Performance Evaluation of VoIP in Mobile WiMAX; Simulation and Emulation studies

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***Abstract*— Worldwide Interoperability for Microwave Access (WiMAX) is an acronym for IEEE 802.16 family which is a leading contemporary broadband wireless Access (BWA) technology. IEEE 802.16e is intended for mobile WiMAX, which supports vehicular mobility with the stringent quality of service (QoS) parameters for various data traffics. Voice over IP (VoIP) provides low cost, modern telephony which can become a better alternative for classical telephony; however there are some issues need to be addressed prior to the deployment of any new technology. Significance of simulation study results can be verified and assessed by emulation testbed results. It is expected that both the results should match closely with each other. This paper makes an effort to study the performance evaluation of VoIP for a mobile user and how the QoS parameters vary for different speeds. The simulation and emulation of a mobile WiMAX system using EXata 2.0.1 are performed. The effectiveness of the comparison of results is discussed.**

Keywords- *emulation, IEEE 802.16e, simulation,WiMAX*

I. INTRODUCTION

WiMAX is a cutting edge technology which follows the IEEE 802.16 [1] family. WiMAX developed based on IEEE 802.16e [2] is capable of providing mobility to users for different data flows pertaining to their QoS requirements. This technology is a connection oriented system which is a centralized architecture. There are two modes of operations; point to multi point (PMP) and mesh mode and this paper concentrates on PMP mode. In PMP mode the data transmission between the MSSs is coordinated by BS. Two directions of communication paths exist between BS and MSS: uplink (from MSS to BS) and downlink (from BS to MSS). In this paper subscriber station (SS) and MSS are used interchangeably.

VoIP is an application which carries the real time voice data over the internet which is highly delay intolerant and needs a high priority transmission. Therefore there are strict set of QoS parameters like delay, jitter, mean opinion score (MOS) etc. which are essentially fulfilled for a meaningful and understandable voice transfer between a source and receiver in a BWA network. The purpose of this study is to examine the effect of mobility of a user on the performance of VoIP.

The performance evaluation of any new emerging technology with different applications is imperative in order to understand and improvise the system to the desired level. Simulation is not always a viable and reliable tool for performance testing and evaluation as simulation is a simple abstraction of the system [3]. On the other hand emulation is a high fidelity model of the physical system which could be a better alternative for simulation. In mobile WiMAX, a user is mobile and hence the channel conditions vary, one of the fundamental aspects of observation is the performance of different applications for different speeds. This paper intends to study the VoIP performance for a MSS under the aforementioned conditions using the strong performance evaluation techniques: simulation and emulation in EXata 2.0.1.

The related work in this area is discussed below; authors in [4] have done a work on VoIP transmission on ertPS. In paper [5] authors have evaluated the performance of different codecs for both WiMAX and WLAN scenario and for an integrated environment. Andres Arjona in [6] has focused on mobile VoIP performance in different BWA networks. Work in [7] also evaluates the performance of different VoIP codecs in a fixed best effort WiMAX network for varying number of VoIP flows. Author of [8] has evaluated the performance of voice and video using a real testbed for fixed nodes. K. Shuib in [9] has evaluated the performance of WiMAX network using Qualnet simulator but the traffic used was constant bit rate (CBR) traffic. Authors of [10] worked on performance evaluation of VoIP over ad hoc networks in emergency scenarios for different obstacle coverage. In papers [11-12] authors have focused on the improvement of VoIP quality by concentrating on R-factor.

The rest of this paper is organized as follows; section II explains about an overview of WiMAX, section III gives the details of WiMAX MAC layer. Section IV explains the VoIP concept. Section V and VI consolidates the simulation and emulation results. Finally Section VII concludes.

II. AN OVERVIEW OF WIMAX

The broadband wireless access (BWA) technology is gaining a tremendous attention of both researchers and academicians. WiMAX is a next generation network (NGN), which is highly potential and capable of fulfilling the users' QoS requirements. IEEE 802.16e-2005 [1] came out as an amendment to the existing IEEE 802.16d-2004 and later in 2009 [2] the further expansion was made. Users can experience ubiquitous computing meeting with the QoS requirements of different data flows. The physical layer of

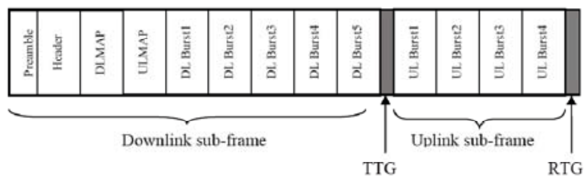


Figure 1. TDD frame structure

WiMAX is based on orthogonal frequency division multiplexing (OFDM) or orthogonal frequency division multiple access (OFDMA) technology. It adopts different modulation and coding schemes (MCS), hence follows adaptive modulation and coding (ACM) scheme as the received signal strength of a user varies in a cell. Initially the operation band of frequencies was 10-66GHz which was line of sight (LOS) and in the later amendments 2-11GHz band is used which is non line of sight (NLOS). All the communications between BS and MSS are regulated by BS. As it has been mentioned earlier, PMP mode of operation is mandatory and follows a centralized architecture. The communication is time division duplex based (TDD), multiple MSSs share a common uplink sub frame to transmit their burst of MAC protocol data units (PDU) to BS as per the requirement using time division multiple access (TDMA). In downlink sub frame BS transmits burst of MAC PDUs to MSSs using time division multiplexing (TDM).
Fig.1 explains the TDD frame architecture of WiMAX. Where transmit/receive transition gap (TTG/RTG) are intended to have short gap between uplink, downlink sub frames and between two successive frames. The downlink sub frame starts with a preamble followed by header, downlink map (DLMAP) and uplink map (ULMAP). Each uplink burst (UL burst) contains a preamble which is used for the synchronisation of BS with specific MSS.

Scheduling algorithms play a vital role in proper resource management and distributing the bandwidth among many users. A scheduling algorithm at BS decides the resource allocation between contending users for bandwidth and a scheduling algorithm at MSS or SS distributes the bandwidth between multiple connections depending on their QoS requirements. Having said that, it becomes very essential to evaluate and assess the performance of any new system with varying network parameters. This would help in designing a new system effectively and also help in developing new protocols and schedulers.

III. WIMAX MAC LAYER

The function of the MAC layer is to provide reliable packet transmission and the proper resource utilization for different type of multimedia traffic. The WiMAX MAC layer consists of service specific convergence sub layer which facilitates to and fro mapping for the MAC layer, various types of network layer protocols such as internet protocol (IP), asynchronous transfer mode (ATM), Ethernet, point to point protocol (PPP). However it also provides an interface to the different WiMAX PHYs. The MAC common part sub layer provides the complete MAC layer architecture; it is defined mainly for PMP mode of operation. Bandwidth request mechanism, admission control, scheduling of resources, QoS parameters mapping for a connection are defined for various data connections. A security support sub layer is also added to the MAC which offers authentication, secure key exchange, and encryption so on. Each connection contains 16 bits connection identifier (CID) which acts as a primary address for all the operations, 48 bits MAC address is mainly used as equipment identifier. Three types of management connections are assigned on entering a network for an each MSS viz. basic, primary and secondary.

GPC and GPSS are two ways of bandwidth requesting mechanisms; the earlier one requests the bandwidth when a connection occurs at MSS. In the recent amendments of WiMAX GPSS is a mandatory type, as it reduces the requesting and granting overhead of the BS and a MSS when there are multiple connections at MSS.

After getting grant from BS, a MSS should manage and redistribute the bandwidth among its connections according to the QoS requirements. There are five distinguished scheduling services have been defined by the standard to handle different multimedia traffic, which is designed to support their stringent QoS requirements.

1. Unsolicited grant service (UGS) is intended to support the real time data like voice over IP (VoIP) without silence suppression, T1/E1. This scheduling type has the highest priority and usually constant bit rate (CBR) data is mapped on UGS, BS allocates fixed size data grants without receiving explicit requests from a requesting entity. Grant is based on the maximum sustained data rate of the connection.

2. Extended real time polling service (ertPS) is designed for real time data, such as VoIP with silence suppression. Variable bite rate (VBR) data is mapped on this scheduling service, which is treated in the same manner as UGS. Bandwidth grant can be negotiated on the basis of maximum and minimum sustained data rate of a connection.

3. Real time polling service (rtPS) is meant for VBR real time data like MPEG video data. In this scheduling service requesting entities send a non contention based unicast request periodically and BS allocates a dedicated grant to the requesting stations. All the above services are real time services and hence are delay intolerant.

4. Non real time polling service (nrtPS) is designed to support VBR non real time services like file transfer protocol (FTP). In this service also MSS send a contention based unicast request and receives grant at longer intervals. This service is delay tolerant but requires being throughput optimal.

5. Best effort service (BE) is intended for the lowest priority non real time traffic like internet HTTP traffic, which does not require any QoS guarantee for its connection.

IV. Voice Over IP

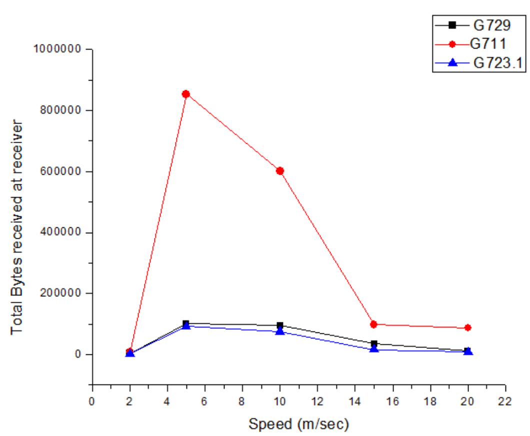
VoIP over WiMAX has been emerging as an infrastructure to provide voice communication over the internet. Voice is an analog signal which is converted to digital form before transmitting over the internet. At the transmitter side voice is encoded and at the receiver side the reverse process decoding is done, which can be accomplished by using a codec. The processed voice signal is communicated over RTP/UDP/IP protocol stack and at the receiver side it is de-packetized and reconstructed back. Perceived voice quality can be measured by a subjective quantity MOS, which varies from 1 (worst) to 5 (best) [4]. G711 codec uses a very high bandwidth and mainly used for classical telephony, G729 offers a very good capacity whereas G723.1 is an optimal multi rate codec which has good compression rate [13]. In this paper performance evaluation is carried out for above mentioned codecs.

V. Simulation and Results

In this paper EXata 2.0.1 [14] is used as the platform for all the performance studies. EXata is a comprehensive suite of tools for emulating large wired and wireless networks. It uses simulation and emulation to predict the behaviour and performance of networks to improve their design, operation, and management [14].

A. *Scenario 1*

In this scenario simulation area is 1500m x 1500m, BS and SS transmission power is set to 30dBm and 20dBm respectively. There are three SSs which are associated with a single BS; an MSS is transmitting a VoIP and file transfer protocol (FTP) to fixed SSs. The mobility model used for MSS is random mobility. Simulation time is set to 5 minutes; performance is evaluated choosing the codecs G711, G723.1 (5.3Kbps) and G729. TDD downlink and uplink time frames are adjusted as 2:1. Multiple simulations are performed and results are plotted for different speeds versus QoS metrics.



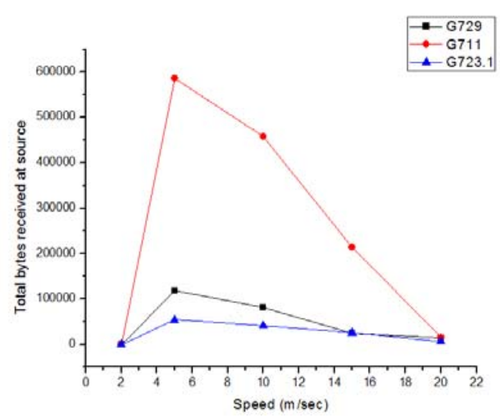


Figure 2. Total bytes received at VoIP receiver

Figure 3. Total bytes received at VoIP source

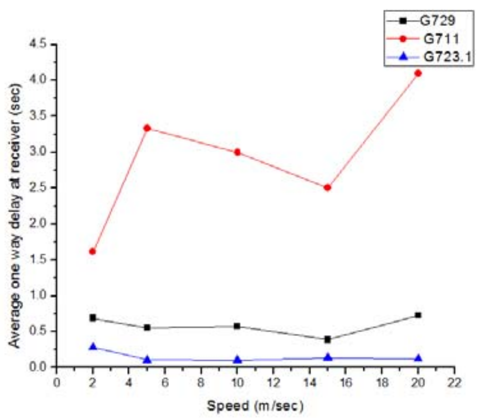


Figure 4. Average one way delay at VoIP receiver

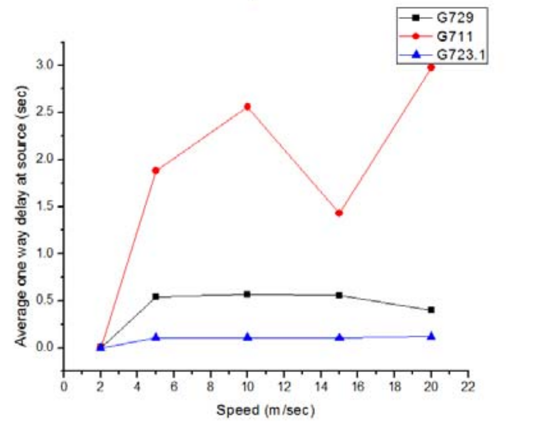


Figure 5. Average one way delay at VoIP source

Fig.2 and 3 reveals the total bytes received at receiver and source and observed that G711 has more in and out data because of its high bandwidth consumption. Figures 4 and 5 give the details of average one way delay at receiver and source. It can be seen that G711 has the maximum and G723.1 has the minimum delay.

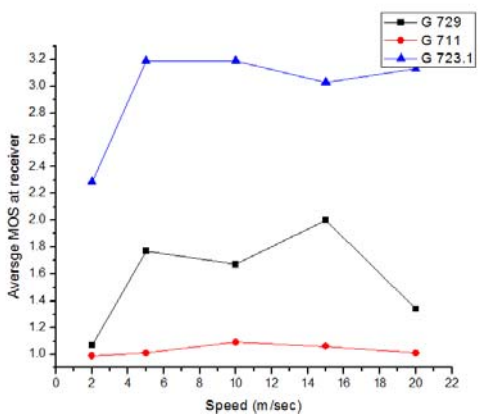


Figure 6. Average MOS of VoIP receiver

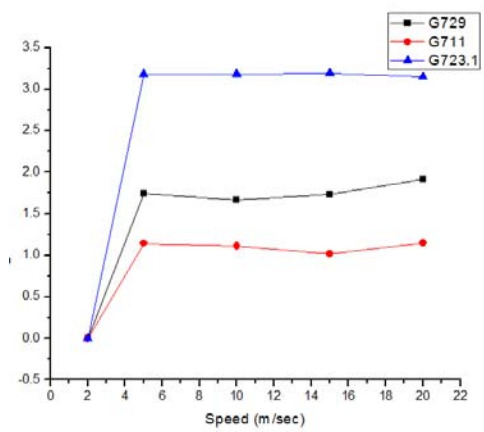


Figure 7. Average MOS of VoIP source

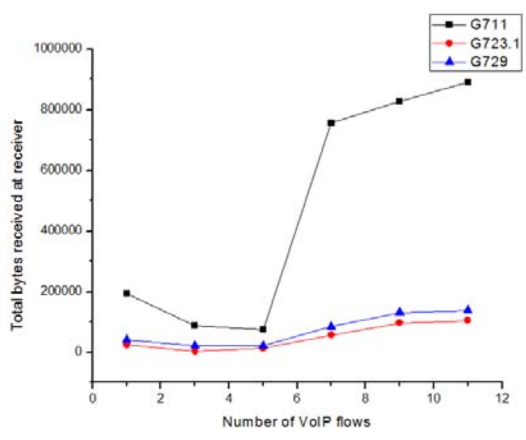


Figure 8. FTP Throughput at receiver and source

Fig. 6 and 7 conveys the average MOS at receiver and source, among all the chosen codecs G723.1 has the best MOS value and G711 has the worst MOS value Fig.8 details the information about FTP traffic; it is very obvious that FTP performance does not depend on codec selection and the speed of the nodes does not have much impact on the FTP performance.

B. Scenario 2

In this scenario only VoIP traffic is considered and all the other settings from scenario 1 are retained. Scenario is designed in such a way that VoIP source is mobile and receiver is fixed. MSS is moving with maximum speed of 15m/sec and minimum speed of 5m/sec, multiple simulations are performed by varying number of VoIP flows and number of SSs carrying them. The results are plotted for G711, G723.1 (5.3 Kbps) and G729 codecs.

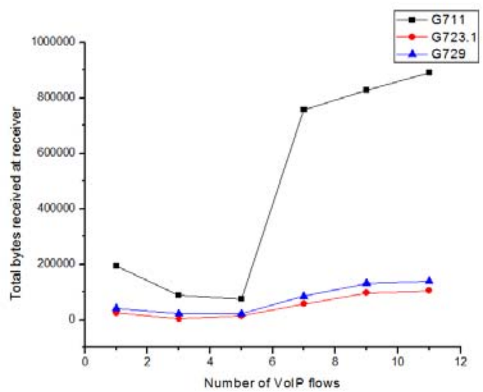


Figure 9. Total bytes received at VoIP receiver

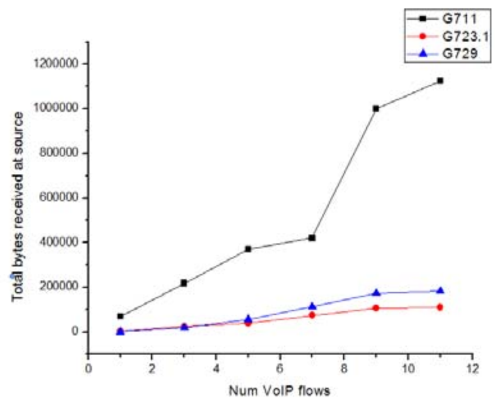


Figure 10. Total bytes received at VoIP source

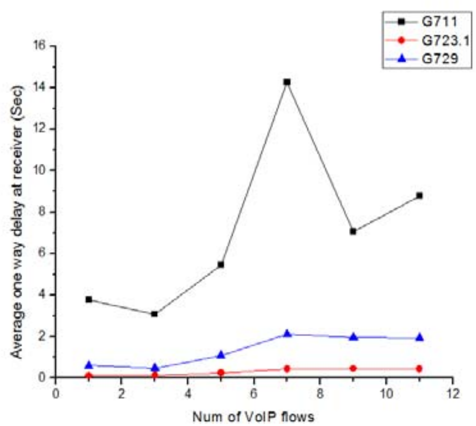


Figure 11. Average one way delay at VoIP receivers

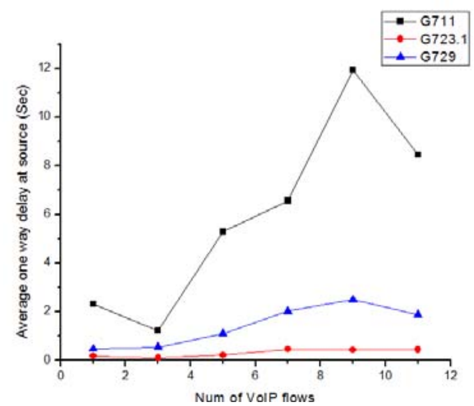


Figure 12. Average one way delay at VoIP sources

Fig.9 and 10 explain the total bytes received at receiver and source, it can be seen that G711 receives more number of bytes but it should also be noticed that it consumes more bandwidth whereas G723.1 and G729 both perform similarly, their bandwidth consumption comparatively less to G711. Average one way delay of VoIP receiver and source can be observed in fig.11 and 12, G723.1 outperforms with least delay.

VI. EMULATION AND RESULTS

Emulation testbed for a WiMAX scenario is established using once again EXata 2.0.1 [14]. A scenario is created consisting of single BS, one MSS and one SS. The ratio of TDD downlink to uplink time frames are adjusted as 2:1. VoIP traffic from MSS to SS is set using Windows NetMeeting utility and this is entirely a real time VoIP traffic which is possible in EXata using the internet gateway configuration. Multiple emulations performed for different speeds of MSS choosing the different codecs for VoIP application. The mobility model chosen is random mobility and the speed with which a MSS moving is set. The packet transfer between VoIP source and receiver is captured using network protocol analyser Wireshark [15]. Since the clocks of laptops were not synchronised with the emulation server on which EXata scenario runs, average end to end delay is not measured in these emulations. Each emulation runs for five minutes. Windows NetMeeting has a limited set of codecs, among which G723.1 (5.3K), G711 (a-law) and G711 (μ-law) are selected for performance evaluation, G729 is not available in Windows NetMeeting. The RTP stream analysis is selected to collect the statistics of VoIP traffic. Major performance metrics are packets sent and mean jitter of to and fro telephony between a VoIP source and receiver.

Fig.13 and 14 explain the packets sent by VoIP receiver and source for above mentioned codecs for different speeds of MSS which is a VoIP traffic source. Since the source is moving it should be noticed that the packets sent by source is consistent for G723.1 codec and at the receiver side though the G711 (a-law) performance is better, G723.1's working is still noticeable.

Fig.15 and 16 bring out the mean jitter of VoIP packets at receiver and source; jitter is the variation in the time between packets arriving, caused by network congestion, timing drift, or route changes. It is noticed that G723.1 performs considerably well under this scenario, actually at the emulation time this was experienced by authors.

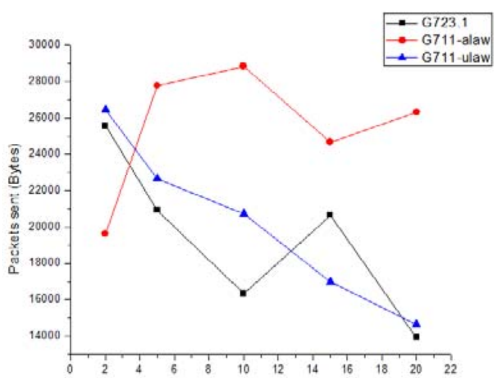


Figure 13. Packets sent by a VoIP receiver

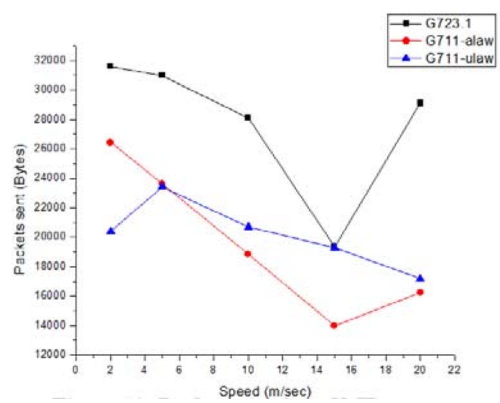


Figure 14. Packets sent by a VoIP source

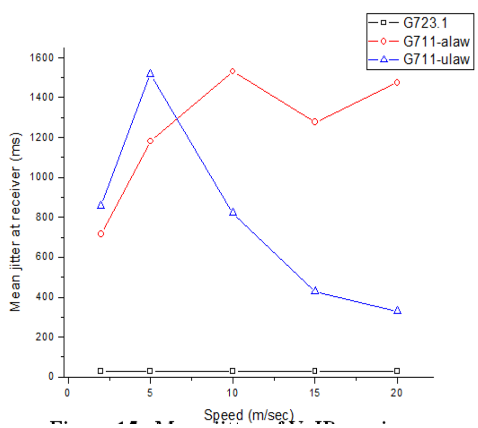


Figure 15. Mean jitter of VoIP receiver

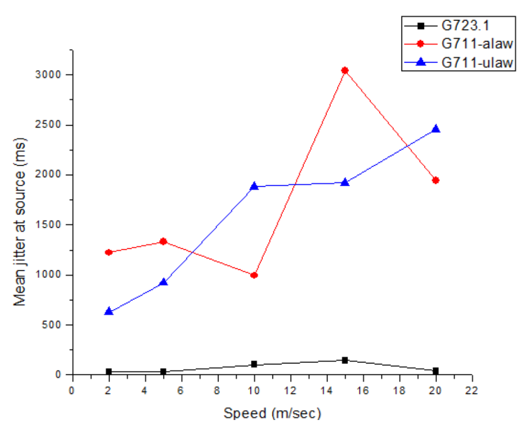


Figure 16. Mean jitter of VoIP source

VII. CONCLUSIONS

In this paper both simulation and emulation studies are carried out for VoIP traffic in a mobile WiMAX scenario. It can be concluded that both simulation and emulation studies are effective and efficient techniques to evaluate the performance of any new upcoming network. The simulation studies proved that G711 has more in and out traffic as its bandwidth consumption is highest among all the codecs, this is justified by emulation results. The delay or jitter performance of G723.1 is good comparatively to other codecs, during the emulation time actually this was experienced.

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References

[1] IEEE802.16e: IEEE Standard for Local and metropolitan area networks Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems, 2005.

[2] IEEE802.16e: IEEE Standard for Local and metropolitan area networks Part 16: Air Interface for Broadband Wireless Access Systems, 2009.

[3] Furqan Haq and Thomas Kunz, "Simulation vs. Emulation: Evaluating Mobile Ad Hoc Network Routing Protocols", In Proc. of IEEE IWWAN, May 2005

[4] Iwan Adhicandra, Rosario G. Garroppo and Stefano Giordano, "Configuration of WiMAX Networks supporting Data and VoIP traffic" . published in proceeding of 11th OPNETWORK Conference, Washington, DC, USA, 25-29 August 2008.

[5] Anindita Kundu, Iti Saha Misra, Salil K. Sanyal, Suman Bhunia, "Voip performance over broadband wireless networks under static and mobile environments", International Journal of Wireless & Mobile Networks (IJWMN) Vol.2, No.4, November 2010.

[6] Andres Arjona, "A study of mobile voip performance in wireless broadband networks", Doctoral dissertation.

[7] Muhammad Imran Tariq, Muhammad Ajmal Azad, Razvan Beuran, Yoichi Shinoda, "Performance Analysis of VoIP Codecs over BE WiMAX Network", 2010 3rd International Conference on Computer and Electrical Engineering (ICCEE 2010).

[8] Kostas Pentikousis, Jarno Pinola, Esa Piri, and Frerk Fitzek, "An Experimental Investigation of VoIP and Video Streaming over Fixed WiMAX", In Proc. Fourth International Workshop on Wireless Network Measurements (WiNMee), Berlin,Germany, Mar. 2008.

[9] Khaled A. Shuaib, "A Performance Evaluation Study of WiMAX Using Qualnet", Proceedings of the World Congress on Engineering 2009 Vol I WCE 2009, July 1 - 3, 2009, London, U.K.

[10] Konstantinos Birkos, Tasos Dagiuklas, Stavros Kotsopoulos "User and Network Level Evaluation of VoIP over Emergency Ad-hoc Networks**",** Mobimedia '09, September 7–9, 2009, London, UK.

[11] Shamik Sengupta, "Improving Quality of VoIP Streams over WiMax", ieee transactions on computers, vol. 57, no. 2, february 2008.

[12] Shamik Sengupta and Mainak Chatterjee, Samrat Ganguly and Rauf Izmailov , "Improving R-Score of VoIP Streams over WiMax",IEEE ICC 2006 proceedings.

[13] Ravi Shankar Ramakrishnan1 and P. Vinod kumar, "Performance Analysis of Different Codecs in VoIP Using SIP", Mobile and Pervasive Computing (CoMPC–2008).

[14] EXata documentation. http://www.scalablenetworks.com

[15] www.wiki.wireshark.org.