



## A QoS Oriented Analysis of ertPS, rtPS and nrtPS flows in WiMAX

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**Abstract-** In the new era of communication, WiMAX (IEEE 802.16) is the most emerging technology that enables ubiquitous delivery of broadband wireless access for fixed and mobile users. It is a more innovative and commercially viable alternative to cable modems and DSL technologies as it is cost effective, easy to implement, high performance and high resource utilization technology. Quality of Service (QoS) is an important consideration for supporting variety of applications that utilize the network resources. This paper focuses on quality of service oriented analysis in WiMAX networks with CBR traffic. Different types of QoS parameters like packet loss, average throughput, average jitter and average end to end delay are investigated with CBR traffic and effect of mobility are evaluated using QualNet 6.1 Wireless network Simulator for rtPS, nrtPS and ertPS flows. The goal is to compare the Quality of Service parameters of the three service flows with respect to mobility so that different types of service flows can be used in optimal way for real life scenarios.

**Keywords-** IEEE 802.16, WiMAX, QoS, CBR, Mobility.

### I. INTRODUCTION

In the new era of communication, WiMAX (IEEE 802.16) is the most emerging technology that enables ubiquitous delivery of broadband wireless access for fixed and mobile users. The IEEE 802.16 standard widely known as WiMAX (Worldwide Interoperability for Microwave Access), has been developed to accelerate the introduction of broadband wireless access into the marketplace. Prior to the introduction of the IEEE 802.16 standard, the most effective ways to obtain access to broadband internet service were mainly through T1, Digital Subscriber Line (DSL), or cable modem. However, these wired infrastructures are considerably more expensive, especially for deployment in rural areas and developing countries. This limitation propelled the industry to devise an alternative means of obtaining broadband internet access and the approach taken was via the wireless medium. The traditional wireless cellular networks have a hierarchical architecture in which centralized controllers facilitate resource management and mobility support in a highly efficient manner, typically for voice call services. Although they are designed primarily for wireless internet access, currently deployed mobile WiMAX networks also adopt this cellular-style hierarchical architecture, but they use less hierarchy. It is one of the most emerging technologies and provides an exciting range of additional features to the existing techniques for the broadband. It can also be treated as an alternative to the existing cable and DSL technologies as it has low cost and can be easily implemented. It also provides high data rate applications with a variety of Quality of Service (QoS) requirements. The famous companies like Motorola and Samsung are already developing WiMAX phones and PDAs and they are already in use in Korea with WiMAX cousin technology, WiBro (Wireless Broadband) [1]. Nowadays WiMAX is considered as one of the main technologies for next generation high speed wireless access networks. It gives a larger coverage compared to Wi-Fi while supporting string QoS and security mechanism because of its optimizable physical layer and too many adaptable capabilities. This latest innovative WiMAX technology is considered as one of the main standards for future wireless networks. Several technologies used by WiMAX, such as Orthogonal Frequency-Division Multiple Access (OFDMA) and resource allocation methods with differentiated QoS are parts of Next Generation Networks (NGN) standards [2]. WiMAX can be convenient for Hybrid Networks, Local Area Networks or long range transmission thanks to MAC relays defined in 802.16j [3].

The name WiMAX was propounded by the WiMAX Forum [4], a consortium consisting of about 420 members incorporating famous companies like AT&T, Fujitsu, Intel and Siemens. It was set up in June 2001 to promote conformance and interoperability of the technology and promote its commercial use. The WiMAX Forum and IEEE 802.16 subcommittee are both involved in the development of open standards based broadband wireless networks. The IEEE 802.16 subcommittee is purely a technical body that defines the 802.16 family of broadband wireless radio interface standards. IEEE 802.16 defines the layer 1 (physical, also referred as PHY) and layer 2 (data link or Media Access Control - MAC) of the (Open Systems Interconnection) OSI seven layer network model. The IEEE 802.16 standard provides specification for the MAC and PHY layers for the air interface. The standard includes details about the

various flavours of PHY layers supported and characteristics of the MAC layer such as bandwidth request mechanisms and the scheduling services supported. It does not define standardized network architecture beyond the base station. Standardized network architecture is essential to ensure inter-working between equipment from different vendors and inter-working between networks of different operators. The WiMAX Forum fills this gap and creates an end-to-end broadband wireless network. It is responsible for preparing profiles for systems that comply with the IEEE 802.16 standard and create interoperability tests to ensure that different vendors' implementation can work together. The first version of the IEEE 802.16 standard was completed in October 2001 and since then several version have emerged addressing issues such as Non-Line of Sight (NLOS) operation, mobility, multiple traffic classes for QoS, operation in the licensed and unlicensed frequency bands [5].

## **II. Quality Of Service**

There is no formal definition of Quality of Service. QoS, in the field of telephony, was defined in 1994 in the International Telecommunication Union (ITU) Recommendation E.800. This definition contains broadly 6 major components such as Support, Operability, Accessibility, Retainability, Integrity and Security. In 1998, the ITU published a document discussing QoS in the field of data networking. The term Quality of Service refers to the probability of the telecommunication network meeting a given traffic contract. Although the name suggests that it is a qualitative measure of how reliable and consistent a network is, there are a number of parameters that can be used to measure it quantitatively. These include throughput, transmission delay or packet delay, delay jitter, percentage of packets lost etc. The use of different kinds of applications in a network, results in heterogeneous traffic load. The traffic from different applications may require certain type of quality of service, for example,

- Streaming multimedia needs timely packet delivery
- IP telephony or Voice over IP (VOIP) needs strict limits on delay and jitter
- Video Conferencing (VTC) need very less time variation in delay
- Dedicated link emulation needs both timely delivery of packets and bounds on maximum delay and jitter
- A safety-critical application, such as remote surgery needs a guaranteed level of availability within stipulated time and with accuracy (this is also called hard QoS).

## **III. Ieee 802.16 Quality Of Service Classes**

To meet all the different QoS requirements such as packet error rate, jitter, data rate, system availability, and the like, WiMAX utilizes different scheduling mechanisms to allocate downlink and uplink transmission opportunities for the different PDUs. The service classes supported by WiMAX are [6]:

Unsolicited Grant Service (UGS): This scheduling service is designed to support applications that generate fixed-size data packets periodically such as T1/E1 and VoIP without silence suppression. To support the real-time needs of such applications and reduce overhead by the bandwidth request-grant process, the BS allocates fixed size data grants without receiving explicit requests from the SS. The size of the grants is based on the maximum rate that can be sustained by the application and is negotiated at connection setup.

real-time Polling Service (rtPS): This scheduling service is designed to support real-time applications that generate variable size packets on a periodic basis such as MPEG video or VoIP with silence suppression. The BS allows the SSs to make periodic unicast requests and allows them to specify the size of the desired grant. Since a dedicated grant request is contention-free, the bandwidth request is guaranteed to be received by the SS in time. SSs belonging to this class are prohibited from using contention request opportunities.

non real-time Polling Service (nrtPS): nrtPS is designed to support non-real time applications that require variable size data grant bursts on a regular basis. This scheduling service supports applications that are delay tolerant but may need high throughput such as File Transfer Protocol (FTP) applications. The BS allows the SS to make periodic unicast grant requests, just like the rtPS scheduling service, but the requests are issued at longer intervals. This will ensure that the SSs receive request opportunities even during network congestion. SSs of this class are also allowed to use contention request opportunities.

Best Effort (BE): This traffic class contains applications such as telnet or World Wide Web (WWW) access that do not require any QoS guarantee. The bandwidth request by such applications is granted on space-available basis. The SS is allowed to use both contention-free and contention based bandwidth requests, although contention-free is not granted when the system load is high.

Extended Real-time Polling Service (ertPS):

The extended real time polling service (ertPS) was introduced at the same time as mobile WiMAX and is a combination of UGS and rtPS. In this QoS class, unsolicited unicast grants are provided by the BS, so in this way the latencies caused by the bandwidth requests are removed. Applications supported offer real-time service flows that generate variable size data packets on a periodic basis such as voice with activity detection (VOIP).

In Table 1, we list some of the applications belonging to each class [6].

Table 1  
Summary of QoS classes [6]

S.No	Quality of Service Class	Application	QoS Specification
1.	Unsolicited Grant Service (UGS)	Voice over IP (VoIP) without silence suppression, T1/E1	Maximum substandard rate, Maximum latency tolerance, Jitter tolerance
2.	Real-time Polling Services (rtPS)	MPEG video	Minimum reserved rate, Maximum substandard rate, Maximum latency tolerance, Traffic priority
3.	Non Real-time Polling Services (nrtPS)	File Transfer Protocol (FTP)	Minimum reserved rate, Maximum substandard rate, Traffic priority
4.	Best Effort (BE)	Web browsing, data transfer	Maximum substandard rate, Traffic priority
5.	Extended Real-time Polling Service (ertPS)	Voice with activity detection (VOIP)	Minimum reserved rate, Maximum substandard rate, Maximum latency tolerance, Jitter tolerance, Traffic priority

#### IV. Simulation Scenario

Simulation is an essential tool in the development and performance evaluation of communication networks. So, we have chosen simulation based methodology for our research. Among the available tools for networks simulation, QualNet 6.1 Wireless network Simulator [7] is employed in order to evaluate the performance of WiMAX. WiMAX scenario is created using nodes and subnets. Two homogeneous networks are considered and one node of each network is assigned to act as a Base Station (BS) whereas all other nodes are assigned to act as a Subscriber Stations (SS). In the first case, WiMAX scenario is simulated for CBR traffic application without any mobility model and the results are noted down. In the second case, it was simulated by enabling random way point mobility at 5 mps for all the nodes of the scenario, similarly for next cases mobility is set as 10, 20, 30, 40 and 50 mps and thereafter results are compared.

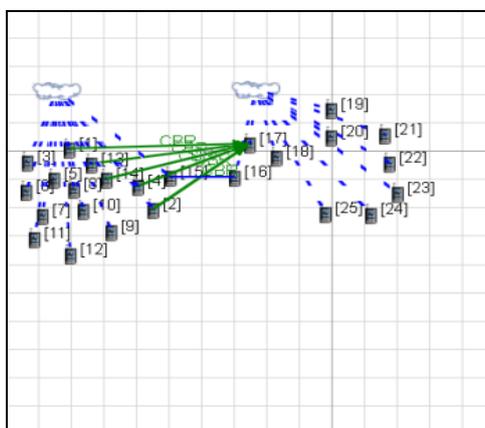


Figure 1: Simulation Scenario

#### V. Simulation Parameters

System parameters like simulation time, channel frequency, bandwidth, transmission power etc were specified as per IEEE 802.16e standard as shown in the Table 2. Bellman Ford routing protocol is used as a default one. Bellman Ford routing protocol is used as a default one, which is a distance vector routing algorithm utilising UDP for control packet transmission. IP Output Queue Scheduler used is Strict Priority. In this scheduler, the selection of packets is based on the

priority assigned to the traffic. The packets are allocated into different priority queues only after being categorized by the scheduler depending on the Quality of Service classes of the traffic. The algorithm first serves the queue with highest priority until it is empty, then it moves to the next highest priority queue. This procedure can also result in bandwidth starvation for low priority QoS classes where the packets may not even get forwarded and no guarantee is offered to flow which has least priority [7]. Eight queues are used in order to avoid queuing packets of different service types into one queue. Even if the application sets a high precedence for its packets with lower numbers of queues in the scenario, they may get blocked by lower precedence packets in network queues. Therefore, in order to fully guarantee the service to the classes, 8 queues are configured at network layer. Once the simulation begins each node under goes the basic registration procedure in order to associate with the Base Station due to this traffic starts after a pause to complete the registration of the mobile node. The traffic starts after a pause of 1 second once the simulation is started in the implemented scenario.

Table 2  
Simulation Parameters

Parameter	Value
Simulator	QualNet 6.1
Simulation Time	10 min
Service Types	ertPS, rtPS, nrtPS
Channel Frequency	2.4GHz
Channel Bandwidth	20MHz
Antenna Model	Omnidirectional
Mobility Model	Random Waypoint
Path loss Model	Two Ray
Max Transmission Power	50dBm
Transmission Power	20dBm
FFT Size	2048
Items to send	1000
Item Size	512 bytes

## VI. Results and analysis

QoS provisioning encompasses providing Quality of Service to the end user in terms of several generic parameters. The perceived quality of service can be quantitatively measured in terms of several parameters. In the analysis, received throughput, packet loss, average end to end delay and average jitter were considered. To make the comparison between rtPS, nrtPS and ertPS presented on the same graph. This makes it easier to see and compare the behavior of different QoS parameters for the same traffic over different type of service flow. Figure: 2-5 show the comparative plots.

### Comparative results for Packet loss

Figure 2 shows comparative plot of the packet loss with respect to mobility. The occurrence of packet loss takes place when a few packets of data moving across the network do not reach the desired destination. This also affects the perceived quality of the service.

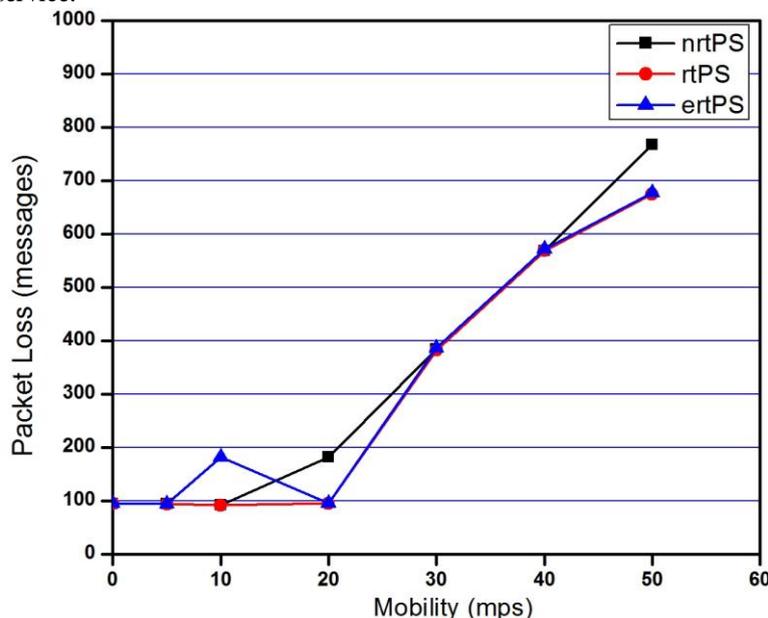


Figure 2: Packet loss wrt mobility

nrtPS flow has maximum packet loss out of the three service flows and it performs worst in this performance parameter as it shows the maximum increase with respect to mobility. Even at no mobility it has maximum number of packet loss. In case of ertPS flow at mobility 10mps an unexpected loss is observed. In all, rtPS and ertPS flows perform better than nrtPS flow.

**Comparative results for received throughput**

Figure 3 shows comparative plot of the received throughput with respect to mobility for nrtPS, rtPS and ertPS service flows. Throughput is a measure of the data rate (bits per second) generated by the application.

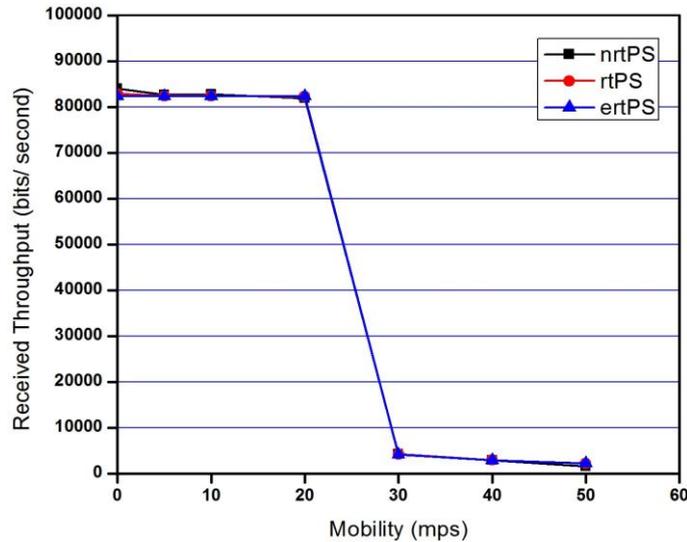


Figure 3: Received throughput wrt to mobility

It is generally observed that ertPS and rtPS flows have maximum received throughput values nrtPS being low priority traffic service have no traffic due to bandwidth starvation in congested environment when the Strict Priority scheduler is applied. As the simulation is configured without having any congestion so the throughput does not show much change in all the service flows in the simulated scenario. At lower values of mobility throughput is high but at higher mobility it is low for all the service flows. Initially when no mobility is assigned to the nodes the value of throughput is high but with the increase in mobility, the received throughput follows a descending trend. The sharp descend in the throughput is observed in all the cases, when the mobility increases from 20 mps to 30 mps this is due to packet loss as the server moves away from the clients at these mobility's and as the packet loss will increase, the received throughput will also decrease.

**Comparative results for Average end to end delay**

Figure 4 shows comparative plot of average end to end delay with respect to mobility for the three service flows. Delay or latency is the time taken by the packets to travel from the client to the server.

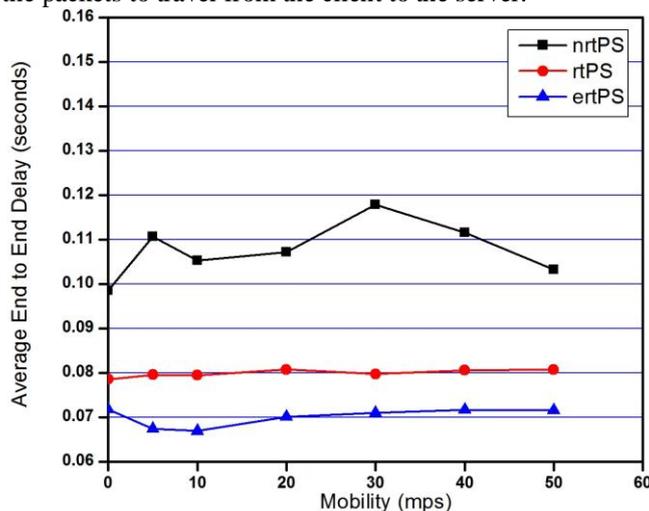


Figure 4: Average end to end delay wrt mobility

The delay is maximum for the nrtPS service flow. So, this is the expected result as this service flow is provided with the least precedence and hence it has encountered the highest delay. ertPS flow has the highest value of precedence so as expected it encounters the least delay. Also, the average end to end delay for rtPS flow remains almost same with increasing mobility. rtPS flow encounters slightly higher delay in comparison to ertPS flow. In case of nrtPS flow, the average end to end delay is higher than rtPS flow.

### Comparative results for average jitter

Figure 5 shows comparative plot of average jitter with respect to mobility for all three service flows. Jitter is the variation in the delay introduced by the components along the communication path. It is the variation in the time between packets arriving.

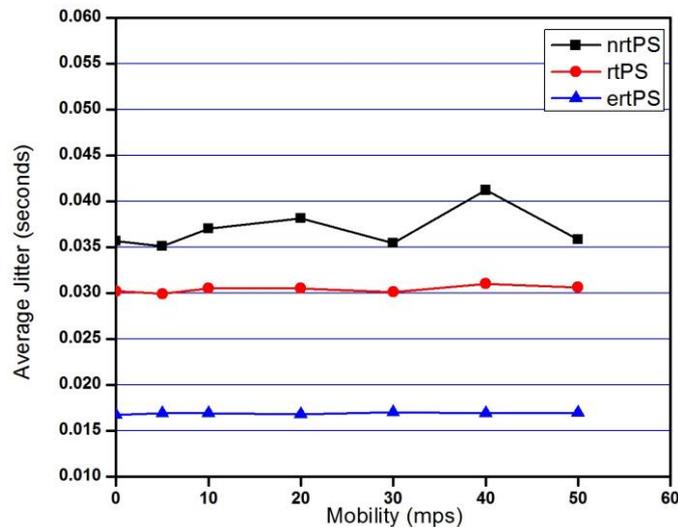


Figure 5: Average jitter for all service flows

Average jitter is the average of delay difference in successive packets. In completely ideal situation, the packets should arrive at the same delay. This will have the delay variation to be zero, implying no jitter. Jitter is commonly used as an indicator of consistency and stability of a network. As shown in Figure 5, nrtPS service flow has the highest jitter. The average jitter for rtPS and ertPS service flows does not vary much as the mobility increases. The value of jitter is very small in ertPS which is desirable.

### VII. Conclusion

In the paper, the general concepts of Quality of Service (QoS) in wireless networks were studied and an extensive comparison of rtPS, nrtPS and ertPS flows with the increase in mobility in WiMAX networks is presented. Comparative results of packet loss as calculated from the total message received shows an increase in all cases with the increase in mobility. In case of nrtPS flow it is maximum and is least in rtPS flow. So, with respect to this performance parameter rtPS flow perform better other service flows. Comparative result for received throughput with respect to mobility for all three service flows shows similar trend. At lower values of mobility throughput is high but at higher mobility it is low for all the service flows. As the simulation is configured without having any congestion so the throughput does not show many changes in all three service flows in the simulated scenario. The average end to end delay is almost similar in all cases and shows negligible changes. It is maximum in case of nrtPS flow. rtPS flow shows minimum delay variation out of all three service flows. ertPS performs best with respect to average end to end delay performance parameter. Average jitter is commonly used as an indicator of consistency and stability of network. The average end to end delay should be constant from one node to another in order to have less jitter. The value of average jitter does not vary much with the increase in mobility in rtPS, nrtPS and ertPS flows. The plot of average jitter shows that nrtPS perform worst whereas ertPS perform best with respect to this performance parameter. Finally, it can be concluded from the results that CBR traffic can be best served by extended real time Polling Service (ertPS) flow as they serve the traffic in the most optimum way.

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