Improvement of the QoS for WiMAX Service Classes

Kire Jakimoski [1], Toni Janevski [2]

[1]Faculty of Information and Communication Technology, FON University, Skopje, Republic of Macedonia
[2]Faculty of Electrical Engineering and Information Technologies, Ss. Cyril and Methodius University, Republic of Macedonia

Abstract – **Mobile users connected to the WiMAX technology can use five different service classes according to their different needs. In this paper we propose priority based uplink scheduling scheme for IEEE 802.16 standard that improves the QoS performances of the WiMAX service classes, especially when the number of connections gets higher. Essential QoS parameters like throughput, delay and jitter have been calculated using the ns-2 simulation tool and changing the number of the ertPS connections. Obtained results prove that our proposed uplink scheduling algorithm gives better results compared with so far known uplink scheduling mechanism.**

Keywords – **delay, jitter, service class, throughput, uplink scheduling scheme, WiMAX.**

1. Introduction

Initially, WiMAX/802.16, the broadband wireless access (BWA) system designed for wireless metropolitan area networks, was designed for small business and residential fixed access users (IEEE 802.16-2004). But soon, in order to support the wireless mobility of the subscriber stations, the Mobile Wimax, known as IEEE 802.16e arrived on the market. IEEE 802.16e offers five service classes to response to the various demands of the users. Three of them are real-time services, Unsolicited Grant Service (UGS), extended real-time Polling Service (ertPS) and real-time Polling Service (rtPS). Non-real-time service classes are nrtPS (non-real-time Polling Service) and BE (best effort). The newest service class is ertPS. It has similar grant mechanism like UGS service class. The rtPS service class guarantees minimum traffic rate and latency bound. The nrtPS service class offers periodic unicast bandwidth request opportunities with more spaced intervals than rtPS and minimum traffic rate guarantee. BE service class shares with the nrtPS contention bandwidth request opportunities.

The quality of service (QoS) in WiMAX can be described by many parameters known as QoS parameters that can affect the performance of the WiMAX network. Main QoS parameters that are used to measure the performances of the WiMAX are throughput, delay, jitter, packet loss, packet delivery ratio. Each QoS parameter has different importance for each of the service classes. Per example, jitter is important as a QoS parameter for UGS and ertPS, but that is not the case with nrtPS and BE. Packet scheduling in the uplink direction at the base station (BS) takes into account all QoS parameters defined by the standard and it doesn't have direct access to the connections queues. It is very important part of the resourse management in WiMAX. It is more challenging than scheduling at the downlink direction, because it is more complex.

There are many proposals in the literature for different versions of uplink scheduling schemes and many authors have worked on the various QoS parameters for the WiMAX service classes. The uplink scheduler presented in [1] does not provide maximum latency guarantees. The one proposed in [2] suggests a priority value computed by the subscriber stations (SSs) in order to provide latency and rate guarantees. Authors in [3] present an uplink scheduler that is standard-compliant scheduling solution for the uplink traffic in IEEE 802.16. It uses three queues, low, intermediate and high priority queue and it supports minimum traffic rate, maximum sustained traffic rate, maximum latency and maximum traffic burst requirements. A fair uplink scheduler is proposed in [4] and it is based on the values of TCP timeout and congestion window and on the channel conditions. Authors in [5] present a cross-layer scheduler that dynamically adjusts the size of the uplink and downlink subframes according to the network conditions. The scheduling scheme presented in [6] classifies packets in 4 classes in order to guarantee latency requirements for real time applications. In [7] authors present an uplink scheduling algorithm that assigns priority values to the connections on the basis of the service class priority, the delay of the packets, the status of the queue and the quality of the channel.

The work presented in this paper extends the uplink scheduler solution proposed in [3], so that we introduce five levels of priority from 1 to 5 (1 is the highest and 5 is the lowest priority value) that are applied respectively to the five service classes ertPS, UGS, rtPS, nrtPS and BE. In this way bandwidth allocation decisions, besides the level of priority of the three queues proposed in [3], also take into account the level of priority of each service class according to our proposed scheme. In [8] the authors analyzed various QoS parameters like packet loss, throughput, average delay and average jitter for VoIP and video traffic using the ns-2 simulator. In [9] authors carried out analysis of a location based performance scenario taking in account delay and throughput. Statistical analysis of the QoS parameters, jitter and end-to-end delay in mobile WiMAX is done in [10].

Simulation results obtained in this paper using the ns-2 simulation tool and WiMAX simulator prove that our proposed uplink scheduling algorithm improves the QoS parameters of the WiMAX service classes. We have calculated the average throughput, average delay and average jitter from the simulations using different scenarios and compared them with the results that can be obtained using the uplink scheduling mechanism proposed in [3]. Our solution especially obtains better QoS performances when the number of the connections gets higher.

The rest of this paper is organized as follows. Section 2 presents the uplink scheduling process in mobile WiMAX. Section 3 describes the proposed uplink scheduling algorithm. Section 4 explains the simulation scenario used to evaluate the proposed uplink scheduling algorithm and presents the simulation results. Section 5 concludes the paper.

2. Uplink Scheduling Process in Mobile WiMAX

In the point-to-multipoint mode in Mobile WiMAX technology the base station (BS) communicates with several subscriber stations (SS) in the WiMAX cell. Connections that are made between the BS and the SS are identified by a CID (Connection Identifier) and are structured in frames. One CID can present one application or a group of applications. When the transmission of the traffic is from the SSs to the BS, it is called uplink transmission (UL). When the transmission is in the opposite direction from the BS to the SSs, it is called downlink (DL) transmission. The admission of the new connection is performed by the BS on the basis of the current situation with the available resources. If the QoS requirement of the requested bandwidth is supported the BS will generate CID and Service Flow Identifier and will notify the SS. The admission control algorithm will accept a new connection when this condition will be fulfilled:

$$C_{reserved} + TR_i^{service} \leq C \qquad (1)$$

$TR_i^{service}$ is the traffic rate of the new connection i of one of the WiMAX service types denoted with *service*. In (1) the already reserved capacity is denoted with $C_{reserved}$ and it is equal to:

$$C_{reserved} = \sum_{i=1}^{n} TR_i^{service} \qquad (2)$$

The whole capacity that is available for the uplink scheduler is denoted with C. It is the amount of uplink bandwidth that can be allocated by the uplink scheduler for transmission and unicast polling.

The scheduling schemes in the MAC layer in Mobile WiMAX are designed to deliver successfully different service classes to the users over the wireless channel. The set of QoS parameters of the five scheduling types at MAC layer (UGS, rtPS, ertPS, nrtPS, BE) are presented in Table 1.

WiMAX service class	Applications	QoS Specification
UGS Unsolicited Grant Service	VoIP	Maximum Sustained Rate Maximum Latency Tolerance Jitter Tolerance
rtPS Real-Time Polling Service	Streaming Audio or Video	Minimum Reserved Rate Maximum Sustained Rate Maximum Latency Tolerance Traffic Priority
ertPS Extended Real-Time Polling Service	Voice with Activity Detection (VoIP)	Minimum Reserved Rate Maximum Sustained Rate Maximum Latency Tolerance Traffic Priority
nrtPS Non-Real-Time Polling Service	File Transfer Protocol (FTP)	Minimum Reserved Rate Maximum Sustained Rate Traffic Priority
BE Best-Effort Service	Data Transfer, Web Browsing	Maximum Sustained Rate Traffic Priority

Table 1. Mobile WiMAX service classes and QoS.

There are three different scheduling categories in the Mobile WiMAX networks. Two of them are located at the WiMAX base stations and one at the subscriber stations. DL-BS and UL-BS are the first two scheduling processes that are located at the BS and the third is used for uplink located at the SSs after receiving grants from the UL-BS scheduler. The UL-BS scheduling process is the most complicated because when the WiMAX base station makes scheduling decisions it has no updated information about the current queue status at the SSs. Due to this, the WiMAX BS estimates the current queue status on the basis of the bandwidth requests received from the SSs. The DL-BS scheduling process is much easier because the base station has the current information about the queue status of all downlink connections.

During the uplink transmission of the data, packets are queued at the SSs. In this position the uplink scheduler operates on a request-grant basis. Each SS sends a message with a bandwidth request to the base station. After receiving the bandwidth request messages, they are classified according to the QoS parameters and service classes in the scheduler and bandwidth allocation process starts. In the UL-MAP an Information Element (IE) is responsible for the new resource assignments and control region that SSs should transmit. The UL-MAP is located in each frame at the beginning of its DL subframe. Each SS identifies CID of the broadcast MAP message and then decodes the IE of the UL-MAP. After this, the packets are sent according to the allocated slots.

The newest class of the WiMAX standard - ertPS combines the efficiency of the UGS and rtPS classes. The allocation of slots is similar to the rtPS class:

$$N_i^{\min} = N_i^{\max} = \begin{cases} 1, & R_i = 0, \\ \left\lceil \dfrac{T_i}{S_i FPS} \right\rceil, & R_i > 0, \end{cases} \quad (3)$$
$$\forall i \mid C_i = ertPS$$

N_i stands for the number of slots within each frame, R_i is the request size, T_i stands for bandwidth requirement of the *i*th connection, S_i stands for the slot size, *FPS* stands for the number of frames the WiMAX BS sends per one second, i.e. the number of bytes a connection can send in one slot, and C_i stands for the *i*th connection class.

It is interesting to note that the ertPS connection can also send the bandwidth requests during the contention period. Bandwidth requests can be updated with a minimum delay. The requested bandwidth should be:

$$T_{ertPS,req} = \max(T_{BW\_REQ}, n_i \left[T_{MAC} / T_{voice}\right] T_p \quad (4)$$

seconds per MAC frame, where n_i is the total number of active connections at the time of making the bandwidth request, $T_{BW\text{-}REQ}$ is the time for transmitting one bandwidth request message, T_{MAC} is the duration of one MAC frame, T_{voice} is the voice packetization time and T_p is the transmission time required for each voice packet. The amount of granted bandwidth is equal to min($T_{ertPS,req}$, $T_{ertPS,max}$) seconds per MAC frame, where $T_{ertPS,max}$ is the maximum amount of time available for the SS in an uplink subframe.

For the ertPS service, the maximum amount of time that each SS is allowed to transmit is given by:

$$T_{ertPS,\max} = \frac{T_{MAC\_UP} - T_{overhead}}{N_S} \quad (5)$$

N_s is the number of connections associated to subscriber stations (SSs) *i*, *i=1,2,…*N_s, T_{MAC_UP} is the duration of one MAC frame in uplink, $T_{overhead}$ is initial ranging period duration.

Packet transmission performance using the ertPS service strongly depends on the MAC frame size. With a short MAC frame duration, using the ertPS service we achieve approximately as good packet transmission performance as using the UGS service. As the frame size increases, the packet transmission performance using the ertPS service degrades.

3. Proposed Uplink Scheduling Algorithm

During the implementation of our proposed priority scheduling scheme we assumed that each SS carries single service flow. In this way the interpretation of the uplink scheduler at the BS is eased. Our proposed solution for the uplink scheduling at the SSs is based on a strict priority scheduling as it is shown in Figure 1.

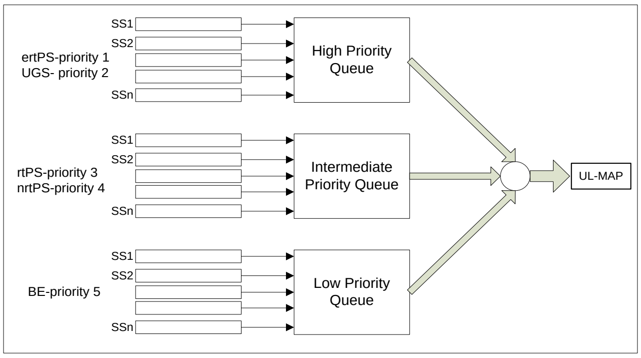

Figure 1. The architecture of the proposed uplink scheduler.

According the proposal ertPS service class has the highest priority and is served first. The proposed solution for uplink scheduling at the SSs is implemented on the basis of the scheduling mechanism explained in [3].

The scheduling mechanism in [3] uses three queues, high priority, intermediate priority and low priority queue. The queue with low priority is used for the BE bandwidth requests, the intermediate queue is reserved for storing the bandwidth request sent by rtPS and nrtPS service classes and the high priority queue is reserved for UGS and ertPS service classes. Bandwidth requests sent by both rtPS and nrtPS connections can migrate to the high priority queue to meet their QoS requirements. After upgrading the scheduling mechanism in [3] that uses queues divided into three categories with our proposed priority-based uplink scheduling, we compared our proposed solution with the solution in [3].

Our proposed uplink scheduler is based on the class' priority level with respect to other classes. Each service class is assigned with the priority as it is shown in Table 2. We use values from 1 to 5 for the class priority level according the ns-2 classification where priority 1 is the highest priority. The priority levels were set as follows: 1 for the ertPS service class, 2 for the UGS service, 3 for the rtPS service, 4 for the nrtPS service and 5 for the BE service. Although ertPS and UGS services are in the same high priority queue, with our added priority when ertPS and UGS are active services in the same time, ertPS traffic will have higher priority for uplink scheduling than UGS. Priority values of 3 and 4 are set for rtPS and nrtPS service connections, respectively, that are in the intermediate queue. This means that rtPS service class has higher priority in uplink scheduling than nrtPS service class, although they are in the same intermediate priority queue. We set higher priority for rtPS service class because it is delay sensitive service. Finally, BE service class that is in the low priority queue has priority value of 5 and is last in the uplink scheduling scheme. Our proposed solution is compared with the performances of the scheduling mechanism presented in [3] in terms of average throughput, average delay and average jitter in different scenarios with different loads of the network. As we can see from the next section our results are better.

The steps that the proposed uplink scheduling algorithm is executing are as follows:

- Insert in the gueue with high priority unicast request opportunities and the periodic data grants that must be scheduled in the next frame;
- Check if rtPS and nrtPS requests can migrate from the queue with intermediate priority to the queue with high priority with the procedures "check the deadline" and "check the minimum bandwidth". In both cases the class priority level of rtPS and nrtPS service classes (3 and 4) remains the same, i.e. they are served after ertPS and UGS service classes with class priority levels of 1 and 2;
- Uplink scheduler distributes among the BE connections the non-allocated bandwidth. BE service class has the last class priority level, that is 5.
- Finally, the uplink scheduler serves all the requests at the queue with high priority firstly and the service class with the highest priority level, that is ertPS.

4. Simulation Scenario and Results

4.1 Simulation Scenario

In the research community one of the most popular tools is the Network Simulator (ns-2). For simulation of IEEE 802.16-based networks using ns-2 several modules were proposed. One of them is implemented by NIST - National Institute of Standards and Technology, but it fails to implements MAC QoS support. Another simulation module for IEEE 802.16 that is presented in [11] provides packet fragmentation and packing, but users cannot configure QoS requirements. There is also a group that developed an 802.16-based simulator for the OPNET tool, but it is a private domain simulator. We have evaluated that the most adequate simulator for WiMAX service classes is the module designed by the group of the University of Campinas [12]. This module is focused on the MAC layer and its mechanisms for bandwidth allocation and QoS support.

The topology of the simulated network for the simulation scenarios done in this paper consists of a BS located at the center of a 250 X 250 meter area, with the SSs uniformly distributed around it. We enabled random motion to the SSs by setting the random-motion to 1 in ns-2, in which case random destinations are assigned to the nodes. Each SS has one uplink flow and one downlink flow, which are mapped to the same service type. Five types of traffic are used: voice, voice with silence suppression, video, FTP, and WEB, which are associated with UGS, ertPS, rtPS, nrtPS, and BE services, respectively. Each of the SSs has only one service flow in order to eliminate the impact of the packet scheduling at the SSs on uplink scheduling.

We used as a voice model for UGS service class an exponential model with mean duration of 1.2 seconds of the "on period" and 1.8 seconds of the "off period", respectively. For UGS service class according this model packets of 66 bytes are generated every 20 milliseconds. For simulating ertPS service class we used in the simulator EVRC – Enhanced Variable Rate Codec as a model of voice with silence suppression [13]. In our scenarios rtPS service class was simulated using 10 different real MPEG traces. nrtPS service class in the simulator that we use is generated with FTP traffic using an exponential distribution with a mean of 512 KBytes. BE service class is simulated with WEB traffic that is modeled in the simulator by a hybrid

Lognormal/Pareto distribution with the body of the distribution modeled by a Lognormal distribution with a mean of 7247 bytes and the tail modeled by a Pareto Distribution with mean of 10558 bytes.

UGS and ertPS service classes have unsolicited grant interval of 20 milliseconds. rtPS has unsolicited polling interval of 20 ms and nrtPS of 1 second. BE service class doesn't have any QoS requirement. The number of UGS, rtPS, nrtPS and BE connections is equal to 10 in all of the simulated scenarios. We only change the number of ertPS connections from 5 to 30 in order to analyze the QoS parameters in various traffic load situations. The duration of each simulation scenario is 200 seconds. Each simulation scenario is repeated 50 times changing the seeds in the ns-2 simulator. We have done 50 iterations of the scenarios in order to have more reliable results. In the figures we show the mean values and the 95% confidence intervals.

The free space model and the two-ray model in ns-2 predict the received power as a deterministic function of distance. They both denote the communication range as an ideal circle. But, in the reality, the received power at some distance is a random variable due to multipath propagation effects, known as fading effects. The free space model and the two-ray model predict the mean received power at distance *d*. Widely-used model that is more general is called the shadowing model. All of the simulated scenarios in our work are using the Shadowing propagation model. This model consists of two parts. The first part is the path loss model. It predicts the mean received power at some distance *d*, denoted by $\overline{P_r(d)}$. Close-in distance d_0 is used as a reference. $\overline{P_r(d)}$ is computed relative to $P_r(d_0)$ as follows:

$$\frac{P_r(d_0)}{\overline{P_r(d)}} = \left(\frac{d}{d_0}\right)^{\beta} \tag{6}$$

β is denoted as path loss exponent which is usually determined empirically by measurements on the field. The value of the path loss exponent is 2 in all of the simulated scenarios. It is typical value for free space propagation. Because the path loss exponent is measured in dB in most cases, from (6) we have:

$$\left[\frac{\overline{P_r(d)}}{P_r(d_0)}\right]_{dB} = -10\beta\log\left(\frac{d}{d_0}\right) \tag{7}$$

The second part of this model used in our simulations reflects the variation of the received power and it can be presented by the following equation:

$$\left[\frac{\overline{P_r(d)}}{P_r(d_0)}\right]_{dB} = -10\beta\log\left(\frac{d}{d_0}\right) + X_{dB} \tag{8}$$

X_{dB} is Gaussian random variable with zero mean at standard deviation denoted with δ_{dB}. This standard deviation δ_{dB} is sometimes called the shadowing deviation. This value is also obtained empirically by measurements. In our simulations this value was set to 4.

In order to analyze the performances of the proposed uplink scheduling algorithm changing the number of ertPS connections without changing the number of connections of the other four services, we varied the number of ertPS connections from 5 to 30 with a step of 5 connections. In this way we also analyze the impact of the number of ertPS connections on the other four service classes. So, we made 6 scenarios where each of the other 4 service classes had 10 connections, while the number of ertPS connections was 5, 10, 15, 20, 25 and 30. The results are shown in average throughput, average delay and average jitter for all five service classes. We haven't presented the average delay and average jitter of nrtPS and BE service classes, because they are non-real-time services, so delay and jitter are not affecting them.

4.2 Simulation Results

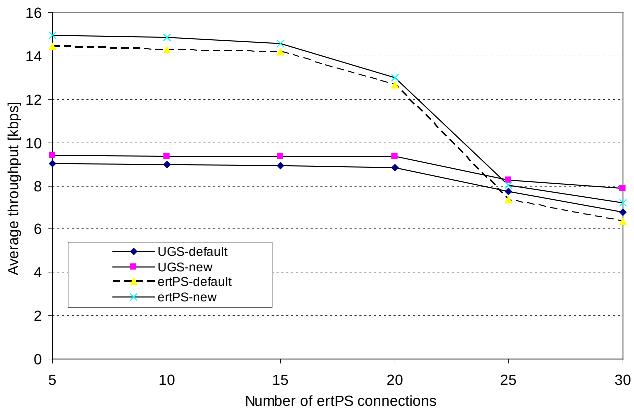

Figure 2. Average throughput of 10 UGS connections and 5 to 30 ertPS connections.

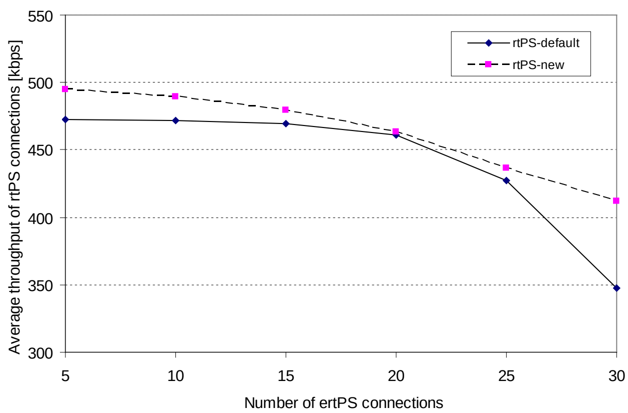

Figure 3. Average throughput of 10 rtPS connections changing the number of ertPS connections.

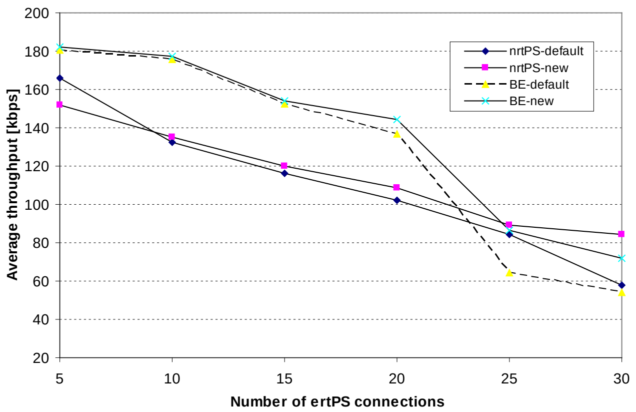

Figure 4. Average throughput of 10 nrtPS and 10 BE connections changing the number of ertPS connections.

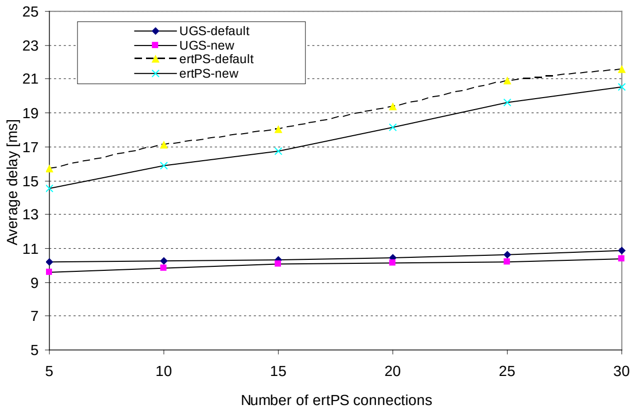

Figure 5. Average delay of 10 UGS connections and average delay of 5 to 30 ertPS connections.

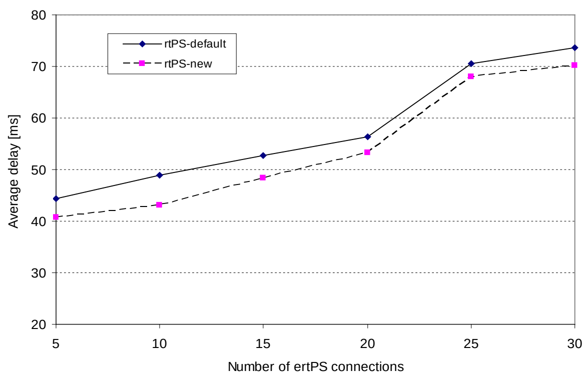

Figure 6. Average delay of 10 rtPS connections under different load of ertPS connections.

Figure 2. presents the average throughput results of 10 UGS connections when the number of the ertPS connections varies from 5 to 30. Results of the average throughput of the variable ertPS connections are also shown on the same graph. The number of rtPS, nrtPS and BE connections is 10 in all cases. UGS-default and ertPS-default in Fig. 2 denote the results obtained with the scheduling mechanism in [3], while UGS-new and ertPS-new denote the results obtained with our modified scheduling scheme. Results show that our solution gives better average throughput results for UGS and ertPS service classes fo all six different traffic loads with ertPS. We can also conclude from the results in Fig. 2 that our proposed uplink scheduling algorithm especially gives better results compared with the one in [3] when the number of ertPS connections gets higher. This is also the case for the average throughput results of rtPS, nrtPS and BE service classes in Fig. 3 and 4.

Furthermore, besides comparing the average results of 10 rtPS connections of our proposed uplink scheduling scheme with other scheme in Fig. 3, we verify in Fig. 3 the minimum and the maximum traffic rate requirements as specified by the standard. In all of the simulated scenarios, the minimum rate requirement of 300 kbps for the average throughput results of rtPS connections and the maximum rate requirement of 500 kbps were in the range regardless of the network load with ertPS connections.

The concluding remarks for Fig. 3 and 4 are also the same as in Fig. 2, for the presented average throughput results of 10 rtPS, 10 nrtPS and 10 BE connections under different number of ertPS connections. Comparing the impact of the increasing the number of ertPS connections on the other four service classes, we can conclude that the increase of the number of the ertPS connections has the lowest impact on the UGS service class. This situation can be explained by the fact that UGS service class is also in the high priority queue as ertPS service class and packets from this service class has more priority than packets from rtPS, nrtPS and BE service classes.

Fig. 5 presents the average delay results for UGS and ertPS service classes. Fig. 6 shows the average delay results of the rtPS service class. After applying our priority based scheme, results for the ertPS service class are obviously better. In the same time average delay of 10 UGS connections is not much degraded when increasing the number of ertPS connections in both cases, before and after the implementation of our proposed scheduling scheme. We explained the reason for this situation in the analysis of the Fig. 2. Average delay of the rtPS service class in Fig. 6 increases when the number of ertPS connections increase in both cases. Reason for this is the fact that the rtPS service class is in intermediate queue and has less priority than ertPS and UGS service classes.

Fig. 7 shows the average jitter results for UGS and ertPS service classes. Similarly to average delay results, average jitter results for ertPS service class are better using the proposed scheduling algorithm compared to the scheduling mechanism in [3]. Average jitter results for UGS service class are in both cases not affected by the increase of the ertPS connections, because this class is in the high priority queue and has class priority 2 according our proposed scheduling order. Their values are around 6 ms in all of the simulated scenarios in both cases.

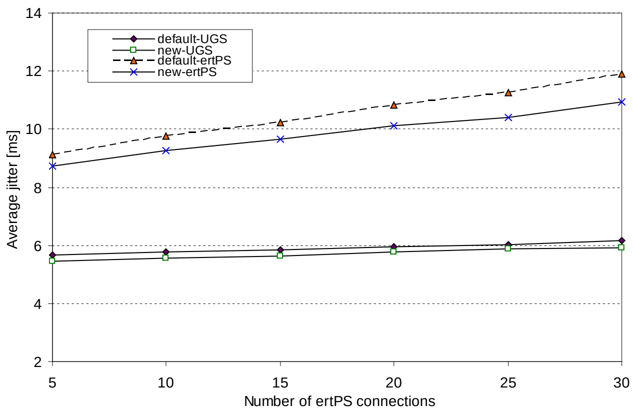

Figure 7. Average jitter for 10 UGS and 5 to 30 ertPS connections.

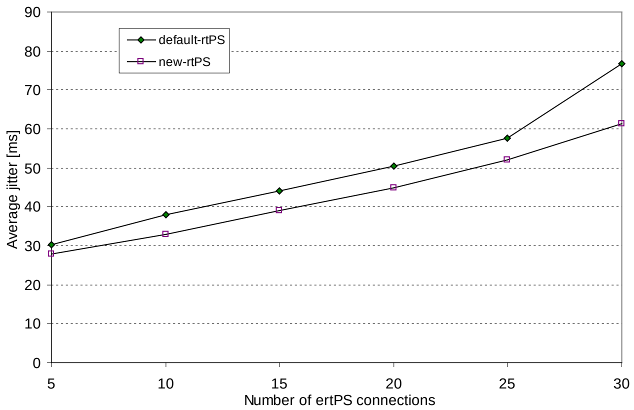

Figure 8. Average jitter for 10 rtPS connections changing the number of ertPS connections.

Fig. 8 presents the average jitter results for 10 rtPS connections changing the number of ertPS connections from 5 to 30 comparing the uplink scheduling mechanism (denoted with rtPS-default) in [3] with our proposed scheduling order (denoted with rtPS-new). Compared results prove that our proposed scheduling algorithm gives better results, especially when the number of ertPS connections gets very high. Hence, we can conclude that the impact of the increase of the number of ertPS connections on degrading the rtPS average jitter is lower when we use our proposed uplink scheduling order.

5. Conclusion

In this paper we have proposed priority based uplink scheduling algorithm for WiMAX service classes. It is implemented using the ns-2 simulator and the WiMAX patch that was developed in [12]. After the modification of the simulator with our added priority based uplink scheduling order in ns-2, we have analyzed the QoS parameters changing the number of ertPS connections from 5 to 30, while keeping the other four service classes with 10 connections. Obtained results are then compared to the already presented effective uplink scheduling mechanism in [3]. Results that are showed in Section 4 prove that the proposed priority based scheduling mechanism gives better results measured in average throughput, average delay and average jitter compared with the uplink scheduling mechanism in [3].

Furthermore, because we have changed the number of ertPS connections from 5 to 30, we have analyzed the impact of the number of ertPS connections on the other four service classes. The impact of the increase of the ertPS traffic load on the UGS service class is negligible in both compared cases, especially for average delay and jitter, because UGS service class is in the high priority queue together with ertPS and has class priority value of 2 according our proposed scheme. The impact on other service classes is noticeable in both compared cases, but our proposed uplink scheduling scheme gives better results for average throughput, delay and jitter, especially when the number of ertPS connections is higher. Hence, using our proposed scheduling order, we decreased the negative impact of the increasing the number of ertPS connections on rtPS, nrtPS and BE service classes.

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Corresponding author: *Kire Jakimoski*
Institution: *Faculty of Information and Communication Technology, FON University, Skopje, Republic of Macedonia*
E-mail: *kire.jakimoski@fon.edu.mk*