New Scheduler with Call Admission Control (CAC) for IEEE 802.16 Fixed with Delay Bound Guarantee

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***Abstract*—The IEEE 802.16 Working Group is developing a standard for broadband wireless access in Metropolitan Area Networks (MAN) known as WiMAX. One of the features of the MAC layer, in this standard, is that it is designed to provide differentiated servicing for traffic with multimedia requirements. Based on these assumptions, and considering that the standard does not specify a scheduling algorithm, a new scheduler with call admission control was proposed based on Latency-Rate (LR) server theory and with system characteristics as specified by the system standard using the WirelessMAN-OFDM (Orthogonal Frequency Division Multiplexing) air interface. The proposed scheduling algorithm calculates the time frame (TF) in order to maximize the number of stations allocated in the system while managing the delay required for each user. Properties of this proposal have been investigated theoretically and through simulations. A set of simulations is presented with both Constant Bit Rate (CBR) and Variable Bit Rate (VBR) traffic, and performance comparisons are made between cases with different delays and different TFs. The results show that an upper bound on the delay can be achieved for a large range of network loads, with bandwidth optimization.**

Keywords-IEEE 802.16; scheduling algorithm; delay bound; optimization; Call Admission Control (CAC).

I. INTRODUCTION

The deployment of high-speed Internet access is often cited as a challenge for the second decade of this century. Known as broadband Internet, it is effective in reducing physical barriers to the transmission of knowledge, as well as transaction costs, and is fundamental in fostering competitiveness. However, wired access to broadband Internet has a very high cost and is sometimes unfeasible, since the investment needed to deploy cabling throughout a region often outweigh the service provider's financial gains. One of the possible solutions in reducing the costs of deploying broadband access in areas where such infrastructure is not present is to use wireless technologies, which require no cabling and reduce both implementation time and cost [1].

This was one of the motivations behind the development by the IEEE (Institute of Electrical and Electronics Engineers) of a new standard for wireless access, called 802.16 [2], also known as Worldwide Interoperability for Microwave Access (WiMAX). It is an emerging technology for next generation wireless networks which supports a large number of users, both mobile and nomadic (fixed), distributed across a wide geographic area.

Motivated by the growing need for ubiquitous high-speed access, wireless technology is an option to provide a cost-effective solution that may be deployed quickly and easily, providing high bandwidth connectivity in the last mile. However, despite the many advantages of wireless access networks, such as low deployment and maintenance costs, ease of configuration and device mobility, there are challenges that must be overcome in order to further advance the widespread use of this type of network.

To achieve this purpose, the IEEE 802.16 standard introduces of a set of mechanisms, such as service classes and several coding and modulation schemes that adapt themselves according to channel conditions. However, the standard leaves open certain issues related to network resource management and scheduling algorithms.

This paper presents a new scheduler with admission control of connections to a WiMAX Base Station (BS). We developed an analytical model based on Latency-Rate (LR) server theory [3], from which an ideal frame size, called Time Frame (TF), was estimated, with guaranteed delays for each user. At the same time, the number of stations allocated in the system is maximized. In this procedure, framing overhead generated by the MAC (Medium Access Control) and PHY (Physical) layers was considered when calculating the duration of each time slot. After developing this model, a set of simulations is presented for constant bit rate (CBR) and variable bit rate (VBR) streams, with performance comparisons between situations with different delays and different TFs. The results show that an upper limit on the delay may be achieved for a wide range of network loads, thus optimizing bandwidth.

The remainder of this paper is organized as follows. In Section II, a brief description of the IEEE 802.16 standard is presented. Our analytical model of packet scheduling is proposed and explained in Section III. Evaluation of the

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capacity of the new scheduler with Call Admission Control (CAC) is shown in Section IV. Conclusions are in Section V.

II. The IEEE 802.16 standard

A. Overview of Fixed WiMAX

The basic topology of a IEEE 802.16 network includes two entities that participate in the wireless link: Base Stations (BS) and Subscriber Stations (SS), as shown in Figure 1.

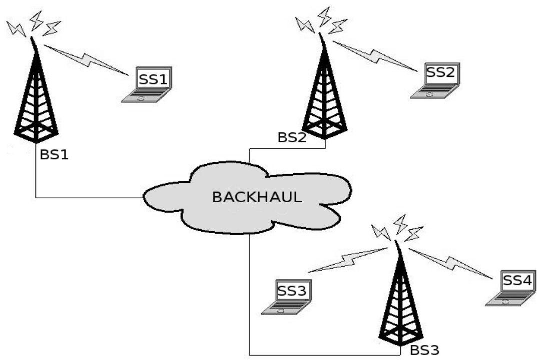

Figure 1. IEEE 802.16 Network Architecture

The BS is the central node, responsible for coordinating communication and providing connectivity to SSs. BSs are kept in towers distributed so as to optimize network coverage area, and are connected to each other by a backhaul network, which allows SSs to access external networks or exchange information between themselves.

Networks based on the IEEE 802.16 standard can be structured in two schemes. In PMP (Point-to-MultiPoint) networks, all communication between SSs and other SSs or external networks takes place through a central BS node. Thus, traffic flows only between SSs and the BS (see Figure 1). In Mesh mode, SSs communicate with each other without the need for intermediary nodes; that is, traffic can be routed directly through SSs. Thus, all stations are peers which can act as routers and forward packets to neighboring nodes. This article only considers the PMP topology.

The communication between a BS and SSs occurs in two different channels: uplink (UL) channel, which is directed from SSs to the BS, and downlink (DL) channel, which is directed from the BS to SSs. DL data is transmitted by broadcasting, while in UL access to the medium is multiplexed. UL and DL transmissions can be operated in different frequencies using Frequency Division Duplexing (FDD) mode or at different times using Time Division Duplexing (TDD) mode.

In TDD, the channel is segmented in fixed-size time slots. Each frame is divided into two subframes: a DL subframe and an UL subframe. The duration of each subframe is dynamically controlled by the BS; that is, although a frame has a fixed size, the fraction of it assigned to DL and UL is variable, which means that the bandwidth allocated for each of them is adaptive. Each subframe consists of a number of time slots, and thus both the SSs and the BS must be synchronized and transmit the data at predetermined intervals. The division of TDD frames between DL and UL is a system feature controlled by the MAC layer. Figure 2 shows the structure of a TDD frame. In this paper, the system was operated in TDD mode with the OFDM (Orthogonal Frequency Division Multiplexing) air interface, as determined by the standard.

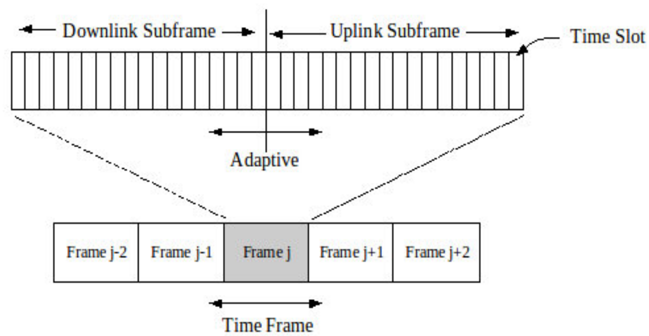

Figure 2. IEEE 802.16 Frame Structure

Figure 3 shows an example OFDM frame structure in TDD mode. As seen earlier, each frame has a DL subframe followed by a UL subframe. In this structure, the system supports frame-based transmission, in which variable frame lengths can be adopted. These subframes consists of a fixed number of OFDM symbols. Details of the OFDM symbol structure may be found in [1].

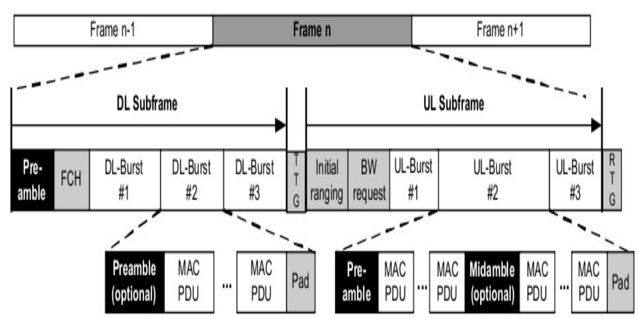

Figure 3. OFDM Frame Structure with TDD

The DL subframe starts with a long preamble (two OFDM symbols) through it the SSs can synchronize with the network and check the duration of the current frame. Instantly after DL long preamble, the BS transmits the Frame Control Header (FCH), which consists of an OFDM symbol and is used by SSs to decode the MAC control messages transmitted by BS.

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The UL subframe consists in contention intervals for initial ranging and bandwidth request purposes and one or several UL transmission bursts, each from a different SSs. The initial ranging slots allow an SS to enter the system, by adjusting its power level and frequency offsets and by correcting its timing offset. Bandwidth request slots are used by SSs to transmit bandwidth request headers.

Two gaps separate the DL and UL subframes: the Transmit/Receive Transition Gap (TTG) and Receive/Transmit Transition Gap (RTG). These gaps allow the BS to switch from the transmit to receive mode, and vice versa.

B. Related Research

Since the standard only provides signaling mechanisms and no specific scheduling and admission control algorithms, some scheduling algorithms have been proposed to provide QoS (Quality of Service) for WiMAX. However, many of these solutions only address the implementation or addition of a new QoS architecture to the IEEE 802.16 standard. A scheduling algorithm decides the next packet to be served on the waiting list and is one of the mechanisms responsible for distributing bandwidth among several streams.

In [5], a packet scheduler for IEEE 802.16 uplink channels based on an hierarchical queue structure was proposed. A simulation model was developed to evaluate the performance of the proposed scheduler. However, despite presenting simulation results, the authors overlooked the fact that the complexity of implementing this solution is not hierarchical, and did not define clearly how requests for bandwidth are made. In [7], authors proposed a QoS architecture to be built into the IEEE 802.16 MAC sublayer, which significantly impacts system performance, but did not present an algorithm that makes efficient use of bandwidth. In [8], authors presented a simulation study of the IEEE 802.16 MAC protocol operating with an OFDM (Orthogonal Frequency Division Multiplexing) air interface and full-duplex stations. They evaluated system performance under different traffic scenarios, varying the values of a set of relevant system parameters. Regarding data traffic, it was observed that the overhead due to the physical transmission of preambles increases with the number of stations. In [9], a polling-based MAC protocol is presented along with an analytical model to evaluate its performance. They developed closed-form analytical expressions for cases in which stations are polled at the beginning or at the end of uplink subframes. It is not possible to know how the model may be developed for delay guarantees. Finally, in [10], the author presents a well-established architecture for QoS in the IEEE 802.16 MAC layer. The subject of this work is the component responsible for allocating uplink bandwidth to each SS, although the decision is taken based on the following aspects: bandwidth required by each SS for uplink data transmission, periodic bandwidth needs for UGS flows in SSs and bandwidth required for making requests for additional bandwidth.

Considering the limitations exposed above, these works form the basis of a generic architecture, which can be extended and specialized. However, in these studies, the focus is in achieving QoS guarantees, with no concerns for maximizing the number of allocated users in the network. This paper presents a scheduler with admission control of connections to the WiMAX BS. We developed an analytical model based on Latency-Rate (LR) server theory [3], from which an ideal frame size called Time Frame (TF) was estimated, with guaranteed delays for each user and maximization of the number of allocated stations in the system. A set of simulations is presented with CBR and VBR streams and performance comparisons are made for different delays and different TFs. The results show that an upper bound on the delay may be achieved for a large range of network loads with bandwidth optimization.

III. ANALYSIS OF THE ANALYTICAL MODEL

A. System Description

Figure 4 illustrates a wireless network operating the newly proposed scheduler with call admission control, which is based on a modified LR scheduler [3] and uses the token bucket algorithm. The basic approach consists on the token bucket limiting input traffic and the LR scheduler providing rate allocation for each user. Then, if the rate allocated by the LR scheduler is larger than the token bucket rate, the maximum delay may be calculated.

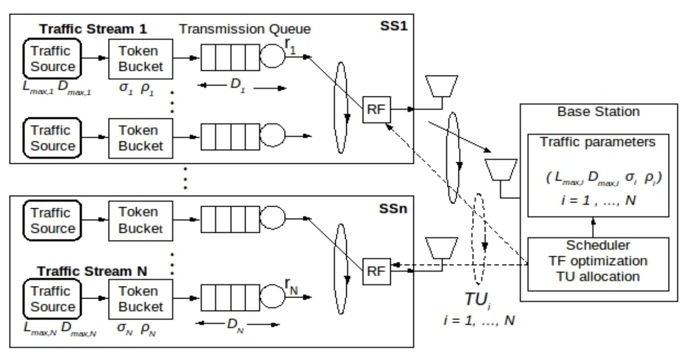

Figure 4. Wireless Network with New Scheduler

The behavior of an LR scheduler is determined by two parameters for each session i: latency θ_i and allocated rate r_i. The latency θ_i of the scheduler may be seen as the worst-case delay and depends on network resource allocation parameters. In the new scheduler with call admission control, the latency θ_i is a TF period, which is the time needed to transmit a maximum-size packet and separation gaps (TTG and RTG) of DL and UL subframes. In the new scheduler, considering the delay for transmitting the first packet, the latency θ_i of is given by

$$\theta_i = T_{TTG} + T_{RTG} + T_{DL} + T_{UL} + \frac{L_{max,i}}{R} \qquad (1)$$

where T_{TTG} and T_{RTG} are DL and UL subframes gaps durations, T_{DL} and T_{UL} are the DL and UL subframes duration, $L_{max,i}$ is the maximum packet size and R is the outgoing link capacity.

Now, we show how the allocated rate r_i for each session i may be determined, and how to optimize TF in order to increase the number of connections accommodated with Call Admission Control (CAC).

B. CAC Description

An LR scheduler can provide a bounded delay if the input traffic is shaped by a token bucket. A token bucket [1] is a non-negative counter which accumulates tokens at a constant rate ρ_i until the counter reaches its capacity σ_i. Packets from session i can be released into the queue only after removing the required number of tokens from the token bucket. In an LR scheduler, if the token bucket is empty, arriving packets are dropped; however, our model ensures that there will always be tokens in the bucket and that no packets are dropped, as described in Section IV. If the token bucket is full, a maximum burst of σ_i packets can be sent to the queue. When the flow is idle or running at a lower rate as the token size reaches the upper bound σ_i, accumulation of the tokens will be suspended until the arrival of the next packet. We assume that the session starts out with a full bucket of tokens. In our model, we consider IEEE 802.16 standard overhead for each packet. Then, as we will show below, the token bucket size will decreased by both packet size and overhead.

The application using session i declares the maximum packet size $L_{max,i}$ and required maximum allowable delay $D_{max,i}$, which are used by the WiMAX scheduler to calculate the service rate for each session so as to guarantee required delay and optimize the number of stations in the network. Incoming traffic passes through a token bucket inside the user terminal during an interval, as shown in Figure 5.

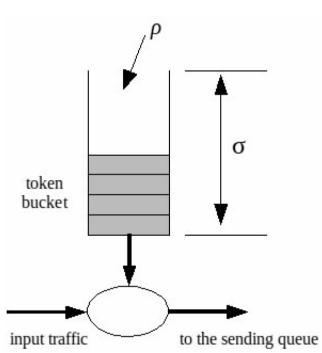

Figure 5. Input Traffic with Token Bucket

This passage of data traffic by the token bucket is bounded by

$$A_i(t) \leq \sigma_i + \rho_i t \qquad (2)$$

where σ_i is the bucket size and ρ_i is the bucket rate.

Then, the packet is queued in the station until it is transmitted via the wireless. Queue delay is measured as the time interval between the receipt of the last bit of a packet and its transmission. In the new scheduler with call admission control, queuing delay depends on token bucket parameters, network latency and allocated rate. In [3], it is shown that if input traffic is shaped by a token bucket and the scheduler allocates a service rate r_i , then an LR scheduler can provide a bounded maximum delay Di:

$$D_i \leq \frac{\sigma_i}{r_i} + \theta_i - \frac{L_{max,i}}{r_i} \qquad (3)$$

where r_i is the service rate, σ_i is the token bucket size, θ_i is the scheduler latency, $L_{max,i}$ is the maximum size of a package. $\frac{\sigma_i}{r_i} + \theta_i - \frac{L_{max,i}}{r_i}$ is the bound on the delay D_{bound}.

Equation (3) is an improved bound delay for LR schedulers. Thus, the token bucket rate plus the overhead transmission rate must be smaller than the service rate to provide a bound on the delay. The upper bound delay D_{bound} should be smaller or equal to the maximum allowable delay:

$$\frac{\sigma_i}{r_i} + \theta_i - \frac{L_{max,i}}{r_i} \leq D_{max,i} \qquad (4)$$

Therefore, three different delays are defined. The first is the maximum delay D_i, the second is the upper bound on the delay D_{bound} and the third is the required maximum allowable delay $D_{max,i}$. The relation between them is $D_i \leq D_{bound} \leq D_{max,i}$.

So, the delay constraint condition of the new scheduler is

$$\frac{(\sigma'_i - L'_{max,i})TF}{r'_i TF - \Delta R + L'_{max,i}} + TF + \frac{L'_{max,i}}{R} + T_{TTG} + T_{RTG} \leq D_{max,i} \qquad (5)$$

where σ'_i is the token bucket size with overhead, $L'_{max,i}$ is the maximum size of a packet with overhead (preamble+pad), TF is the time frame, r'_i is the rate allocated by the server with overhead, R is the outgoing link capacity, T_{TTG} is the gap between downlink and uplink subframes, T_{RTG} is the gap to between uplink and downlink subframes, $D_{max,i}$ is the maximum allowable delay and Δ is the sum of initial ranging and BW request, which is the uplink subframe overhead. Physical rate, maximum packet size and token bucket size are parameters declared by the application. However, TF and total allocated service rate must satisfy Equation (5).

Figure 6 shows a frame structure with TDD allocation formulas as described by Equation (5). Physical rate, maximum packet size and token bucket size are parameters

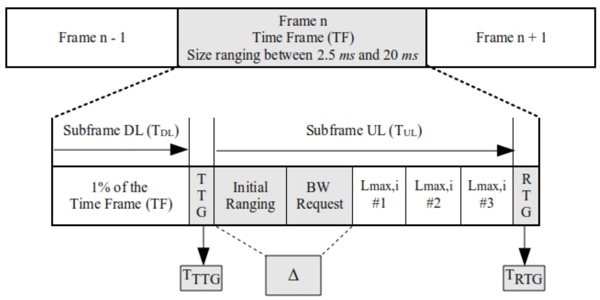

Figure 6. Frame structure with TDD allocation formulas of Equation (5)

declared by the application. However, TF and total allocated service rate must satisfy Equation (5).

Equation (6) is the second constraint condition to TF and service rate. Token bucket rate plus the rate to transmit overhead and a maximum-size packet must be smaller than the service rate to place a bound on delay. Thus, the second constraint condition is

$$\rho_i + \frac{\Delta R + L'_{max,i}}{TF} \leq r'_i \qquad (6)$$

where ρ_i is the bucket rate, Δ is the uplink subframe overhead, R is the outgoing link capacity, $L'_{max,i}$ is the maximum packet size with overhead, TF is the time frame and r'_i is the rate allocated by the service with overhead.

Previous schedulers do not provide any mechanism to estimate the TF needed to place a bound on delay or to maximize the number of stations, because each application requires a TF without the use of criteria to calculate the time assigned to each user. TF estimation is important because a small TF reduces maximum delay, but increases overhead at the same time. On the other hand, a large TF decreases overhead, but increases delay. Therefore, we must calculate the optimal TF to allocate the maximum number of users under these both constraint. The maximum number of users is achieved when the service rate for each user is the minimum needed to guarantee the bound on the delay D_{bound}. Different optimization techniques may be used to solve this problem. In this study, we have used a step-by-step approach, which does not change the scheduler's essential operation. We start with a small TF, for example, $2.5ms$, calculate r'_i and repeat this process every $0.5ms$ until we find the minimum r'_i that satisfies both equations.

IV. Performance Analysis

To analyze the IEEE 802.16 MAC protocol behavior with respect to the new scheduler with call admission control, this section presents numerical results obtained with the analytical model proposed in the previous section. Then, with a simulation tool, the proposed analytical model is validated by showing that the bound on the maximum delay is guaranteed. In this section, two types of delays are treated: required delay, in which the user requires the maximum delay, and the guaranteed maximum delay, which is calculated with the analytical model.

A. Calculation of Optimal Time Frame

All PHY and MAC layer parameters used in simulation are summarized in Table I.

Table I
PHY and MAC parameters

PARAMETER	VALUE
Bandwidth	20MHz
OFDM Symbol Duration	13,89 μs
Delay	5 / 10 / 15 and 20 ms
Δ (Initial Ranging and BW Request) $\rightarrow$ 9 OFDM Symbols	125,10 μs
TTG + RTG $\rightarrow$ 1 OFDM Symbol	13,89 μs
UL Subframe (preamble + pad) $\rightarrow$ 10% OFDM Symbol	1,39 μs
Physical Rate	70 Mbps
DL Subframe	1% TF

Performance of the new scheduler with call admission control is evaluated as the delay requested by the user and assigned stations. Station allocation results, in the system with an optimal TF, limited by the delay requested by the user, are described in sequence. The first step is define token bucket parameters, which are estimated in accordance with the characteristics of incoming traffic and are listed on Table II.

Table II
Token bucket parameters

	Audio	VBR video	MPEG4 video
Token Size (bits)	3000	18000	10000
Token Rate (kb/s)	64	500	4100

Thus, the optimal TF value is estimated according to the PHY and MAC layer's parameters (see Table I), token bucket parameters (see Table II), required maximum allowable delay, physical rate and maximum package size.

The graph in Figure 7 shows the optimal TF value, for four delay values required by users (5, 10, 15 and 20 ms).

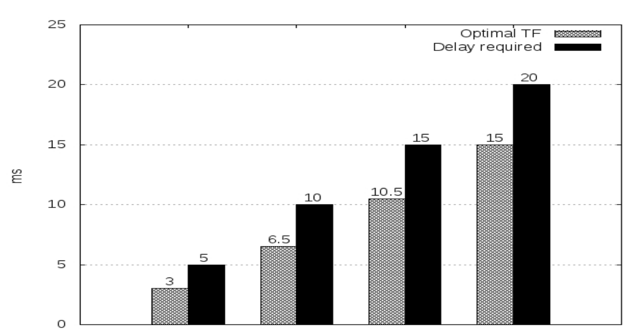

Figure 7. Optimal TF

Next, we show the number of SSs assigned to each traffic type. As an example, Figure 8 show that when the user-requested delay is of 20 ms, an optimal TF of 15 ms is calculated and 50 users can be allocated for audio traffic, or 30 users for VBR video traffic, or 13 users for the MPEG4 video traffic.

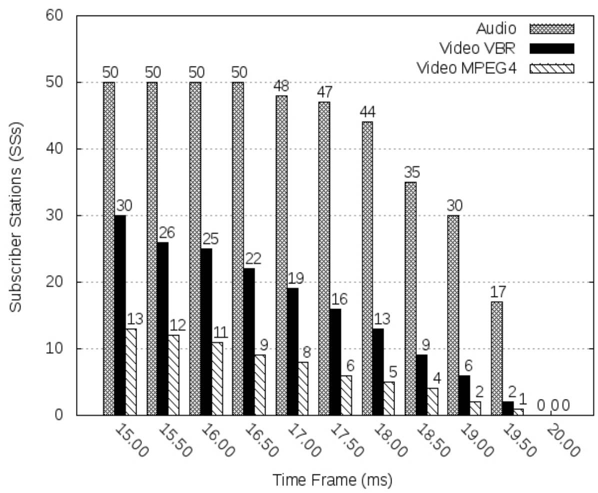

Figure 8. Number of subscriber stations for 20 ms of delay

Two important observations from Figure 8 should be highlighted:

1) With a requested delay of 20 ms, we cannot choose a TF of less than 15 ms, since the restrictions placed by Equation (5) (which regards delay) and Equation (6) (which regards the token bucket) are not respected and thus no bandwidth allocation guarantees exist.
2) We also cannot choose a TF greater than 15 ms, even though it complies with Equations (5) and (6) with respect to guaranteed bandwidth, because there will be a decrease in the number of users allocated to each traffic flow due to increase of the delay.

The same philosophy holds true for other delay values of 5, 10 and 15 ms.

B. Guaranteed Maximum Delay

In this article, only UL traffic is considered. To test the new scheduler's performance, we have carried out simulations of an IEEE 802.16 network consisting of a BS that communicates with eighteen SSs, with one traffic flow type by SS and the destination of all flows being the BS. In this topology, six SSs transmit on-off CBR audio traffic (64 kb/s), six transmit CBR MPEG4 video traffic (3.2 Mb/s) and six transmit VBR video traffic. Table III summarizes the different types of traffic.

On Figure 9, with an optimal TF of 3 ms and an user-requested delay of 5 ms, the average guaranteed maximum delay for audio traffic is 1.50 ms. For VBR video

Table III
Description of the different traffics

Node	Application	Arrival Period (ms)	Packet size (max) (bytes)	Sending rate (kb/s) (mean)
$1 \rightarrow 6$	Audio	4.7	160	64
$7 \rightarrow 12$	VBR video	26	1024	≈ 200
$13 \rightarrow 18$	MPEG4 video	2	800	3200

traffic, whose packet rate is variable, the average maximum delay is 1.97 ms. For MPEG4 video traffic, the average maximum delay is 2.00 ms.

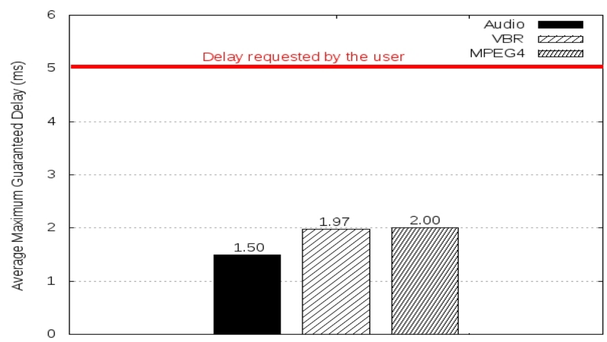

Figure 9. Maximum Guaranteed Delay

C. Comparison with other Schedulers

The new scheduler with call admission control, here called *New Scheduler*, was compared to those of [9], here called $Scheduler\_1$, and [5], here called $Scheduler\_2$. The comparison was accomplished through the ability to allocate users in a particular time frame (TF). Table IV shows the parameters used for comparisons.

Table IV
Parameters used for comparisons

PARAMETER	$Scheduler\_1$	$Scheduler\_2$
Bandwidth	20 MHz	20 MHz
OFDM symbol duration	13.89 μs	13.89 μs
Time Frame (TF)	5 ms	10 ms
Delay Requested by the user	0.12 ms	20 ms
Maximum Data Rate	70 $Mbps$	70 $Mbps$
Traffic type	Audio	Audio

In the graph of Figure 10, we compare the *New Scheduler* with the $Scheduler\_1$. A maximum delay of 0.12 ms was requested by the user, and the duration of each frame (TF) was set at 5 ms. Other parameters are listed in Table IV. In comparison, the *New Scheduler* allocates 28 users in each frame, while the $Scheduler\_1$, allocates 20 users. Thus, the *New Scheduler* presents a gain in performance of 40% when compared with the $Scheduler\_1$.

In the graph of Figure 11, we compare the *New Scheduler* with the $Scheduler\_2$. A maximum delay of 20 ms was requested by the user, and the duration of each frame (TF) was set at 10 ms. Other parameters are listed in Table IV.

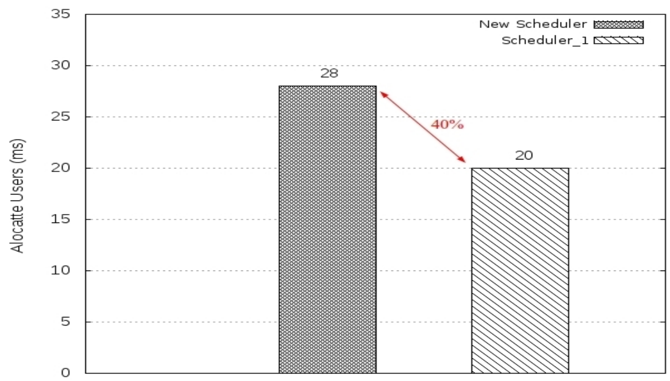

Figure 10. Comparison of allocation of users with $Scheduler\_1$

The comparison was extended by also considering frame duration values of 7.00 ms, 8.00 ms and 9.00 ms to demonstrate the efficiency of the *New Scheduler*. For a TF of 10 ms, the *New Scheduler* allocates 41 users in each frame, while the $Scheduler\_2$ allocates only 33 users. This represents 24.24% better performance for the *New Scheduler*. Similarly, the *New Scheduler* also allocates more users per frame in comparison with the $Scheduler\_2$ for all other frame duration values.

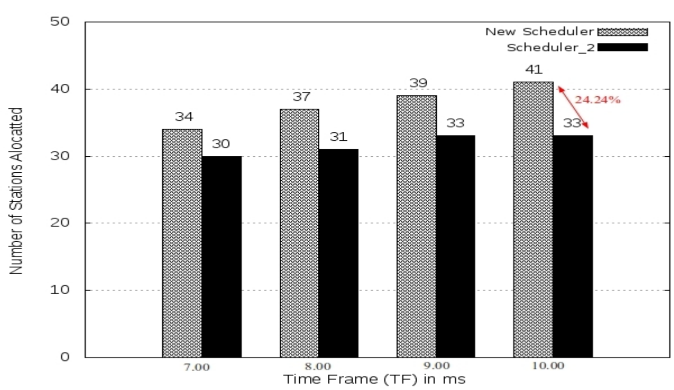

Figure 11. Comparison of allocation of users with $Scheduler\_2$

V. Conclusion

This work has presented the design and evaluation of a new scheduler with call admission control for IEEE 802.16 fixed networks, that guarantees different maximum delays for traffic types with different QoS requisites and optimizes bandwidth usage. Firstly, we developed an analytical model to calculate an optimal TF, which allows an optimal number of SSs to be allocated and guarantees the maximum delay required by the user. Then, a simulator was developed to analyze the behavior of the proposed system.

To validate the model, we have presented the main results obtained from the analysis of different scenarios. Simulations were performed to evaluate the performance of this model, demonstrating that an optimal TF was obtained along with a guaranteed maximum delay, according to the delay requested by the user. Thus, the results have shown that the new scheduler with call admission control successfully limits the maximum delay and maximizes the number of SSs in a simulated environment.

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