ELECTRONICS AND ELECTRICAL ENGINEERING

ISSN 1392 – 1215 2010. No. 2(98)

ELEKTRONIKA IR ELEKTROTECHNIKA

T 120 SYSTEM ENGINEERING, COMPUTER TECHNOLOGY

SISTEMŲ INŽINERIJA, KOMPIUTERINĖS TECHNOLOGIJOS

QoS Management for Broadband IEEE 802.16 based Networks in FDD Mode

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Introduction

Broadband Wireless Access (BWA) has become the best way to meet residential and small business demand for high speed Internet access and multimedia services. As an emerging technology for broadband access, it provides the following advantages over its wired competitors: 1) rapid deployment and ease to implement; a BWA network can be installed rapidly without extensive underground cable infrastructure as is the case of Cable or DSL networks, 2) high scalability; carriers can expand the BWA network as subscriber demand for bandwidth grows by adding channels, or cells, 3) lower maintenance and upgrade costs and 4) higher data rates. However, the wide-scale adoption of BWA systems will be determined by its ability to overcome cost and performance barriers. If BWA can meet these challenges it could easily be the next revolution in wireless networks systems such as WLANs.

As far as performance is concerned, BWA systems will provide services for diverse types of traffic, while allowing for different Quality of Service (QoS) requirements. Although the IEEE 802.16 MAC protocol includes QoS guarantees, it does not provide a complete solution and does not detail how to schedule traffic to fulfill QoS requirements, specifically [1, Section 6.1]. In recent years, several scheduling algorithms for BWA networks have been published [2-5]. However most of these studies focus on the Time Division Duplex (TDD) mode. Despite this, the Frequency Division Duplex (FDD) mode is expected to be a widely used solution for a number of reasons. First, even though it is generally accepted that TDD systems offer cost advantages over their FDD counterparts, most licensed bands intended for data applications operate with FDD systems in mind. Second, MAC-level software tends to have a more complicated scheduler than an FDD system since it must deal with synchronizing many subscriber time slots in both TX (Transmission) and RX (Reception) mode.

Currently, there is little research on scheduling algorithms specifically targeted for IEEE 802.16 systems that provides a flexible assignment of UL channel resources according to network operator needs. In a real scenario, IEEE 802.16 network operators would like to distribute the channel resources according to the QoS agreement acquired by subscribers. However, many scheduling algorithms found in the literature primarily allocate all UGS bandwidth resources in the current UL-frame. Then, in the remaining UL-frame (if any), rtPS bandwidth requirements are allocated, and so on. This results in starvation of lower priority classes such as nrtPS and BE.

In this paper we explore a combination of various scheduling schemes to closely match QoS service classes defined in the IEEE 802.16 MAC protocol, considering a FDD radio. To the best of our knowledge, there is little research regarding scheduling solutions specifically targeted for IEEE 802.16 FDD systems. We present a scheduling algorithm that supports diverse traffic classes, such as Constant Bit Rate (CBR) and Variable Bit Rate (VBR) with different QoS requirements. The scheduling algorithm combines Prioritization, Early Deadline First (EDF) [6], Round Robin (RR) [7], and Weighted Fair Queueing (WFQ) [8] strategies to closely match real time and non-real time traffic services, such as Voice over IP (VoIP), multimedia services and high speed Internet access. We call this scheduling mechanism the EDF-BWA Scheduling Algorithm (EBSA).

The rest of the paper is structured as follows. In Section 2, we provide an overview of the IEEE 802.16 MAC protocol and provide detailed information of the four different types of QoS agreements supported by the standard. In Section 3, we describe the proposed scheduling algorithm. In Section 4, we present a performance analysis of EBSA for different traffic sources. Finally, in Section 5, we present our conclusions and future work.

IEEE 802.16 standard overview

The IEEE 802.16 group produced a standard that was approved in December 2001 [9]. This standard, Wireless MAN-SC, specifies a physical layer that uses single-carrier modulation techniques and a media access control (MAC) layer with a burst time division multiplexing (TDM) structure that supports both frequency division duplexing (FDD) and time division duplexing (TDD).

After completing this standard, the group started work on extending and modifying the standard to work in both licensed and license-exempt frequencies in the 2-11 GHz range, which would enable non line of sight (NLOS) deployments. Further revisions to 802.16 were made and completed in 2004. IEEE 802.16-2004 [1] has replaced 802.16, 802.16a, and 802.16c with a single standard, which has also been adopted as the basis for HIPERMAN (high-performance metropolitan area network) by ETSI (European Telecommunications Standards Institute).

In 2003, the 802.16 group began work on enhancements of the specifications to allow for vehicular applications. That revision, 802.16e, was completed in December 2005 and was published formally as IEEE 802.16e-2005 [10]. It specifies scalable OFDMA (orthogonal frequency division multiple access) for the physical layer and makes further modifications to the MAC layer to accommodate high-speed mobility. In general, the three versions use a generalized MAC structure described below.

A. IEEE MAC Protocol

Requests for resource allocations and data transmissions from the Subscriber Station (SS) to the Base Station (BS) are carried in an uplink (UL) frame. Transmissions from the BS to SSs are carried by a downlink (DL) frame. A typical signaling frame for TDD includes a UL-frame (see Fig. 1a) and a DL-frame (see Fig. 1b) using a single channel frequency as illustrated in Fig. 1c. In FDD, these frames are transmitted at the same time using different channel frequencies as illustrated in Fig. 1d.

The IEEE 802.16 MAC protocol regulates uplink (UL) channel access using Time Division Multiple Access (TDMA). Upon entering the BWA network, each

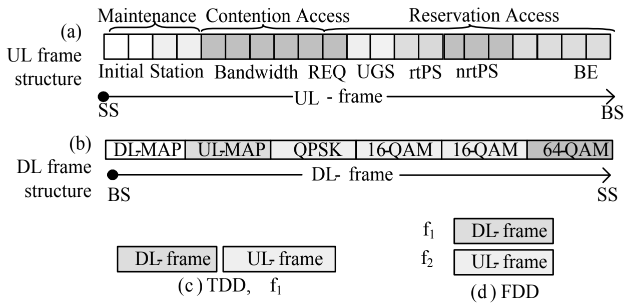

Fig. 1. Frame structure for TDD and FDD access

Subscriber Station (SS) has to go throughout the initialization process setup, described as follows:

Subscriber stations need to synchronize with a downlink channel (DL-ch) and an uplink channel (UL-ch). When a SS has tuned to a DL-ch, it obtains the frame structure of the UL-ch, called a UL-MAP frame. Then the ranging procedure is performed, where the round-trip delay and power calibration are determined for each SS, so that SS transmissions are aligned to the correct mini-slot boundary. Then the SS negotiates basic capabilities to the BS, which is the phase where the SS and the BS exchange their supported parameters. Next, the SS should use the Privacy Key Management (PKM) protocol to obtain authentication from the BS. Then the SS performs the registration process by establishing a security association that allows the SS entry into the network. The next step is to establish IP connectivity, the BS uses the DHCP mechanism in order to obtain an IP address for the SS and any other parameters needed to establish IP connectivity. Then, the SS establishes the time of the day, which is required for time-stamping logged events and key management. In the next step, the SS transfers control parameters via TFTP, such as boot information, QoS parameters, fragmentation, and packing, among others. The last step is to establish connections for pre-provisioned service flows belonging to the SS.

After the initialization process is completed, a SS can create one or more connections over which its data is transmitted to and from the BS. SSs contend for transmission opportunities using the contention access period (or contention block) of the current UL-frame. The BS collects these requests and determines the number of slots (grant size) that each SS will be allowed to transmit in the next UL-frame, using a UL_MAP sub-frame, as shown in Figure 1b. The UL-MAP frame contains Information Elements (IE), which describe the maintenance, contention or reservation access of the UL-frame. The UL-MAP is broadcasted in the DL channel by the BS in each DL-frame. After receiving the UL-MAP, a SS can transmit data in the predefined reserved slots indicated in the IE. These reserved slots are transmission opportunities assigned by a scheduling algorithm using the following QoS service agreements.

B. QoS in IEEE 802.16

The IEEE 802.16 defines four different QoS types, which are based on those defined in the DOCSIS v.1.1 standard [11]. These classes of services are described below.

Unsolicited Grant Service (***UGS)***: This service supports real-time service flows that generate fixed-size data packets on a periodic basis (CBR-like services), such as T1/E1, VoIP or videoconferencing. At the beginning of the connection setup, a SS transmits its requirements to the BS, such as grant size (G), grant interarrival time (λ), tolerated grant jitter (j) and *Poll* bit. The UGS service also includes Activity Detection (AD) to examine the flow state. If the state is inactive, then the UGS-AD Service sets the *Poll* bit to 1 and periodically provides a *unicast transmission opportunity* (*utxop*), in which a SS can tell the BS to reestablish its UGS service, thus saving bandwidth.

Real-Time Polling Service (*rtPS*): This service is oriented for real-time service flows that generate variable size data packets on a periodic basis (VBR-like services), such as MPEG video streams. The *rtPS* service offers periodic *utxop*, which meet the flow's real-time needs and allow the SS to specify the size of the desired channel reservation. A SS should indicate to the BS at the beginning of the session about its requirements for this service, such as polling interval (λ) and tolerated poll jitter (j).

Non Real-Time Polling Service (*nrtPS*): This type of service is like the *rtPS*, however polling will typically occur at a much lower rate and may not necessarily be periodic. This applies to applications that have no requirement for real time service but may need an assured high level of bandwidth. An example of this may be bulk data transfer (via FTP) or an Internet gaming application. The parameters required for this service are the polling interval (λ), along with the minimum and maximum sustained data rate.

Best Effort (*BE*): This kind of service is for standard Internet traffic, where no throughput or delay guarantees are provided.

The IEEE 802.16 MAC protocol can identify the type of service flow required by a SS using the following fields of the IEEE 802.16 protocol stack: source or destination MAC address, EtherType, source and destination IP address or network, IP protocol type, source or destination port number, IP type of service bits and any combination thereof. A simple example of how a classification might be used would be to match VoIP traffic from a particular source IP address and UDP port and direct that traffic into a dynamically created service flow that has a QoS parameter set providing a UGS mode of data transmission.

Once the service flows have been identified, the BS uses two modes of operation to allocate grants: 1) Grants per Connection (GPC) and Grants per Subscriber Station (GPSS). In the first case, the BS grants bandwidth explicitly to each connection, whereas in the second case bandwidth is granted to all of the connections belonging to the SS. The latter case (GPSS) allows smaller UL-MAPs and requires more intelligent SSs to make last moment decisions and perhaps utilize the bandwidth differently than it was originally granted by the BS. This may be useful for real-time applications that require a faster response from the system.

Edf-Bwa Scheduling Algorithm (EBSA)

The *UGS*, *rtPS* and *nrtPS* have specific requirements, for instance, *UGS* and *rtPS* have a deadline (jitter), and late packets that miss the deadline will be useless, but these two services can tolerate packet loss. However, for *nrtPS* and BE packet loss is not permitted but can handle larger delays. In order to guarantee these types of services with their specific requirements, we have implemented a scheduling algorithm to match CBR-like and VBR-like traffic. For all types of services, when a request for bandwidth is received at the BS, the QoS policy of the SS is first analyzed to make sure it is not violating the QoS contract (e.g. maximum bandwidth requirement). For all types of requests, EBSA will provide to the SS a *UGS*, *rtPS*, *nrtPS* or *BE* service using classifiers as defined in [1].

If the request is for a *UGS* service, EBSA will provide periodic grants to the SS. These grants are allocated in a *UGS* queue and ordered using EDF. The tolerated grant jitter is taken as the ordering parameter. If the request is for a *rtPS* or n*rtPS* service, EBSA will provide periodic *utxop* to the SS, as defined above. These *utxop* are allocated in an *rtPS* or *nrtPS* queue, respectively, and ordered using EDF. For these two services, the tolerated poll jitter is taken as the ordering parameter. Grants are ordered using a FIFO scheme for *BE* requests as illustrated in Fig. 2.

Finally, EBSA will dispatch grants or *utxop* using a WFQ scheme, where the weights of reservation for each type of service (W_{ugs}, W_{rtp} , W_{nrtps}, and W_{be}) can be configured according to the traffic needs by the network operator or dynamically assigned by EBSA using a reservation ratio calculator, based on current network traffic. In this paper, we base our study using fixed ratios. In general, EBSA is easy to implement and is compatible with IEEE 802.16 QoS requirements. In continuation, we present 6 steps to implement EBSA.

1. The first step is to identify the SS requirements by the following vector that is registered at the connection setup or updated as the SS requests bandwidth.

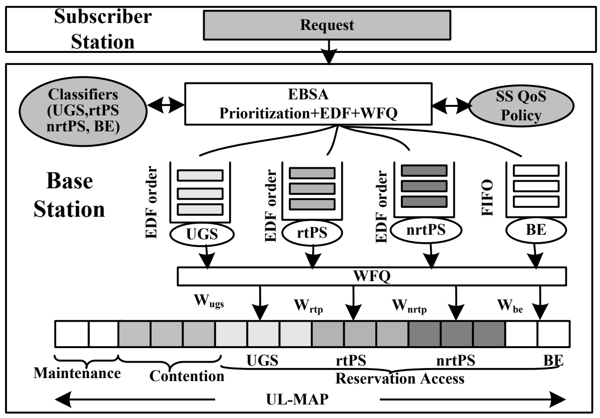

Fig. 2. UL and DL channel structure

$$SS_n = \{QoS_i, s, l, j, Poll, p, G\}, \qquad (1)$$

where '***QoS_i***' is the service type required by a SS, and defined by

$$QoS_i = \{0 \text{ for } UGS, 1 \text{ for } rtPS, 2 \text{ for } nrtPS\}, \qquad (2)$$

'***l***' is the interarrival time or polling interval measured in ***UL-frames***, (e.g., if interarrival time or polling interval = 10 ms and *UL-frame* = 2 ms, then *l = 10ms/2ms = 5 UL-frames),* '***j***' is the jitter measured, also in *UL-frames*, '***p***' is the user transmission (tx) priority, '***G***' is the grant size and '***s***' is the state of the connection. For every UL-MAP sent, the BS increases the user state (SSs_n) by 1. When SSs_n becomes equal to the interarrival time (SSs_n=SSl_n), the BS sends a grant or unicast transmission opportunity (*utxop*) to user SS_n depending on the '***Poll***' bit.

2. Secondly, the maximum number of priorities 'Max_p_i' that will be used for the service type QoS_i is obtained as follows:

$$Max\_p_i = \frac{\max(SSj_1\_QoS_i, SSj_2\_QoS_i, ..., SSj_n\_QoS_i)}{UL\_frame}, \quad (3)$$

where $SSj_n\_QoS_i$ refers to the jitter of user SS_n requiring QoS_i.

3. Next, each SS_n with QoS_i obtains its tx priority '$SS\boldsymbol{p}_n\_QoS_i$' using expression (4).

$$\begin{array}{ll}
\text{if } SSs_n_QoS_i - SSl_n_QoS_i = 0 \text{ and Poll} = 1 & \text{(a utxop is required)} \\
\quad SSp_n_QoS_i = Max_p_i & \text{(set SSp to highest tx priority)} \quad (4) \\
\text{else if } SSs_n_QoS_i - SSl_n_QoS_i \rangle 0 & \text{(a grant is required)} \\
\quad \text{(get tx priority based on current state and tolerated delay)} & \\
\quad SSp_n_QoS_i = SSs_n_QoS_i - SSj_n_QoS_i - SSl_n_QoS_i + Max_p_i & \\
else & \text{(a grant is not required in the current UL - frame)} \\
\quad SSp_n_QoS_i = 0 & \text{(set tx priority to 0)}
\end{array}$$

4. Following this, any maintenance regions are scheduled first as indicated in [1], and then the minislots that are to be used during contention access (for bandwidth request transmissions) are scheduled, using a Contention Slot Allocator (CSA). We have presented a good approximation for this in [12]. If there are some SS_n requiring a *utxop* in the current UL-frame, these are scheduled first. Users with a *UGS* service and the *Poll* bit set to 1 will use the *utxop* to tell the BS to reestablish its service. Users with an *rtPS* or *nrtPS* service always have the *Poll* bit set to 1, and these subscriber stations use the *utxop* for bandwidth requirements.
5. After this, in the remaining space of the UL-frame, the SS_n is scheduled according to its tx priority in the following order: *UGS*, *rtPS, nrtPS, BE*. In order to avoid scheduling unfairness, W_{ugs}, W_{rtp}, W_{nrtps}, and W_{be} are used to weigh bandwidth allocation. A high priority has precedence upon a low priority, where priority ($SSp_n\_QoS_i$)= 1 is the lowest and priority = Max_p_i is the highest. If there are two or more SS_n with the same priority, the Round Robin method is used to grant bandwidth allocation to the users. For *rtPS* services, if the available space in the current UL-frame < G, then the BS should use the fragmentation technique and send continuous grant opportunities until G is complete. For *nrtPS* services, if the available space in the current UL-frame < G, then the BS should use the fragmentation technique and users may request further bandwidth using piggyback requests. Every grant allocated in the UL-MAP for SS updates its state as follows:

$$SSs_n\_QoS_i = SSs_n\_QoS_i - SSl_n\_QoS_i\,. \quad (5)$$

6. Finally, any *BE* request using fragmentation + piggyback is scheduled to better utilize the UL channel.

The following section presents our scheduling algorithm, which is well suited to support QoS as it provides very low access delays for VBR-like and CBR-like traffic for IEEE 802.16 based networks.

Performance Analysis

We implemented a detailed simulation model of the IEEE 802.16 MAC protocol using the OPNET Package v. 11. A hierarchical design was used as shown in Fig. 3. At the top level of the BWA network topology, the network components, for example the BS and SS, along with their connectivity are shown in Fig 3a. The next level, Fig. 3b, defines the functionality of a SS in terms of components such as traffic sources, MAC, interfaces, etc. The operation of each component is defined by a state machine (an example of which is shown in Fig. 3c). The actions of a component at a particular state are defined in Proto-C code as shown in Fig. 3d. This approach allows modifications to be made to the operation of the IEEE 802.16 protocol and different optimizations and enhancements to be tested. The parameters used are given in Table 1.

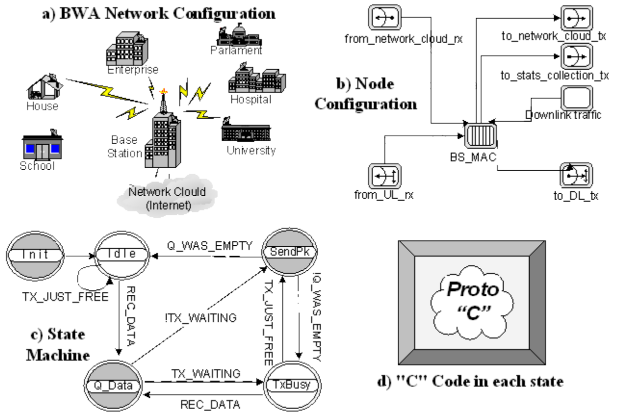

Fig. 3. OPNET simulation model

Table 1. Simulation Parameters

Parameter	Value
UL data rate (QPSK, m=2)	9.6 Mbps
UL Channel Bandwidth (BW_{UL})	6000 kHz
Roll of factor (γ)	0.25
DL data rate (16-QAM, m=4)	22.4 Mbps
Minimum contention slots per UL-frame (c)	7 slots
UL minislot size	16 bytes
UL-frame Duration (F) (≈2ms)	150 minislots
Simulation time for each run	60s
Distance from nearest/farthest SS to the BS	0.1 – 2.3 km
Reed Solomon (short grants/long grants)	6 / 10 bytes
Limit between short and long grants	245 bytes
Maximum number of users in the network	200

A. Traffic scenarios

In all simulations, one uplink channel with a capacity of 9.6 Mbps and one downstream channel with a capacity of 22 Mbps were used. Only four traffic sources were considered for the performance of the proposed scheduling algorithm. These traffic sources are described as follows.

1. VoIP- G.723-UGS. This traffic type emulates a speech codec "G.723.1", which according to the ITU, IETF and the VoIP Forum is the preferred speech codec for Internet telephone applications. This codec generates a data rate of 5.3 kbps, where 20-byte data packets are generated and encoded every 30 ms by adding the complete headers as illustrated in Table 2; one obtains a VoIP stream of 38.4 kbps at the physical (PHY) layer.

2. Vo IP- G.711. Codec G.711 was used to stress the BWA network. This codec was also employed because it is commonly used for quality voice calls. G.711 is the mandatory codec according to the ITU-T H.323 conferencing standard, which uses Pulse Code Modulation to produce a data rate of 64 kbps. This codec creates and encapsulates an 80-byte VoIP frame every 10 ms, and

demands a stream of 166.4 kbps at the PHY layer.

Table 2. VoIP Codecs; G.711 and G.723.1

Parameters	G.711 - 64 kbps	G.723.- 5.3 kbps
Frame size	10 ms	30 ms
Voice frame	80 bytes	20 bytes
RTP	12 bytes	12 bytes
UDP	8 bytes	8 bytes
IP	20 bytes	20 bytes
LLC	3 bytes	3 bytes
SNAP	5 bytes	5 bytes
Ethernet MAC	18 bytes	18 bytes
IEEE 802.16 MAC	6 bytes	6 bytes
PHY: (Prea+GB+FEC)	10+FEC bytes	10+FEC bytes
Total PacketSize	202bytes/13slots	86bytes/9slots
Net rate at MAC/PHY	116.6 /166.4 kbps	22.9 /38.4 kbps

Prea = Preable, GB = Guarband, and FEC = 6* No_CodeWords

3. MPEG4-rtPS. For this type of service we used the traces of 10 MPEG-4 movies as defined in [13]. The traces were digitalized using QCIF (Quarter Common Intermediate Format) with 176*144 pls, at 25 fps (frames per second). Table 3 shows the selected MPEG-4 movies. The MPEG-4 movie transmitted was selected by the SS in a random fashion. The mean data rate at the PHY layer was 157.7 kbps.

Table 3. MPEG-4 movies

No	Movie Name	Mean video Frame (bytes)	Grant Size (Slots)	Video frame Rate (kbps)
1	Aladdin	297.61	25	80
2	Die Hard III	587.06	44	14.08
3	Futurama	1106.30	70	224
4	Jurassic	684.74	50	160
5	Mr. Bean	437.91	34	108.8
6	Robin_Hood	460.18	36	115.2
7	Silence	1871.20	130	416
8	Start Trek	209.22	19	60.8
9	Starwars	530.59	40	128
10	The Simpsons	1464.60	103	329.6

4. Internet Traffic-IP: The Internet traffic distribution utilized is the one introduced by the IEEE 802.14 working group [12]. The message size distribution was as follows: 64-byte Pk. 60%, 128-byte Pk. 6%, 256-byte Pk. 4%, 512-byte Pk. 2%, 1024-byte Pk. 25% and 1518-byte Pk. 3%. The inter-arrival times were set in such a way that the Internet offered load per active station was 38.4 kbps at the physical layer.

B. Performance of EBSA for Voice, Video and Internet Traffic

In this traffic scenario we study how the proposed algorithm scales with an increased number of traffic sources producing voice, video and Internet traffic.

The network was configured in groups of 6 SSs, in the following manner: 1-VoIP$_{G711}$ stream (SS 1, 7, 13, etc), 1-VoIP$_{G723}$ stream (SS 2, 8, 14, etc), 1 MPEG4 stream (SS 3, 9, 15, etc) and 3 Internet users (SS 4-6, 10-12, etc). For this traffic scenario, the weighted ratios were configured as follows: W_{ugs} =0.60, W_{rtp} = 0.38, W_{nrtps}, and W_{be} = 0.02. The performance analysis of EBSA is provided in terms of throughput, access delays and cumulative probability.

Fig. 4 presents the throughput achieved by each service flow. In the figure, the maximum system throughput achieved by the network is approximately 9.1 Mbps, which corresponds to 95 % of the UL channel capacity (of 9.6 Mbps), the other 5% was assigned to contention and maintenance access. From this system throughput, 7.2 Mbps (75% of UL capacity) was used for SS data transmission. The other 2 Mbps (20% of UL capacity) was consumed by the Radio Link Protocol (RLP) of the IEEE 802.16, which consists of MAC and PHY headers.

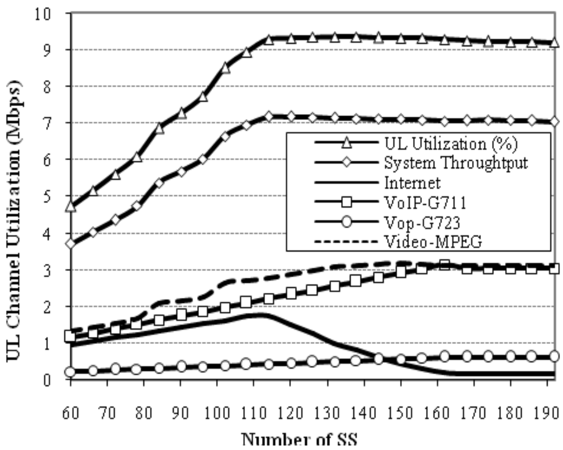

Fig. 4. Throughput for all traffic types

In order to validate our results, we derived a formula to estimate the maximum number of supported SSs. MaxSS in a 6000 kHz UL channel is as follows:

$$\text{MaxSS} = \frac{6R_{UL}}{VoIP_{G711} + VoIP_{G723} + MPEG4 + 3Internet}, \quad (6)$$

where the mean data rate (at PHY) of $VoIP_{G711}$, $VoIP_{G723}$, MPEG4, and *Internet* traffic is 166.4kbps, 38.4kbps, 157.7kbps and 38.4kbps respectively. R_{UL} is the maximum PHY data rate for reservation access in the UL direction that is given by,

$$R_{UL} = \frac{mBW_{UL}}{1+\gamma} \cdot \frac{F-c}{F}. \quad (7)$$

Thus, using the values given in Table 1 we can find that MaxSS is,

$$\text{MaxSS} = \frac{6\dfrac{2\cdot 6000}{1+0.25}\cdot\dfrac{150-7}{150}}{166.4+38.4+157.7+3\cdot 38.4} \approx 115$$

From simulation results, the number of supported SSs was of 114 (formed by 57 Internet users, 19-VoIP$_{G711}$ users, 19-VoIP$_{G723}$ users and 19-MPEG4 users). This value represents the maximum number of users that the network can support before it starts dropping packets, and is close to the maximum number estimated by Eq. (6). In Fig. 4 we can appreciate that Internet throughput starts to decrease when the network is loaded with 114 SSs. At this point, the portion of Internet traffic transmitted was 24% ((114/2)SS*38.4kbps/9152kbps) of the bandwidth for reservation access R_{UL} = 9.152Mbps. This portion was much higher than W_{be} = 2%. This is because EBSA allocates un-scheduled rtPS bandwidth to BE traffic and

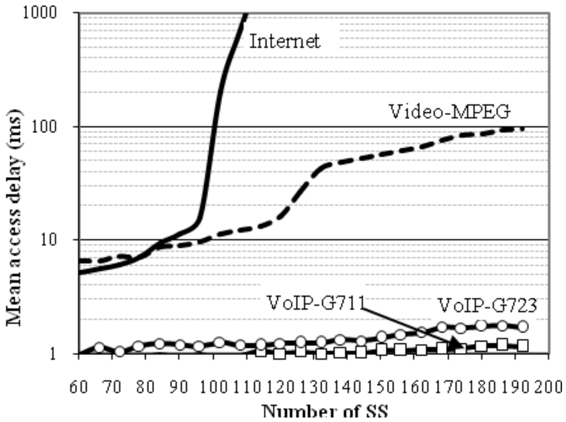

Fig. 5. Mean Access Delays

un-scheduled UGS bandwidth is allocated to rtPS traffic and then to BE traffic. However, during high congestion periods (> 160 SS), EBSA insures that each service type obtains the portion assigned to it (2% for BE, 38% for rtPS and 60% for UGS).

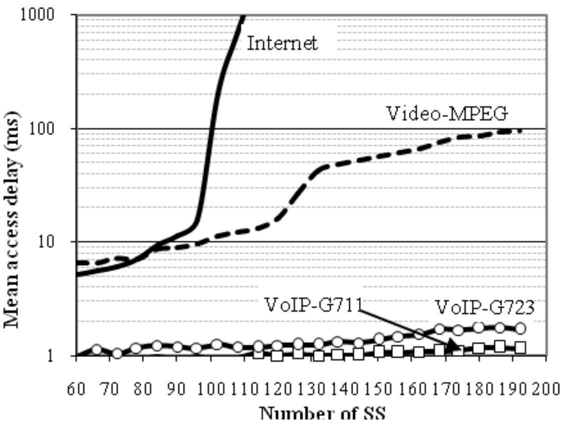

Fig. 6. Mean Access Delays

In terms of mean access delays, EBSA always provides balanced access delays, according to the tolerated jitter. The maximum mean access delays for VoIP-G711, VoIP-G723 and MPEG4 streams during high congestion periods (> 150 SSs), were of 2 ms, 4 ms and 95 ms, respectively, as shown in Fig. 5.

These access delays were lower than the tolerated jitter for each UGS and rtPS stream. However, with a network population >114 SSs Internet frames had mean access delays of over 200 ms. The direct consequence of having W_{be} = 2% is that during congestion periods, Internet traffic is assigned a decreased scheduling priority in order to guarantee low transmission delays for UGS and rtPS streams. However, this behavior can be controlled by the network operator by setting the rations according to user needs.

The maximum VoIP streams (without Internet traffic) that a UL channel can support from Eq. (6) is 44-G.711 and 44-G.723 streams. Fig. 6a shows that for VoIP streams, 100% of frames transmitted on the BWA network had access delays under 4ms for G.711 streams and 8ms for G.723 streams. All VoIP streams were transmitted without packet loss. Fig. 6b shows the performance when the BWA network is over-loaded with 45-G.711 streams, 45-G.723

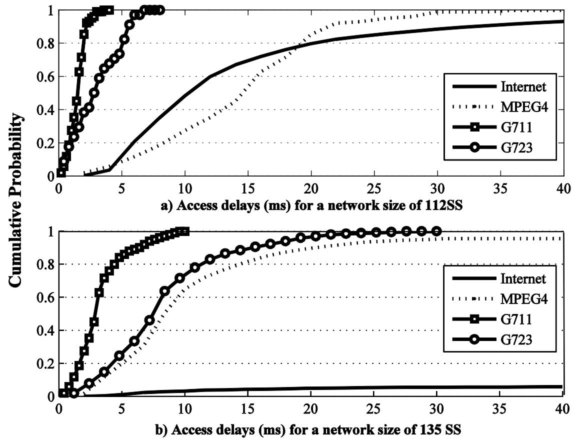

Fig. 7. Cumulative Probability vs Access delay

streams and 45 Internet users. Here, simulation results reported a packet loss of ≈3 %, due to late packets, which is still acceptable for the support of VoIP streams.

This paper does not include a discussion about the impact of channel errors in the previous analysis of EBSA. Channel errors can degrade the QoS observed by SSs in various ways, depending of the particular service class being considered. For *UGS* and *ntPS* classes for instance, losing packets due to channel errors may represent a violation of the QoS flow agreement. We are currently investigating ways to overcome this problem within the EBSA framework. One simple solution already proposed by other researchers [14] is to assign additional transmission opportunities (slots in our case) to flows facing channel errors in order to keep up with QoS requirements.

Finally, EBSA can easily be modified so that users exceeding the maximum bandwidth allowed are redirected to other UL channels or cells.

Conclusions

This paper has presented a scheduling algorithm for IEEE 802.16 based networks in FDD mode. The proposed algorithm is practical, compatible with IEEE QoS requirements, and is easy to implement. The purpose of *EBSA* is to provide a higher transmission priority to service flows with minimum tolerated jitter. It provides tight delay guarantees for *UGS* and *rtPS,* and minimum bandwidth reservations for *nrtPS* and *BE* flows, according to the weighted ratios. Simulation results of *EBSA* show that real-time services, such as VoIP and video, can be supported with very low access delays even during high congestion periods. Results found by the simulation model were in good agreement with a simple theoretical model that estimated the maximum number of SSs in the UL channel. The performance of *EBSA* with mixed traffic sources (*UGS*, *rtPS*, *nrtPS* and *BE*) and with channel errors will be further investigated through simulations and theoretical analysis. The results of such performance analysis will be provided in future publications.

Acknowledgments

This work was supported by DGAPA, National

Autonomous University of Mexico (UNAM) under Grant PAPIIT IN104907 and PAPIME PE 103807.

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Received 2009 10 15

V. Rangel, Y. Macedo, L. Ortiz, J. Gómez, R. Aquino, A. Edwards. QoS Management for Broadband IEEE 802.16 based Networks in FDD Mode // Electronics and Electrical Engineering. – Kaunas: Technologija, 2010. – No. 2(98). – P. 3–9.

The design and performance analysis of a scheduling technique for the provision of QoS over Broadband Wireless Access Networks (BWA) is presented. The proposed scheduling algorithm is based on the MAC protocol of the IEEE 802.16 standard and focuses on the uplink channel, which is the limiting factor of BWA networks and is critical in the delivery of services to individual users. Although the IEEE 802.16 standard had proposed several QoS service classes for various types of applications, they do not suggest how to schedule traffic to fulfill timing critical services such as compressed/uncompressed voice, audio and video streams. We have derived a mechanism called EBSA that combines several scheduling algorithms to closely match VBR-like and CBR-like traffic over the IEEE 802.16 air interface. Simulation results show that EBSA provides real-time services with very low access delays even during congestion periods. Ill. 7, bibl. 14 (in English; summaries in English, Russian and Lithuanian).

В. Рангел, Ю. Маседо, Л. Ортиз, Ю. Гомес, Р. Акино, А. Эдвардс. Анализ эффективности планирования QoS в широкополосных IEEE 802.16 сетях в режиме FDD // Электроника и электротехника. – Каунас: Технология, 2010. – № 2(98). – С. 3–9.

Представлен анализ эффективности планирования обеспечения QoS более широкополосных беспроводных сетей (BWA). Предложенный алгоритм планирования на основе MAC-протокола IEEE 802.16, а основное внимание уделяется наземному каналу, который является фактором, ограничивающим сетеи BWA и играет решающую роль в оказании услуг для индивидуальных пользователей. Хотя по IEEE 802.16 предложено несколько класс служб QoS для различных типов приложений, они не свидетельствуют о том, как планировать потоки данных для выполнения важнейших задач, таких как передача сжатого и несжатого голоса, аудио-и видеопотоков. Получен механизм, называемый EBSA, который сочетает в себе несколько алгоритмов планирования очередей. Результаты моделирования показывают, что EBSA обеспечивает оказание услуг в реальном времени с очень низкими задержками доступа даже во время периодов заторов. Ил. 7, библ. 14 (на английском языке; рефераты на английском, русском и литовском яз.).

V. Rangel, Y. Macedo, L. Ortiz, J. Gómez, R. Aquino, A. Edwards. QoS eilių sudarymo plačiajuosčiuose IEEE 802.16 FDD tipo tinkluose našumo analizė // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2010. – Nr. 2(98). – P. 3–9.

Pateiktas eilių sudarymo metodas QoS užtikrinti plačiajuosčiuose belaidės prieigos tinkluose (BWA) ir jo našumo analizė. Siūlomas eilių sudarymo algoritmas remiasi IEEE 802.16 standarto MAC protokolu ir yra skirtas duomenų siuntimo kanalui. Šis kanalas yra efektyvumą ribojantis BWA tinklų veiksnys, vaidinantis lemiamą vaidmenį teikiant paslaugas individualiems vartotojams. Nors pagal IEEE 802.16 standartą, atsižvelgiant į taikymo sritį, skiriamos kelios QoS paslaugos klasės, nėra reglamentuojama, kaip turi būti planuojamas duomenų srautas, tiksliai laiku teikiant paslaugas, tokias kaip suglaudintos arba nesuglaudintos balso, muzikos ir vaizdo srautų informacijos perdavimas. Sukurtas EBSA mechanizmas, tarpusavyje suderinantis kelių eilėms sudaryti naudojamų algoritmų pranašumus. Modeliavimo rezultatai rodo, kad, naudojant EBSA, paslaugos teikiamos realiu laiku – prieigos vėlinimo trukmė yra itin maža netgi esant perkrautam tinklui. Il. 7, bibl. 14 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).