 366–384.
[7] M. Narbutt and M. Davis, "Effect of Free Bandwidth on VoIP Performance in 802.11b WLAN Networks", In Proceeding of IEE Irish Signals and Systems, Dublin, **(2006)** June 28-30, www.cnri.dit.ie/rpublications/NARBUTT_ISSC_06.pdf.
[8] M. Narbutt and M. Davis, "An Assessment of the Audio Codec Performance in Voice over WLAN (VoWLAN) Systems", In Proceeding of the International Conference on Mobile and Ubiquitous Systems: Networking and Services, (MOBIQUITOUS'05), **(2005)**, pp. 461– 467.
[9] "ITU-T recommendations P.59", International Telecommunications Union (ITU-T), 1993 A. Azarfar and A. H. Jahangir, Voice quality measurement in a typical router-based network, In Proceeding of 32nd IEEE Conference on Local Computer Networks (LCN 2007), **(2007)**, pp. 274–275.
[10] ITU-T Recommendation H.323, "Visual telephone systems and equipment for local area networks which provide a non guaranteed quality of service", **(1996)**.
[11] Z. Jiansong, T. Kun, Z. Jun, W. Haitao and Z. Yongguang, "A Practical SNR-Guided Rate Adaptation", Microsoft Research Asia, Beijing, China, **(2007)**, pp. 146–150.
[12] C. William, "IEEE 802.11g – New draft standard clarifies future of Wireless LAN. Wireless Networking Business Unit", Texas Instruments, **(2004)**, pp. 1–5.
[13] S. Niranjan, C. Sayantan and D. G. Jerry, "Voice Capacity under Quality Constraints for IEEE 802.11a based WLANs", In Proceedings of the 2006 international conference on Wireless communications and mobile computing (IWCMC '06), **(2006)**, pp. 1345–1350.

Authors

Imran Edzereiq Kamarudin is currently lecture networking subjects in Universiti Malaysia Pahang, Malaysia. A certified CCNA and CCAI with 5 years working experience in network and telecommunications industry prior to lecturing His research area includes wireless network technology, mobile data network and wireless sensor network

Syahrizal Azmir Md Sharif, certified in CEH and ENSA, currently lecture security subjects in Universiti Malaysia Pahang, Malaysia His research area includes wireless network security and hacking method.

Tutut Herawan received a B.Ed degree in year 2002 and M.Sc degree in year 2006 degree in Mathematics from Universitas Ahmad Dahlan and Universitas Gadjah Mada Yogyakarta Indonesia, respectively. He obtained a PhD in Theoretical Data Mining from Universiti Tun Hussein Onn Malaysia in year 2010. Currently, he is a lecturer with Department of Mathematics Education, Universitas Ahmad Dahlan, Indonesia**.** He currently supervises four PhD and had successfully co-supervised two PhD students and published more than 120 papers in various international journals and conference proceedings. He has appointed as an editorial board member for IJDTA, TELKOMNIKA, IJNCAA, IJDCA and IJDIWC. He is also been appointed as a reviewer of several international journals such as Knowledge-Based Systems, Information Sciences, European Journal of Operational Research, Applied Mathematics Letters, and guest editor for several special issues of international journals. He has served as a program committee member and co-organizer for numerous international conferences/workshops including Soft Computing and Data Engineering (SCDE 2010-2011 at Korea, SCDE 2012 at Brazil), ADMTA 2012 Vietnam, DTA 2011-2012 at Korea, DICTAP 2012 at Thailand, ICDIPC 2012 at Lithuania, DEIS 2012 at Czech Republic, NDT 2012 at Bahrain, ICoCSIM 2012 at Indonesia, ICSDE'2013 at Malaysia, ICSECS 2013 at Malaysia, SCKDD 2013 at Vietnam and many more. His research area includes Knowledge Discovery in Databases, Educational Data Mining, Decision Support in Information System, Rough and Soft Set theory.