Performance Analysis of VoIP in Multi-Hop Wireless Network

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Abstract—**This paper presents the performance evaluation of Voice over Internet Protocol (VoIP) in Multi-hop Wireless Network (MWN) developed using Multi-radio Access Relay (MAR). The MWN is deployed using 3 MARs in Universiti Teknikal Malaysia campus. The performance of VoIP are investigated using Real Time Protocol (RTP) and Compress Real Time Protocol (CRTP) header techniques. RTP and CRTP are used to transport voice packets using G711.1, G723.1 and G729.2 codec. The performance of VoIP is analyzed based on three important elements which are delay, jitter and packet loss.**

Index Terms—**VoIP, multi-hop wireless network, RTP, CRTP, delay, jitter, packet loss.**

I. Introduction

Voice over Internet protocol [1] is a technology that enables the transport of voice over data networks such as the public Internet. The idea of VoIP has been discussed since the early 1970s when the idea and technology were developed. However, at that time VoIP did not find wide acceptance and deployment both among users and telecommunication providers, mainly due to the lack of Internet Protocol (IP) infrastructure. The circuit-switched calling was still a much more reliable alternative, especially due to the poor quality of early VoIP calls.

However, following the rapid growth of the Internet and the Web in the mid 1990s along with the huge investments in the IP networking infrastructure by businesses, vendors and carriers, VoIP is increasingly becoming an alternative to send voice over public switched telephone network (PSTN). The basic idea behind VoIP involves the transmission of voice as data packets using IP. The user's voice is converted from analogue form into a digital signal, compressed or uncompressed and broken down into a series of packets. These packets are then routed through private or public IP networks from one user to another and reassembled and decompressed at the receiving side. Factors that affect the adoption of VoIP technology include cost savings and improved network utilization.

Multi-hop Wireless Network (MWN) is one of the characteristics of Wireless Mesh Networks (WMN) [2,3,4]. It is a communications network made up of radio nodes organized in a mesh topology. The MWN often consist of clients, routers and gateways Wireless networks are great for extending network and internet connections through buildings, campuses, and other spaces. But even though the technology frees users from needing to connect wires from their computer to the network, the wireless routers and access points must ultimately connect to the underlying network over Ethernet cables. Thus, a big challenge in any wireless deployment is running the wires to the wireless equipment, often in areas without power and certainly without Ethernet connections. MWN can provide solutions to this challenge.

The main purpose of MWN topology is to extend the coverage of LAN network and can provide more Internet access to many users. Since VoIP is one of popular real time applications used by Internet users, it is important to analyze the performance of VoIP in MWN. The aim of this paper is to analyze the performance of VoIP for different types of voice codec using RTP and CRTP header techniques.

The rest of the paper is organized as follows: VoIP is introduced in Section 2. Section 3 describes the network configuration for the MWN. The results obtained are discussed in Section 4. Finally, the conclusions of the study are given in Section 5.

II. Voice over Internet Protocol

VoIP means that calls are transmitted over an IP network such as the Internet instead of Public Switched Telephone Networks. Since access to the Internet is available at more and more places in the world, it is possible to use VoIP in a higher degree. VoIP converts standard telephone voice signals into compressed data packets that can be sent over IP. Before transmitted over packet switched networks, the speech signal has to be digitized at the sender; the reverse process is performed at the receiver. The digitalization process is composed of sampling, quantization and encoding.

Codec is a short name for coder-decoder. Codec is used to compress analog voice signal to digitally encoded version that can travel on computer networks. Sound quality, bandwidth required, and resource requirements all depends on the choice of codec [5,6]. G.7xx, including G.711, G.722, G.723.1, G.726, G.727, G.728, G.729, is a suite of International Telecommunications Union (ITU) standards for audio compression and de-compression.

G.711 is a codec that was introduced by ITU for use in digital telephony, such as in Integrated Services Digital Network (ISDN), T1 and E1 links. G.711 is primarily used for encoding telephone audio signal at a rate of 64kbps with a sampling frequency of 8kHz and 8 bits per sample. G.711 represents logarithmic pulse-code modulation (PCM). There are two main compression algorithms defined in ITU standard which are the μ-law and A-law. A typical algorithmic delay is 0.125ms with no look-ahead delay. In an IP network, voice is converted into packets with durations of 20ms of sampled voice.

G.723.1 is an audio codec for voice that compresses voice audio in 30ms frames. With algorithmic look-ahead of 7.5ms duration, the total algorithmic delay is 37.5ms. There are two bit rates at which G.723.1 can operate: 6.3kbps (using 24byte frames) and 5.3kbps (using 20byte frames) with Algebraic Code Excited Linear Prediction (ACELP) algorithm. The coder operates on speech frames of 30ms corresponding to 240 samples at a sampling rate of 8000 samples per second. G.723.1 is mostly used in VoIP applications due to its low bandwidth requirement.

G.729 codec belongs to the Code Excited Linear Prediction (CELP) model speech coders and uses Conjugate Structure - Algebraic Code Excited Linear Prediction (CS-ACELP). G.729 operates at 8 kbps, but there are extensions which provide rates of 6.4kbps and 11.8kbps. The coder compresses voice in packets of 10ms duration and required look-ahead delay of 5ms. The total algorithmic delay for the coder is 15ms. G.729.2 is for G.729 codec with 2 samples per packet.

Each VoIP packet includes the headers at the various protocol layers such as RTP, User Datagram Protocol (UDP), IP, 802.11 and the payload comprising the encoded speech for certain duration depends on the codec deployed.

The Real Time Transport Protocol (RTP) is used to transport the voice packets. RTP is carried over UDP. The IP, UDP, and RTP headers that encapsulate voice traffic have a total size of 40 bytes [7]. A variant of RTP is Compressed RTP (CRTP), which eliminates much of the overall packet header. By eliminating this overhead, a more efficient packet is placed onto the network. If CRTP is used, then the 40 bytes of overhead incurred by the IP/UDP/RTP headers can typically be compressed down to 2 to 4 bytes.

III. Network Setup

This project starts with deployment of the multi-hop topology network using Multi-radio Access Relay (MAR) in a Cafeteria [8]. By using 3 MARs with 802.11b/g wireless system, the WMN was deployed. The MARs are designated as A, B and C and located as shown in Fig. 1. The distance between MAR A and MAR B is 46.9 meter while the distance between MAR B and MAR C is 37 meter. The height of all three MARs is 3.65 meter from the ground. MAR A is located near to a wired backbone to make sure the Internet connection can be easily established later.

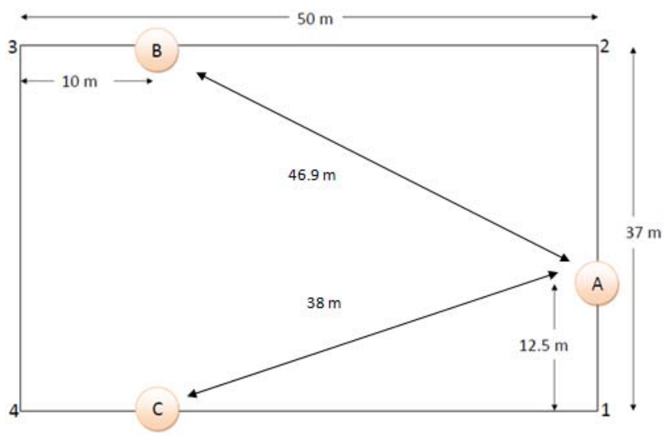

Fig. 1. Testbed layout

All MARs are configured using Linux command under Ubuntu 11.04 operating system. The configurations for all MARs are shown at Table I. Non overlapping channels number 1, 6 and 11 are used to avoid signal interference between MARs. Inside each MAR, there are two wireless network radio interfaces which are ath0 and ath1.Using proper Linux commands, two different interfaces each at a different MAR can be connected to create wireless multi-hop network. In other words, ath0 in MAR A is connected to ath1 in MAR B. Ath0 in MAR B is connected to ath1 in MAR C.

TABLE I. Settings for Multi-radio Access Router

MAR Name	**IP Address**	**Channel**	
		ath0	**ath1**
A	192.168.30.1	1	-
B	192.168.20.1	6	1
C	192.168.10.1	11	6

The performance of VoIP is analyzed using DITG 2.8.0-rc1 [9], which is an open source Internet traffic generator. In this work, three functions being used are ITGSend, ITGRecv and ITGDec. ITGSend is used at server side to send data to client. There are many components that available in ITGSend option such as destination address, destination port, source port and etc. Meanwhile, the ITGRecv is used by client to received data sent from the server. The results are saved in log files and can be decoded using ITGDec function.

RTP and CRTP header techniques and three types of voice codecs which are G711.1, G723.1 and G729.2 are used in the tests. The voice code options are available in D-ITG 2.8.0-rc1. The performance of VoIP in the network is analyzed based on three important elements which are latency, jitter and packet loss. The values of these elements are contained in the log file and can be displayed using ITGDec.

In this project, the terms that are used for all tests are 1 hop, 2 hops and 3 hops. The 1 hop configuration is where the server connected to the MAR A using an RJ45 cable while the client is connected to the same MAR through wireless connection. In 2 hops configuration, the server remains connected to MAR A while the client now connects to MAR B. Lastly, for the 3 hops condition, the server still connected the MAR A and the client is connected to MAR C.

For the first test, the location of client is near the MAR where the RSSI value is much higher than -60dBm. For the lower signal strength test, the client is located at a place where RSSI value of -60 dBm is recorded. Only CRTP header is used in the lower signal strength condition. In all test configurations, the server sends VoIP data to the client for 15s.

IV. RESULTS AND DISCUSSION

A. Delay

Delay or latency refers to the amount of time it takes to transmit data from source to destination. Fig. 2 shows the delay for VoIP traffic using RTP header sent from the server to the client. As expected, the value of delay increase from 1 hop to 3 hops. The voice codec of G711.1 recorded the lowest value of average delay while the voice codec of G729.2 recorded the highest value of average delay.

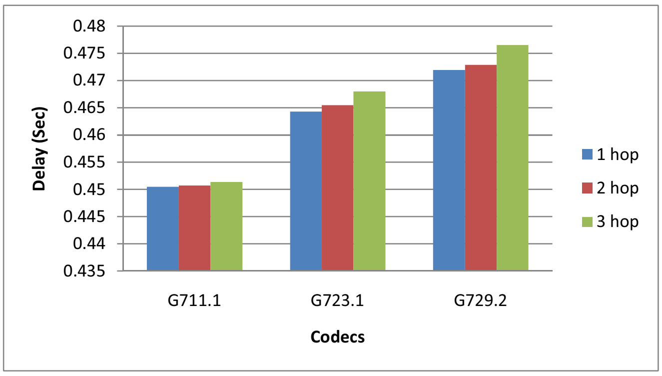

Fig. 2. Delay comparison with RTP header

The delay pattern by using CRTP scheme is similar with RTP as shown at Fig. 3. The average delay is increased with increasing number of hops, but there is a slight difference between the values of average delay. From the observation, the values of average delay for CRTP are always higher than RTP. It is because the CRTP compression process will take more time compared to RTP header process. The delay of CRTP header process and plus with voice codec compression time make G729.2 having the highest value of average delay.

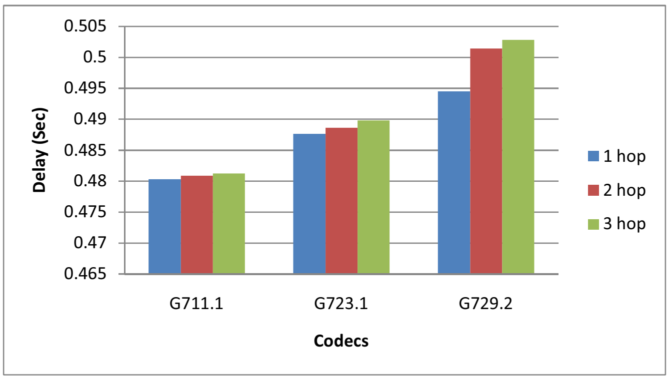

Fig. 3. Delay comparison with CRTP header

The average delay value measured at lower signal strength with CTRP scheme is shown in Fig. 4. When VoIP communication is taking place at lower signal strength, we can see that there is significant increase of delay recorded. For each voice codec, the delay value is more than 1s and as usual the value is increasing as the number of hops increase. With RSSI value of -60dBm, the distance between the client and the MARs are farther and this caused the average delay to increase to more than twice compared to previous results.

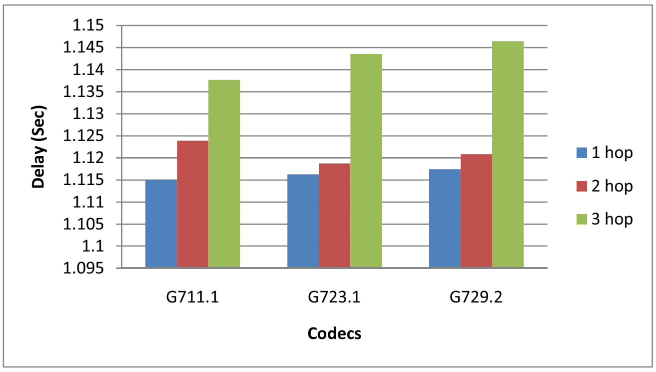

Fig. 4. Delay comparison with CRTP header at lower signal strength

B. Jitter

Jitter is defined as a variation in the delay of received packets. From the Fig. 5, 6 and 7, the values of average jitter are increased from 1 hop to 3 hops but it is decrease with the difference for voice codec, contrary with the previous three graphs. The jitter value for G711.1 codec is higher than G723.1 and G729.2. The results proved that the voice quality for G729.2 is better than G711.1 and G723.1 in term of jitter during data transmissions. The G729.2 is known as the best quality of voice compression. With the low of bit rate which is 8kbps, the G729.2 also can reduce the size of voice data that client receive.

Fig. 5. Jitter comparison with RTP header

Fig. 6. Jitter comparison with CRTP header

Fig. 7. Jitter comparison with CRTP header at lower signal strength

The signal strength test is used to determine the effect of lower signal strength to the performance VoIP in the multi-hop network. The jitter value for RTP and CRTP are slightly the same compare to at lower signal strength test with CRTP header as shown at Fig. 7. For lower signal strength, the maximum jitter value is reach to 2.3ms while for RTP and CRTP header test maximum jitter value is 1.7ms. From the results, higher average jitter are recorded at lower signal strength condition.

C. Packet Loss

Packet loss occurs when one or more packets of data travelling across a network fail to reach their destination. Fig. 8 shows the packet loss for RTP header test is increase from 1 hop to 3 hops. The highest packet loss occurs during the test is 1.5%, when using voice codec G711.1. There is no packet loss recorded for a single hop communication. The results of CRTP test is shown at Fig. 9. The highest value for packet loss is 4.5% at 3 hops for voice codec G723.1. Similar with using RTP, no packet loss recorded when tested in single hop configuration. Lastly at lower signal strength, the result shows that the highest percentage of packet loss is 6% for 3 hops configuration for voice codec G723.1 as shown at Fig. 10.

Fig. 8. Packet loss comparison with RTP header

Fig. 9. Packet loss comparison with CRTP header

Fig. 10. Packet loss comparison with CRTP header at lower signal strength

At lower signal strength, the results showed that the increasing of number of hops will cause the increasing number of packet loss no matter what type of voice codec is used during the VoIP communication. The result shows that the number of packet loss will affect the performance of voice codec G 723.1 if using CRTP header technique. The value of packet loss does not meet the minimum requirement of 1% as shown at the graphs.

V. CONCLUSION

In this paper, we analyzed the performance of VoIP in multi-hop network using RTP and CRTP header techniques and the effect of lower signal strength. The measurements were done for G711.1, G723.1 and G729.2 voice codecs. To analyze the VoIP performance, three important elements which are delay, jitter and packet loss are considered.

For this particular wireless multi-hop network deployment, each codec with the number of hops and header techniques recorded a high value of delay. ITU-T G.114 recommended that the value of delay should be less than 150 ms (one way). The average jitter values recorded are very low compared to the recommend value given by National Institute of Standards and Technology (NIST) which is less than 40ms. In term of packet loss, the VoIP communication using different codecs with RTP header technique recorded the lowest packet loss value. The recommended value of packet loss recommended by NIST is 1%. The percentage of packet loss increases as the number of hop increases and also at lower signal strength condition.

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REFERENCES

[1] B. Goode, "Voice over Internet protocol (VoIP)," Proceedings of the IEEE, vol.90, no.9, pp. 1495- 1517, Sep 2002.

[2] I.F. Akyildiz, X. Wang, and W. Wang, “Wireless mesh networks: a survey,” Elsevier, 47(4), pp. 445–487, 2005.

[3] A. Hamidian, C.E Palazzi, T.Y. Chong, J.M. Navarro, U. Korner, and M. Gerla, "Deployment and evaluation of a Wireless mesh network," Second International Conference on Advances in Mesh Networks, pp. 66-72, 18-23 June 2009.

[4] S. Yuhe, and X. Jie, “New solutions of VoIP on multi-hop wireless network,” International Conference on Control, Automation and Systems Engineering, pp. 199-202, 11-12 July 2009

[5] S. Narayan, M. Gordon, C. Branks, and L. Pan, “VoIP network performance evaluation of operating systems with IPv4 and IPv6 network implementations,” IEEE International Conference on Computer Science and Information Technology, vol.5, no., pp. 669-673, July 2010.

[6] H. Sathu and M. A. Shah, “Performance monitoring of VoIP with multiple codecs using IPv4 and IPv6to4 tunnelling mechanism on Windows and Linux,” International Journal of Modeling and Optimization. vol.2, no.3, June 2012.

[7] N.Sulaiman, R.Carrasco, and G.Chester “Performance evaluation of voice call over an IP based betwork,” 41th Annual Conference on Information Sciences and Systems, vol., no., pp. 392-395, March 2007.

[8] F. Idris, M. Z. A. M. Nizam, M. F. Rahmat, H. P. Hou and L. K. Chuan, " A Practical Analysis of Multi-Hop Wireless Network Deployment in UTeM Campus," 3rd International Conference on Engineering and ICT (ICEI2012), pp. 481-483, 4-5 April 2012.

[9] A. Dainotti, A. Botta, A. Pescapè, "A tool for the generation of realistic network workload for emerging networking scenarios", Computer Networks (Elsevier), vol.56, issue 15, pp. 3531-3547, 2012.