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 subscribers 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 for performance is concerned, BWA systems will provide services for diverse traffic classes, with 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 tell how to schedule traffic to fulfill QoS requirements specifically [1, Section 6.1]. In recent years, several scheduling algorithms for BWA networks were 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; nonetheless, most licensed bands intended for data applications operate with FDD systems in mind. Second, The MAC-level software tend to have a more complicated scheduler than an FDD system since it must deal with synchronizing many subscribers´ 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 the UL channel resources according to network operators needs. In a real scenario, IEEE 802.16 network operators would like to distribute the channel resources according to the QoS agreement acquired by subscriber users. However, many scheduling algorithms found in the literature allocate, in the first place, 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 mechanisms in order 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. In this paper, 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 realtime and non-realtime 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, specified a physical layer that used single-carrier modulation techniques and a media access control (MAC) layer with a burst time division multiplexing (TDM) structure that supported 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] replaces 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 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 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 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 gets 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, this 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 get authenticated by the BS. Then the SS performs the registration process by establishing a security association that allows the SS to entry into the network. The next step is to establish IP connectivity, the BS uses the DHCP mechanisms 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, packing, among others. The last step is to set up connections for

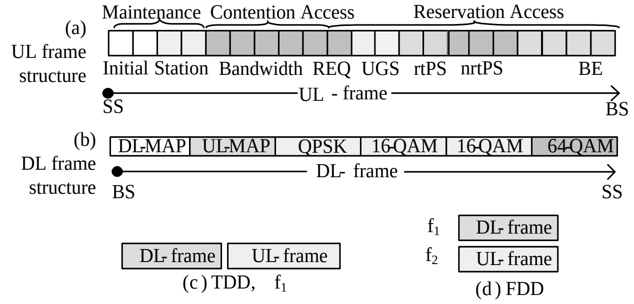

Fig. 1. Frame structure for TDD and FDD access.

preprovisioned 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 subframe, 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 is oriented for the support of real-time service flows that generate fixed- size data packets on a periodic basis (CBR-like services), such as T1/E1, VoIP or videoconference. At the beginning of the connection setup, a SS indicates the BS about its requirements for this service, 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 provides periodically a *unicast transmission opportunity* (*utxop*), in which a SS can indicate 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 a 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), 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 to direct that traffic into a dynamically created service flow that had 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 the bandwidth is granted to all 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 accommodates larger delays. In order to guarantee these types of service with their specific requirements, we have implemented a scheduling algorithm to match CBR-like and VBR-like traffic. For all types of service, 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 request, EBSA will provide to 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. For *BE* requests, the grants are ordered using a FIFO scheme 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 based our study using fixed ratios. In general, EBSA is easy to implement and is compatible with IEEE 802.16 QoS requirements. Bellow we present 6 steps to implement EBSA.

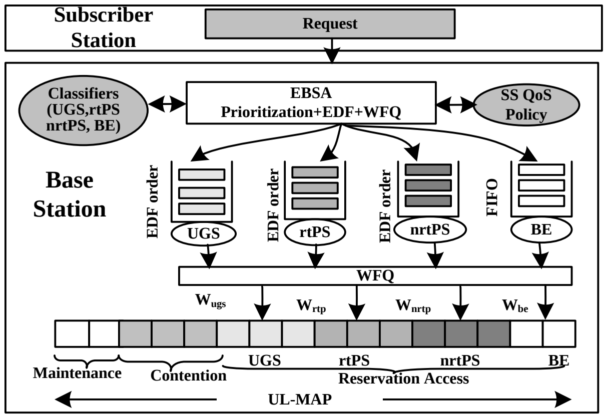

Fig. 2. UL and DL channel structure.

1) We have identified the SS requirements by the following vector that is registered at connection setup or updated as the SS asks for bandwidth.

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

where '$\boldsymbol{QoS_i}$' 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 ($SS\boldsymbol{s}_n$) by 1. When $SS\boldsymbol{s}_n$ becomes equal to the interarrival time ($SS\boldsymbol{s}_n$=$SS\boldsymbol{l}_n$), it is time for the BS to send a grant or unicast transmission opportunity (*utxop*) to user SS_n depending on the '***Poll'*** bit.

2) Get the maximum number of priorities 'Max_p_i' that will be used for service type QoS_i as follows:

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

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

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

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

4) Schedule first any maintenance regions as indicated in [1], then schedule the minislots that will be used during contention access (for bandwidth request transmissions),

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, schedule these slots first. Users with a *UGS* service and the *Poll* bit set to 1 will use the *utxop* to indicate the BS to reestablish its service. Users with a *rtPS* or *nrtPS* service always have the *Poll* bit set to 1, and these subscriber stations use the *utxop* for bandwidth requirements.

5) In the remaining space of the UL-frame, schedule the SS_n according to its tx priority in the following order: *UGS, rtPS, nrtPS, BE*. In order to avoid scheduling unfairness, use the weigh (W_{ugs}, W_{rtp}, W_{nrtps}, and W_{be} respectively) for bandwidth allocation. A high priority have 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, use Round Robin to grant these 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 for further bandwidth using piggyback requests. For every grant allocated in the UL-MAP for SS, update its state as follows.

$$\mathrm{SSs}_n\_QoS_i = \mathrm{SSs}_n\_QoS_i - SSl_n\_QoS_i\,. \qquad (5)$$

6) Finally, schedule any *BE* request using fragmentation + piggyback for a better utilization of the UL channel.

In the following section we demonstrate that our scheduling algorithm is well suited for the support of QoS providing 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 and this is 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 such as that in Fig. 3d. This approach allows modifications to be applied to the operation of the IEEE 802.16 protocol and different optimizations and enhances to be tested. The parameters used are given in Table 1.

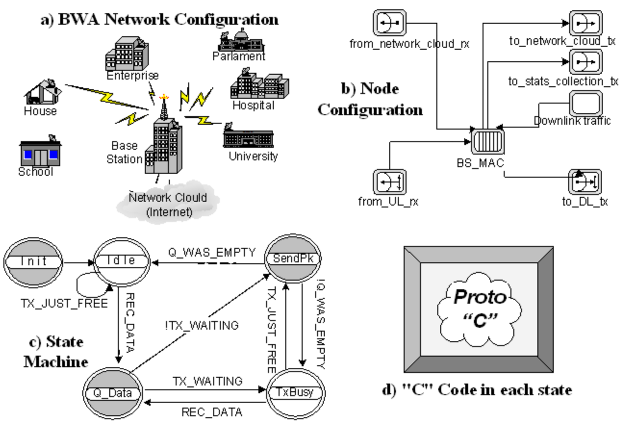

Fig. 3. OPNET simulation model.

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 telephony 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.

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 = Guardband, and FEC = 6* No_CodeWords

2)***Vo IP- G.711***. Codec G.711 was considered to stress the BWA network and also because this codec will be 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 a 80-byte VoIP frame every 10 ms, and demands a stream of 166.4 kbps at the PHY layer.

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 to transmit is selected by the SS in a random fashion. The mean data rate at the PHY layer becomes of 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 is 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 are set in such a way that the Internet offered load per active station is 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 will be 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 of 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.

In order to validate our results, we derived a formula to estimate the maximum number of supported SS

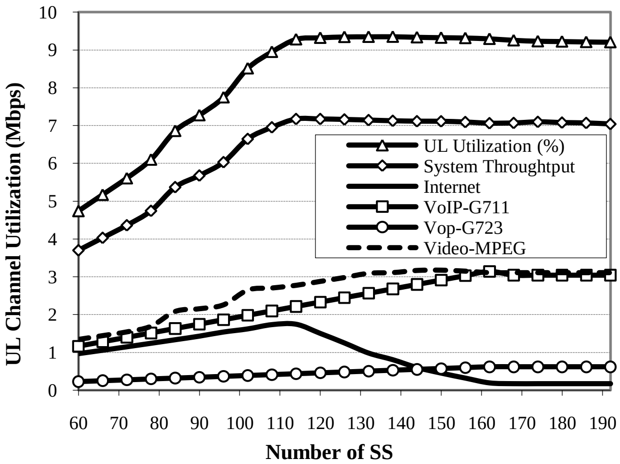

Fig. 4. Throughput for all traffic types.

(MaxSS) in a 6000 kHz UL channel, 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, as explained above. 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,

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

From simulation results, the number of supported SS 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 start dropping packets, and it is close to the maxim number estimated by Eq. (6). In Fig. 4 we can appreciate that Internet throughput starts to decrease when the network is loaded with 114 SS. At this point, the portion of Internet traffic transmitted was of 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 allocated un-scheduled rtPS bandwidth to BE traffic and un-scheduled UGS bandwidth was first allocated to rtPS traffic and then to BE traffic. However on high congestions periods (> 160 SS), EBSA makes sure that each service type gets its portion assigned (2% for BE, 38% for rtPS and 60% for UGS).

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 on high congestions

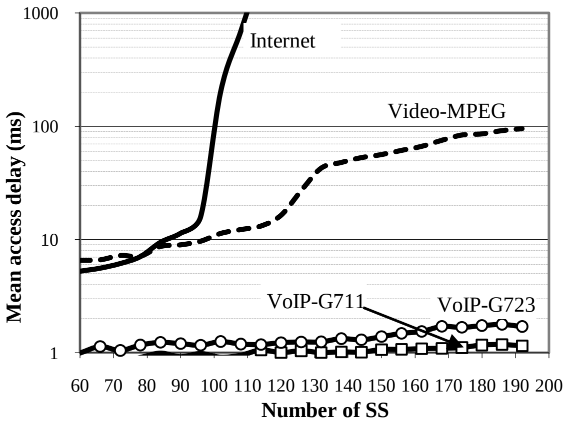

Fig. 5. Mean Access Delays.

periods (> 150 SS), 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 >114SS Internet frames had mean access delays over 200 ms. The direct consequence of having W_{be} = 2% is that on congestions periods, Internet traffic gets a decreased scheduling priority in order to guarantee low transmission delays for UGS and rtPS streams. However, this unfairness can be controlled by the network operator by setting the rations according to the 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 streams and 45 Internet users. Here, simulations results reported a packet loss of ≈3 %, due to late packets, which is still acceptable for the support of VoIP streams.

In this paper, we did not include a discussion about the impact of channel errors in the previous analysis of EBSA. Channel errors can degrade the QoS observed by SS in various ways depending of the particular service class being considered. For *UGS* and *ntPS* classes for instance, losing packets in the air 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

In this paper we have presented a scheduling algorithm for IEEE 802.16 based networks in FDD mode. The proposed algorithm is practical, compatible with IEEE

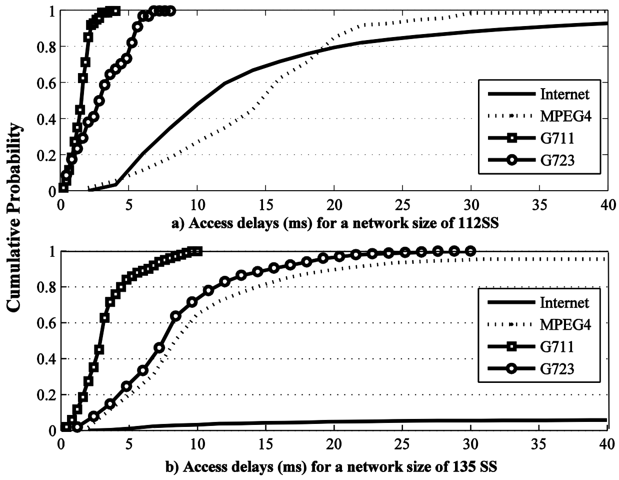

Fig. 6. Cumulative Probability vs Access delay.

QoS requirements, and easy to implement. The purpose of *EBSA* is to provide a higher transmission priority to service flows with minimum tolerated jitter. It provides tight delays 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 SS in the UL channel. The performance of *EBSA* with mixed traffic sources, (*UGS*, *rtPS*, *nrtPS* and *BE*) 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

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V. Rangel, Y. Macedo, L. Ortiz, J. Gómez, R. Aquino, Performance Analysis of QoS Scheduling in Broadband IEEE 802.16 Based Networks // Electronics and Electrical Engineering

This paper presents the design and performance analysis of a scheduling technique for the provision of QoS over Broadband Wireless Access Networks (BWA). 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 compresed/uncompresed 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.