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Performance Evaluation of VOIP Codecs Over Wi-Fi WIMAX

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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.***

Keywords: VOIP, WIMAX, WIFI, Codec, OPNET

1. Introduction

Recent Voice over IP (VoIP)applications such as Skype,Google Talk, and Face Time 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]. TheIEEE802.11 (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 75Mbps) 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 17.5. A simulator to analyze the QoS of VoIP application under various codecs.

2. Background

The performance of VoIP applications using various technologies have been addressed in the liter ature. Selection of the appropriate VoIP codec was in vestigated using the OPNET 16.0.Asimulator 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 to lerable 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].Thes imulation results show that the through puts 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 networksunderthreecodecs:G.711, G. 723,and G. 729. We use OPNET16.0.Atosimulateand analyze the QoS of VoIP performance. MOS, end-to-end delay, and jitter are examined as performance metrics.

a) VoIPOVERWI-FI

Wi-Fi is commonly used in residential, business, and public areas. It is not able that the perceived throughput in Wi-Fi does not match the real through put .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. Be side the high traffic generated by users, both protocols VoIP and Wi-Fi create large headers, which result in degraded VoIP performance [3]

b) 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].

c) 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.

3. Method

To evaluate the QoS in VoIP, we designed two models using OPNET Modeler16.0.A.ThefirstmodelisVoIPover 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 1Gbps. All links have 10% to 20% background traffic load. The IP cloud is connected to VoIP server as shown in Figure 1.

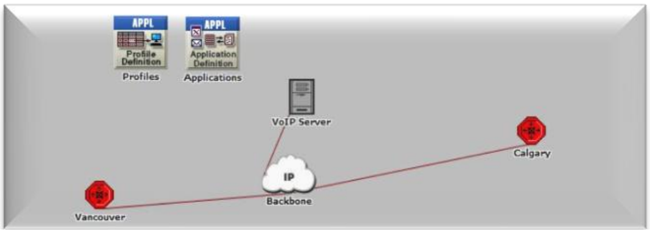

Figure 1: Wi-Fi network model.

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 IEEE802.11g protocol (54Mbps)as shown in Figure3.

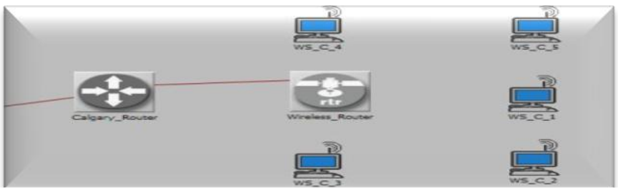

Figure 2: Calgary Wi-Fi subnet.

The wireless router provides connectivity to five clients in each subnet. These clients are located within a circle of radius 20m. 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].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

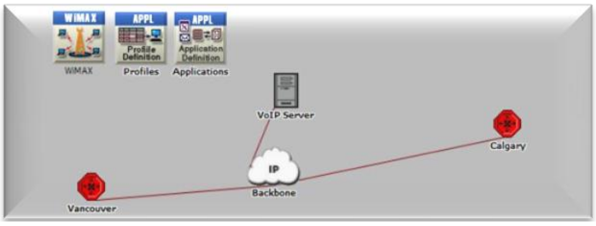

Figure 3: WiMAX network model.

The Calgary WiMAX subnet is shown in Figure 9. It consists of a main router and a BS. The BS connects five stations, which are located within a circle of radius 15 km.

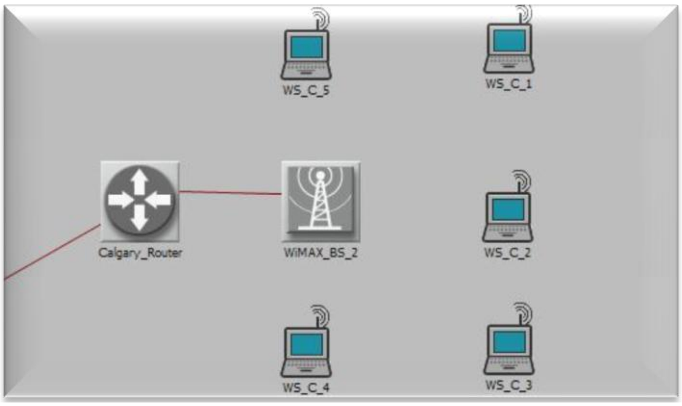

Figure 4: Calgary WiMAX subnet.

Asimulation 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.

Table 4: Simulation scenarios

Scenario	*Scenario Name*	*Codec*	*Clients Number*
1	VoIP overWi-Fi	G. 711	10
2	VoIP overWi-Fi	G. 723	10
3	VoIP overWi-Fi	G. 729	10
4	VoIP overWiMAX	G. 711	10
5	VoIP overWiMAX	G. 723	10
6	VoIP overWiMAX	G. 729	10

4. Results and Discussions

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 codec's (G. 711,G. 723, and G.729).

A. VOIPOVERWI-FI

Three simulation scenarios for VoIP over Wi-Fi are considered. The MOS values are shown in Figure5.G.711has the highs average MOS value of 4.35. Codecs G.723 and G.729 also have acceptable MOS values between 3.95 and 4.0, respectively.

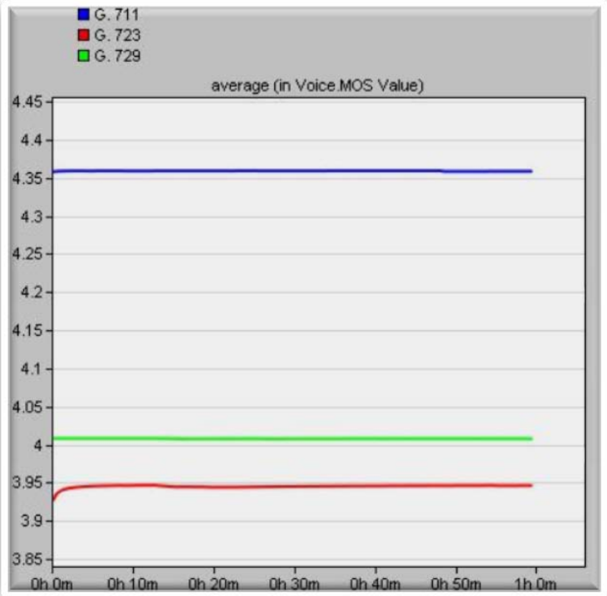

Figure5: Wi-Fi: Average MOS.

Although G. 711 has the highest data-rate, it shows the lowest average end-to-end delay as shown in Figure 6. However, average end-to-end delays of G. 723 and G. 729 are larger than300 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.

B. VOIPOVERWIMAX

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

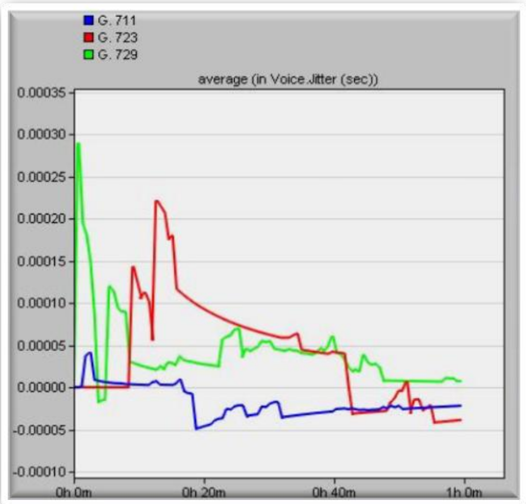

Figure 6: Wi-Fi: Average jitter

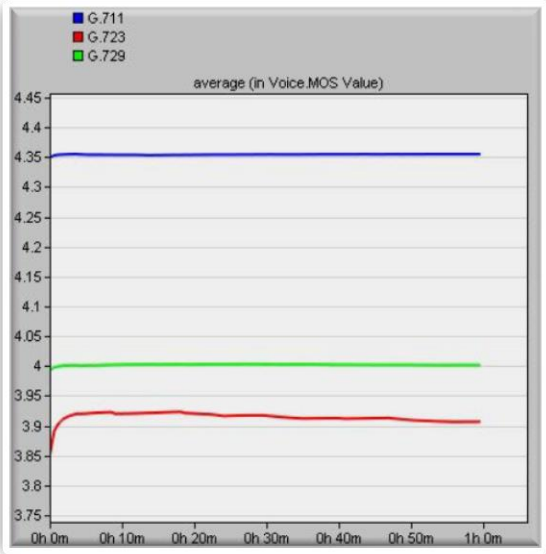

Figure 7: WiMAX: Average MOS

All codecs have average end-to-end delays less than 140 ms as shown in Figure 8. They are in the range of a good voice connection. All three codecs experience very small jitter as shown in Figure 9. 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.

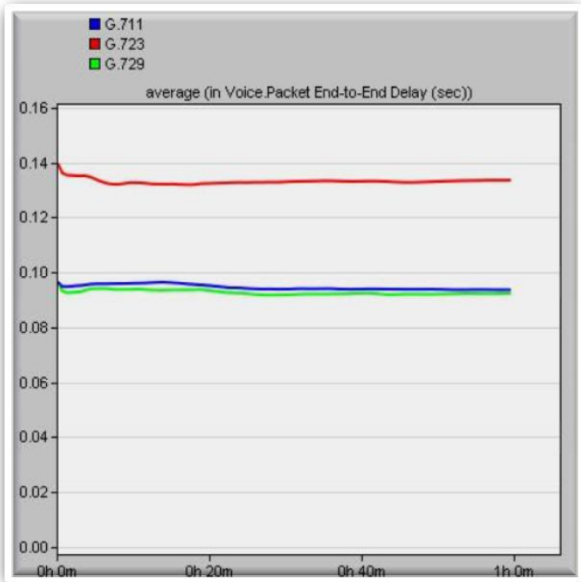

Figure 8: WiMAX: Average end-to-end delay

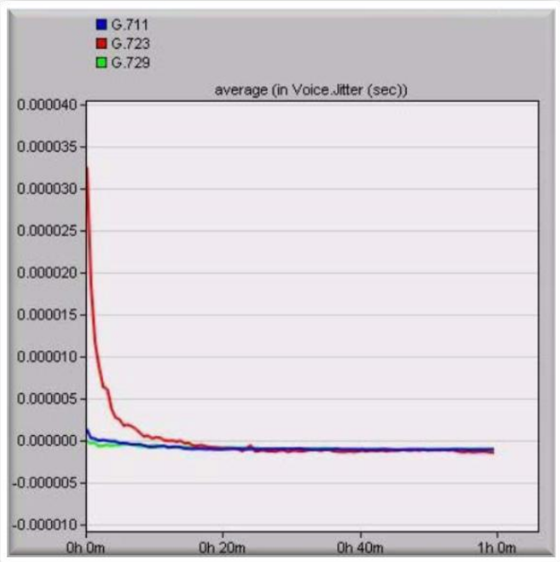

Figure 9: WiMAX: Average jitter.

5. Conclusion

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.

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