Performance Analysis of VoIP Codecs over Wi-Fi and WiMAX Networks

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Abstract

Voice over IP (VoIP) applications such as Skype, Google Talk, and FaceTime are promising technologies for providing low cost voice calls to customers over the existing data networks. Wireless networks such as Wi-Fi and WiMAX focus on delivering Quality of Service (QoS) for VoIP applications. However, there are numerous aspects that affect quality of voice connections over wireless networks. In this paper, we evaluate performance of three VoIP codecs over Wi-Fi and WiMAX networks. OPNET Wi-Fi and WiMAX simulation models are designed to generate and evaluate performance metrics such as Mean Opinion Score (MOS), average end-to-end delay, and jitter.

1. Introduction

Recent Voice over IP (VoIP) applications such as Skype, Google Talk, and FaceTime have changed the way we communicate. Due to the low cost, VoIP has become a viable alternative to the expensive traditional Public Switched Telephone Networks (PSTNs). VoIP parameters define its Quality of Service (QoS) such as Mean Opinion Score (MOS), end-to-end delay, jitter, packet loss, and throughput [1].

The existing Wi-Fi and WiMAX wireless networks offer flexibility to support real-time applications such as VoIP [1]. The IEEE 802.11 (Wi-Fi) technology shows great success as inexpensive wireless Internet access while the IEEE 802.16 (WiMAX) provides large coverage area (approximately 50 km) and high data rates (up to 75 Mbps) using radio links [2].

In this paper, we examine the required QoS for VoIP applications in both Wi-Fi and WiMAX technologies. We use OPNET 16.0.A simulator to analyze the QoS of VoIP application under various codecs. The paper is organized as follows: Section 2 gives an overview about VoIP over Wi-Fi and WiMAX and surveys related research reports. Section 3 describes the simulation scenarios. We present and discuss the simulation results in Section 4. Finally, the conclusion and suggested future work are given in Section 5.

2. Background

The performance of VoIP applications using various technologies have been addressed in the literature. Selection of the appropriate VoIP codec was investigated using the OPNET 16.0.A simulator in an integrated WiMAX/Wi-Fi network [1]. It was shown that VoIP under GSM Enhanced Full Rate (GSM-EFR) and GSM Full Rate (GSM-FR) codecs achieves desirable speech quality with tolerable delay and jitter. However, G.726 performs poorly in terms of MOS value, delay, jitter, and packet loss.

Another simulation study compares VoIP over Wi-Fi and VoIP over WiMAX [3]. The simulation results show that the throughputs of Wi-Fi and WiMAX networks are affected by VoIP application. However, jitter and packet loss are experienced only in Wi-Fi networks.

Various buffer sizes were deployed to improve the performance of real-time applications over WiMAX [4]. Queues are required in this type of applications because they reduce the overall delay. The impact of channel bandwidth, time division duplex (TDD) frame size, and retransmission in real-time applications over WiMAX were simulated using OPNET [2]. The study indicates that WiMAX may deliver sufficient bandwidth with packet delays and jitter that meet QoS requirements.

Various QoS configurations were used to improve the performance of VoIP over Best Effort (BE) WiMAX [5]. The extended real-time polling service (ertPS) scheduling class that was designed to support variable rate real-time services significantly improves the performance of VoIP over BE WiMAX.

Performance metrics such as MOS, end-to-end delay, jitter, and packet delay variation of WiMAX and Universal Mobile Telecommunications System (UMTS) were also analyzed using OPNET simulations [6]. The results confirm that VoIP over WiMAX performs better than VoIP over UMTS. Performance evaluation of various VoIP codecs over the WiMAX network shows that both the size of the jitter buffer and packetization time significantly affect the performance of VoIP over WiMAX networks [7].

In this paper, we evaluated the performance of VoIP applications over Wi-Fi and WiMAX networks under three codecs: G. 711, G. 723, and G. 729. We use OPNET 16.0.A to simulate and analyze the QoS of VoIP performance. MOS, end-to-end delay, and jitter are examined as performance metrics.

2.1 VoIP over Wi-Fi

Wi-Fi is commonly used in residential, business, and public areas. It is notable that the perceived throughput in Wi-Fi does not match the real throughput. Furthermore, all users share the access to the channel, which is very critical for all real-time applications in general and especially for VoIP. The low capacity of Wi-Fi connections has a high impact on the QoS in VoIP. Beside the high traffic generated by users, both protocols VoIP and Wi-Fi create large headers, which result in degraded VoIP performance [3].

2.2 VoIP over WiMAX

WiMAX as a broadband wireless technology is considered as an alternate solution to wired networks. It provides up to 75 Mbps data rate and has a coverage area of up to 50 km [2]. It also supports QoS requirements by various applications especially real-time applications such as VoIP. WiMAX supports its applications through four distinct traffic classes:

- *Best Effort* (BE) is designed for applications such as web browsing [3] that do not require QoS.
- *Non Real-Time Polling service* (nrtPS) supports non real-time applications such as File Transport Protocol (FTP) [3] that require variable size of data.
- *Unsolicited Grant service* (UGS) supports Constant Bit Rate (CBR) application such as VoIP without silence suppression [3], [8] where Base Station (BS) assigns a fixed bandwidth to users.
- *Real-Time Polling service* (rtPS) supports real-time applications with variable size data such as MPEG [8] where BS allocates bandwidth based on Subscriber Station (SS) request.

Although WiMAX has been designed to provide broadband Internet service, VoIP applications have a high impact on performance of WiMAX networks [5].

2.3 QoS of VoIP Applications

Users currently take advantage of the existing data networks through text messages, voice calls, and video calls. The traditional phone networks cannot compete with these types of services due to their low equipment and operating costs and the ability to integrate voice and data applications [1]. The QoS for VoIP is measured by performance metrics such as Mean Opinion Score (MOS), end-to-end delay, and jitter.

- *Mean Opinion Score* (MOS) scale varies from 1 to 5and measures the quality of the voice. The value of the worst quality is 1 and the best quality is 5 [6], as shown in Table 1.

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

Table 1: Mean Opinion Score (MOS) [9].

- *Jitter* is the variation in arrival time of consecutive packets [10]. Before decoding, packets arrive to buffers of limited size and some packets may be lost or arrive out of order. Jitter is calculated by computing the difference in delay of packets over a period of time [6].
- *Packet end-to-end delay* is measured by calculating the delay from the speaker to the receiver. It includes network delay, encoding and decoding delays, and compression and decompression delays [10].

The guidelines for voice quality measurement for both end-to-end delay and jitter, shown in Table 2, are provided by the Telecommunication Standardization Sector of the International Telecommunications Union (ITU-T) [10]. A good quality voice call should have a delay between 0 ms and 150 ms and a jitter between 0 ms and 20 ms. However, if a call experiences a delay greater than 300 ms or a jitter greater than 50 ms, it is considered to be of a poor quality. Otherwise, calls are considered to be of acceptable quality.

Network parameter	Good	Acceptable	Poor
Delay (ms)	0□150	150–300	> 300
Jitter (ms)	0□20	20–50	> 50

Table 2: Guideline for the voice quality [10].

2.4 VoIP Codecs

VoIP relays on several codecs, which are used to compress and decompress audio samples. Each codec applies different algorithms. The most popular codecs are listed in Table 3 [9]. In this paper, we evaluate three VoIP codecs: G. 711, G. 723, and G. 729.

Codec	Data rate (kbps)	MOS score
G. 711	64	4.3
G. 723	5.3	3.6
G. 726	32	4.0
G. 728	16	3.9
G. 729	8	4.0
GSM	13	3.7

Table 3: Common VoIP codecs [9].

2.4.1 G. 711

G. 711 is a public domain codec widely used in VoIP applications. It was introduced in 1972 by the ITU. It employs a logarithmic compression that compresses each 16-bit sample to 8-bits. As a result, its bit-rate is 64 kbps, which is considered the highest bit-rate among the codecs. G. 711 offers very good audio quality and the MOS value of 4.3 [11].

2.4.2 G. 723

G. 723 is a licensed codec. It is designed for calls over modem links with data-rates of 28.8 and 33 kbps. Therefore, it has two versions with distinct bit-rates: 5.3 and 6.4 kbps [11]. In this paper, we consider the 5.3 kbps, which is based on the Algebraic Code-Excited Linear Prediction (ACELP) algorithm. Its MOS value is 3.6 [12].

2.4.3 G. 729

G. 729 is also a licensed codec designed to deliver good call quality without consuming high bandwidth [11]. It is built based on the Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP) algorithm with bit-rate of 8 kbps and MOS value of 4.0 [11], [12].

3. OPNET Model

To evaluate the QoS in VoIP, we designed two models using OPNET Modeler 16.0.A. The first model is VoIP over Wi-Fi network. It simulates a wireless network that consists of two mobile subnets: Vancouver and Calgary. These subnets are connected via IP cloud using Ethernet links at 1 Gbps. All links have 10% to 20% background traffic load. The IP cloud is connected to VoIP server as shown in Figure 1.

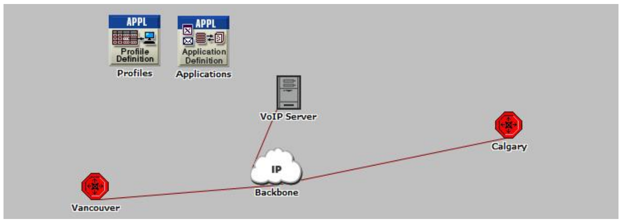


Figure 1: Wi-Fi network model.

Each subnet has a main router, as shown in Figure 2. The main router is connected to a wireless router, which is configured to support IEEE 802.11g protocol (54 Mbps) as shown in Figure 3.

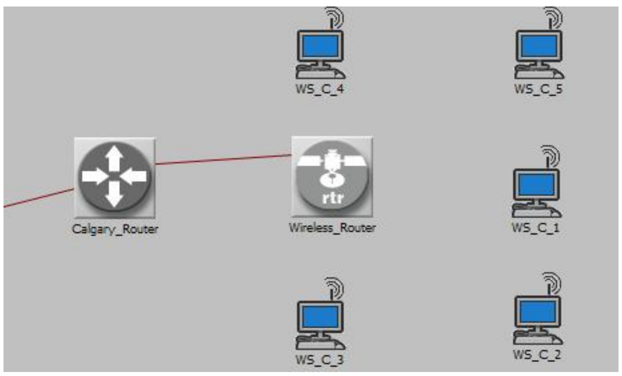
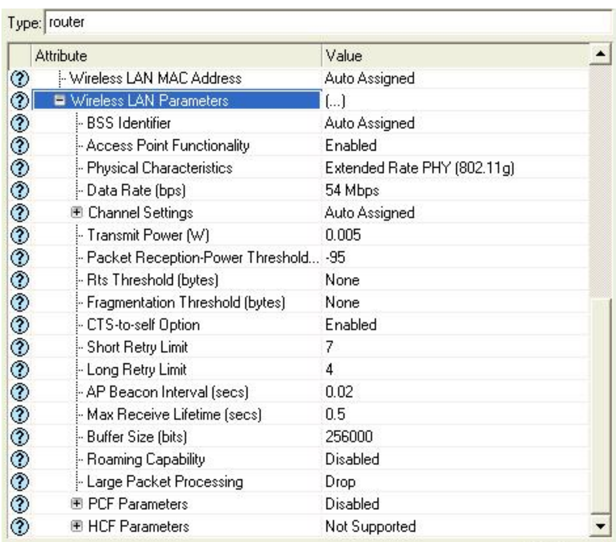


Figure 2: Calgary Wi-Fi subnet.



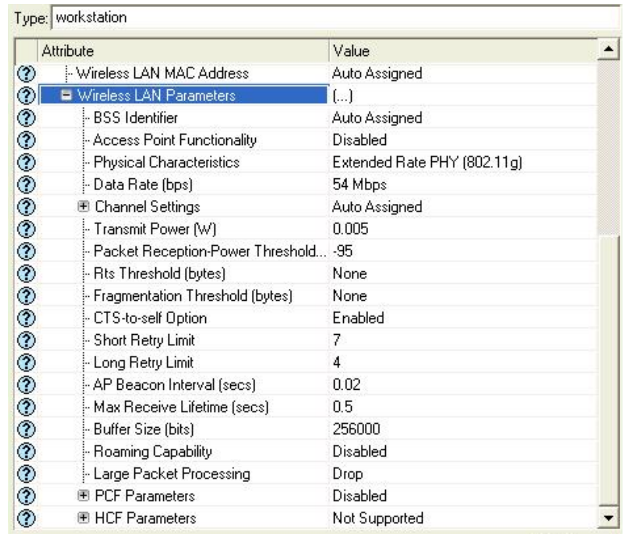
Type: router

Attribute	Value
Wireless LAN MAC Address	Auto Assigned
Wireless LAN Parameters	(...)
BSS Identifier	Auto Assigned
Access Point Functionality	Enabled
Physical Characteristics	Extended Rate PHY (802.11g)
Data Rate (bps)	54 Mbps
Channel Settings	Auto Assigned
Transmit Power (W)	0.005
Packet Reception-Power Threshold...	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (bytes)	None
CTS-to-self Option	Enabled
Short Retry Limit	7
Long Retry Limit	4
AP Beacon Interval (secs)	0.02
Max Receive Lifetime (secs)	0.5
Buffer Size (bits)	256000
Roaming Capability	Disabled
Large Packet Processing	Drop
PCF Parameters	Disabled
HCF Parameters	Not Supported

Figure 3: Wi-Fi: Wireless router configuration.

The wireless router provides connectivity to five clients in each subnet. These clients are located within a circle of radius 20 m. They are configured to initiate several voice calls during the simulation time. The configuration of the client workstations is shown in Figure 4.

The second OPNET model is VoIP over WiMAX. The WiMAX model is composed of Base Stations (BSs) and Subscriber Stations (SSs). The BSs provide air interface to the SSs to enable VoIP calls [6].



Type: workstation

Attribute	Value
Wireless LAN MAC Address	Auto Assigned
Wireless LAN Parameters	(...)
BSS Identifier	Auto Assigned
Access Point Functionality	Disabled
Physical Characteristics	Extended Rate PHY (802.11g)
Data Rate (bps)	54 Mbps
Channel Settings	Auto Assigned
Transmit Power (W)	0.005
Packet Reception-Power Threshold...	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (bytes)	None
CTS-to-self Option	Enabled
Short Retry Limit	7
Long Retry Limit	4
AP Beacon Interval (secs)	0.02
Max Receive Lifetime (secs)	0.5
Buffer Size (bits)	256000
Roaming Capability	Disabled
Large Packet Processing	Drop
PCF Parameters	Disabled
HCF Parameters	Not Supported

Figure 4: Wi-Fi: Workstation configuration.

We created a WiMAX network model with two mobile subnets: Vancouver and Calgary. Both subnets are connected via IP cloud using Ethernet links at 1 Gbps. This IP cloud is connected to VoIP application server as shown in Figure 5. All links in this model have 10% to 20% background traffic load.

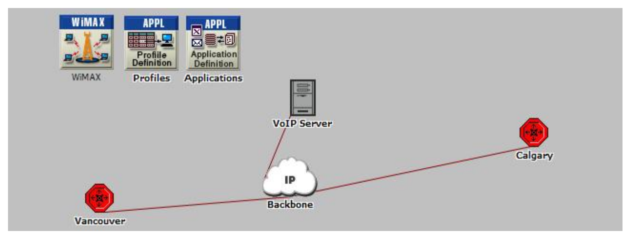


Figure 5: WiMAX network model.

The Calgary WiMAX subnet is shown in Figure 6. It consists of a main router and a BS. The BS connects five stations, which are located within a circle of radius 15 km.

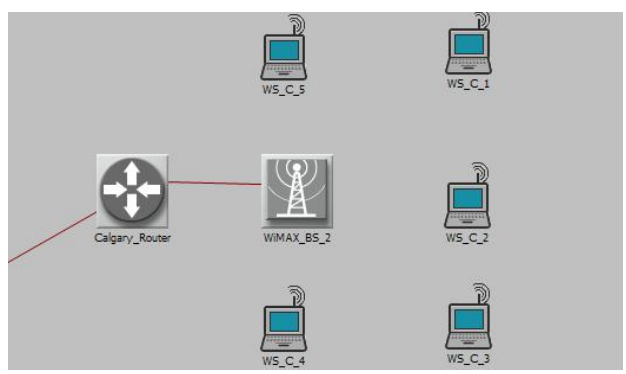


Figure 6: Calgary WiMAX subnet.

These stations are configured to make several voice calls during the simulation time. We used transmission power of 0.5W, receiver sensitivity of -200 dBm, and PHY profile wireless Orthogonal Frequency-Division Multiplexing (OFDMA) with 20 MHz. The WiMAX configurations of BSs and SSs are shown in Figure 7 and Figure 8, respectively.

We created a service class Gold with UGS allocation for VoIP application and deployed service flows and classifiers on all SSs.

UGS connections were configured for both uplink and downlink. QPSK modulation with ½ initial coding rate is used for the setup.

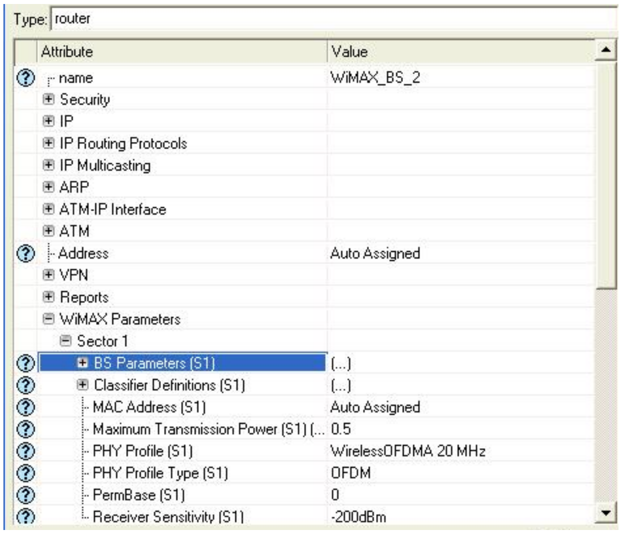


Figure 7: WiMAX: BS configuration.

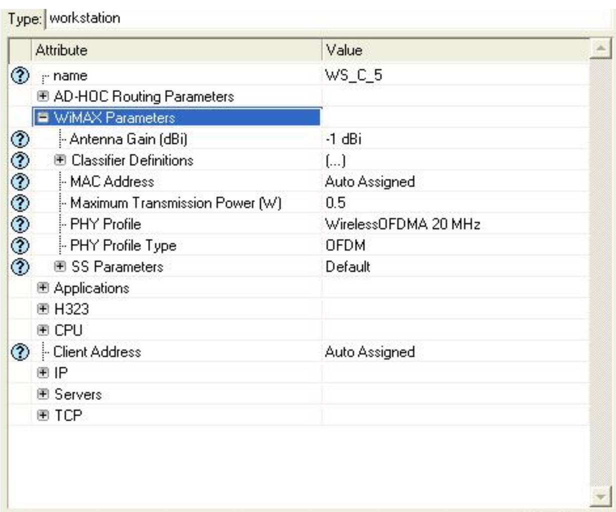


Figure 8: WiMAX: SS configuration.

3.1 Application Configuration

Clients in both Wi-Fi and WiMAX networks are configured to run VoIP application. This VoIP application is defined as voice application shown in Figure 9. It runs Interactive Voice service and generates one voice frame per packet. This application runs in a serial mode as defined in the application profile shown in Figure 10. The call arrivals from clients are exponentially distributed. The average duration of each call is 3 min [13]. A call to another client is generated randomly, as shown in Figure 10. All call inter-arrival times are exponentially distributed [10].

3.2 Simulation Scenarios

Each model is tested under three different simulation scenarios. Each scenario is configured to use one voice codecs: G. 711, G. 723, or G. 729. Six scenarios that are used in this paper are shown in Table 4. The simulation time for each scenario is 60 min.

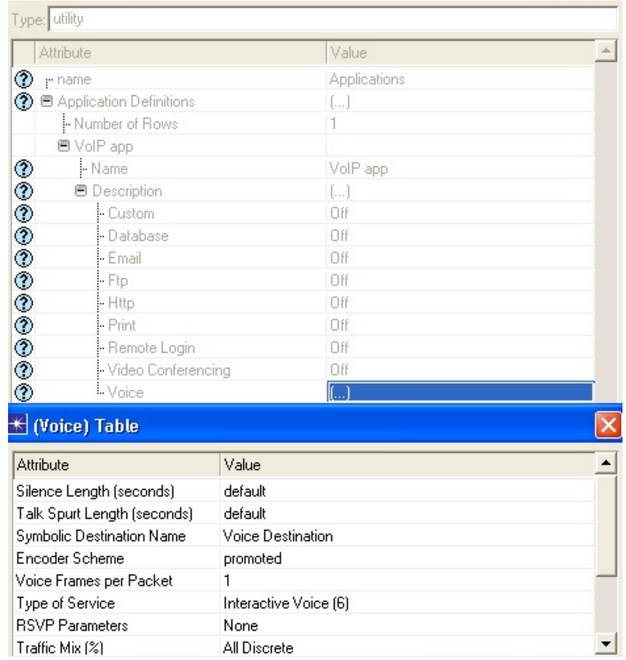


Figure 9: Application definition.

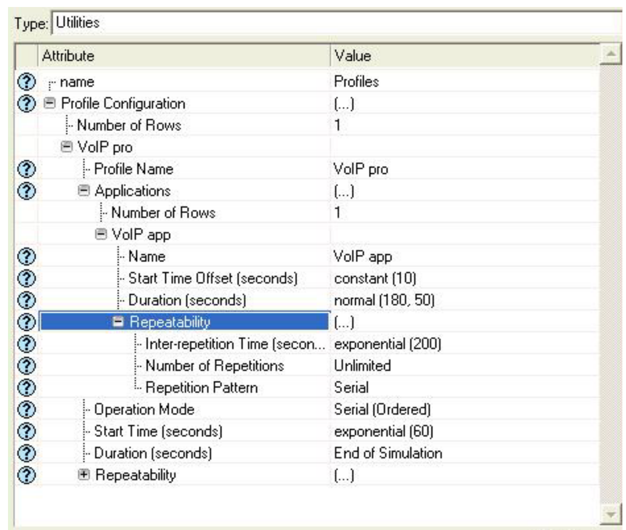


Figure 10: VoIP application profile.

Scenario	Scenario Name	Codec	Clients Number
1	VoIP over Wi-Fi	G. 711	10
2	VoIP over Wi-Fi	G. 723	10
3	VoIP over Wi-Fi	G. 729	10
4	VoIP over WiMAX	G. 711	10
5	VoIP over WiMAX	G. 723	10
6	VoIP over WiMAX	G. 729	10

Table 4: Simulation scenarios.

4. Simulation Results

In this Section, we discuss the simulation results for VoIP over Wi-Fi and VoIP over WiMAX models. Each model is tested with the three codecs (G. 711, G. 723, and G. 729).

4.1 VoIP over Wi-Fi

Three simulation scenarios for VoIP over Wi-Fi are considered. The MOS values are shown in Figure 11. G. 711 has the highest average MOS value of 4.35. Codecs G. 723 and G. 729 also have acceptable MOS values between 3.95 and 4.0, respectively.

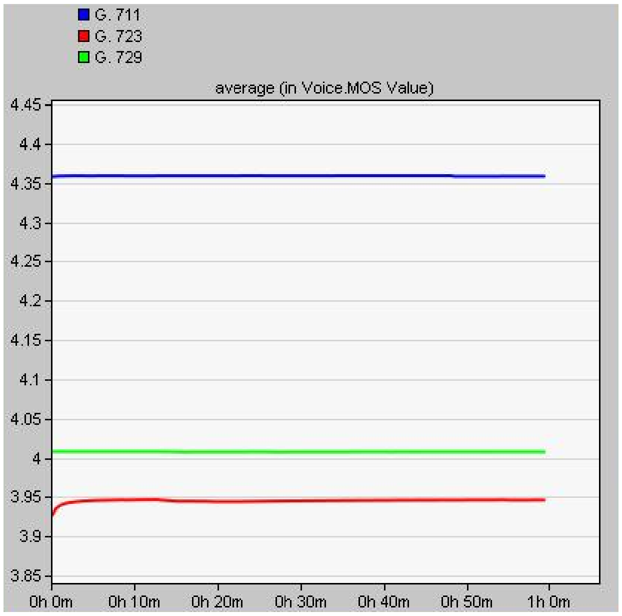


Figure 11: Wi-Fi: Average MOS.

Although G. 711 has the highest data-rate, it shows the lowest average end-to-end delay as shown in Figure 12. However, average end-to-end delays of G. 723 and G. 729 are larger than 300 ms, which is considered a poor voice quality. The calls have slight jitter under G. 729 codec as shown in Figure 13. G. 711 codec shows the best performance for VoIP applications over Wi-Fi networks.

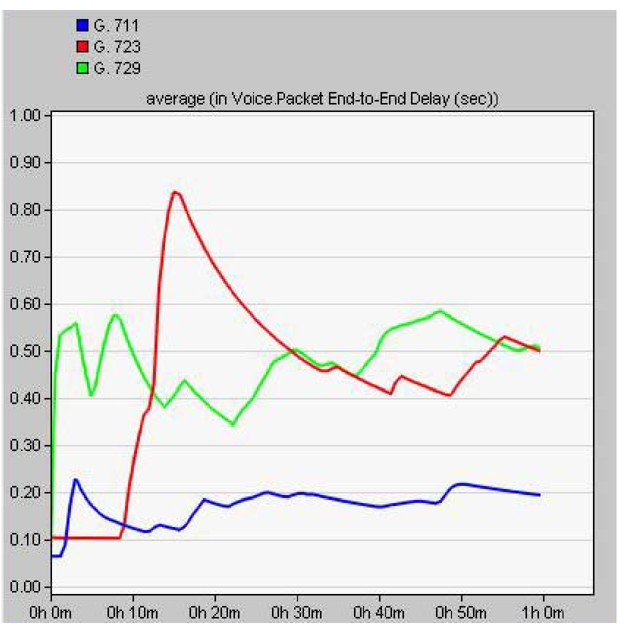


Figure 12: Wi-Fi: Average end-to-end delay.

4.2 VoIP over WiMAX

The performance of VoIP over WiMAX is tested using G.711, G. 723, and G. 729 codecs. The average MOS value for the three codecs is shown in Figure 14. Codecs G. 711 achieves the best MOS value of 4.35 followed by G. 723 and G. 729 with MOS values of 3.9 and 4.0, respectively. These values are acceptable, as indicated in Table 1.

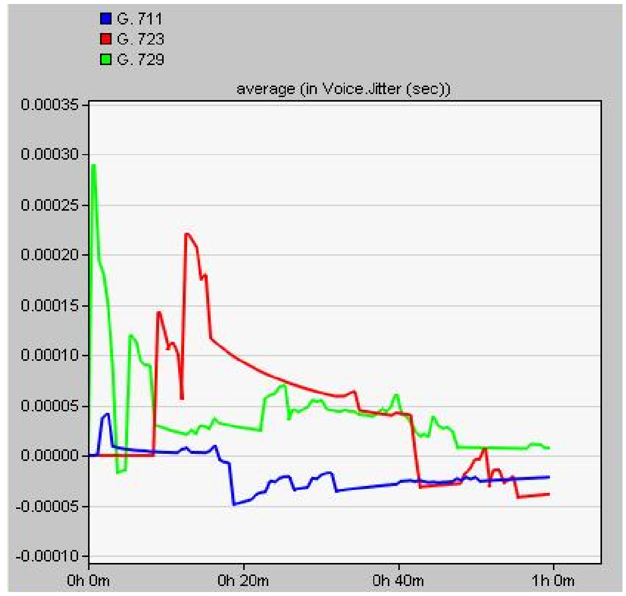


Figure 13: Wi-Fi: Average jitter.

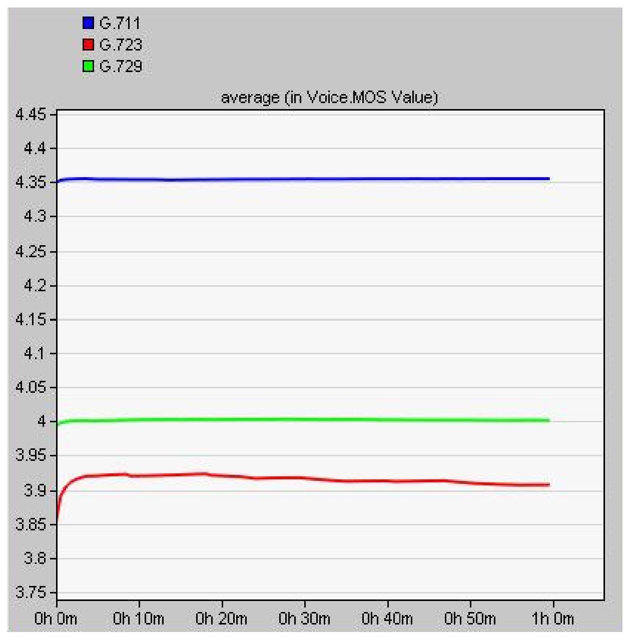


Figure 14: WiMAX: Average MOS.

All codecs have average end-to-end delays less than 140 ms as shown in Figure 15. They are in the range of a good voice connection. All three codecs experience very small jitter as shown in Figure 16. These simulation results indicate that G. 711, G. 723, and G.729 are appropriate for VoIP application over WiMAX.

The overall results indicate that the VoIP application performs better over WiMAX network than over Wi-Fi network. The WiMAX average end-to-end delay and average jitter are smaller than in case of Wi-Fi because WiMAX provides broadband service to support heavier traffic load over the network. Both Wi-Fi and WiMAX networks have similar MOS values.

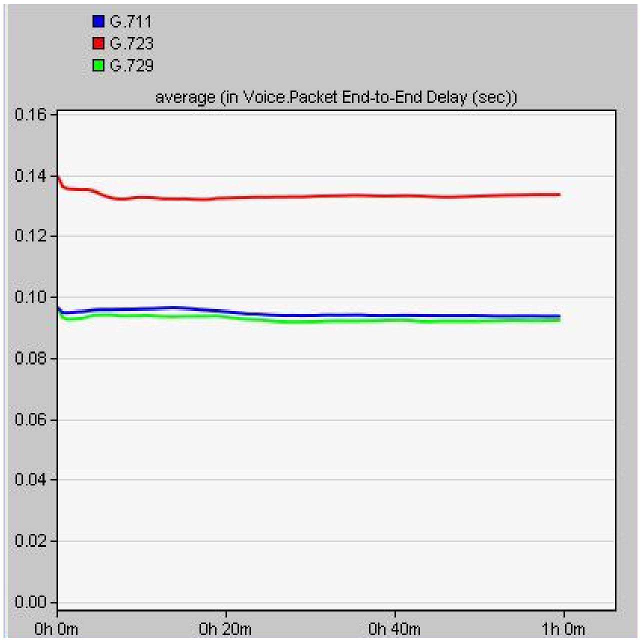


Figure 15: WiMAX: Average end-to-end delay.

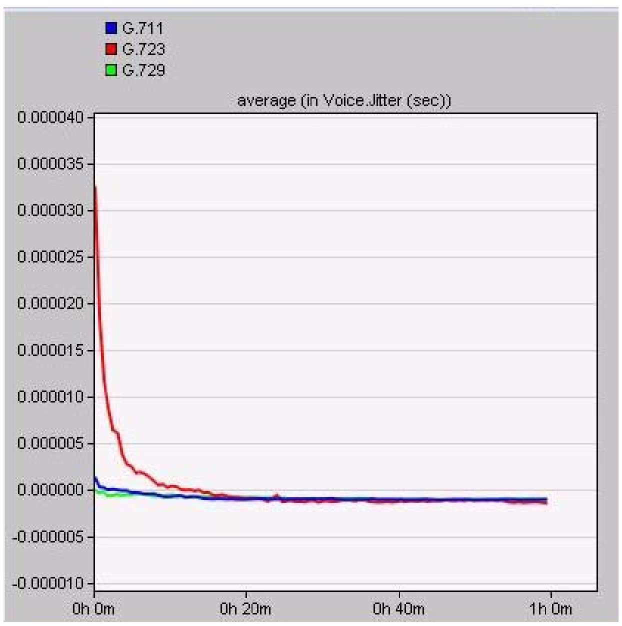


Figure 16: WiMAX: Average jitter.

5. Conclusions

In this paper, we evaluated the performance of three VoIP codecs over Wi-Fi and WiMAX networks. The VoIP performance was simulated via six simulation scenarios using OPNET 16.0.A. We considered voice calls from fixed nodes. The MOS, average end-to-end delay, and jitter were used as performance parameters that define VoIP QoS. The G. 711 codec offers the best performance for VoIP over Wi-Fi. However, all three codecs G. 711, G. 723, and G. 729 show acceptable performance quality for VoIP over WiMAX. Since the mobility also affects VoIP performance, its impact could be examined further. Wi-Fi and WiMAX networks may also employ other VoIP codecs such as G. 722, G. 726, and G.728.

References

[1] S. Islam, M. Rashid, and M. Tarique, "Performance analysis of WiMAX/Wi-Fi system under different codecs," *International Journal of Computer Applications*, vol. 18, no. 6, pp. 13–19, Mar. 2011.

[2] W. Hrudey and Lj. Trajkovic, "Mobile WiMAX MAC and PHY layer optimization for IPTV," Special issue on Recent Advances in Simulation and Mathematical Modeling of Wireless Networks, *Mathematical and Computer Modelling*, Elsevier, vol. 53, pp. 2119-2135, Mar. 2011.

[3] M. A. Qureshi, A. Younus, M. Saeed, F. A. Sidiqui, N. Touheed, and M. S. Qureshi, "Comparative study of VoIP over WiMAX and Wi-Fi," *International Journal of Computer Science*, vol. 8, no. 1, pp. 433–437, May 2011.

[4] R. Gill, T. Farah, and Lj. Trajkovic, "Comparison of WiMAX and ADSL performance when streaming audio and video content," *OPNETWORK 2011*, Washington, DC, Aug. 2011

[5] I. Adhicandra, "Measuring data and VoIP traffic in WiMAX networks," *Journal of Telecommunications*, vol. 2, no. 1, pp. 1–6, Apr. 2010.

[6] S. Jadhav, H. Zhang, and Z. Huang, "Performance evaluation of quality of VoIP in WiMAX and UMTS," in *Proc. 12th International Conference on Parallel and Distributed Computing, Applications and Technologies (PDCAT2011)*, Gwangju, Korea, Oct. 2011, pp. 375–380.

[7] M. I. Tariq, M. A. Azad, R. Beuran, and Y. Shinoda, "Performance analysis of VoIP codecs over BE WiMAX network," in *Proc. 3rd International Conference on Computer and Electrical Engineering (ICCEE 2010)*, Chengdu, China, Nov. 2010, pp. 47–51.

[8] E. Haghani and N. Ansari, "VoIP traffic scheduling in WiMAX networks," in *Proc. Global Telecommunications Conference (GLOBECOM 2008)*, New Orleans, LA, USA, Dec. 2008, pp. 1–5.

[9] R. S. Ramakrishnan and P. V. Kumar, "Performance analysis of different codecs in VoIP using SIP," in *Proc. Conference on Mobile and Pervasive Computing (CoMPC2008)*, Chennai, India, Aug. 2008, pp. 142–145.

[10] J. Yu and I. Al-Ajarmeh, "Call admission control and traffic engineering of VoIP," *The Second International Conference on Digital Telecommunications (ICDT 2007)*, San Jose, California, USA, July 2007, pp. 11–16.

[11] V. Toncar, "VoIP basics: overview of audio codecs." (April 2012). [Online]. Available: http://toncar.cz/Tutorials/VoIP/VoIP_Basics_Overview_of_Audio_Codecs.html.

[12] VoIP codecs. (April 2012). [Online]. Available: http://www.voip-sip.org/voip-codecs.

[13] A. M. Noll, "Cybernetwork technology: Issues and uncertainties," *Communications of the ACM 39*, vol. 39, no. 12, pp. 27–31, Dec. 1996.

[14] OPNET Modeler software. [Online]. Available: http://www.opnet.com/products/modeler/home.html.

[15] V. Vassiliou, P. Antoniou, I. Giannakou, and A. Pitsillides, "Requirements for the transmission of streaming video in mobile wireless networks," in *Proc. The 16th International Conference on Artificial Neural Networks*, ICANN 2006, Athens, Greece, Sept. 2006, pp. 528–537.