Journal of Mobile Multimedia, Vol. 9, No.3&4 (2014) 230-241

ANALYSIS OF VOIP AND VIDEO TRAFFIC OVER WIMAX USING DIFFERENT SERVICE CLASSES

TARIK ANOUARI

Computer, Networks, Mobility and Modeling laboratory

Department of Mathematics and Computer

FST, Hassan 1st University, Settat, Morocco
tarik.anouari@gmail.com

ABDELKRIM HAQIQ

Computer, Networks, Mobility and Modeling laboratory

Department of Mathematics and Computer

FST, Hassan 1st University, Settat, Morocco

e-NGN Research group, Africa and Middle East
ahaqiq@gmail.com

WiMAX, the acronym for Worldwide Interoperability for Microwave Access is a set of technical standards based on IEEE 802.16 standard. It provides wireless connection of companies or individuals over long distances at high speed, it's an adequate response to some rural or inaccessible areas. Unlike DSL (Digital Subscriber Line) or other wired technology, WiMAX uses radio waves and can provide point-to-multipoint (PMP) and Mesh modes. In parallel, Voice over IP (VoIP) technology is the biggest revolution in communication technology. It replaces the traditional telephone service and offers free long distance calls. Video stream traffic is characterized by the ability to transmit real-time and interactively visual and auditory information. VoIP and Video traffic are highly delay intolerant and need a high priority transmission. In this paper, we analyze the performances of the most common VoIP codec, namely G.711, and video streaming H.263 format using BE, rtPS, UGS, ertPS and nrtPS service classes and NOAH routing protocol. NS-2 simulator is used to analyze the QoS parameters. Our objective is to analyze different WiMAX service classes with respect to the QoS parameters such as, average jitter, throughput and average delay while increasing mobile nodes.

Key words: WiMAX, VoIP, Video Stream, BE, rtPS, UGS, ertPS, nrtPS, NS2.

1 Introduction

Many related works were published on analyzing different QoS parameters using WiMAX service classes. In [12], Mohamed, Zaki and Elfeki evaluate the performance of different VoIP codecs in WiMAX network with respect to network performance metrics such as MOS (Mean Opinion Score), packet end-to-end delay, jitter and packet delay variation. In [4], the impact of voice codec schemes and statistical distribution for VoIP in WiMAX has been analyzed. The simulation results show that better choice of voice codec and statistical distribution have important impact on VoIP performance in WiMAX network. In [3], the paper compares the performance of two different QoS service classes namely UGS and ertPS service classes. Joshi and Jangale [8] focus on analyzing the performance of different VoIP codecs using BE service class in WiMAX network with respect to Qos parameters such as throughput, average delay and jitter. Abid, Raja, Munir, Amjad, Mazhar and Lee [2] analyze the performance of WiMAX network when multimedia contents are transferred using BE and ertPS service classes. Vikram and Gupta [16] analyze the QoS parameters like jitter, thrpughput, delay, PDR (Packet delivery Ratio) and PLR (Packet Loss Ratio) in WiMAX network using UGS service class. In [18], simulation study was conducted to evaluate the user's QoE (Quality of Experience) when video is streamed from a source to a Mobile Station (MS) via a WiMAX Base Station (BS) in term of the following parameters, namely, the reserved rate at the BS for the video stream, the Modulation and Coding Scheme employed, the distance between BS and MS, and the tolerable end-to-end delay. Zhang, Hu, Le and Nguyen [19], evaluate the transmission performance of multimedia streams, especially SVC (Scalable Video Coding), in mobile WiMAX network by comparing the throughput and the packet delay in different scenarios, and count the frame loss of the received video. The simulation results indicate that, in terms of frame loss, the number of MS is critical to the performance of video transmission.

In the previous work [5], our simulation study was limited on analyzing QoS performance of VoIP traffic using UGS, BE and rtPS service classes in term of throughput, jitter and delay. In this paper we analyze both VoIP and Video traffic using different WiMAX service classes. We have reproduced the same simulation scenarios as in [5] to carry out the QoS parameters.

The rest of this work is organized as follows. Sections 2 and 3 give short descriptions of the WiMAX technology and the VoIP technology respectively. Section 4 describes the Video Streaming technology. Simulation environment and performance parameters are described in Section 5. Section 6 shows simulation results and analysis. Finally, Section 7 concludes the paper.

2 WiMAX Technology

WiMax is a set of technical standards based on the 802.16 standards [6, 7]. It's an alternative solution for the deployment of broadband networks in large areas, whether or not covered by other technologies such as DSL and can provides a high speed connection by radio waves.

WiMAX can be used in PMP connection: from a central base station, serving multiple client terminals is ensured and in point-to-point (P2P) mode, in which there is a direct link between the central base station and the subscriber.

PMP mode is less expensive to implement and operates while P2P mode can provide greater bandwidth.

2.1 QoS in WiMAX Networks

WiMAX technology natively implements the concept of QoS [14]. It may satisfy QoS requirements for a wide range of services and data applications especially with the high speed connection, asymmetric capabilities UL and DL and the flexible mechanisms for resource allocation. Some applications like Video streaming and VoIP require a short response time and cannot tolerate congestion in terms of throughput, transmission delay, jitter, packet loss and rate.

The concept of QoS obviously depends on the service considered, its requirements of response time, which is its sensitivity to transmission errors... etc. For video streaming, we will need a near real-time transfer, with very low latency and low jitter, while VoIP traffic is intolerant of network delay and retransmission.

A complete definition of QoS often refers to the mode of transport of information, although the solution adopted by the network to provide the service must remain transparent to the user.

Respecting QoS requirements becomes very important in IEEE802.16 systems to guarantee their performance, in particular in the presence of various types of connections, namely the current calls, new calls and the handoff connection.

2.2 WiMAX Network Architecture

WiMAX operates in infrastructure mode, it consists in a base station named BTS (Base Transeiver Station) or BS (Base Station) that sends to clients, receives their requests and forwards them to the network provider, it can provide various levels of QoS over its queuing, scheduling, control signaling mechanisms, classification and routing. Figure 1 shows the architecture of WiMAX network [6, 7].

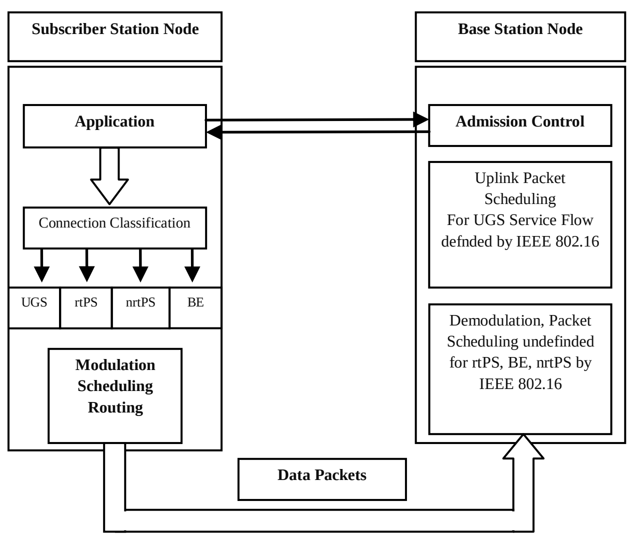

Figure 1 WiMAX Network Architecture

2.3 Different Service Classes in WiMAX

Each connection on the Uplink is mapped to a specific service. Each service is associated with a set of rules imposed by the scheduler of the BS responsible for assigning the capacity of the uplink and other parameters between SS and BS. Detailed rules and service use for particular connection uplink specifications are negotiated during the connection establishment. To satisfy different kinds of applications, WiMAX standard has defined four service classes of quality, namely BE, rtPS, nrtPS and UGS. The ertPS service class was added specifically for the mobile version [1].

Some services are very demanding in term of QoS, while others have fewer requirements. VoIP cannot tolerate delay in data transmission and so that for the video streaming.

Table 1 classifies different service classes of WiMAX and gives their description and QoS parameters.

Table 1. Service classes in WiMAX

Service	**Description**	**QoS parameters**
UGS	Real-time data streams comprising fixed size data packets at periodic intervals	Maximum Sustained Rate Maximum Latency Tolerance Jitter Tolerance
rtPS	support real-time service flows that periodically generate variable-size data packets	Traffic priority Maximum latency tolerance Maximum reserved rate
ertPS	Real-time service flows that generate variable-sized data packets on a periodic basis.	Minimum Reserved Rate Maximum Sustained Rate Maximum Latency Tolerance Jitter Tolerance Traffic Priority
nrtPS	Support for non-real-time services that require variable size data grants on a regular basis	Traffic priority Maximum reserved rate Maximum sustained rate
BE	Data streams for which no data minimum service level is required.	Maximum Sustained Rate Traffic Priority

3. VoIP technology

3.1. VoIP Transport System

VoIP is a technique that allows communicating by voice through the Internet or any other network supporting TCP/IP. It offers an alternative that works by routing digitized voice signals over IP networks, such as company’s intranet or internet in some cases. One of the principal advantages of VoIP is its capacity to reduce costs, since the calls are established through the data network instead of the network operator's telephony.

VoIP is based on several "standards", H.323 is now the most popular in VoIP, even if it’s gradually being replaced by the SIP protocol, both SIP and H323 protocols are the standards defined for signaling about Internet telephony. They have different approaches to solve the same problem. Those protocols are responsible for defining data formats, methods, dialog control infrastructure and terminals, as well as identifying interlocutors [9]. VoIP communications needs these signaling systems to signaling, codec negotiation, transporting information, establishment of the connection, control and ending a call.

SIP and H.323 use RTP (Real Time Transport Protocol) as the protocol to transfer multimedia data and the UDP (User Datagram Protocol) to carry the voice stream.

3.2. VoIP Codecs

The term "codec" is an abbreviation for (encoder-decoder). The voice streams are encoded at the transmitting end, sent over the network, and then decoded at the receiving end, where they are played through speakers or headphones.

The Codecs G.711, G.729 and G.723.1 are frequently used, although there are many others. The differences between these codecs consist in the energy required to perform the compression and decompression, and in the size of the compressed audio file or stream, which has an impact on the amount of bandwidth required to transport data between the both sides [9, 10, 20].

Below, a short description of the most used codecs:

• **G.711:** This codec is the first to be used in VoIP. The principle used is the coding of the signal according to a logarithmic scale. This codec produces a stream with a size of 64 kbps and that its MOS score obtained is for 4.2.

• **G.723.1:** This is the default codec when communicating at low flow, two modes are available. The first one provides rate of 6.4 kbps and the second a rate of 5.3 kbps, the mode can be changed during the communication.

• **G.729:** Unlike G.723 it still not used in Windows. The flow rate is 8 kbps. His MOS score is 4.0.

Table 2 gives some properties of the most used codecs: G.711, G.723.1 and G.729.

Table 2. Characteristics of VoIP Codecs

IUT-T Codec	Algorithm	Codec Delay (ms)	Bit Rate (kbps)	Packets Per Second	IP Packet Size (bytes)
G.711	PCM	0 .375	64	100	120
G .729A	ACELP	35	8	100	50
G.723.1	CS-ACELP	97.5	5.3	33	60

4. Video Streaming Technology

Video Streaming is a principle used essentially for sending content in near real time, A client media player can begin playing the data (such as a movie) before the entire file has been transmitted.

3.2. Video Stream Codecs

Video codecs encode stream or signal for transmission, storage or encryption of data. On the other hand, they can decode the stream or signal for editing or restitution.

H.263 is a recommendation of the video coding standard developed by ITU-T Q.6/SG16 (International Telecommunication Union). It was initially developed for the transmission of video at very low rates lines, for applications of videophone on public switched telephone network.

At the start of a video communication between two devices with this codec, they exchange their characteristics through H.245 [15] and they choose the modes of H.263 they use when communicating.

5. Simulation Environnement

5.1. Simulation Model

In this work, we analyze the performance of VoIP and Video traffic using WiMAX service classes, a 64kbps G.711 codec and H.263 data stream was used within the Network Simulator (NS-2) [11], they are commonly used for VoIP traffic and video-conferencing applications respectively. Our simulation scenario consists of creating a number of mobile nodes (SS, subscriber stations) and connecting them to a base station (BS). A sink node is created and attached to the base station to accept incoming packets. A traffic agent is created and then attached to the source node. For generating the video traffic we use raw video files (YUV files) from the video trace repository of Arizona State University [17]. A trace file (binary format) is generated and attached to the UDP agent as traffic source, this file contains information of the time and packet size.

The network simulator NS-2.32 was used, we have implemented the NIST (National Institute for Standards and Technologies) WiMAX 2.6 module patch [13]. Parameters as VoIP codec, Video format, number of mobile nodes and service class was passed while running the simulation scenario. For each service class under consideration, number of mobile nodes is varied from 2, 4, 6, 8 and 10. The main parameters used in our simulation are listed in table 3.

The generated trace files are interpreted and filtered based on a PERL script, it’s an interpretation scripts software used to extract datas from trace files in term of throughput, jitter and delay. The extracted analysis results are plotted in graphs using EXCEL software.

5.1. Simulation Parameters

Simulation parameters are shown in table 3:

Table 3. Simulation parameters

Parameter	**Value**
Network interface type	Phy/WirelessPhy/OFDMA
Propagation model	Propagation/OFDMA
MAC type	Mac/802_16/BS
Routing protocol	NOAH
Antenna model	Antenna/OmniAntenna
Link layer type	LL
Frame size (msec)	5
Duplex scheme	TDD
Packet Rate	4 packet/s
Modulation Technique	BPSK
Simulation time	200s

5.3. The Performance Parameters

Our simulation focuses on analyzing the main QoS parameters for WiMAX Network, namely average throughput, average jitter and average delay.

6 Simulation Results and Analysis

We have performed various simulation scenarios, the main objective is to analyse and compare the average throughput, average delay and average jitter of BE, rtPS, nrtPS, ertPS and UGS service classes using G.711 VoIP codec and H.263 video stream format .

The figures 2(a) and 2(b) show the variations of average throughput against the number of mobile nodes for VoIP and Video traffic respectively under various service classes. The Average throughput of VoIP traffic increases for all service classes as the number of nodes increases before arriving at six nodes, then it decreases. For the Video traffic, the average throughput increases for all classes before it decreases while reaching four nodes for rtPS and ertPS, six nodes for nrtPS and BE and eight nodes for UGS service class.

On the two figures, arriving at the fourth node, the average throughput of the rtPS and ertPS classes decreases quickly compared with the other service classes and has the lowest average throughput. Average throughput values of BE and nrtPS traffic are similar.

UGS service class gives better performances compared with the other service classes for both VoIP and Video traffic. The reason for this is that UGS service class has low percentage of packet drops, and the bandwidth mechanism used in rtPS (lot of overhead in requesting bandwidth from the base station).

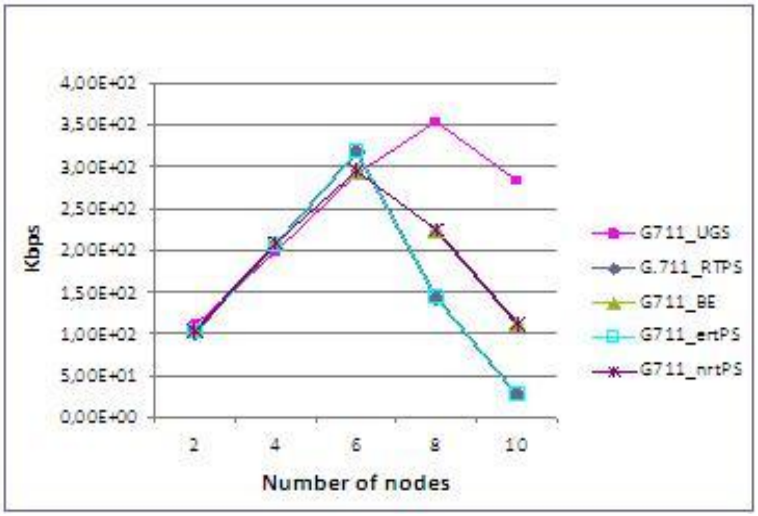

Figure 2(a): Throughput for VoIP traffic under various service classes

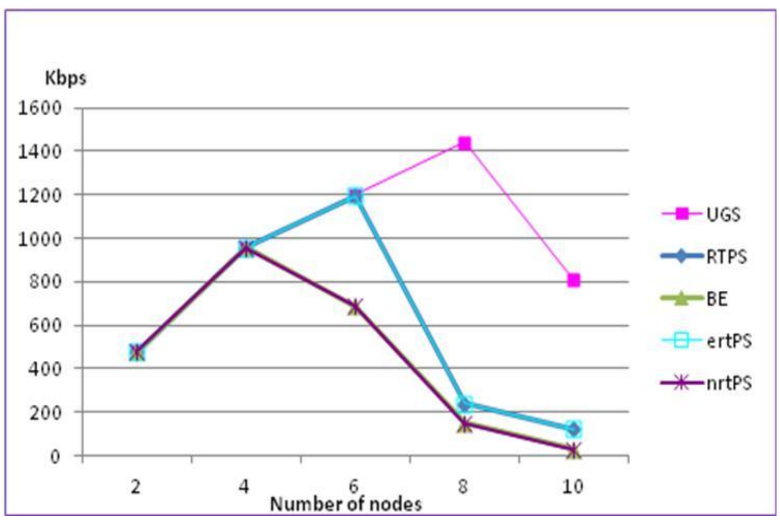

Figure 2(b): Throughput for Video traffic under various service classes

By analyzing the figures 3(a) and 3(b), BE and nrtPS service class has the highest jitter values. They have similar values as the number of nodes increases, the same thing is observed for ertPS and rtPS service classes.

On figure 3(a), average jitter values of all service classes under consideration increases from the sixth node while the rtPS and ertPS classes decrease from the eighth node. The average jitter values for UGS service class do not vary as much as the number of nodes increases compared with the other service classes. UGS flows are configured to send fixed size packets at regular intervals with minimal jitter. It's best suited for VoIP traffic.

On the 3(b), average jitter values of all service classes under consideration increases while increasing number of mobile nodes. rtPS and ertPS service classes has the lowest jitter. It can be concluded that the rtPS flow is best suited for Video traffic.

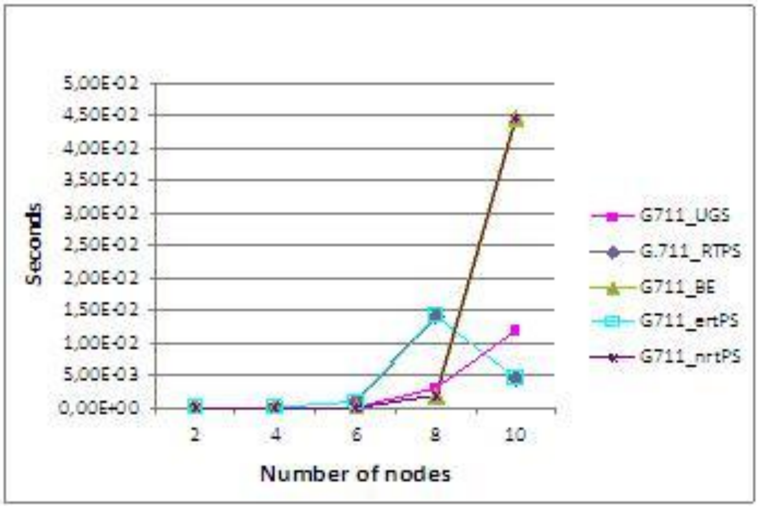

Figure 3(a): Average Jitter for VoIP traffic under various service classes

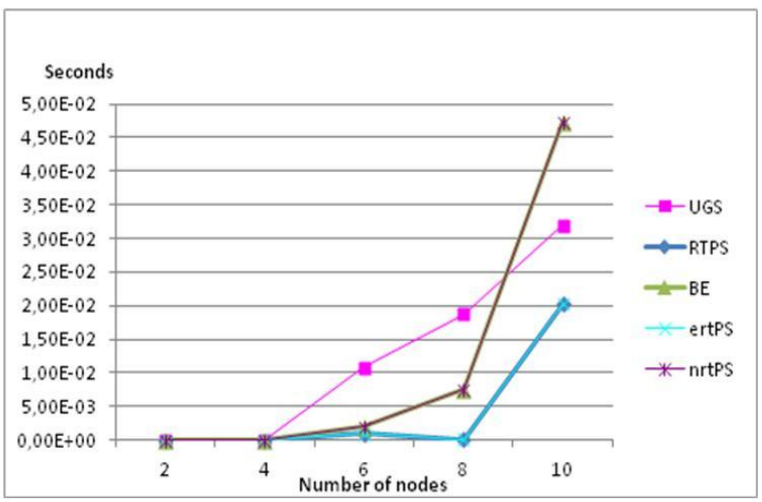

Figure 3(b): Average Jitter for Video traffic under various service classes

The figures 4(a) and 4(b) show the variations of average delay against the number of mobile nodes for VoIP and Video traffic respectively under various service classes. The average delay of rtPS and ertPS vary similarly and still identical as the number of nodes increases, the same behaviour is observed for BE and nrtPS.

On the 4(a) figure, rtPS and ertPS have the highest values of delay while they have the lowest delay values in the case of Video traffic. From node 2 to 6, average delay is insignificant for all service classes, but from the sixth node, the average delay of rtPS, ertPS, nrtPS and BE traffic increases quickly. However, the average delay values for UGS traffic keep insignificant in comparison to the others service classes.

From the 2(a), 3(a) and 4(a) figures, it's observed that UGS service class has the highest throughput, lowest average jitter and lowest delay. This makes it to be the most suitable for VOIP traffic. However from the 2(b), 3(b) and 4(b) figures, it's observed that rtPS and ertPS service classes have lower jitter and delay, UGS service class appears to perform better in terms of throughput. However UGS service class has the bandwidth already attributed to transmit data on a periodic basis, even when there is no data being sent. So the network resources are not effectively exploited with UGS service class for Video traffic.

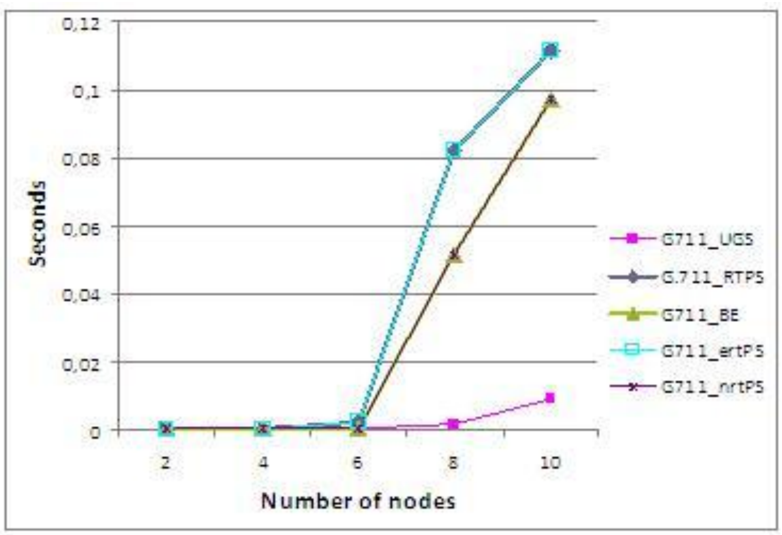

Figure 4(a): Average Delay for VoIP traffic under various service classes

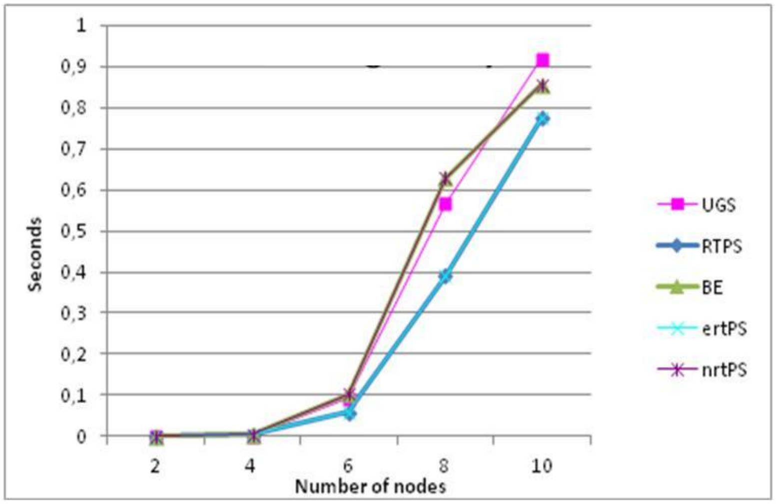

Figure 4(b): Average Delay for Video traffic under various service classes

7 Conclusion

In this paper, performances of BE, UGS, rtPS, ertPS and nrtPS service classes have been analysed using VoIP and Video traffic in terms of throughput, average jitter and average delay, G.711 VoIP codec and video streaming H.263 format were used.

For VoIP traffic, rtPS and ertPS perform better than nrtPS and BE service classes, UGS perform better than all the other service classes in term of jitter, delay and throughput. For Video traffic, rtPS and ertPS perform better than the other service classes in term of jitter and delay. In the case where the number of nodes does not exceed four nodes, all service classes show very similar performances in both VoIP and Video traffic. Simulation results show that performance parameters are impacted by changing the size of the network.

In conclusion, it's observed that UGS service class has the best performance parameters serving VoIP. In fact, UGS service class is dedicated to handle real-time service flows. The frames are generated in fixed sizes at regular interval, like for VoIP. In parallel rtPS and ertPS are the most suited service classes serving Video Streaming, which is variable bit rate traffic. The bandwidth can be periodically requested in the rtPS and ertPS service classes instead of fixed bandwidth already being allocated.

This performance analysis can be improved by using different mobility models with different WiMAX service classes.

References

1. Z. Abichar, Y. Peng and J. Morris Chang, "WiMax: The Emergence of Wireless Broadband", IT Professional, Vol. 8, Issue. 4, pp. 44-48, Doi:10.1109/MITP.2006.99, July-Aug. 2006.
2. H. Abid, H. Raja, A. Munir, J. Amjad, A. Mazhar, D. Lee, "Performance Analysis of WiMAX Best Effort and ertPS Service Classes for Video Transmission", ICCSA, Issue 3, pp. 368-375, 2012.

3. I. Adhicandra, “Measuring data and VoIP traffic in WiMAX networks”, Arxiv Preprint arXiv: 1004.4583, 2010.
4. S. Alshomrani, S. Qamar, S. Jan, I. Khan and I. A. Shah, “QoS of VoIP over WiMAX Access Networks”, International Journal of Computer Science and Telecommunications, Vol. 3, Issue 4, April 2012
5. T. Anouari and A. Haqiq, “Performance Analysis of VoIP Traffic in WiMAX using Various Service Classes”, International Journal of Computer Applications (0975 – 8887), Volume 52, No.20, August 2012.
6. IEEE 802.16-2004, “IEEE Standard for Local and Metropolitan Area Networks Part 16: Air Interface for Fixed Broadband Wireless Access Systems”, October, 2004.
7. IEEE standard 802. 16-2005, “IEEE standard for Local and Metropolitan Area Networks-Part16: Air Interface for Fixed and Mobile Broadband wireless Access systems Amendment 2” February 28, 2006.
8. D. Joshi, and S. Jangale, “Analysis of VoIP traffic in WiMAX using NS2 simulator” International Journal of Advanced Research in Computer Science and Electronics Engineering, Vol. 1, Issue 2, April 2012.
9. S. Karapantazis and F.-N. Pavlidou, “VoIP: A comprehensive survey on a promising technology”, Computer Networks, Vol. 53, Issue 12, pp. 2050-2090, 2009.
10. A. Kumar and S. Ganesh Thorenoor, “Analysis of IP Network for different Quality of Service”, International Symposium on Computing, Communication, and Control (ISCCC 2009) Proc. of CSIT Vol.1 (2011) ©IACSIT Press, Singapore, 2011.
11. Marc Greis, “Tutorial for Network Simulator NS”, http://www.scribd.com/doc/13072517/tutorial-NS-full-byMARC-GREIS.
12. M.A. Mohamed, F.W. Zaki and A.M. Elfeki, “Performance Analysis of VoIP Codecs over WiMAX Networks”, IJCSI International Journal of Computer Science Issues, Vol. 9, Issue 6, No 3, November 2012.
13. NIST: IEEE 802.16 module for NS-2 (Feb 2009), http://www.antd.nist.gov/seamlessandsecure/pubtool.shtml#tools.
14. P. Rengaraju, C.H. Lung, A. Srinivasan, R.H.M. Hafez, “Qos Improvements in Mobile WiMAX Networks”, AHU J. of Engineering & Applied Sciences, Vol. 3, Issue 1, pp. 107-118. ALHOSN University, 2010.
15. H. Schwarz, D. Marpe, T. Weigand, ―Overview of the scalable video coding extension of the H.264/AVC standard‖, IEEE Trans. Circuits Syst. Video Technol, vol. 17, no. 9, pp.1103-1120, Sept. 2007.
16. M. Vikram and N. Gupta, “Performance Analysis of QoS Parameters for Wimax Networks”, International Journal of Engineering and Innovative Technology, Vol. 1, Issue 5, May 2012.
17. “Video Traces Research Group of Arizona State University”, http://trace.eas.asu.edu, 2009.
18. A. Vishwanath, P. Dutta, M. Chetlur, P. Gupta, S. Kalyanaraman, A. Ghosh, “Perspectives on Quality of Experience for Video Streaming over WiMAX “,Mobile Computing and Communications Review, v. 13. n. 4, October 2009.
19. H. Zhang, M. Hu, T. A. Le, H. Nguyen, “Performance Evaluation of Video Streaming over Mobile WiMAX Networks”, GLOBECOM Workshops (GC Wkshps), 2010 IEEE, pp. 898 - 902 , France, 2010.
20. T. Zourzouvillys and E. Rescorla, “An introduction to standards-based VoIP: SIP, RTP, and friends”, Internet Computing, IEEE, Vol. 14, Issue 2, pp. 69 –73, 2010.