

# Performance Comparison of G.711 in different Efficiency Modes using ERTPS service flow

Amandeep Kaur<sup>#1</sup>, Harsukhpreet Singh<sup>\*2</sup>, Anurag Sharma<sup>#3</sup>  
<sup>#</sup>Electronics & Communication Department, Punjab Technical University  
CTITR, Jalandhar, India

**Abstract-** WiMAX (Worldwide Interoperability for Microwave Access) is currently one of the hottest technologies in broadband wireless access. The IEEE 802.16 wireless technology provides high throughput broadband connections over long distance. One of killer applications for the 802.16 is Voice over IP (VoIP) service; Voice over Internet Protocol (VoIP) is a vastly growing technology which enables the transport of voice data over Internet Protocol based networks. In this research paper we investigate the performance of VoIP traffic over WiMAX network. We analyzed the performance of VoIP codec G.711 in terms of jitter, delay & throughput by comparing different modes of efficiency. The simulation was done using OPNET Modeler 14.5 and simulation results found the efficiency mode in which VoIP codec performed better.

## I. INTRODUCTION

IEEE 802.16 group was established to develop WiMAX and grouped the air-interface standard into two general networks covering fixed and mobility [1]. Initially, IEEE 802.16 was designed for small business and residential fixed access users. After that Mobile WiMAX known as IEEE 802.16e come into existence which provides wireless mobility and supports subscriber stations moving at vehicular speeds [2]. Fixed WiMAX provides large bandwidth communication and networking keys up to 48 Mbps (fixed downlink) and 7 Mbps (fixed uplink), even in the isolated areas with a coverage area of 8 Km [3]. Mobile WiMAX provides 9.4 Mbps for the downlink and 3.3 Mbps for the uplink across the coverage area of 3 km. It supports high bandwidth and hundreds of users per channel at high speeds & promises to provide a range of 30 miles as a substitute to wired broadband like cable and DSL [7]. IEEE 802.16 is used not only as xDSL but it can also used as a mobile internet access technology. One of killer applications for the 802.16 is VoIP service to support bidirectional speech conversation. Since its introduction, VoIP has been gaining huge popularity and some services have broadened their coverage [4]. VoIP provides the use of Internet or intranet for voice calls by sending voice data in packets using Internet Protocol (IP) to users. The transmission technology for VoIP is essentially digital since VoIP is based on

IP. In digital transmission system, voice signal is digitized first and then separated in packets using complex algorithms known as codecs [5]. For encoding and decoding different codecs like G.711, G.723 and G.729 are used. The aim of this study is to investigate the performance of VoIP over WiMAX networks for the purpose to identify the best suitable VoIP codec and statistical distribution in such scenarios. In This paper two codecs G.711 & G.729 are compared over ertps service class.

## II. VOICE OVER INTERNET PROTOCOL

VoIP is a technology designed for communication over the internet using IP (Internet Protocol). VoIP converts the analog voice signal from source like computer into digital signal packets that can be transmitted over the internet. Basically, VoIP can turn a standard internet connection into a way to place a free voice calls. The bandwidth is used more efficiently in VoIP technology as the codecs are used for compression and decompression of analog voice signal into digitized packets. In the telecommunication industry the most popular voice codecs used are G.711, which uses narrowband speech and  $\mu$ -law or a-law encoding, G.729 which uses Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP). The G.729 codec uses the popular Conjugate- Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP) algorithm and produces a lower quality voice at a reduced data rate of 24 kbps. This codec also operates on narrowband voice and is widely used in voice over IP systems when carrying traffic on the WAN where bandwidth is costly.

## III. RELATED WORK

A rapid growth has been noticed in various wireless technologies in recent years. This has resulted in an increase in demand for wireless data services and multimedia application such as VoIP, streaming audio and video [6]. In order to provide good service and to meet the user demands, research has been in progress both in wireless technologies and VoIP network system. VoIP is becoming more and more popular day by day especially after the deployment of WiMAX network in many countries. Authors in [8] discuss the quality of VoIP calls with respect to the

delay and loss of packets. The aim of their work was to analyse the quality of service (QoS) on long distance data transfer between two locations with VoIP over WiMAX. Authors in [1]-[9] analysed the performance of VoIP over mobile WiMAX over Best Effort class in order to identify the best codec. Authors in [5] compared three voice codecs i.e.G.711, G.723 and G.729 were simulated in order to find the most appropriate voice codec for VoIP over WiMAX network under different traffic distributions.

**IV. SIMULATION SETUP**

Scenarios were designed to evaluate the performance of VoIP over the WiMAX network in the network simulator OPNET Modeler 14.5 with the assumption that only VoIP traffic is generated in this network model. Figure 1 illustrates the network model considered in simulations. WiMAX network consists of one base station (BS) & 10 subscriber stations (SS). The parameters for BS & MS are shown in figures 2 & figure 3 respectively. Voice quality is important for VoIP system because of the user’s high demands for good quality voice services [5]. In these scenarios, we considered the use of various efficiency modes in the same WiMAX network in order to investigate the performance of G.711 voice codec over ertps service flow at different node densities. The node densities used are 10,20,30,40. Figure 2 shows the table for simulation parameters.

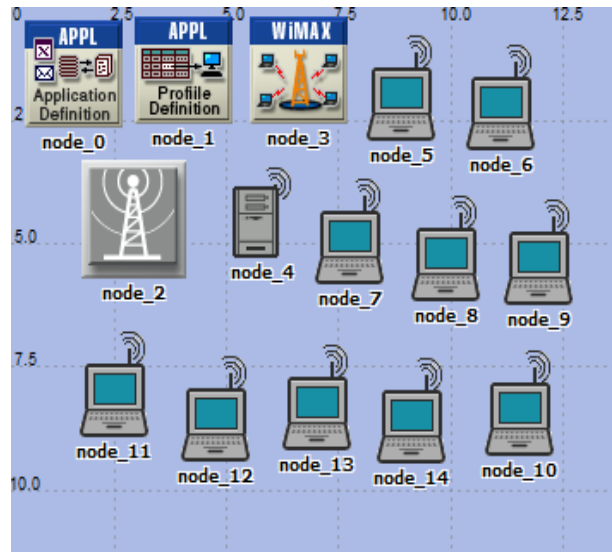


Figure1. Network model

**V. RESULTS & DISCUSSIONS**

The duration of the simulation for all scenarios was 5 minutes to each scenario. The parameters that are most interesting in this paper are throughput, jitter & delay.

Parameter	Value
Simulator	OPNET 14.5
Simulation Time	5 Minutes
Service Flow	ertps
Duplex Mode	TDD
Channel Frequency	5Ghz
Channel Bandwidth	20Mhz
FRAME DURATION	2ms
Modulation Coding	Adaptive
Path Loss Model	Free Space
Efficiency Mode	Physical Layer Enabled, Efficiency Enabled, Framing Module Enabled
Antenna gain (BS)	15dBi
Antenna gain (MS)	14dBi

Figure2. Simulation Parameters

Figure4 shows graph for jitter by codec G.711 in all three efficiency modes; efficiency enabled, framing module enabled, physical layer enabled, used in scenario. It has been observed that the jitter in efficiency enabled mode is higher than other two modes. Framing modules enabled & physical layer enabled mode has almost same values for jitter. At node density 10 the jitter in physical layer enabled mode is little lower than framing module enabled mode. As the density increases the values of jitter became same at each node density for both modes.

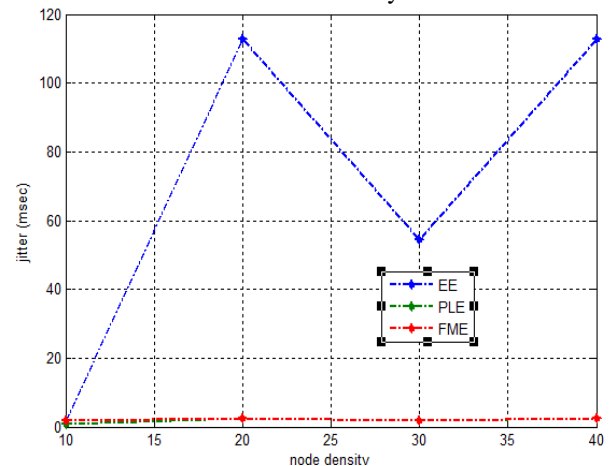


Figure4. Jitter (sec)

Figure 5 shows the delay for all efficiency modes for codec G.711 over ertps service flow. From the graph it has been analyzed that delay is increasing as the node density increases in all the three efficiency modes. In efficiency enabled mode the delay the very high about 16 seconds. Physical layer enabled mode outputs lower delay than both the efficiency modes. Figure 6 shows the graph for throughput in bytes per second for codec in different efficiency modes. Throughput increases steadily as the node density increases. This increase is higher in physical layer enabled mode as the graph shows.

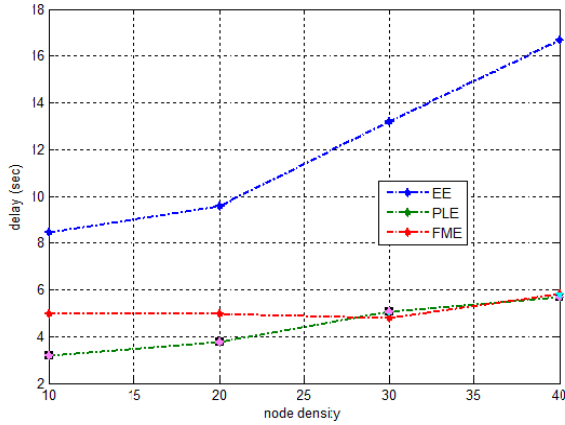


Figure5. Delay

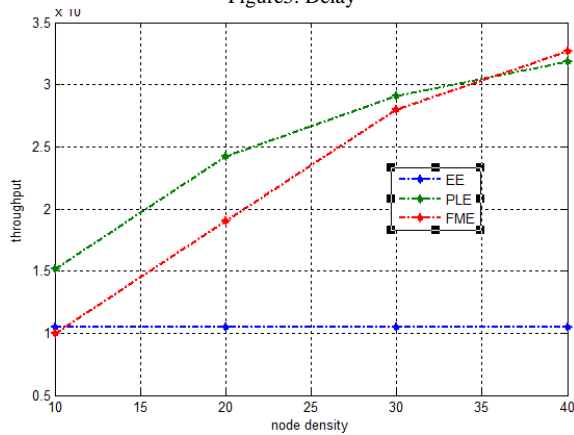


Figure6. Throughput

Efficiency enabled mode outputs lower throughput than other two modes. At node density 50 the throughput of framing module enabled efficiency mode is higher than the physical layer enabled mode. But the overall throughput at all node densities is better in physical layer enabled mode.

## VI. CONCLUSION

In this paper, the performance of the VoIP codec G.711 which is mapped to the ertps service class is analyzed and evaluated over different efficiency modes. Various scenarios are simulated to investigate the best performing efficiency mode for codec G.711 over ertps service class. In general, it is important for VoIP calls to deliver a minimum delay and jitter maximum throughput. From the simulation results we can conclude that in Physical Layer Enabled efficiency mode the performance of G.711 is better than Efficiency Enabled & Framing Module Enabled efficiency modes with minimum jitter, delay & maximum throughput. Physical Layer Enabled efficiency mode is best suitable for real time applications.

## REFERENCES

1. HR Othman, DM. Ali, N AMohdYusof, S Syahidah, A Idris, "Performance Analysis of VoIP over Mobile WiMAX (IEEE 802.16e) Best Effort Class", IEEE 5th Control and System Graduate Research Colloquium, Aug. 11 - 12, UiTM, Shah Alam, Malaysia, 2014, pp: 130-135.
2. K Jakimoski, T Janevski, "Priority Based Uplink Scheduling Scheme for WiMAX Service Classes", IJ. Information Technology and Computer Science, 2013, pp: 66-74.
3. S Saha, A Chakraborty, AK Mukhopadhyay, A K Bhattacharjee, "Performance Analysis of Service Classes for IEEE 802.16m QoS Optimization", ICETACS 2013, pp: 149-155.
4. I Adhicandra, "Measuring Data and VoIP Traffic in WiMAX Networks", Journal of Telecommunications, Volume 2, Issue 1, April 2010, pp:1-6.
5. 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 [Volume 3, Issue 4, April 2012, pp: 92-98.
6. S. Sengupta, M. Chatterjee and S. Ganguly, "Improving quality of VoIP streams over WiMax," Computers, IEEE Transactions on, vol. 57, 2008, pp: 145-156.
7. M Seyedzadegan and M Othman, "IEEE 802.16: WiMAX Overview, WiMAX Architecture", International Journal of Computer Theory and Engineering, Vol. 5, No. 5, October 2013, pp: 784-787.
8. MA Mohamed, FW Zaki and RH Mosbah, "Improving Quality of VoIP over WiMAX", IJCSI International Journal of Computer Science Issues, Vol. 9, Issue 3, No 3, May 2012 ISSN (Online): 1694-0814, pp: 85-91.
9. PD Joshi, S Jangale, "Analysis of VoIP traffic in WiMAX using NS2 simulator", International Journal of Advanced Research in Computer Science and Electronics Engineering Volume 1, Issue 2, April 2012, pp: 109-114.