Adaptive Contention Resolution for VoIP Services in the IEEE 802.16 Networks

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Abstract

In the IEEE 802.16 networks, a subscriber station can use the contention slots to send bandwidth requests to the base station. The contention resolution mechanism is controlled by the backoff start/end values and a number of the request transmission opportunities. These parameters are set by the base station and are announced to subscriber stations in the management messages. In the case of the VoIP services, it is critical that the contention resolution occurs within the specified time interval to meet the VoIP QoS requirements. Thus, it is the responsibility of the base station to set correct contention resolution parameters to ensure the QoS requirements. This paper presents analytical calculations for the parameters that control the contention resolution process in the IEEE 802.16 WiMAX networks. The simulation results confirm the correctness of theoretical calculations. They also reveal that the adaptive parameter tuning results in a better resource utilization when compared to the static configuration. At the same time, all the delay requirements are ensured.

1 Introduction

The explosive growth of the Internet demands more advanced and reliable multimedia services. One of these services is Voice-over-IP (VoIP). Being connected to the Internet backbone networks through wired, local and broadband wireless connections, customers can use the VoIP services at considerably lower price when compared to the conventional telephone lines and mobile networks.

WiMAX is an IEEE standard for the wireless broadband access network [1]. The main advantages of WiMAX when compared to other network access technologies, such as WLAN, are the longer range and more sophisticated support for the Quality-of-Service (QoS) at the MAC level. Various application and service types can be used in the WiMAX networks and the MAC layer is designed to support this convergence. A good overview of the key WiMAX features is given in [5]. The standard defines two basic operational modes: point-to-multipoint (PMP) and Mesh. While a subscriber station (SS) can communicate with other SSs in the Mesh mode, it is allowed to communicate only through the base station (BS) in the PMP mode. It is anticipated that providers will use the PMP mode to connect customers to the Internet [10]. In this case, a provider can control the environment to ensure the QoS requirements of customers. Recently, the IEEE 802.16e standard [2] defined mobile extensions which opened a possibility to build mobile wireless VoIP terminals and embed WiMAX into mobile phones and personal data assistants.

WiMAX provides a rich set of scheduling classes to support services with various traffic profiles and requirements. As a result, there is no unique choice for the scheduling class to support the VoIP services. As stated in [1], a provider can use the Unsolicited Grant Service (UGS) to support the VoIP applications *without* silence suppression. Indeed, since the scheduler at the BS allocates fixed data grants at the periodic time intervals to the UGS connection, such a behaviour fits ideally the nature of the VoIP codec. Of course, the UGS class can also support the VoIP codecs *with* silence suppression, but some bandwidth resources will be always wasted during the silence phase because the BS will allocate slots while an SS will not utilize them. To avoid wasting bandwidth resources, a provider can rely upon the real-time Polling Service (rtPS). However, the premise idea behind the rtPS class is that the traffic profile is of the *variable* bit-rate nature, such as video data. Consequently, the scheduler at the BS allocates resources according to the bandwidth request sizes [13]. Such a resource allocation differs from the VoIP traffic profile because the VoIP codec either outputs data at some constant rate (active phase), or does not send data at all (silence phase). Another problem relates to the fact that the rtPS QoS profile has the minimum and maximum bandwidth. However, since the VoIP application generates the constant rate traffic, there is no notion of the minimum/maximum bandwidth. The 802.16e standard came up with the extended real-time Polling Service (ertPS) that accounts for all these peculiari-

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ties. It combines efficiency of the UGS and rtPS classes and aims to support the VoIP codecs with silence suppression. A comparison of these classes is presented in [16].

It is important to note that unlike the UGS and rtPS classes, the ertPS class is allowed to take part in the WiMAX *contention resolution* process. Since the VoIP application activity is characterized by the active and silent phases, the scheduler at the BS is free to choose a number of slots to allocate for the ertPS connections in the silent phase. On the one hand, if the BS performs polling, i.e., allocates at least one slot, then the ertPS connection can send the bandwidth request at any time resulting in a small medium access delay, but some resources are wasted for polling. On the other hand, the BS can achieve better resource utilization by reserving no polling slots and allocating free slots to other connections. However, every time the active phase starts, the ertPS connection has to go through the contention resolution mechanism to send the bandwidth request.

In [14], we presented an analysis of various solutions for the VoIP services. It revealed that a provider can achieve a considerably better resource utilization if the ertPS connections use the contention mechanism to send the bandwidth requests. However, since the VoIP application cannot tolerate large delays, it is crucial that the contention resolution is completed within the required time interval. As presented in [14], it is achievable if the correct contention resolution parameters are used.

The WiMAX contention resolution mechanism is controlled by two sets of parameters: a number of the request contention slots and the backoff start/end window values. While a number of the request contention slots determines the overall number of the transmission opportunities during which an SS can send the bandwidth request, the backoff start/end window values control *when* an SS sends the bandwidth request. These parameters are set by the BS and announced to SSs in the uplink map (UL-MAP) and the uplink channel descriptor (UCD) messages. Thus, the BS can control flexibly SSs and the contention mechanism. The 802.16 specification left open concrete values for the backoff start/end window values, neither does it specify how a number of the request transmission opportunities should be calculated. Thus, the task of the BS is to use some algorithm to set these parameters. On the one hand, knowing a mean number of the inactive connections, a provider can always choose an appropriate static configuration. On the other hand, a better resource allocation is achieved if the parameters are tuned adaptively. Besides, it eliminates the need to find an optimal static configuration, especially when a number of competing connections changes drastically. It is especially crucial for the VoIP services that pose strict delay requirements.

In this paper, we extend our previous research work [12], in which we presented the analytical calculations to determine the backoff start/end values and the number of the request transmission opportunities. Though an idea of adapting these parameter dynamically has been considered in [11, 6, 4], either only a subset of parameter was considered or the simulation environments were too simple to capture the properties of the WiMAX network and the transport protocols. We present a complete solution that calculates the backoff start/end values and the number of the contention slots. Previously, we showed by the means of the NS-2 simulator that the adaptive approach improves the resource utilization of the TCP applications. However, we did not study the delay-critical and non-responsive services, such as VoIP. Thus, this paper focuses on the adaptive contention resolution and the VoIP services. To run simulations in the NS-2 simulator, we implemented the WiMAX MAC and PHY layers.

The rest of the article is organized as follows. Section II presents the theoretical background for the VoIP resource allocation in the WiMAX networks. The WiMAX contention resolution mechanism and the adaptive contention resolution are also presented. Next, Section III describes the simulation environment to test the adaptive approach in conjunction with the VoIP services. This section also presents and analyses the simulation results. Section IV presents other WiMAX contention resolution solutions and how they can benefit from the adaptive approach. Finally, Section V concludes the article and outlines further research directions.

2 VoIP and adaptive backoff

2.1 VoIP resource allocation

The basic approach for allocating resources in the WiMAX network is that the BS does the scheduling for both the uplink and downlink directions. In other words, an algorithm at the BS translates the QoS requirements of SSs into the appropriate number of slots within the WiMAX frame. To avoid inefficient resource allocation, the BS accounts for the *bandwidth request size* that specifies size of the SS output buffer. Furthermore, the BS accounts for the scheduling class to interpret correctly the QoS requirements and the bandwidth request size. When the BS makes a scheduling decision, it informs all SSs about it by using the UL-MAP and DL-MAP messages in the beginning of the WiMAX frame. These special messages define explicitly slots that are allocated to each SS in both the uplink and downlink directions.

According to [13], one can calculate a number of slots N_i within a frame by using the following basic expression:

$$N_i = \left\lceil \frac{B_i}{S_i \text{FPS}} \right\rceil, \quad (1)$$

where B_i is the bandwidth requirement of the ith connection, S_i stands for the slot size, and FPS stands for a number of frames the WiMAX BS sends per one second. The idea behind (1) is that we determine the overall number of slots necessary to ensure the bandwidth requirements, and then divide it by a number of frames to calculate a number of slots we have to allocate in each WiMAX frame.

As mentioned earlier, the purpose of the ertPS class is to combine efficiency of the UGS and rtPS classes. If the speech is active, then a codec outputs data at some constant rate resulting in the non-zero bandwidth request size R_i. From this point of view, resource allocation is similar to UGS because we have to allocate data grants at fixed time intervals. At the same time, the request size R_i equals zero during a silence period.[1] In this case, the BS can allocate one slot thus enabling an SS to ask for more bandwidth when the active phase of the speech starts:

$$N_i = \begin{cases} 1, & R_i = 0, \quad (2a) \\ \left\lceil \frac{B_i}{S_i \text{FPS}} \right\rceil, & R_i > 0. \quad (2b) \end{cases}$$

It is important to note that no contention occurs in this case. In the active phase, an SS can piggy-back the bandwidth request to the data packets or send it as a standalone message to ask the BS to allocate slots in the next frame. In the silence phase, the BS polls an SS thus letting it a possibility to send the bandwidth request during the polling slot. Simulation results for this approach are presented in [14].

As mentioned earlier, the ertPS connection can also send the bandwidth requests during the contention period. Thus, if the request size equals zero, then the BS may avoid reserving one slot thus achieving a better resource allocation:

$$N_i = \begin{cases} 0, & R_i = 0, \quad (3a) \\ \left\lceil \frac{B_i}{S_i \text{FPS}} \right\rceil, & R_i > 0. \quad (3b) \end{cases}$$

However, 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.

2.2 WiMAX contention resolution procedure

When an SS wants to enter the contention resolution process, it sets its internal backoff window equal to the backoff start value defined in the UCD message. Then, the SS selects randomly a number within its backoff window. This random value indicates the number of contention transmission opportunities that the SS will wait before transmitting the bandwidth request. The SS considers the contention transmission lost if no data grant has been given within the required time interval.[2] If so, the SS increases its backoff window by a factor of two and selects a number within its new backoff window. If the backoff window reaches the backoff end value, then the SS drops the protocol data unit and starts from the beginning.

Fig. 1 presents a sample contention resolution procedure for four SSs. Suppose, there are four request transmission opportunities and all the SSs have the initial backoff window value of 1. Then, at the first stage, they will generate random values in the range of $[0..0]$. In this case, all the SSs will experience collisions. At the next stage, all SSs will increase their backoff window values, the resulting range will be $[0..1]$. Once again, there is still a high probability that all the SSs will experience collisions (we assume that all SSs generate uniformly distributed random numbers). Finally, as the backoff window value becomes four, there is a higher probability that all SSs will send successfully the bandwidth requests. However, there is also a probability that some SSs will generate the same backoff value and the contention resolution procedure will be repeated.

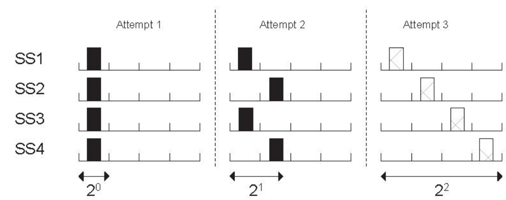


Figure 1. Sample contention resolution.

The example presented in Fig. 1 illustrates that if the *initial* backoff window value were four, then SSs would need less time to finish the contention resolution. One could think that large backoff start values are the key for an efficient contention resolution mechanism. However, it is not so. The reason is that if there are only a few SSs taking part in the contention resolution, then a small backoff start value suffices. The large backoff start value will result in a longer contention resolution which is not acceptable for the delay-critical VoIP application. Thus, the optimal backoff start value depends on a number of SSs.

It is also important to note that increasing a number of the contention slots, i.e., transmission opportunities, does not increase the probability of the successful transmission.

[1] If an SS does not have data, it just does not send any bandwidth request. From the viewpoint of the BS scheduler, it can be treated as the bandwidth request of zero bytes.

[2] While the 802.16d specification defines this interval in time units, the 802.16e specification defines it in a number of frames within which no data grants are allocated. This number is also announced in the UCD message.

Indeed, for the example presented above, it does not matter whether the BS allocates space for 4 transmission opportunities or 8. The contention resolution process will be the same. However, as we will present later, the number of the contention slots determines how fast the contention is resolved, i.e., it determines the medium access delay.

2.3 Adaptive backoff

The optimal backoff start/end values and a number of the contention transmission opportunities are considered [12]. Briefly, the worst-cast estimation of a probability for a successful transmission of the bandwidth request can be determined as follows:

$$p = \left(\frac{W-1}{W}\right)^{N_c - 1}, \tag{4}$$

where N_c stands for a number of the inactive connections and W is the backoff window value. Based on this, the backoff start value S can be calculated by using the following expression:

$$S = \begin{cases} \min\left\{\left\lceil -\log_2\left(1 - 2^{\frac{\log_2 p}{N_c - 1}}\right)\right\rceil, 15\right\} & N_c > 1, \quad \text{(5a)} \\ 0 & N_c \leq 1. \quad \text{(5b)} \end{cases}$$

The backoff start value is represented as a power-of-two number, where only the power value S is kept. Since the power value is encoded with the four bits in the UCD message, the maximum value is 15. The basic idea behind (5) is that a provider specifies the desired probability p, while the BS tracks a number of the inactive connections N_c and updates the backoff start value. For instance, when the VoIP application does not send data, the bandwidth request size equals zero. The BS treats this connection as the inactive one. Later, when the VoIP connection is allotted slots, a number of inactive connections declines. Since p has a direct relationship to the packet drops, its value is governed by the service level agreement signed between a provider and a customer. The bigger p is, a fewer packets will be dropped.

There is also the backoff end E value, but in the case of the VoIP services it is reasonable to set it to the same value as for the backoff start parameter. As considered in [1, 12], once the backoff start and backoff end have the same values, an SS has only one attempt to send the bandwidth request. If an attempt fails, the VoIP packet is dropped and the contention resolution is started from the beginning for the next packet. Of course, it may affect the quality of speech but, on the contrary, it will preserve the timing requirements that are very important for the real-time interactive applications. It is worth mentioning that such an approach is not suitable for the TCP applications performance of which depends significantly on packet drops.

Once the BS determines the backoff start value, it can calculate the number of the bandwidth request transmission opportunities. If the backoff start and end values are the same, then the following expression can be used [12]:

$$N_T = \left\lceil \frac{2^S}{\mathrm{FPS} \cdot T} \right\rceil, \tag{6}$$

where T is the time within which the BS must allocate an uplink data grant, and N_T is the number of the bandwidth request transmission opportunities to allocate in each WiMAX frame.

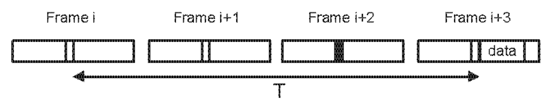


Figure 2. Medium access delay

Fig. 2 presents the graphical interpretation of T. There is one request transmission opportunity in each WiMAX frame, and an SS has to defer two transmission opportunities before sending the bandwidth request, i.e., the bandwidth request is sent in the $i{+}2$th frame. As shown in Fig. 2, T is the time to complete the contention resolution plus one frame duration. We should account for an additional frame because even though the BS receives successfully the bandwidth request in the $i{+}2$th frame, the uplink data grant will be allocated only in the next frame. This is due to the fact that the BS does the scheduling decision in the begining of a new frame. The value of T must be less or equal to the packet inter-arrival time that can be determined from the VoIP codec parameters.

3 Simulation

This section presents the simulation analysis of the adaptive backoff approach for the VoIP services. To run simulations, we have implemented the WiMAX MAC and PHY levels in the NS-2 simulator. The MAC implementation contains the main features of the WiMAX standard, such as downlink and uplink transmission, connections, MAC PDUs, packing, fragmentation, the contention and ranging periods, the MAC level management messages. The implemented PHY is OFDM. Since we focus on the contention resolution that is done on the MAC level, error generation at the PHY level is disabled.

The BS tracks a number of the inactive connections and calculates new backoff start/end values and the amount of the request contention slots based on expressions (5) and (6). New values are announced to the SSs in the UCD and UL-MAP messages respectively.

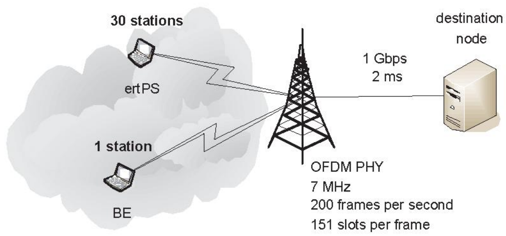


Figure 3. Network structure.

Fig. 3 presents the network structure we use in the simulation scenario. There is the BS controlling the WiMAX network, parameters of which are presented in Table 1, 30 WiMAX SSs, and one wired node. The SSs exchange data with the wired node through the BS, while the BS allocates resources to the SSs based on their QoS requirements and the bandwidth request sizes. Free slots, if any, are shared between the BE connections. The details of the scheduling algorithm are presented in [13].

For the sake of clarity, an SS hosts exactly one application and establishes one uplink and one downlink *transport* connection to the BS (there is also the basic management connection to exchange the management messages). All the SSs establish the ertPS transport connection to send VoIP data except the last SS that generates the background traffic over the BE transport connection. The VoIP application is represented by the G.711 audio codec [7] that has the highest mean opinion score [9]. Its packet size including the IP/UDP/RTP overhead is 300 bytes, the resulting sending rate is 80,000 bps. To emulate silence suppression, each VoIP application runs the ON/OFF model, parameters of which are 1.004 and 1.587 [8]. We simulate the *bidirectional* VoIP transmission, i.e., the VoIP applications at the SSs send data to the wired node, while the latter hosts the VoIP applications that send data to the wireless SSs by using the same G.711 codec and the ON/OFF model. However, it is important to note that the contention resolution occurs for the uplink transmission only because all the downlink packets reside at the BS. Thus, the purpose of the downlink VoIP packets is to create downlink traffic and to study a realistic scenario.

The background traffic is simulated by the FTP-like application that sends constantly data over the TCP protocol. The purpose of this BE connection is to test the efficiency of the resource utilization. Since the TCP protocol adjusts its window and the transmission rate to the available bandwidth, i.e., to the available slots allocated by the BS, it is a good way to test how the overall bandwidth is utilized in the WiMAX network. It does not matter how many background BE connections there are. If there were several ones, then the available slots would be shared fairly between them [13]. It is also worth mentioning that the BE connection takes part in the contention resolution process only in the beginning of a simulation run. As the FTP application sends continuously data, an SS can always piggy-back the bandwidth request to the user packets.

Table 1. WiMAX parameters.

Parameter	Value
PHY	OFDM
Bandwidth	7 MHz
Cyclic prefix length	1/32
Duplexing mode	TDD
Frames per second	200 (5 ms per frame)
Slots per frame	151
MCS (BE)	QPSK1/2 (24 B/slot)
MCS (VoIP)	16-QAM3/4 (72 B/slot)
Ranging opportunities	4
Ranging backoff start/end	5/15
Fragmentation/packing	ON
PDU size	as large as possible
CRC/ARQ	OFF

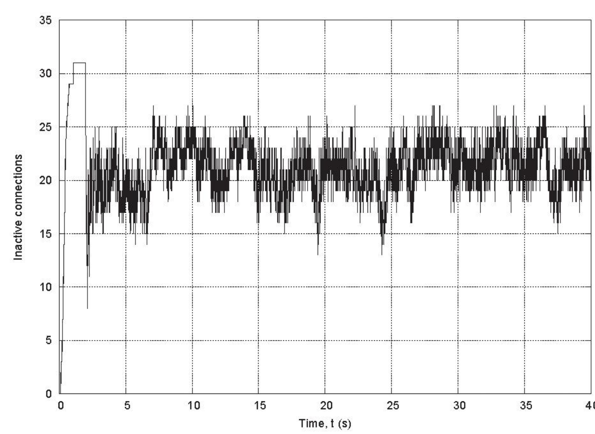


Figure 4. Inactive uplink VoIP connections.

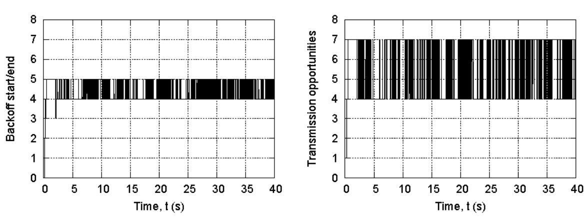
Due to the ON/OFF model running on the top of each VoIP application, there are active and silent phases. It results in the varying number of the inactive uplink connections as seen by the WiMAX BS. Fig. 4 presents the dynamics of the inactive uplink connections during the simulation run.

When a number of the inactive connections changes, the BS has two choices: either take no actions and rely upon the static configuration, or update the backoff parameters and the number of the request transmission opportunities. To compare these approaches, we run the same simulation scenario with the static and adaptive configuration. In addition, we run a simulation scenario when the BS performs polling, i.e., when there is no contention resolution.

Table 2. Simulation results.

Value	Polling	Static backoff		Adaptive backoff
		Case I	Case II	
Parameters	1 slot/ frame	E/S=4, TO=2	E/S=5, TO=7	p=0.25, T=30ms
BE data (MB)	6.984	15.288	13.406	14.031
VoIP data (MB)	4.528	4.308	4.427	4.350
Total data (MB)	11.512	19.596	17.833	18.381
VoIP loss (%)	0	0.89	0.39	0.69
Max.delay (ms)	–	42.18	28.2	27.67

As follows from the simulation results presented in Table 2, the best results are achieved when the parameters are changed adaptively. Firstly, when the polling is in effect, no packet drops occur. However, since the BS has to allocate at least one slot for each uplink VoIP connection to poll for data, the resulting resource utilization is not efficient. Depending on the static backoff configuration, either the BE connection can send more data (case I) or the VoIP packets experience low loss and delay (case II). In case I, there are only two request transmission opportunities. Thus, more free slots remain for the BE connection. However, the price for it is a larger number of dropped VoIP packets and the largest medium access delay. The static configuration in case II provides better results from the viewpoint of the VoIP packet loss. Indeed, since the static configuration relies upon the worst-case scenario, it always overprovides the number of the bandwidth request transmission opportunities which results in a faster contention resolution process and lower VoIP loss. However, there is an inefficient resource allocation because a fixed number of slots is always reserved for the transmission opportunities regardless of a number of connections that may take part in the contention. The adaptive approach is the tradeoff between the efficient resource utilization and the delay guarantees. The BS calculates the backoff start/end values and allocates the request transmission opportunities based on a number of connections thus achieving a better resource utilization. Fig. 5(a) and Fig. 5(b) present the calculated parameters.



(a) Backoff start/end (b) Transmission opportunities

Figure 5. Dynamically calculated parameters.

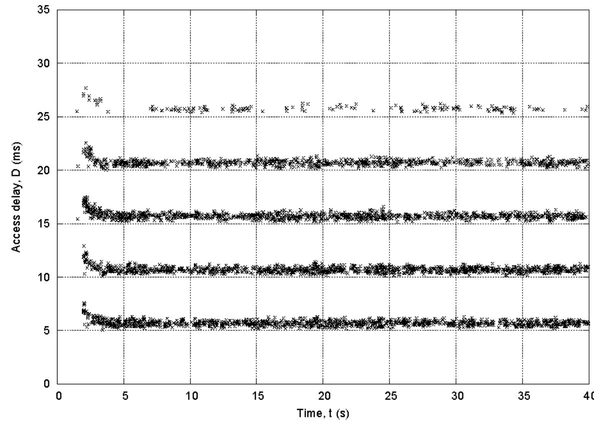


Figure 6. VoIP medium access delay (adaptive backoff case).

Furthermore, as follows from Table 2, the VoIP loss is insignificant.

Fig. 6 presents the medium access delay of all the ertPS connections. The x-axis plots the time when an SS enters the contention resolution process, while the y-axis plots the time difference between that event and the first data packet sent within the uplink data grant allocated by the BS. As follows from Fig. 6, the largest medium access delay is less than 30 ms. All the access delays fall into several clusters which is explained by the chosen WiMAX frame duration of 5 ms. Small fluctuations within a cluster are due the fact that an SS chooses randomly a transmission opportunity and that the VoIP uplink data bursts are allocated at different places within the uplink part of the WiMAX frame. Table 3 also presents the percentile medium access delay values. Thus, the adaptive backoff is capable of ensuring the timing requirements.

Table 3. Percentile medium access delay

Value	95th	90th	85th	80th
Access delay (ms)	21.02	20.79	20.63	20.36

Based on the simulation results, it is possible to arrive at the conclusion that the adaptive backoff provides a good tradeoff between an efficient resource utilization and ensuring the delay QoS requirements of the VoIP connections.

4 Other solutions

An appealing feature of the adaptive approach is the fact that a provider can apply it only to a group of the VoIP connections. Since it is the BS who controls and decides whether an SS may take part in the contention, the BS can change the default behaviour of the ertPS class by setting an appropriate bit in the *request/transmission policy* parameter in the DSA-REQ and DSA-RSP messages. For instance, a provider can define two groups of the VoIP applications: the premium VoIP customers and ordinary VoIP customers. The contention resolution will be disabled for the premium VoIP customers because the BS will poll them, while the ordinary VoIP customers will be allowed to use the contention resolution (the default ertPS behaviour).

Another interesting solution to boost the contention mechanism is the multicast polling. It enables the BS to allocate the contention slots for a group of SSs, rather than for all the SSs. In this case, one contention domain splits into several ones resulting in better delay characteristics of some service classes. The most logical way to use the multicast polling is to assign a separate multicast polling group to a service class or to a group of connections belonging to some service class. Referring back to the example above, the BS may allocate separate multicast polling groups for the premium and ordinary VoIP connnections. The limitation of the multicast polling is the fact that the backoff start/end parameters are assigned globally to the common contention period and the muticast polling groups. Besides, the multicast polling is not mandated by the WiMAX Forum recommendations [3].

5 Conclusions

In this paper, we have analysed the adaptive calculations of the backoff start/end values and the number of the transmission opportunities in conjunction with the VoIP services. The simulation results have presented that the adaptive parameter tuning provides better results when compared to the static configuration. Due to the better resource allocation the nodes are capable of transmitting more data and all the VoIP timing requirements are ensured. Besides, the adaptive approach eliminates a need to find the optimal static parameters, which is crucial when a number of connections changes drastically.

Though the simulation results were presented for the OFDM PHY working in the TDD mode, the exactly the same approach can be applied to the FDD mode and the SC PHY. However, it is worth mentioning that the OFDMa PHY contention resolution works a little bit differently. Thus, our future research work will aim to consider the OFDMa PHY. Though it has the same contention resolution mechanism, an SS sends a special CDMA code instead of the bandwidth request. Some performance results for the OFDMa PHY are presented in [15].

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