A PARAMETRIC QOS ANALYSIS OF WIMAX SCHEDULING SERVICES

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ABSTRACT: *In recent era, ubiquitous, high Quality of Service (QoS), pledged mobile data is exceedingly demanded. With the mass enhancement of research activities QoS guarantee mechanisms are becoming significant. IEEE 802.16 promises to provide wireless access over long remoteness in a variety of momentums and becoming challenging in wireless environments. We have analyzed and focused different QoS parameters on various scheduling services observed in our simulated WiMAX network. The simulation study considers throughput, packet delivery ratio, number of packets dropped, delay and jitter parameters with three core services for handling essential traffic of ERTPS, RTPS and UGS . We have considered various packet sizes analyzed with each of the service separately. The study also considered the data rate change effects amongst these service parameters. On the basis of our simulations we propose to use the packet size of 200 to ensure the delivery of maximum packets to the destination. We suggest the fewer throughputs of dropped packets in order to improve the QOS. Comparative performance of UGS and RTPS are also studied. The highest priority of UGS is investigated amongst all the studied services. These investigation studies and achievements have never been reported in the literature till date to the best of our knowledge. We address the trade-off problem of longer delay and packet drops in ERTPS connection during contention period in the current study and propose a new strategy for admission control design at BS for its resolution as a future study.*

Keywords: QoS, WiMAX, Scheduling Services, ERTPS, RTPS, UGS

1. INTRODUCTION

Wifi sparked the wireless broadband revolution, unleashing the users from wires and providing opportunity to access internet smoothly. On the other hand Wifi has remained constrained to homes, offices, hotspots or coffee shops due to its short range wireless technology, extending limits only within hundred meters [4]. Whereas Worldwide Interoperatibility for Microwave Access (WiMAX) is a very well known standard, IEEE 802.16, "last mile" broadband technology, which ensures to crack distance limitations of Wireless Fidelity (Wi-Fi) by offering 70 Mbps speeds over 50 km radius[1] Rcent updates providing upto 1 Gbps. The IEEE 802.16 family of wireless-networks standards is endorsed by the WiMAX Forum. A service named WiBro, which was earlier marketed in Korea, is technically adopted to be WiMAX [3] . Fixed WiMAX which is known as the original IEEE 802.16 standard was published in 2001 and later on updgraded to be Mobile WiMAX (originally based on 802.16e-2005), which is the revision deployed in many countries, and is basis of future revisions such as 802.16m-2011.

Over the last decade there has been a major boost in communication networks, specifically the challenge for .Broadband Wireless Access(BWA)networks, in providing Quality of Service (QoS) along with the services with very different characteristics. QoS support in wireless networks is a much more difficult task than in wired networks, mainly because the characteristics of a wireless link are highly variable and unpredictable, both on a time-dependent basis and a location dependent basis. To cope with such issues, QoS in wireless networks is usually managed at the medium access control (MAC) layer[5].

In this research work four QoS parameters for three WiMaX scheduling services namely Unsolicited Grant service(UGS),Real Time Polling service (RTPS) and Enhanced real time Polling service (ERTPS) are scrutinized. The Packet delivery ratio, End to End delay, Throughput, Jitter and cumulative sum of dropped packets is are examined. Two network parameters, data rate and packet size, are taken as metrices on the basis of which the QoS service parameters are explored for each service. The rest of the paper is arranged as following. Section II describes QoS architecture, section III presents the simulation setup, the next section presents the results and analysis and section V concludes the paper.

2. QOS ARCHITECTURE

QOS support is one of the essential features of WiMaX, due to its connection oriented MAC architecture, where BS is responsible for serving all uplink and downlink connections[3]. A uni-directional logical link is established before any kind of transmission takes place. Temporary addresses are assigned for data transmission over particular link which are termed as connection Identifier (CID), through which connections are identified [5].

Since 802.16 is a connection-oriented MAC therefore it assigns traffic to a service flow and maps it to MAC connection using a CID. In this way, even connectionless protocols, such as IP and UDP, are transformed into connection-oriented service flows, which is defined by a particular Service Flow Identifier SFID). SFIDs are assigned by Base Station depending upon the type of traffic application which is also responsible for mapping it with unique CIDs[2]. In order to achieve QoS requirements for diverse traffic applications WiMAX comes up with five scheduling services which are defined below[8].

2.1 UGS

This service is completely designed to support Constant bit rate (CBR) traffic, such as audio streaming and Voice over IP (VOIP). The BS provides fixed size data grants at periodic intervals to UGS flow thus it does not send the bandwidth request.

2.2 RTPS

supports real time data streams such as MPEG videos. It allows SS to specify the size of desired grants. Polling mechanism is used in order to request the transmission resources.

2.3 ERTPS

combines the efficiency of UGS as well as RTPS Before it sends request for transmission resources its connection may experience longer delays or even packet drops. This service flow appears in IEEE 802.16e (Mobile WiMAX)..

2.4 NRTPS

comes as bandwidth demanding non real time service flow with variable packet size such as large file transfers. Used for those applications which are not sensitive to delay and jitter.

2.5 Best Effort (BE)

supports the low priority elastic traffic such as telnet or HTTP. It gives no guarantee in terms of throughput and packet delays. The foremost task in this work is to analyze the performance of three significant WiMAX scheduling services which are UGS, RTPS and ERTPS .

3. ANALYSIS SCENARIO AND SCRUTINIZING PARAMETERS

We have simulated three of these services using Network Simulator 2 (NS-2) model.. The simulation code is the main tcl file to be run in NS-2.29 which produces two more files. Nam file with .nam suffix shows the animated model of the scenario. In our coding a single BS at center location rounded by remaining eight nodes SS. This model is illustrated in Fig 1. The transmission signals in form of circles generated from nodes requesting bandwidth; while the packets roaming in the network are shown by small dots.

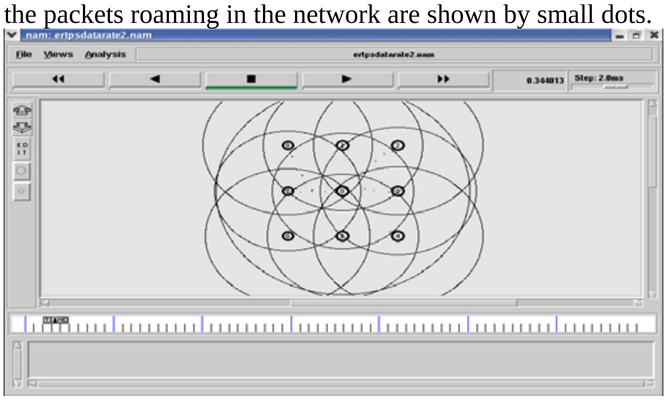

Figure 1: The animated model simulation scenario.

The trace file with .tr suffix gives out the resultant facts which are the traces of all nine nodes from which one BS, that is node 0, and the remaining eight SSs. However, there is a bandwidth request send, from which SS, that is node 1 to node 8, how many times, what bandwidth is allocated by BS and to which node, all of this information is contained one trace file. These traces are studied by “MGRTOOL” for concluding the results. The next section discusses the results obtained by the simulation carried out using this scenario model and their analysis.

As discussed earlier that four different QoS parameters are simulated in this work, which scrutinize the services under observation. These parameters are characterized in the sequel.

3.1 Packet Delivery Ratio

Packet Delivery Ratio can be defined as ratio of the packets delivered to the destination to the packets generated by the traffic sources [17].

3.2 Average End-to-End Delay

Average End-to-End Delay includes all possible delays caused by buffering during the route discovery, queuing at the interface queue, delaying at the MAC layer and the propagation times [17].

3.3 Throughput

In communication networks, throughput is the amount of digital data per time unit that is delivered over a physical or logical link, or that is passing through a certain network node [17].

3.4 Jitter

Jitter is an unwanted variation of one or more signal characteristics in electronics and telecommunications. Jitter may be seen in characteristics such as the interval between successive pulses, or the amplitude, frequency, and phase of successive cycles [17]. Jitter period is the interval between two times of maximum effect (or between two times of minimum effect) of a jitter characteristic, for a jitter that varies regularly with time.

4. SIMULATION RESULTS

This section presents the simulated results and analysis for the parameters detailed in the previous section. In order to calculate the packet delivery ratio, initially the number of “sent packets“, in a given format of the trace file, is calculated. In the similar fashion the “received packets” are calculated, and finally their ratio is obtained. It is observed that by increasing the data rate the cumulative sum of sent packets in all services is also increased. Therefore more packets are assumed to be sent to the subscriber stations from the base station with the increase of data rate. However, when the packet size is increased the cumulative sum of sent packets is not that much effected as it was assumed with the data rate. The sum of sent packets and the sum of received packets is found to calculate the packet delivery ratio. The packet delivery ratio for ERTPS, RTPS and UGS is compared in Fig.

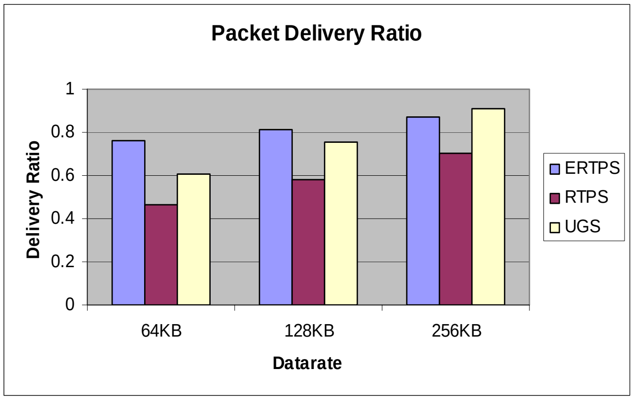

Figure 2: Comparison of the Packet Delivery Ratio Changing Data rate

It is observed from Fig 2. that increasing the data rate effects the packet delivery ratio, as soon as more packets are received at the destination from the base station. ERTPS gives us better performance than other two services. Fig. 5 shows packet delivery ratio for the changing packet size. It is evident that UGS is far better than the RTPS as far as delivery ratio of packets is concerned.

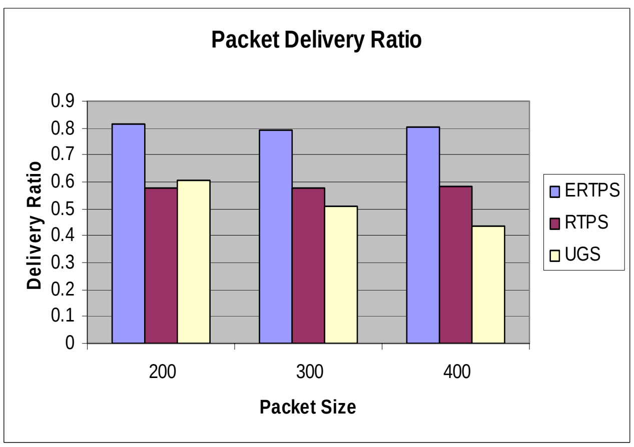

Figure 3: Packet Delivery Ratio by Changing Packet Size

Varying the packet size, it is observed that ERTPS has almost no effect on the packet delivery ratio. The packets are sent to their destination regardless of the packet size. But in RTPS and UGS the case is not the same. When the packet size is increased RTPS is effected a little while UGS is much more effected than other two services. It concludes that that ERTPS gives us best performance as it has higher number of sent packets and consequently more packets are received. If the data rate is increased then more packets are sent and received. The packet delivery ratio of ERTPS is better than RTPS and UGS. It is construed to use the packet size of 200 to ensure the delivery of maximum packets to the destination. Fig. 4 shows the cumulative sum of dropped packets during the entire simulation time.

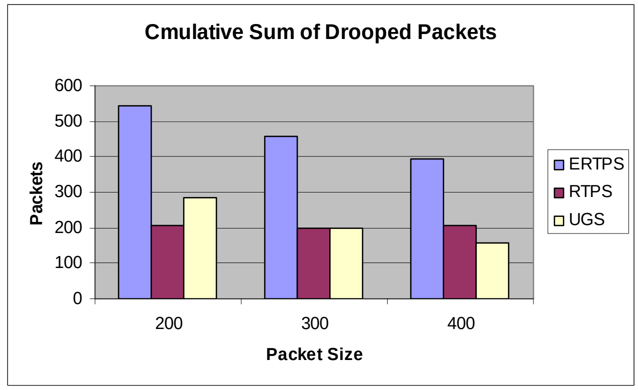

Figure 4: Cumulative Sum of Dropped Packets by Changing Packet Size

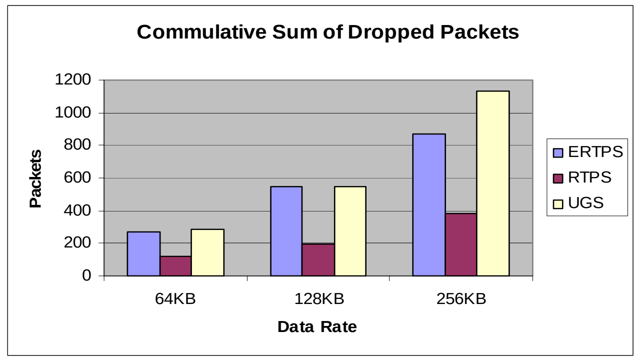

Figure 5: Cumulative Sum of Dropped Packets by Changing Data rate

With the increase of the data rate more packets are dropped. Packets are dropped when the packet queues are full due to congestion or due to bit errors. Whereas with the increase of the packet sizes the number of dropped packet are fewer as compared to the small sized packets. However increasing the packet size requires more time to be transmitted to the proper destination and consequently delay is increased. It is therefore concluded to set the packet size of 300 in order to reduce the amount of dropped packets. .

For calculating the End-to-End delay, we have to calculate the send(s) and receive(r) time for each packet which is calculated with id (li) of Trace Level (AGT) and type CBR, and then average is calculated. The End-to-End delay for ERTPS, RTPS and UGS is compared in Fig 6.

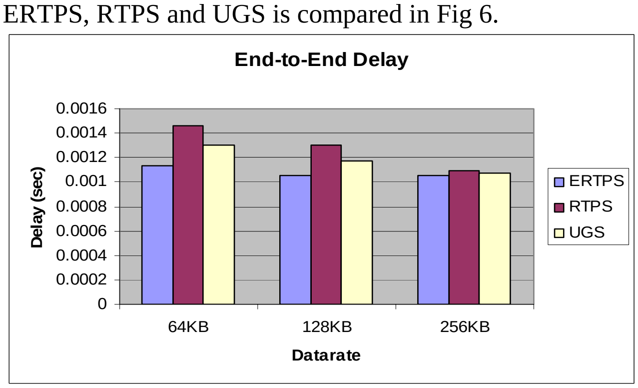

Figure 6 : Average End to End Delay by Changing Data rate

It is observed that increasing data rate effects the end to end delay. If the data rate is increased the subscriber stations get faster data than the delay of receiving packet which decreases. Increasing the packet size effects the delay which is shown in Fig 7.

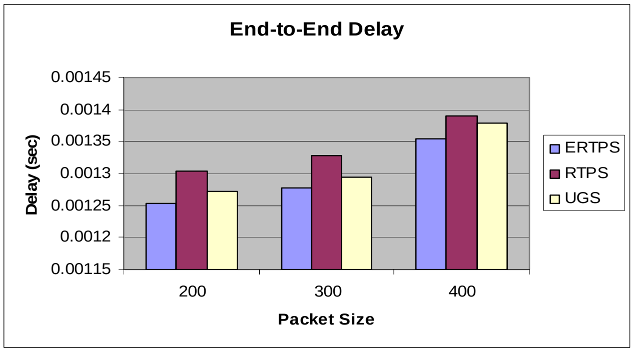

Figure 7: Average End to End Delay by packet size

From the simulation scenario it is pragmatic that by increasing the packet size the subscriber stations have to suffer from delay. Because packet size is large, it introduces some delay to reach at their proper destination. Again the ERTPS gives us best results in both the cases whether we increase the data rate or we increase the packet size, the end to end delay is very much less than RTPS and UGS.

The throughput of sending and receiving packets is also analyzed. Throughput is observed to be dissimilar for different services. Fig. 8 presents the throughput of sending packets when data rate is increased.

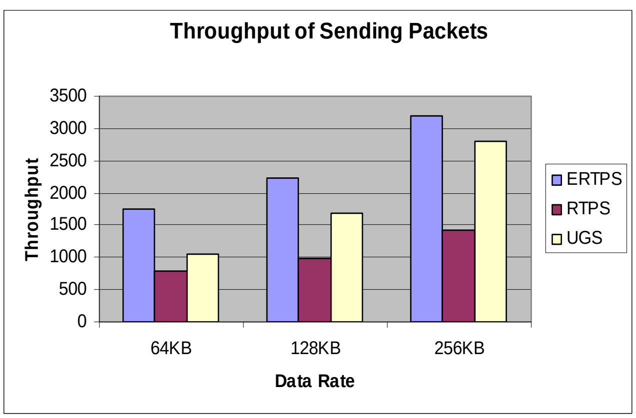

Figure 8: Throughput of Sending Packets by Changing Data rate

It is observed that throughput increases proportionally with data rate. If the data rate is increased the throughput increases. However it varies for different services. In ERTPS, it is better than other two, shown in Fig. 8.

When the packet size is increased the throughput decreases in UGS whereas increasing the packet size doesn't effect much in RTPS. In ERTPS decrease in throughput is observed when the packet size is increased.

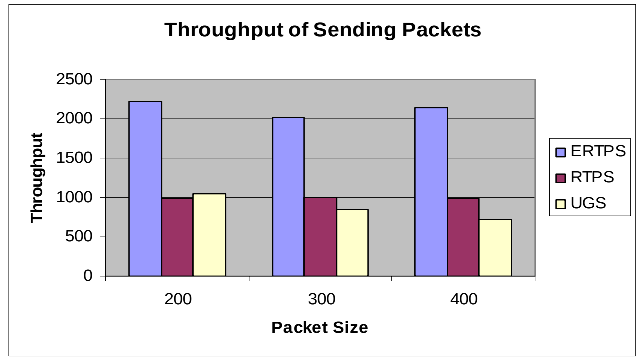

Figure 9: Throughput of Sending Packets by Changing Packet Size

Fig. 10 depicts the throughput of receiving packets when data rate is increased.

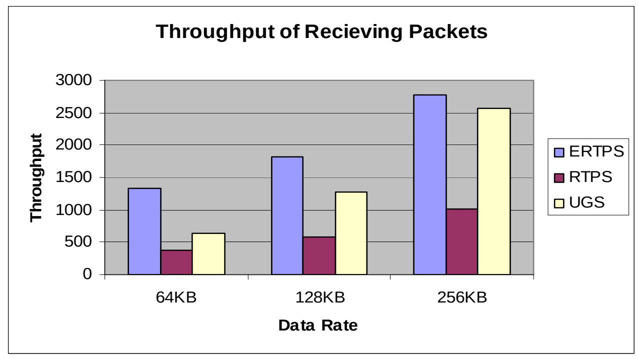

Figure 10: Throughput of Receiving Packets by Changing Data rate

By increasing the data rate throughput increased but it also varies for different services. ERTPS is again found to be better than other two. While UGS gives better results than RTPS.

Fig. 11 analyzes the throughput by changing the packet size ranging from 200,300 and 400.

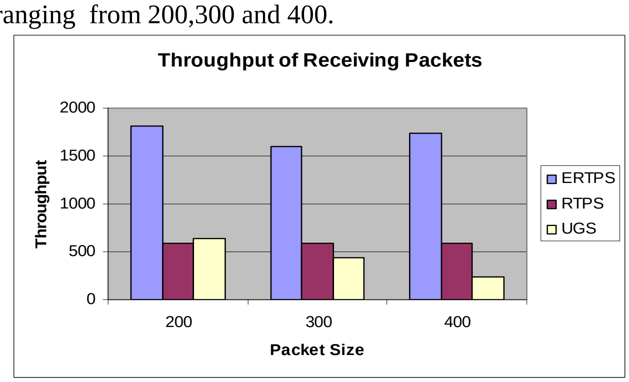

Figure 11: Throughput of Receiving Packets by Changing Packet Size

Increasing the packet size effects much on UGS but RTPS service is not much effected by this increase. Throughput remains almost the same in RTPS. In ERTPS the variation in throughput is observed. Therefore in order to improve the throughput packet size of about 200 bytes is recommended which provides the optimal throughput in all the three services. In order to achieve the optimal results both parameters must be balanced in combination. Increasing the data rate results in increasing the throughput of sending and receiving this concludes that set the data rate up to that level that the receiving throughput achieves its maximum value. A packet size of 200 is suggested which gives the maximum throughput in all the observed cases. The throughput of dropped packets is minimized while receiving throughput is maximized.

Considering jitter NRTPS class is not analyzed due to its ill sensitiveness to delay and jitter. To analyze jitter, the difference in time between two consecutive packets that leave the BS is calculated. Taking the UGS connection traffic parameters into account, the ideal distance between two UGS packets is recommended to be 1ms. By calculating the jitter it is easy to notice that the maximum jitter is smaller when the BS interleaves the slots.

If the BS uses a larger frame size and puts all the slots consecutively without implementing the interleaving function, then the maximum jitter is increased. The jitter by increasing the data rate is shown in Fig 12. and by increasing the packet size is presented in Fig 13.

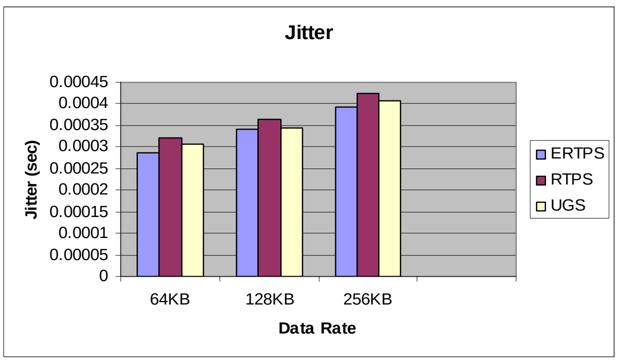

Figure 12: Jitter by Changing Data rate

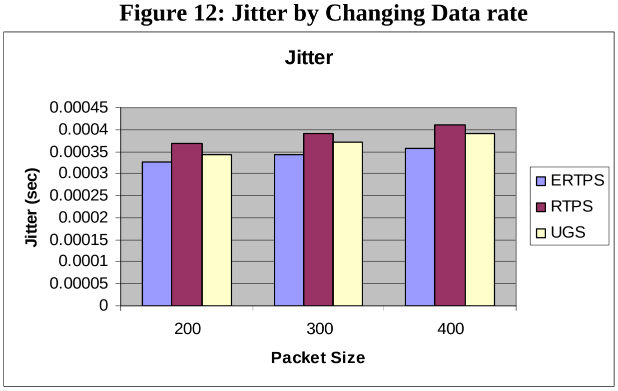

Figure 13: Jitter by Changing Packet Size

It is observed that increasing data rate results in increase of jitter. When the packet size is increased jitter is increased. There is another thing which effects the jitter that is number of maximum connections from BS to SS. If all the slots are assigned to few connections then remaining connections will suffer from longer delays and jitter. Furthermore, both jitter and delays will remain constant until those connections are assigned the slots.

5. CONCLUSION

Our simulation takes into account throughput, packet delivery ratio , number of packets dropped, delay and jitter parameters with three different core services for handling essential traffic of ERTPS, RTPS and UGS . We used various packet sizes and analyzed with different services separately. Furthermore, we changed the data rate and re-analyzed the four quality of service parameters.

It is obvious from the results that minimum packet size gives us better throughput and increasing the data rate increases it. In all the three services we found that ERTPS gives us better performance and throughput while its jitter and delay is less than other two services. On the basis of our simulations we propose to use the packet size of 200 to ensure the delivery of maximum packets to the destination. In order to improve the throughput packet size of about 200 bytes is recommended which provides the optimal throughput in all the three services. To improve the QOS the fewer throughputs of dropped packets are suggested. A packet size of of 200 is also suggested which gives the maximum throughput in all the observed cases. UGS gives us better results than RTPS because it needs not to send bandwidth request and its performance is found to be better than RTPS. Moreover, UGS has highest priority than other services. If multiple service requests have sent to the BS then UGS will be served first. This type of achievements has not been achieved in the literature till date to the best of our knowledge.

It is understandable that the ERTPS connection may experience longer delays or even packet drops before it can send the bandwidth request during the contention period. We address this trade-off problem to our future studies. A new strategy for admission control design at BS to resolve this issue is proposed for the future.

6. ACKNOWLEDGMENT

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