Performance Analysis of VOIP over WiMAX

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***Abstract*—WiMAX is a assuring technology for providing wireless last-mile connectivity, because of its wide coverage area, high speed data rates and low deployment cost. MAC and Physical layer of WiMAX relate to the IEEE 802.16e standard, which describes 5 dissimilar data delivery service classes that can be employed for satisfying Quality of Service (QoS) needs of various applications i.e. Videoconference, VoIP, Web, FTP etc. The primary objective of the paper is to analyze a case of deployment of QoS over a cellular WiMAX network. Particularly, the paper compares the performance found utilizing two dissimilar QoS configurations is differed from the delivery service class employed to carry VoIP traffic such as ertPS or UGS. Results shows that for delay-sensible traffic that fluctuates outside its related data rate, having the probability to provide back some of its preserved bandwidth, ertPS has the benefit to allow the transmission of BE traffic.**

Keywords: VoIP, ertPS, QoS,FTP, UGS, WiMAX

I. INTRODUCTION

The IEEE 802.16e technology (WiMAX) is a assuring choice to wireless LAN or 3G for supplying last mile connectivity by radio link because of its wide coverage domain, high speed data rates and low deployment cost. The standard defines the air-interface between Base Station (BS) and a Subscriber Station (SS). IEEE 802.16 (WiMAX) can be employed not only as XDSL substitution for small business clients but also as a mobile internet access technology. One of important application for the 802.16 is Voice over IP (VoIP) service to provide support to two way voice communication. There have been many studies concentrating on performance measurement of IEEE 802.16 WiMAX Networks by using OPNET modeler. Ramachandran et al [3] analyzed performance of IEEE 802.16 for Broadband Wireless Access. They employed OPNET's DOCSIS simulator to model the IEEE 802.16 MAC. Rangel et al [4] analyzed performance of QoS in Broadband IEEE 802.16 Based Networks by utilizing OPNET WiMAX simulator, they concentrated primarily on enforcing their own scheduling algorithms. Dang et al [5] analyzed the scheduling algorithms performance for WiMAX networks. they concentrated primarily on implementing some available scheduling algorithms. The aim of this study was to analyze a case of QoS enforcement over a cellular WiMAX network and to analyze the ability of WiMAX network to provide enough QoS to data and voice applications. The steps taken involve generating the WiMAX network, enforcing the needed applications, enforcing QoS topology within the WiMAX, setting the QoS topology within the WiMAX cells to fulfill voice needs, and further setting the QoS topology to enhance data application performance, without decreasing the voice application performance. The paper is formed as follows. QoS in IEEE 802.16 is explained in Section 2. Section 3 gives a summary of VoIP involving MOS and R-Score. Section 4 describes system model design. The results are represented in Section 5. At last, the paper conclusion is in section 6.

II. QUALITY OF SERVICE IN IEEE 802.16

Earlier, four service types were supported in the 802.16 standard: rtPS, nrtPS, UGS and BE. The UGS (Unsolicited Grant Service) is like the CBR (Constant Bit Rate) service in ATM, which creates a fixed size burst at regular interval of time. This service can be employed to substitute a constant rate service or T1/E1 wired line. It also can be utilized to provide the support to real time applications i.e. video streaming or VoIP applications. Still the UGS (Unsolicited Grant Service) is not complex, it may not be the best option for the VoIP applications in that it can waste bandwidth during the voice calls off period. The rtPS (real-time polling service) is used for varying bit rate real-time applications i.e. VoIP. At each polling interval, Base Station polls a mobile and the polled mobile sends bandwidth request (bwrequest) if it has packets to send. The base station accepts the data burst by using UL-MAP-IE on its reception. The nrtPS (non-real-time polling service) is very like to the rtPS with the exception that it permits contention based polling. The BE (Best Effort) service can be employed for applications i.e. FTP or e-mail, where there is no hard latency need. other service type is ertPS (Extended rtPS) [6] was presented to provide support to varying rate real-time services i.e. video streaming and VoIP. It has a benefit over rtPS and UGS for VoIP applications because it contains lesser overhead as compared to rtPS and UGS.

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III. VOICE OVER IP (VOIP) PERFORMANCE

This section will provide the summary of Voice over IP and E-Model. VoIP application generally works as follows. First, a voice signal is sampled, digitalized, and encoded by using a given coder/algorithm. The converted data (called as frames) is put into packets and transported by using UDP/RTP/IP. Receiver de-packetized the data and sent to a payout buffer, which eliminates the delay found in the network. Eventually, the data is decrypted and the voice signal is rebuild. The E-model is a communication planning tool formulated by the ITU-T. It gives required voice quality prediction as would be detected by a distinct telephone user taking part in a whole end-to-end voice call. A broad range of bad situations are also considered i.e. end-to-end delay, codec impairments, packet drop, jitter as well as echo and noise. The ITU-T E-model is based on simulating the results of a wide number of subjective tests performed to evaluate the sensed voice call quality. This is not a genuine model it means that it cannot correctly predict the complete and perfect view of a single user, but, for a large number of users, the results are sufficiently correct to allow for planning and measurement uses. The output of the E-model is a value is called Transmission Rating Factor or R-value. Using this value, other quality measures can also be found, such as MOS (Mean Opinion Score). The MOS is a subjective attribute score whose ranges from 1 (worst) to 5 (best) and is received by carrying subjective surveys. The individual transmission parameters are converted into various impairment factors that are integrated to generate the R-value ranging from 0 to 100

$$R = 100 - Is - Ie - Id + A \qquad (1)$$

where Is represents the signal-to-noise impairments linked with distinct switched circuit networks routes, Ie is for equipment impairment factor linked with the losses because of the network and codec's, Id presents the impairment generated by the mouth-to-ear delay, and A adjusts for the above impairments under different user conditions and is called as the expectation factor. Once an R-value has been computed, an estimated MOS can be estimated for the voice call quality by using the formulae that follow.

For $R < 0$, $MOS = 1$

For $0 < R < 100$,

$$MOS = 1+0.035R + 7 \times 10-6R(R - 60)(100 - R) \qquad (2)$$

For $R > 100$, $MOS = 4.5$

Variables generallly considered in VoIP are only Ie and Id. Thus, if default values are employed for all other factors, the expression for R-factor from equation (1) can be write as:

$$R = 94.2 - Ie - Id \qquad (3)$$

IV. SYSTEM MODELS

For We utilized OPNET Simulator version 14.5 with WiMAX Module ability [9] in this experiment. We constructed two scenarios involving Improve Data scenario and Improve Voice scenario. We consider there are two companies working on the systems. First, a Service Provider company which is a wireless connectivity supplier in that region. Second, a Client company that has several employees which are mobile across a specified area. The Service Provider company requires to measure whether it can satisfy the needs of the Client company involving measuring the corporate servers through a data application and being capable to talk to head office through voice application.

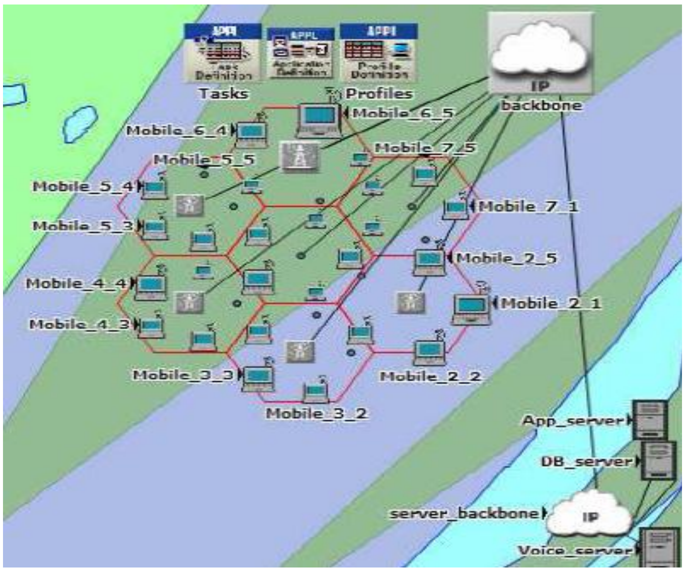

Fig 1. Network Model

The network contains an IP backbone and seven cells and Cell radius is adjust to 0.2 km. Every cell has 5 nodes. There is a server backbone consisting three servers i.e. Voice server, Application server and Database server. These nodes show the Service Provider company network (Fig 1). Subscriber Node Transmission Power is adjust to 0.5 W and Base Station Transmission Power is adjust to be 5W. The Multipath and Pathless Model are adjust to Pedestrian.

A. Application Model

The Client company will operate two applications: a voice application contains the calls which are made to the company head office and a data application contains Oracle transactions. The Service Provider company has caught the 3- tier Oracle transaction and supplied it to the client company as an ACE trace. The PCM is voice traffic. All client employees will operate the data applications, but only one from four will operate the voice application in each cell. The data application is positioned between the Service Provider company at its head office and the Client Company, by using the ACE trace of caught traffic. Then, the PCM-quality voice application is

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positioned between the Service Provider company's head office and the client company.

Table 1: Mac Service Class Parameters

Service Class	Type	Maximum Sustained Traffic Rate	Minimum Reserved Traffic Rate
Gold	UGS	64 Kbps	64 Kbps
Silver	rtPS	1 Mbps	0.5 Mbps
Bronze	BE	384 Mbps	384 Kbps
Platinum	UGS	2.5 Mbps	2.5 Mbps

B. QoS configurations model

WiMAX QoS strictly requires to be configured. We consider that the Service Provider company is planning to take UGS connections with solid QoS assurance for its voice applications. The data application is imparted to utilize Best Effort connections without any QoS assurance. The Client company requires to show their already offered bandwidth as a constant UGS bandwidth distribution. We generated a service class Platinum with UGS distribution to preserve the bandwidth for which the client company has committed to their available employees. They can be admitted within the left bandwidth. The size of Gold service class is changed to the 64Kbps rate of the voice codec (Table 3). Then, we positioned classifiers and service flows over the WiMAX mobile nodes. For every node, we deployed UGS connections for both downlink and uplink directions on which the voice application is distributed. We adjust Service Class to Gold, Initial Coding Rate to ½ and Initial Modulation to QPSK on Downlink Service Flows, While in the Uplink Service Flows, we joined one extra profile and adjust Service Class Name to Platinum, Initial Coding Rate to ½ and Initial Modulation to QPSK, (Table 4).

IV. RESULTS

A. Improve Voice Scenario

In the general process, the voice connection has a load of 96 Kbps leading in the throughput of 64Kbps. This variation between throughput and load is leading inadequate delays for voice. For solving this problem, we have employed the Improve Voice scenario. We changed the size of the UGS rate of the Gold service class from 64Kbps to 96Kbps.

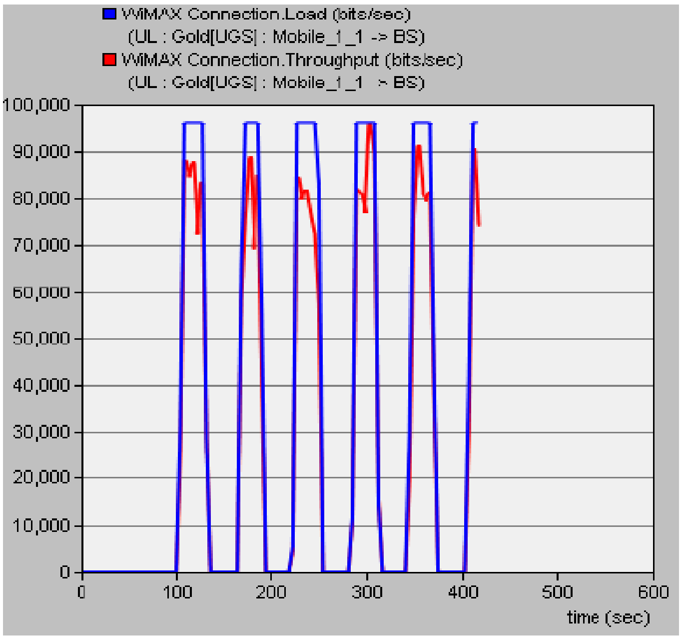

Fig. 2. Voice behaviour over UGS after improvement

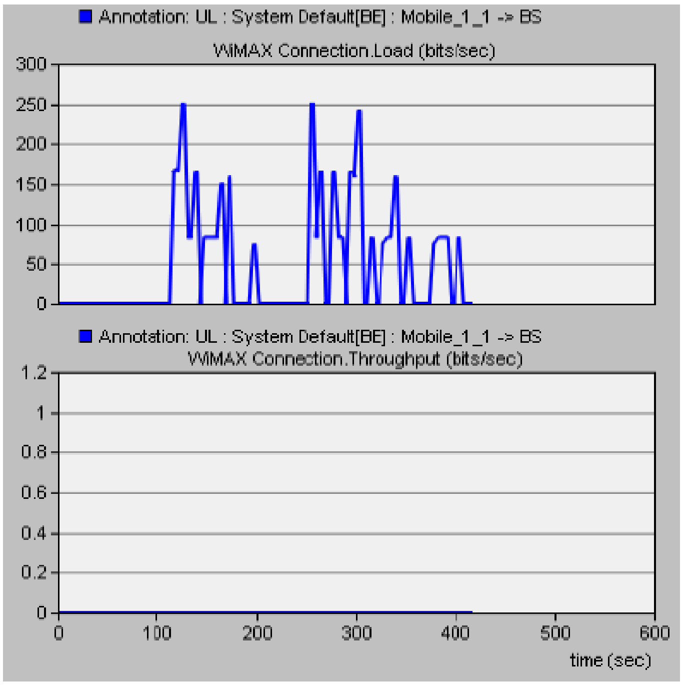

Fig. 3: Behaviour over Best Effort

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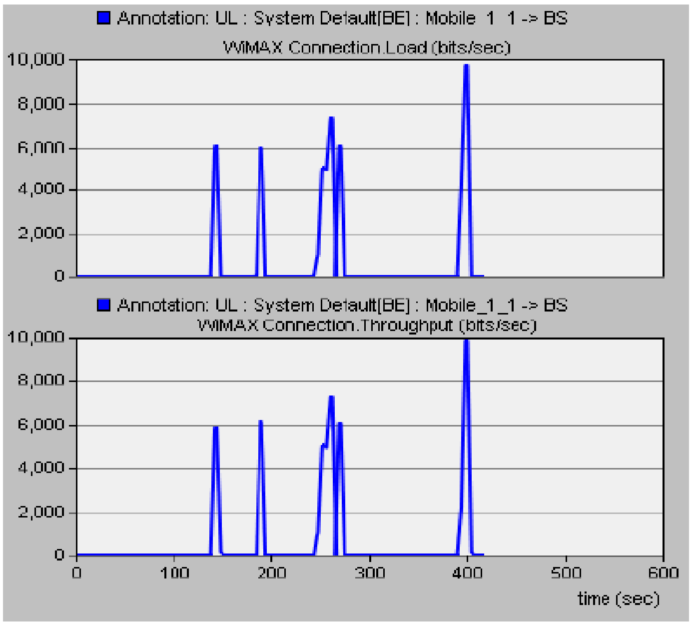

Fig. 4. Data behaviour over Best Effort after improvement

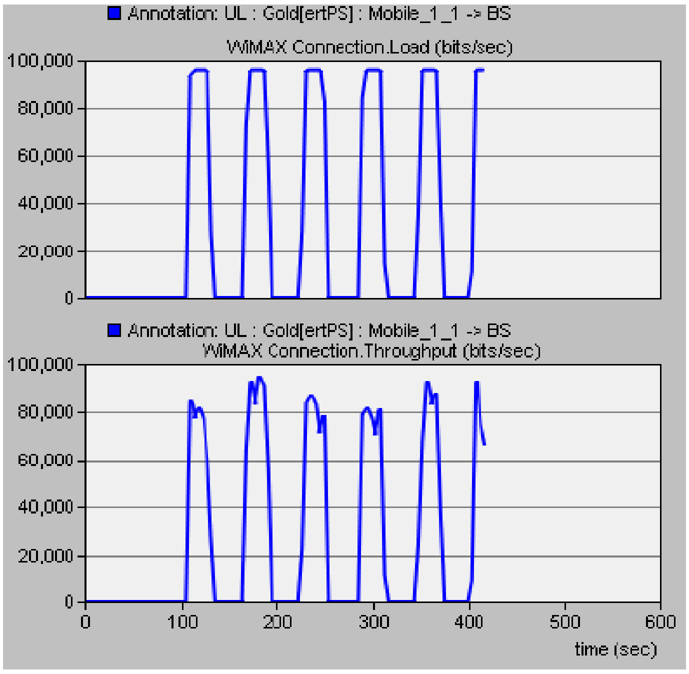

Fig. 5. Voice behaviour over ertPS

The throughput is more drawn out in time as compared to the load, proposing some queuing. The UGS connection size is 64Kbps, but it obtains 96Kbps in load. The comparison is made up by overhead among the MAC layer and the application layer (Fig 2). However, the voice application provides a high load at the WiMAX MAC of 96 Kbps, accordingly we require to re-size the Gold UGS connection. Improving the UGS capability permits the voice load at the MS to be checked by the voice throughput at the BS. Losses because of disturbance throughout the air are considered for the difference. Due to the additional capacity provided in the UGS flows containing voice, the ACE application operating in Best Effort ceases flowing, even though there is load provided to the Best Effort connection, there is no traffic routed and the connection queue keeps holding data packets (Fig 3).

B. Improve Data Scenario

We modified the voice QoS in the Improve Data Scenario, so that the Best Effort traffic even has a probability to flow. Load is provided to the default Best attempt connection in the uplink. Traffic collects in the Best Effort connection waiting line. No traffic is routed out over the air to save the Best Effort waiting line. This is because of the deficiency of grants being planned for this Best Effort connection. Relief will be supplied by switching the planning class of the voice connections from UGS to ertPS for the data traffic. The latter gives an elastic distribution, make suitable to the flow of traffic. If there is no traffic load, then the ertPS distribution is decreased temporarily. This decrement gives relief to the Best Effort connection utilized for the data traffic, without worse effect to the voice application. In the former scenario, the Best Effort connection was suffered from grants. Here, as a result of shifting voice from UGS to ertPS, traffic began flowing once again throughout the Best Effort connection. Traffic that goes into the connection as load on one side goes out as throughput from another side (Fig 4). In summation, the voice traffic is not negatively influenced (Fig. 5).

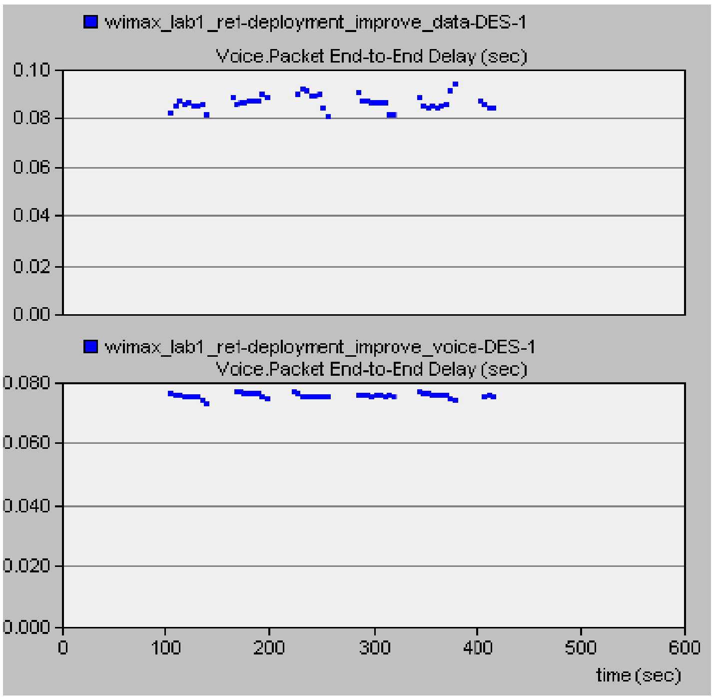

Fig. 6. End-to-end delay for different scenarios

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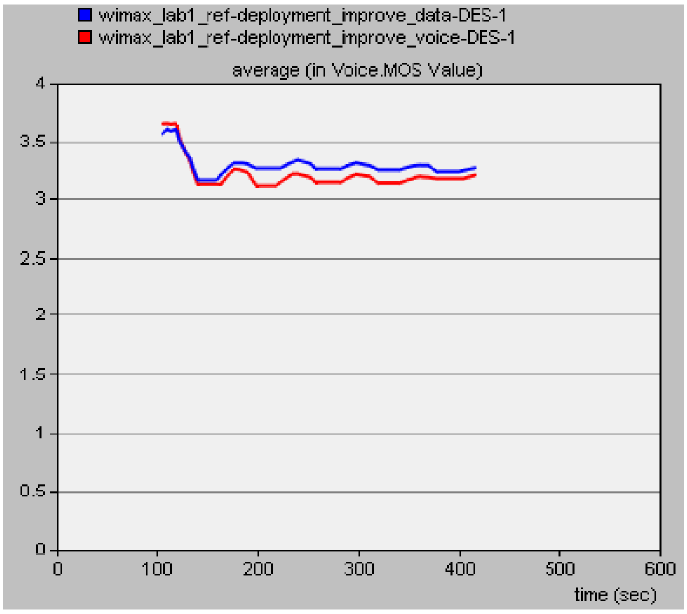

Fig. 7: MOS over different scenarios

The voice delays in the improve data scenario are only quite higher in comparison of improving voice scenario. Thus, increasing UGS capability decreases the voice end to end delays below 80ms. Shifting from UGS to ertPS does not have worse effect on the voice delays (Fig 6). For voice quality over various scenarios, we can view that after enhancing voice and data application, the performance of both data and voice quite steadily and less similar for MOS at approx. 3.2 until 400 seconds (Fig 7). Still, the voice performance decrease slightly as compared data performance. The reason is, there were some packet drops and jitters in the transactions

CONCLUSION

This paper represents our growth or development in measuring the performance of IEEE 802.16. We have utilized the WiMAX Connection statistics (e.g. throughput, load) to derive the behaviour of traffic mapped to service flows whether the load provided to a connection from the higher layer is checked by the throughput the connection provides back to the higher layer on the another side of the WiMAX hop and whether the connection is suffered of grants and traffic from the higher layer develops in the connection's waiting line. For delay-sensible traffic that shows variation outside its related rate, we have utilized ertPS scheduling class which has the benefit of providing back some of its preserved bandwidth, if there is no traffic to be served by this reserved bandwidth. For voice quality over various scenarios, the performance of both data and voice is quite steadily and similar for MOS at approx. 3.2 until 400 seconds.

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