Performance Evaluation of Quality of VoIP in WiMAX and UMTS

Sheetal Jadhav, Haibo Zhang and Zhiyi Huang
Department of Computer Science
University of Otago, New Zealand
Email: {sheetal;haibo;hzy}@cs.otago.ac.nz

***Abstract*—Next Generation Wireless Networks (NGWNs) focus on convergence of different Radio Access Technologies (RATs) providing good Quality of Service (QoS) for applications such as Voice over IP (VoIP) and video streaming . The voice applications over IP networks are growing rapidly due to their increasing popularity. To meet the demand of providing high-quality of VoIP at anytime and from anywhere, it is imperative to design suitable QoS model. In this paper we conduct simulation study to evaluate the QoS performance of WiMAX and UMTS for supporting VoIP traffic. We designed simulation modules in OPNET for WiMAX and UMTS, and carried out extensive simulations to evaluate and analyze several important performance metrics such as MOS, end-to-end delay, jitter and packet delay variation. Simulation results show that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications compared with UMTS.**

I. Introduction

Recent advances in Internet technology have changed the way people communicate. With the rapid growth of wireless packet-switched networks, sending data through the Internet rather than the Public Switched Telephone Network (PSTN) has become a better option in terms of cost for users and service providers, leading to huge growth of voice applications over IP networks. With the new emerging set of mobile phones such as iPhones, VoIP has become a de facto standard for voice applications in the Internet. Mobile phone users can make a voice/video call through the Internet anywhere anytime with better communication quality and less cost than PSTN. With the telecom industry moving towards the next generation wireless networks which are going to provide high-quality service and higher down-link/up-link speed, VoIP continues to improve its QoS, especially for long distance calls. This improvement is going to impact businesses like call centers, multinational companies, as well as the normal users to a great extent than ever imagined.

An attractive wireless technology for VoIP is Worldwide Interoperability for Microwave Access (WiMAX) specified by IEEE 802.16 standard aimed at providing wireless access over long distances in a variety of ways from point-to-point communication to mobile cellular access. WiMAX provides wide coverage area with lower cost of network deployment. The coverage area of a single WiMAX cell is around 30 to 50 km, and its speed is up to 40 Mbps [1]. Moreover, WiMAX supports Quality of Service (QoS) by providing different service classes for both real-time and non-real-time traffic. Thus WiMAX is a very attractive technology for providing integrated voice, video services for VoIP.

Another emerging wireless technology is the development and deployment of Universal Mobile Telecommunications System (UMTS) as a part of 3G network. As a complete network system, it provides wider coverage and high mobility to fulfill the user demands in any places including office, home, urban and rural areas. UMTS supports packet-based applications including real-time multimedia applications such as VoIP with a peak down-link data rate of 14.4 Mbps.

Considering the current large deployment of WiMAX and UMTS networks and the promising integration of the two networks sooner or later, it is necessary to study their QoS differences and possible ways to resolve the differences between their QoS models. We believe that the classification of different QoS requirements from the real-time multimedia applications will help choose the best available network without degrading the QoS of the applications. These classifications can be used for implementing the resource scheduling system of UMTS and WiMAX when they are integrated. For example, depending upon the network congestion and available resources, a call can be transferred from UMTS to WiMAX or vice versa to provide better QoS to the user.

In this paper, we take VoIP as an application scenario to study the differences of QoS between UMTS and WiMAX, in order to investigate how well these two networks cope with real-time multimedia applications. This study will help identify the strengths and weaknesses of the two networks in terms of QoS and can guide the applications to choose the best available network in a heterogeneous environment. We have designed and implemented WiMAX and UMTS simulation modules in OPNET and carried out extensive simulations to analyze the Mean Opinion Score (MOS), packet end-to-end delay, jitter and packet delay variation for different type of VoIP traffic in these two networks. Our simulation results show that WiMAX has better QoS to support VoIP compared with UMTS.

The rest of the paper is organized as follows. Section II briefly gives background VoIP, UMTS and WiMAX. Section III deals with the simulation setup used in OPNET for both UMTS and WiMAX. Section IV evaluates and analyzes the simulation results of the VoIP application running on UMTS and WiMAX. Section V discusses the related work. Finally, in Section VI we conclude this paper.

II. Background

A. VOIP

VoIP community uses Session Initiation Protocol (SIP) protocol for signaling. SIP is an RFC (Request For Comment) standard from the IETF (Internet Engineering Task Force), responsible for administering and developing protocols that define the Internet. SIP translates the user name to the current network address, manages the call admission, dropping, or transferring mechanisms, allows for changing the features of a session, etc. Another popular protocol for a voice/video call on an IP network is H.323. VoIP is one of the most common and cheap technology to communicate for short and long distances [2]. Many VoIP providers also offer the service free of charge regardless of the distance. The analog voice data is digitized and transferred as packets over the IP network. These packets are decoded and converted back to the analog voice signal. Detailed VoIP illustration in UMTS and WiMAX is explained in Section III.

B. UMTS and WiMAX

UMTS is proposed to converge packet-switched and circuit-switched networks. Its IP Multimedia System (IMS) is used for multimedia communications. IMS was originally defined by the Third Generation Partnership Project (3GPP) for the next generation mobile networking applications and uses SIP as the signaling protocol. With the availability of UMTS, more and more phones can use different wireless networks other than WiFi. One example is Fring [3] that uses VoIP over UMTS.

So far, four service types have been proposed and incorporated into the QoS model of UMTS:

- Conversational class - for voice/video telephony, with low end-to-end delay and low jitter, two-way
- Streaming class - for streaming video, with low jitter, one-way
- Interactive class - for web browsing, with low loss/error rate, two-way
- Background class (Best Effort) - for email and background download, with low loss/error rate, one-way

The WiMAX wireless technology is called the last-mile solution for wireless broadband access. One of the features of the MAC (Media Access Control) layer of WiMAX is that it is designed to differentiate services among traffic categories with different multimedia requirements [4]. WiMAX offers some flexible features that can potentially be exploited for delivering real-time services. Though the MAC layer of WiMAX has been standardized, there are certain features that can be tuned for specific applications and channels. For example, the MAC layer does not restrict itself to fixed-sized frames, but allows variable-sized frames to be constructed and transmitted. This is very useful for framing VoIP packets [5, 6].

So far, five service types have been proposed and incorporated into the QoS model of WiMAX:

- Unsolicited Grant Service (UGS) - Supports real-time data streams.
- Real-time Polling Service (rtPS) - Supports real-time data streams.
- Non real-time Polling Service (nrtPS) - Supports delay tolerant data with variable packet sizes.
- Best Effort (BE) - Supports data streams where no minimum data rate is required and packets are handled based on available bandwidth.
- Extended real-time Polling Service (ertPS): Scheduling algorithm for VoIP services with variable data rates and silence suppression.

Both WiMAX and UMTS have advantages and disadvantages compared to each other. WiMAX is the first truly open mobile standard (IEEE802.16e) governed by the IEEE's fair licensing practices and open to participation. This is in fact revolutionary since 3GPP and 3GPP2 are consortium and do not allow open participation. This open process should lead to greater innovation and hence a better performance when moving forward and can potentially reduce intellectual property licensing fees, because it provides for a quicker improvement of the technology compared to existing mobile technologies. WiMAX is also the first major mobile standard to offer all-IP network. UMTS will get there in subsequent releases but it still employs a complicated and ultimately expensive core network [7]. Table I shows comparison of the two networks [8].

TABLE I
Comparison of WiMAX and UMTS

Paramater	WIMAX	UMTS
Peak down-link data rate	46Mbps with 3:1 DL-to-UL ratio TDD; 32Mbps with 1:1	14.4Mbps using all 15 codes; 7.2Mbps with 10 codes
Peak up-link data rate	7Mbps in 10MHz using 3:1 DL-to-UL ratio; 4Mbps using 1:1	1.4Mbps initially; 5.8Mbps later
Bandwidth	3.5MHz, 7MHz, 5MHz, 10MHz, and 8.75MHz initially	5MHz
Modulation	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM
Multiplexing	TDM/OFDMA	TDM/CDMA
Duplexing	TDD initially	FDD
Frequency	2.3GHz, 2.5GHz, and 3.5GHz initially	00/900/1,800/1,900/ 2,100MHz

III. Simulation Setup

To evaluate the performance of WiMAX and UMTS for VoIP traffic, we have designed and implemented WiMAX and UMTS simulation modules in OPNET network simulator [9] based on OPNET's discrete event simulation model library.

A. WiMAX Simulation Module

As illustrated by Figure 1, the WiMAX simulation module is composed of User Equipments (UEs), Base Station (BS), and Access Service Networks-Gateway (ASN-GW). The BS provides air interface to the UEs for VoIP call and is also responsible for tunnel establishment and radio resource management. The BS is connected to the ASN-GW which is responsible for connection management, location management, radio resource management, admission control, caching of subscriber profiles and AAA client functionality.

When a mobile user (e.g. UE0) makes a VoIP call to another user (e.g. UE1), the signaling protocols like H.323, SIP are used to setup the route for the transmission over the IP network. The call is authenticated at the AAA server, and special services will be granted based on the subscription of the user. A channel is then opened on which the actual media will travel using UDP for transport. The Gateway protocols like the Media Gateway Control Protocol are used to establish control and status in the media and signaling gateways. Routing (UDP, TCP) and transport protocols (RTP) are used once the route is established for the transport of the data stream.

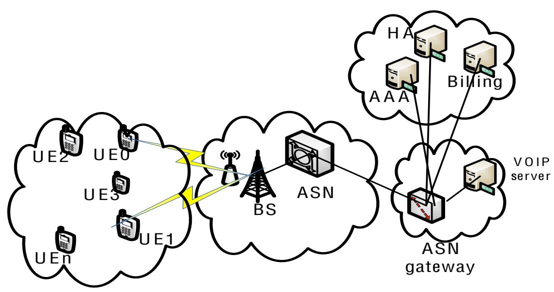

Fig. 1. VoIP in WiMAX

B. UMTS Simulation Module

As illustrated by Figure 2, the designed UMTS simulation module consists of the Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). CN provides routing, switching, and network management functions. The radio interface for UE is provided by URTAN which comprises of Radio Network Controller (RNC) and Node B (base station) [10]. A UE can be mobile handset, laptop, desktop or any device that can provide access to the network.

Similar to WiMAX, when a mobile user (e.g. UE0) makes a VoIP call to another user (e.g. UE1), the voice packets are carried from UE to RNC through Node-B over a protocol called Real-time Transport Protocol (RTP) which is encapsulated in User Datagram Protocol (UDP). First the Radio Resource Control (RRC) connection is established over the channel. Then, Radio Network Controller (RNC) sets up a point-to-point radio connection as well as the signaling connection to the network before sending acknowledgment back to the UE. The accounting information (time usage, type of service) and subscriber status is forwarded to the home AAA server for authorization, authentication and accounting of the call. After that, the UE will start the attach process. Then, the PDP context will be set up. The PDP context contains mapping and routing information for packet transmission between the UE, SGSN and the gateway GSN (GGSN). In UMTS, VoIP traffic is routed directly from the Gateway to the VoIP server. Voice signal is sampled, digitized, encoded, and decoded in UE. SIP is used as a signaling protocol in 3GPP. Pulse Code Modulation (PCM) quality voice has been generated over IP. The BE type of service and the weighted round-robin queuing have been selected. Figure 2 shows an overview of VoIP in UMTS.

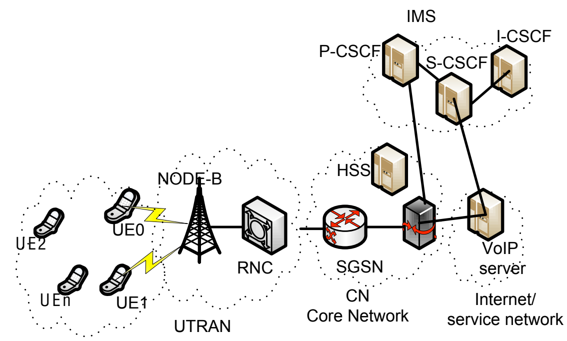

Fig. 2. VoIP in UMTS

C. Simulation Configuration

To make fair comparisons, we use the same VoIP configurations for UMTS and WiMAX Networks. Table II shows the setup. The service flow is designed to support best effort type of service with variable size data packets as VoIP with silence suppression. There are differnt codec such as G.711, G.721, G.722.etc. We use G.711 encoder scheme which supports PCM. We use one voice frame per packet as it increases the call handling capacity of the network compared to using two or three voice frames per packet. The algorithmic compression delay for G.711 is 0.02 and decompression delay is 0.02 seconds.

Pulse Code Modulation (PCM) quality voice has been generated over IP. The Best Effort (BE) type of service with bronze service class and initial QPSK modulation with 12 initial coding rate is used for the setup. The MAC address is distance based. The average Service Data Unit (SDU) is 120 bytes and the buffer has the size of 64KB.

TABLE II
WIMAX AND UMTS NETWORK PARAMETERS

Attribute	Values
Silence Length (seconds)	default
Talk Spurt Lenght (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Best Effort (0)
RSVP Parameters	None
Traffic Mix	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02

D. Performance Metrics

In our simulations, we use the following four metrics to evaluate the performance of WiMAX and UMTS in terms of end-to-end QoS for VoIP.

- **Mean Opinion Score (MOS):** MOS provides a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality. In our simulation, we compute MOS through a non-linear mapping from R-factor as in [11]:

$$MOS = 1 + 0.035R + 7 * 10^{-6} R(R - 60)(100 - R) \quad (1)$$

where $R = 100 - I_s - I_e - I_d + A$. I_s is the effect of impairments that occur with the voice signal; I_e is the impairments caused by different typesof losses occured due to codecs and network, and I_d represents the impairment caused by delay particularly mouth-to-ear delay. Using the default setting for I_s and A, Eqn 1 can be reduced to $R = 94.2 I_e I_d$.

- **Packet end-to-end delay:** The total voice packet delay is calculated as:

$$D_{e2e} = D_n + D_e + D_d + D_c + D_{de} \quad (1)$$

where D_n, D_e, D_d, D_c and D_{de} represent the network, encoding, decoding, compression and decompression delay, respectively.

- **Jitter:** In OPNET, jitter is computed as the signed maximum difference in one way delay of the packets over a particular time interval. Let $t(i)$ and $t'(i)$ be the time transmitted at the transmitter and the time received at the receiver, respectively. Jitter is calculated as follows:

$$jitter = \max_{i=1}^{n}([t'(n) - t'(n-1)] - [t(n) - t(n-1)]) \quad (2)$$

- **Packet delay variation(PDV):** PDV in OPNET is defined as the variance of the packet delay, which is computed as follows:

$$PDV = \frac{\sum_{i=1}^{n}([t'(n) - t(n)] - u)^2}{n} \quad (3)$$

where u is the average delay of the n selected packets.

IV. Simulation Results and Analysis

In this section we compare the performance of VoIP in WiMAX and UMTS through extensive simulations. To effectively analyze the performance, we measure the four metrics presented in Section III over a set of simulations with different number of homogeneous mobile users, i.e. all mobile users in the same simulation use the same configurations.

A. MOS

Figure 3 plots the average MOS with different number of homogeneous VoIP connections. A major observation is that the average MOS decreases with the increase of number of connections in UMTS, whereas the average MOS remains roughly steady in WiMAX irrespective of the number of VoIP connections. With 25 connections, the MOS in WiMAX is almost 3 times larger that that in UTMS. As can be seen from Eqn (1), the higher the packet loss rate, the lower the MOS value. This indicates that, compared with UMTS, WiMAX has less congestions, less traffic burst and better bandwidth allocation strategies, and thus low packet loss rate. It is clear that WiMAX can provide better voice quality than UMTS, especially in scenarios with large number of VoIP connections.

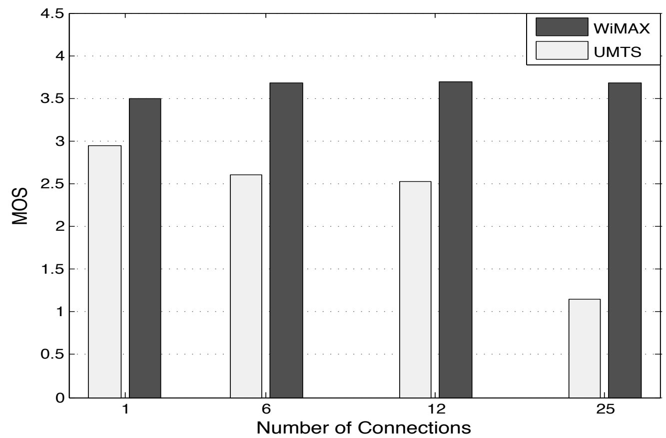

Fig. 3. MOS

B. Packet End-to-End Delay

Packet end-to-end delay is one of the most important performance metric in VoIP. Figure 4 and Figure 5 show the average packet end-to-end delay in UMTS and WiMAX, respectively. We plot the measurement from the time (i.e. 160 sec) when the communications become stable as it takes some time to set up the VoIP connections. As can be seen from the figures, the average delay in WiMAX is much more steady that that in UMTS. With 25 homogeneous connections, WiMAX has a maximum average packet end-to-end delay of 0.12 second, which is more than 50% less than the average delay in UMTS. When the number of connections is increased from 2 to 6 and 12, the average end-to-end delay is increased by 33% and 25%, respectively in UMTS, whereas in WiMAX it is increased only by 14.2% and 12.5%. These simulation results indicate that WiMAX can provide better VoIP services in terms of end-to-end packet delay. The reason is that WiMAX is an all-IP network, whereas UMTS is still a combination of circuit and packet switched technologies. A VoIP call in UMTS has to go through a selection procedure to choose the circuit switched network or the packet switched network, which takes considerable amount of time contributing to the end-to-end delay of the network. The above results of QoS will immensely help in deciding RAT for multiple-interface mobile devices which will become common rather sooner than later.

C. Jitter

According to Eqn (2), the jitter value can be negative which means that the time difference between the packets at the destination is less than that at the source. Figure 6 and Figure 7

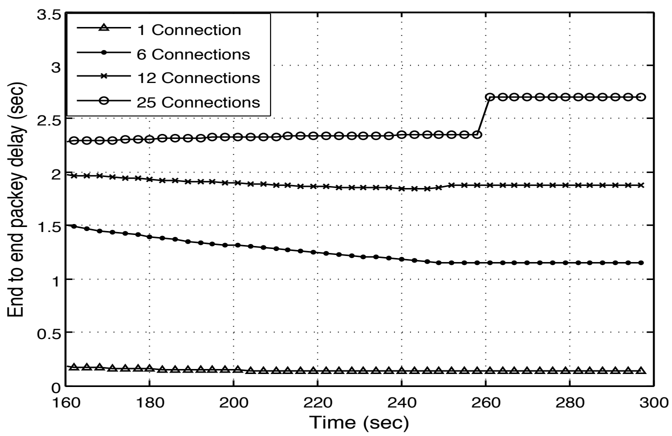

Fig. 4. Packet end-to-end delay in UMTS

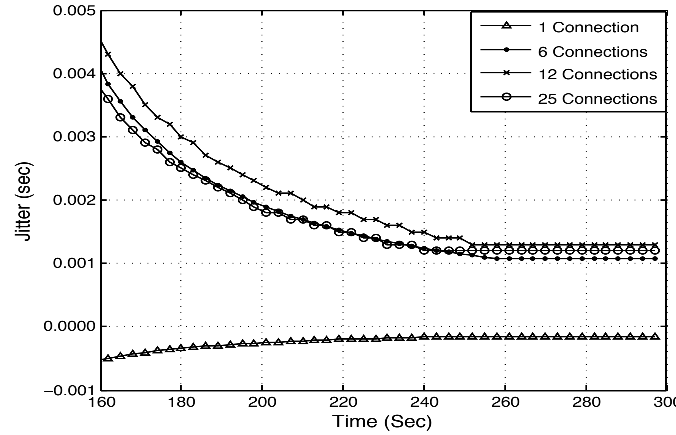

Fig. 6. Jitter in UMTS

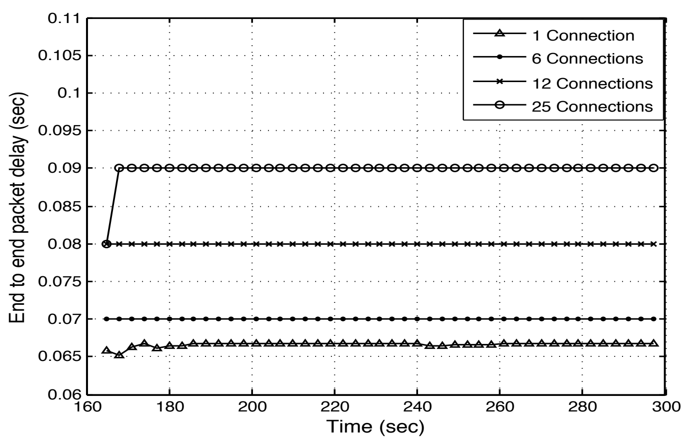

Fig. 5. Packet end to end delay WiMAX

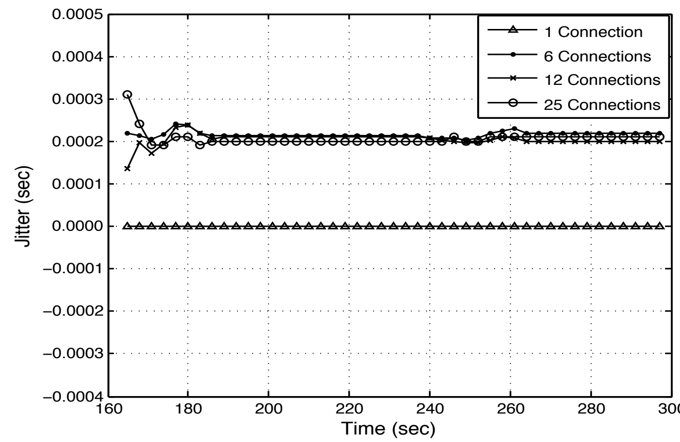

Fig. 7. Jitter in WiMAX

plot the jitter in UMTS and WiMAX, respectively. It can be seen that UMTS has a large range of jitter variation, ranging from -0.0005 to 0.0045, and takes longer time to converge to the stable stage. For WiMAX, it has a narrow range between 0.0000 to 0.0003, accounting for only 6% of that for UMTS. Moreover, it has a fast convergence to the stable state. This phenomenon can be explained as follows: as the number of users increases in UMTS, the congestion in the system also increases due to the slow packet scheduling. The multimedia sessions such as streaming will end up in more increased time for buffering the online videos. This will cause delay in transferring packet at receiving terminal, thereby leading to a poor quality video for the user. The two figures show a maximum jitter value of 0.0045 for UMTS and 0.0003 for WiMAX, respectively. The jitter factor is also a reason for an decreased MOS value for the UMTS.

D. Packet Delay Variation

Packet delay variation plays a crucial role in the network performance degradation and affects the user-perceptual quality. Higher packet delay variation results in congestion of the packets which can results in the network overhead. Figure 8 and 9 shows that, WiMAX is having a smaller delay variation of 0.00015 which can be tolerated because of buffering and jitter compensation within the voice decoder, thereby providing a stable QoS for the service. UMTS on the other hand is having a larger delay variation of 0.21, and this results in disturbed QoS particularly in streaming services.

V. Related Work

Most of the existing work was done to evaluate the performance of VoIP in WiMAX and UMTS, respectively. The authors in [12, 13] evaluated the capacity of VoIP services on High-Speed Down-link Packet Access (HSDPA), in which frame-bundling is incorporated to reduce the effect of relatively large headers in the IP/UDP/RTP layers. This work concludes that the capacity of VoIP service on HSDPA is attractive for transmission of voice. The work in [14] analyzes the efficiency of resource utilizationand VoIP capacity in IEEE 802.16e, and shows that UGS and rtPS algorithms have some problems to support VoIP, such as the waste of up-link resources in the UGS algorithm and the additional access delay and MAC overhead due to bandwidth request process in the rtPS algorithm. It is stated that ertPS algorithm

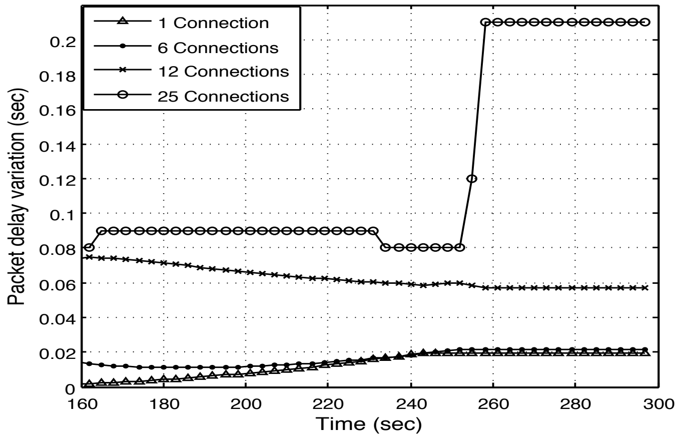

Fig. 8. Packet Delay Variation in UMTS

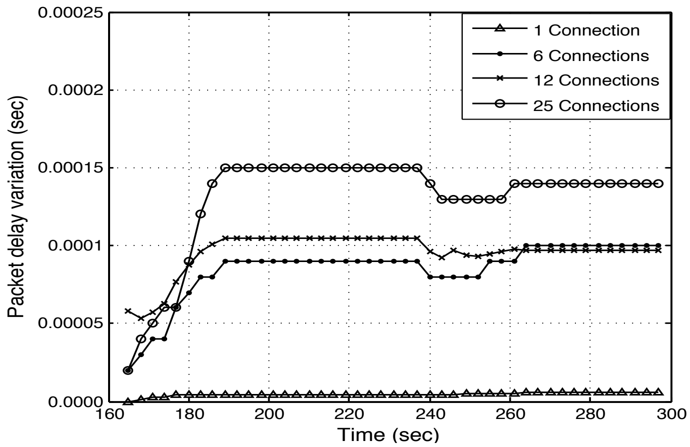

Fig. 9. Packet Delay Variation in WiMAX

can support 21% and 35% more voice users compared with the UGS and rtPS algorithms. The authors in [15] examine QoS deployment over cellular WiMAX network, and compare the performance of VoIP application using two different QoS configurations (UGS and ertPS). Their results show that ertPS has advantages in scenarios with delay-sensitive traffic. The work in [16] reports on the measurements on a real WiMAX network through synthetic VoIP traffic generation. Although some work has been devoted to understand different QoS models of a particular network with respect to VoIP, there is not much work on comparing the performance of QoS of VoIP traffic in different networks. Different with the existing work, we focus on evaluating QoS parameters for VoIP on two popular and widely deploying networks, UMTS and WiMAX, which will be inter-operating with each other in the near future.

VI. CONCLUSIONS AND FUTURE WORK

Next generation networks with multiple technologies offer different multimedia services to the user. It also provides the luxury of utilizing the best available technology for the required service to a user, companies and business organizations. In this study we have conducted extensive simulation study to evaluate the performance of WiMAX and UMTS for supporting VoIP traffic. We have analyzed several important critical parameters such as MOS, end-to-end delay, jitter and packet delay variation. Simulation results show that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications, compared with UMTS. This study is our first step towards exploring possible implementations of the next generation wireless networks. Future work includes the suitable model for mapping of QoS between UMTS and WiMAX, and the auto-configuration mechanism for the guarantee of QoS requirement during network switching.

REFERENCES

[1] M. Chakraborty and D. Bhattacharyya, "Overview ofend-to-end wimax network architecture," *WiMAX Security and Quality of Service: An End-to-End Perspective*, 2010.
[2] A. V. . T. Velte, *Cisco: A Beginner's Guide*, 4th ed., 2006.
[3] D. Shinder. (2007, November) Mobile voip. [Online]. Available: http://www.fring.com/
[4] C. Cicconetti, C. Eklund, L. Lenzini, and E. Mingozzi, "Quality of service support in ieee 802.16 networks," *IEEE Network Magazine*, vol. 20, no. 2, May 2006.
[5] *Voice over Internet Protocol. Definition and Overview*.
[6] A. Ghosh, D. Wolter, J. Andrews, and R. Chen, "Broadband wireless access with wimax/802.16: Current performance benchmarks and future potential," *IEEE communications magazine*, vol. 43, no. 2, p. 129, 2005.
[7] [Online]. Available: http://www.visantstrategies.com/analysis/wimax_advantage.html
[8] J. Andrews, A. Ghosh, and R. Muhamed, *Fundamentals of WiMAX*. Prentice Hall, 2007.
[9] [Online]. Available: www.opnet.com
[10] H. Holma and A. Toskala, Eds., *HSDPA/HSUPA for UMTS*, 1st ed., 2006.
[11] "The e-model, a computational model for use in transmission plannning. itu-t recommendation g.107," May 2000.
[12] Y. Kim, "Voip service on hsdpa in mixed traffic scenarios," in *Computer and Information Technology, 2006. The Sixth IEEE International Conference on*, 2006.
[13] P. Lunden, J. Aijanen, K. Aho, and T. Ristaniemi, "Performance of voip over hsdpa in mobility scenarios," in *Proceedings of the 67th IEEE Vehicular Technology Conference*, May 2008, pp. 2046 – 2050.
[14] H. Lee, T. Kwon, D. H. Cho, G. Lim., and Y. Chang, "Performance analysis of scheduling algorithms for voip services in ieee 802.16 e systems," in *IEEE 63rd Vehicular Technology Conference*, 2006, pp. 1231 – 1235.
[15] I. Adhicandra, "Measuring data and voip traffic in wimax networks," *Arxiv preprint arXiv:1004.4583*, 2010.
[16] N. Scalabrino, F. De Pellegrini, I. Chlamtac, A. Ghittino, and S. Pera, "Performance evaluation of a wimax testbed under voip traffic," in *WiNTECH '06: Proceedings of the 1st international workshop on Wireless network testbeds, experimental evaluation & characterization*. New York, NY, USA: ACM, 2006, pp. 97–98.