

A Call Admission Control with Dynamic Bandwidth Allocation for providing delay guarantees in IEEE 802.11e Networks

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Abstract— The 802.11e working group has recently proposed innovative functionalities in order to support delay-sensitive multimedia applications, such as real-time voice and video, in Wireless Local Area Networks (WLANs). This paper proposes an innovative dynamic bandwidth allocation algorithm along with a measurement based CAC algorithm for providing delay guarantees to real-time media flows in IEEE 802.11e networks. These algorithms have been implemented in the *ns-2* simulator and extensive computer simulations have been carried out to assess their validity. Simulation results have shown that proposed algorithms guarantee bounded delays for multimedia flows, whereas, analogous algorithms proposed by the 802.11e working group fail in presence of high network loads.

Keywords— QoS, WLANs, 802.11e networks, Feedback Control

I. INTRODUCTION

The widespread deployment of IEEE 802.11 Wireless Local Area Networks (WLANs) as a fundamental tool for enabling ubiquitous wireless networking is mainly due to their easy installation, flexibility and robustness against failures [1].

The 802.11 Medium Access Control (MAC) employs a mandatory contention-based channel access scheme called Distributed Coordination Function (DCF), which is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) [1] and an optional centrally controlled channel access scheme called Point Coordination Function (PCF) [1].

With PCF, the time is divided into repeated periods, called *SuperFrames* (SFs), which consist of a Contention Period (CP) and a Contention Free Period (CFP). During the CP, the channel is accessed using the DCF whereas, during the CFP, is accessed using the PCF. Each superframe must contain a CP long enough to transmit at least one Mac Protocol Data Unit (MPDU) [1].

The fundamental building block of the IEEE 802.11 architecture is the Basic Service Set (BSS), which is composed by a group of stations (STAs), located in the same geographical area, that access the radio channel under the direct control of a single coordination function (i.e., DCF or PCF) [1].

The 802.11 standard allows two topological configurations that are the Independent Basic Service Set (IBSS) and the Extended Service Set (ESS)[1]. An IBSS is a particular BSS where any station can establish a direct communication session with any other station in the IBSS (i.e., an ad hoc network architecture). An ESS, instead, is composed by one or more interconnected BSS, which are linked to each other via a common distribution system, generally wired. In the case of ESS, stations within each BSS cannot communicate to each other directly but the traffic is channeled through a central station, which is referred to as Access Point (AP) and is responsible also for inter-BSS communications [1]. In the sequel we will always assume ESS configurations.

At present, the use of WLANs has been essentially focused on best effort data transfer because the basic DCF and PCF access methods cannot provide delay guarantees to real-time multimedia flows [2], [3]. Recently, in order to support also delay-sensitive multimedia applications, such as real-time voice and video, the

802.11e working group has proposed innovative functionalities, which enable service differentiation in WLANs [4]. In particular, the 802.11e proposal introduces: (1) the Hybrid Coordination Function (HCF), which is an enhanced access method to distribute the WLAN bandwidth; (2) a Call Admission Control (CAC) algorithm; (3) specific signalling messages for service request and Quality of Service (QoS) level negotiation.

This paper proposes an innovative dynamic bandwidth allocation algorithm along with an associated CAC algorithm for providing delay guarantees to real-time media flows in IEEE 802.11e networks. The dynamic bandwidth allocation algorithm has been designed using classic feedback control theory [5] whereas the CAC scheme is an extension of the admission control algorithm proposed by the 802.11e working group. Both of them exploit the HCF Controlled Channel Access (HCCA) centralized access method. The developed schemes have been implemented in the *ns-2* simulator and extensive computer simulations have been carried out to assess their performance. Simulation results have shown that the proposed algorithms are able to provide bounded delays to multimedia flows, whereas, the standard algorithms proposed by the 802.11e working group fail in presence of high network loads.

The rest of the paper is organized as follows: Section II describes the main functionalities of the 802.11e QoS enhancements; Section III proposes and analyzes the dynamic bandwidth allocation algorithm, and illustrates the proposed CAC scheme; Section IV shows simulation results; finally, the last Section draws the conclusions.

II. IEEE 802.11E QOS ENHANCEMENTS

In order to obtain service differentiation in 802.11 WLANs, the 802.11e working group has introduced new functionalities for QoS service level negotiation, Call Admission Control, and an improved access method, which is the Hybrid Coordination Function (HCF). Stations operating under 802.11e specifications are usually known as enhanced stations or QoS Stations (QSTAs). 802.11e defines 8 Traffic Categories (TCs) with the priority values of the IEEE 802.1D standard [6]. Four Access Categories (ACs) have been introduced in order to support the mentioned eight TCs [4]. To satisfy the QoS requirements of each AC, the concept of TXOP (Transmission Opportunity) is introduced, which is defined as the time interval during which a station has the right to transmit. The contiguous time during which TXOPs are granted to the same QSTA is called Service Period (SP). The interval T_{SI} between two successive SPs is called Service Interval [2], [4].

A. The Hybrid Coordination Function (HCF)

The HCF is made of a contention-based channel access, known as the Enhanced Distributed Coordination Access (EDCA), and of a HCF Controlled Channel Access (HCCA). The use of the HCF requires a centralized controller, which is called the Hybrid Coordinator (HC) and is generally located at the access point.

The **EDCA method** operates as the basic DCF access method but using different contention parameters per access category. In this way, a service differentiation among ACs is statistically pursued [7].

The **HCCA method** combines some of the EDCA characteristics with some of the PCF basic features as it is shown in Fig. 1. Each superframe starts with a beacon frame after which, for legacy purpose, there could be a contention free period for PCF access. The remaining part of the superframe forms the CP, during which the QSTAs contend to access the radio channel using the EDCA mechanism.

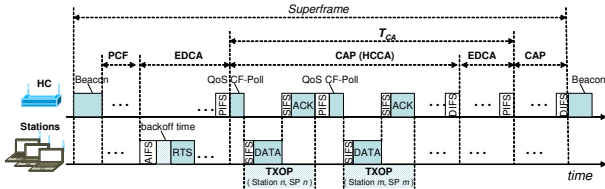


Fig. 1. Scheme of a superframe using the HCF controlled access method.

After the medium remains idle for at least a PIFS interval during the CP, the HC can start a Contention Access Phase (CAP)¹. During the CAP, only QSTAs polled and granted with a special frame, known as *QoS CF-Poll frame*, are allowed to transmit during their TXOPs. Thus, the HC implements a prioritized medium access control. Moreover, at least one CP interval, long enough to transmit a maximum size data frame at the minimum BSS rate, must be contained in a superframe; this CP interval can be used for management tasks, such as associations of new stations, new traffic negotiations, and so on. CAP length cannot exceed the value of the system variable *dot11CAPLimit*, which is advertised by the HC in the Beacon frame when each superframe starts [4].

The IEEE 802.11e specifications allow QSTAs to feed back queue lengths of each AC to the HC. This information is carried on a 8 bits long subfield contained in the *QoS Control Field* of each frame header, during data transmission both in the CAPs and in the CPs; queue lengths are reported in units of 256 octets. This information can be used to design novel HCCA-based dynamic bandwidth allocation algorithms using feedback control [5], [8], as will be shown in this paper. In fact, the 802.11e draft does not specify how to schedule TXOPs in order to provide the required QoS; it only suggests a simple scheduler that uses static values declared in TSPECs for assigning fixed TXOPs (more details about this scheduler can be found in [4] and [9]). This scheduler provides a CBR service, thus it is not well suited for bursty media flows [9]. An improved bandwidth allocation algorithm has been proposed in [10], which schedules transmission opportunities by taking into account both the average and the maximum source rates declared in the TSPECs. However, also this scheme does not consider the dynamic behavior of multimedia flows. An adaptive version of the simple scheduler, which is based on the Delay-Earliest Due-Date algorithm, has been proposed in [9]. However, this algorithm does not exploit the explicit queue length to assign TXOPs, but implements a trial and error procedure to discover the optimal TXOP to be assigned to each station.

In the following section, we will design a novel closed-loop control scheme based on feeding back queue levels to guarantee the typical QoS requirements of real-time audio/video applications.

B. QoS signalling

Service request and Service level negotiation are new functionalities introduced by the 802.11e proposal [4] in order to support

QoS in 802.11 networks. Each Traffic Stream (TS), i.e., a data flow with QoS needs, is described by a *Traffic SPECification* (TSPEC), which indicates the main characteristics of the TS. When a new TS has to be started, the following steps are executed considering that each node has a MAC layer and a Management Entity:

- 1) The *Station Management Entity* (SME) of the QSTA with the new TS request issues a setup phase by generating a message, which is known as *Mac Layer Management Entity (MLME)-ADDTS request* and contains the TSPEC of the TS.
- 2) The QSTA MAC layer transmits this request to the HC and starts a response timer (known as *ADDTS timer*), the duration of the *ADDTS Timer* is defined by the system variable *dot11ADDTSResponseTimeout*.
- 3) After receiving the request, the MAC layer of the HC generates the message *MLME-ADDTS indication* for its SME which contains the TSPEC.
- 4) The SME in the HC decides whether to admit the TS with the specified TSPEC, or to refuse it or to not admit it suggesting an alternative TSPEC.
- 5) The HC MAC transmits the ADDTS response.
- 6) The QSTA MAC layer receives this management frame and reset the *ADDTS timer*. Then it sends a *MLME-ADDTS confirm* message to its SME. The QSTA SME decides whether the response meets its needs or not.

C. Call Admission Control

An IEEE 802.11 network may use admission control to administer the available bandwidth resources. The HC is used as admission control unit. Since the QoS facility supports two access mechanisms, there are two distinct admission control mechanisms: one for contention-based access and the other for controlled-access. Herein, we will focus on the latter mechanism since the algorithms proposed in this paper are based on the controlled access mechanism HCCA. Details regarding the contention-based admission control can be found in [4].

By assuming that k is the number of admitted flows, when a new TS requests admission, the *Admission Control Unit* (ACU), calculates the static TXOP duration that needs to be allocated for the stream ($TXOP_{k+1}$) as stated by the simple scheduler algorithm. The stream is admitted if the following inequality is satisfied:

$$\frac{TXOP_{k+1}}{T_{SI}} + \sum_{i=1}^k \frac{TXOP_i}{T_{SI}} \leq \frac{T - T_{CP}}{T} \quad (1)$$

where T indicates the superframe duration and T_{CP} is the time used for EDCA traffic during the superframe.

III. HCCA-BASED DYNAMIC BANDWIDTH ALLOCATION

This section describes the HCCA-based bandwidth allocation algorithm that has been adopted within this paper. The algorithm, which has been designed using classical feedback control theory and will be referred to as *Feedback Based Dynamic Scheduler* (FBDS), distributes the WLAN bandwidth among all the multimedia flows by taking into account the queue levels fed back by the QSTAs [8].

We will refer to a WLAN system made of an Access Point (AP) and a set of quality of service enabled mobile stations (QSTAs). Each QSTA has N queues, with $N \leq 4$, one for any AC in the 802.11e proposal. Let T_{CA} be the time interval between two successive CAPs. Every time interval T_{CA} , assumed constant, the AP must allocate the bandwidth that will drain each queue during the next CAP. We assume that at the beginning of each CAP, the AP is aware of all the queue levels q_i , $i = 1, \dots, M$, at the beginning of the previous CAP, where M is the total number of

¹HCCA can be also enabled during the CFP with a procedure similar to the one described in this Section.

traffic queues in the WLAN system. The dynamics of the i^{th} queue can be described by the following discrete time linear model:

$$q_i(k+1) = q_i(k) + d_i(k)T_{CA} + u_i(k)T_{CA}; \quad i = 1, \dots, M, \quad (2)$$

where $q_i(k) \geq 0$ is the i^{th} queue level at the beginning of the k^{th} CAP; $u_i(k) \leq 0$ is the average depletion rate of the i^{th} queue (i.e., the bandwidth assigned to drain the i^{th} queue); $d_i(k) = d_i^s(k) - d_i^{CP}(k)$ is the difference between $d_i^s(k) \geq 0$, which is the average input rate at the i^{th} queue during the k^{th} CAP, and $d_i^{CP}(k) \geq 0$, which is the amount of data transmitted by the i^{th} queue during the k^{th} CAP divided by T_{CA} . The signal $d_i(k)$ is unpredictable since it depends on the behavior of the source that feeds the i^{th} queue and on the number of packets transmitted during the contention periods. Thus, from a control theoretic perspective, $d_i(k)$ can be modelled as a disturbance. Without loss of generality, the following piecewise constant model for the disturbance $d_i(k)$ can be assumed [11]:

$$d_i(k) = \sum_{j=0}^{+\infty} d_{0j} \cdot 1(k - t_j) \quad (3)$$

where $1(k)$ is the unitary step function, $d_{0j} \in \mathbb{R}$, and t_j is a time lag. Due to the assumption (3), the linearity of the system (2), and the superposition principle that holds for linear systems, we will design the feedback control law by considering only a step disturbance: $d_i(k) = d_0 \cdot 1(k)$.

A. The control law

Our goal is to design a control law that drives the queuing delay τ_i experienced by each frame going through the i^{th} queue to a desired target value τ_i^T that represents the QoS requirement of the AC associated to the queue.

In particular, we consider the following control law:

$$u_i(k+1) = -k_i \cdot q_i(k) \quad (4)$$

which gives the way to compute the $\mathcal{Z}$ -transform of $q_i(k)$ and $u_i(k)$ as follows:

$$Q_i(z) = \frac{z \cdot T_{CA}}{z^2 - z + k_i \cdot T_{CA}} \cdot D_i(z) \quad (5)$$

$$U_i(z) = -\frac{k_i \cdot T_{CA}}{z^2 - z + k_i \cdot T_{CA}} \cdot D_i(z) \quad (6)$$

whit $D_i(z) = \mathcal{Z}[d_i(k)]$. From eq. (5) and ((6)) the system poles are $z_p = \frac{1 \pm \sqrt{1 - 4k_i \cdot T_{CA}}}{2}$, which give an asymptotically stable system if and only if $|z_p| < 1$, that is:

$$0 < k_i < \frac{1}{T_{CA}}. \quad (7)$$

In the sequel, we will always assume that k_i satisfies this asymptotic stability condition stated by (7).

To investigate the ability of the control system to provide a queuing delays approaching the target value τ_i^T , we apply the final value theorem to Eqs. (5) and (6). By considering that the $\mathcal{Z}$ -transform of the step function $d_i(k) = d_0 \cdot 1(k)$ is $D_i(z) = d_0 \cdot \frac{z}{z-1}$, the following results turn out:

$$u_i(+\infty) = \lim_{k \rightarrow +\infty} u_i(k) = \lim_{z \rightarrow 1} (z-1)U_i(z) = -d_0;$$

$$q_i(+\infty) = \frac{d_0}{k_i},$$

which implies that the the steady state queuing delay is:

$$\tau_i(+\infty) = \left| \frac{q_i(+\infty)}{u_i(+\infty)} \right| = \frac{1}{k_i}. \quad (8)$$

As a consequence, the following inequality has to be satisfied in order to achieve a steady-state delay smaller than τ_i^T :

$$k_i \geq \frac{1}{\tau_i^T}. \quad (9)$$

By considering inequalities (7) and (9) we obtain that the T_{CA} parameter has to fulfill the following constraint:

$$T_{CA} < \min_{i=1..M} \tau_i^T. \quad (10)$$

In order to transform the bandwidth u_i into a $TXOP_i$ assignment we have to consider that if the i^{th} queue is drained at data rate C_i , the following relation holds:

$$TXOP_i(k) = \frac{|u_i(k) \cdot T_{CA}|}{C_i} + O \quad (11)$$

where $TXOP_i(k)$ is the TXOP assigned to the i^{th} queue during the k^{th} service interval and O is the time overhead due to ACK packets, and interframe space time intervals (see Fig. 1). The extra quota of TXOP due to the overhead O depends on the number of MSDUs corresponding to the amount of data $|u_i(k) \cdot T_{CA}|$ to be transmitted. O could be estimated by assuming that all MSDUs have the same nominal size specified into the TSPEC. Moreover, when $|u_i(k) \cdot T_{CA}|$ does not correspond to a multiple of MSDUs, the TXOP assignment will be rounded in excess in order guarantee a queuing delay always equal or smaller than the target delay τ_i^T .

B. Channel saturation

In order to avoid channel saturations, we will adopt a CAC scheme, which will be described in the next Section and has been obtained by improving the one proposed by the 802.11e working group. However, since transient network overloads cannot be avoided due to the burstiness of the multimedia flows, it is necessary to reallocate the TXOPs to avoid exceeding the *dot11CAPLimit*. This task is performed as follows: when $\sum_{i=1}^M TXOP_i(k_0) > \text{dot11CAPLimit}$, each computed $TXOP_i(k_0)$ is decreased by an amount $\Delta TXOP_i(k_0)$, so that the capacity constraint $\sum_{i=1}^M [TXOP_i(k_0) - \Delta TXOP_i(k_0)] = \text{dot11CAPLimit}$ is satisfied. In particular, the generic amount $\Delta TXOP_i(k_0)$ is evaluated as a fraction of the total amount $\Delta = \sum_{i=1}^M TXOP_i(k_0) - \text{dot11CAPLimit}$, as follows:

$$\Delta TXOP_i(k_0) = \frac{TXOP_i(k_0)C_i}{\sum_{i=1}^M [TXOP_i(k_0)C_i]} \Delta. \quad (12)$$

Notice that Eq. (12) provides a $\Delta TXOP_i(k_0)$, which is proportional to $TXOP_i(k_0)C_i$, in this way connections transmitting at low rates are not penalized too much.

C. Call Admission Control

When the number of multimedia flows sharing the WLAN increases and the channel saturates and delay bounds cannot be guaranteed [12]. Under these conditions, a Call Admission Control scheme is required to guarantee QoS. Herein we describe the proposed CAC scheme, which exploits the 802.11e CAC proposal (see Sec. II).

In particular, starting from the TXOPs allocated to the active traffic streams in each CAP, a new flow request is admitted if the following inequality is satisfied:

$$\frac{TXOP_{k+1}}{T_{CA}} + \sum_{i=1}^k \frac{TXOP_i}{T_{CA}} \leq \frac{T - T_{CP}}{T} \quad (13)$$

where T indicates the superframe duration and T_{CP} is the time used for EDCA traffic during the superframe.

Notice that the proposed CAC test (13) has been obtained from the standard CAC test (1), by replacing the constant TXOPs used by the simple scheduler with the time-varying ones assigned by our

bandwidth allocation algorithm. In this way, the CAC test takes into account the bandwidth actually used by the flows and not just the sum of the average source rates declared in the TSPECs.

IV. PERFORMANCE EVALUATION

In order to test the proposed bandwidth allocation algorithm in realistic scenarios, computer simulations involving audio, video and FTP data transfers have been run using the *ns-2* simulator [13]. We have considered a reference scenario where a 802.11a WLAN network is shared by a mix of α video flows encoded with the MPEG-4 standard [14], α video flows encoded with the H.263 standard [15], 3α audio flows encoded with the G.729 standard [16], and α FTP best effort flows. For the video flows, we have used traffic traces available from the video trace library [17]. For audio flows, we have modeled G.729 sources using Markov ON/OFF sources, where the ON period is exponentially distributed with mean value 3 s, and the OFF period has a truncated exponential pdf with an upper limit of 6.9 s and an average value of 3s [18]. During the ON period, the source sends 20 Bytes sized packets every 20 ms (i.e., the source data rate is 8 kbps) however, by considering the overheads of the RTP/UDP/IP protocol stacks, the rate becomes 24 kbps. During the OFF period the rate is zero because we assume a Voice Activity Detector (VAD). We have used the HCCA access method during the CFP, and the EDCA access method during the CP. EDCA parameters have been set as suggested in [4].

In the *ns-2* implementation, the T_{CA} is expressed in Time Unit (TU), which in the 802.11 standard [1] is equal to $1024\mu s$. We assume a T_{CA} of 29 TU. The proportional gain k_i is set equal to $1/\tau_i^T$. Main characteristics of the considered multimedia flows are summarized in Table 1. Before starting data transmission, a

Type of flow	Nominal (Maximum) MSDU Size [Byte]	Mean (Maximum) Data Rate [kbps]	Target Delay [ms]	Inactivity Interval [s]
MPEG-4	1536(2304)	770 (3300)	40	3
H.263 VBR	1536(2304)	450 (3400)	40	3
G.729 VAD	60(60)	8.4 (24)	30	10

Tab. 1. Main features of the considered multimedia flows.

multimedia source has to set up a new Traffic Stream as specified in Sec. II-B. For that purpose, it sends an *ADDTS request* and waits for an *ADDTS confirm*. If this reply is not received within a Δ_{TO} timeout interval, the admission request is repeated up to a maximum number of times, N_{Adm} (we have chosen $N_{Adm} = 10$ and $\Delta_{TO} = 1.5s$). If after the N_{Adm}^{th} request no reply is received back, then the *ADDTS request* is considered lost and a new admission procedure is initiated after an exponentially distributed random time Δ_{defer} with average value equal to 1 min.

The duration of video flows is deterministic and equal to 10 min, whereas audio flows durations are exponentially distributed with average value of 2 min. After that a multimedia flows terminates, a new one is generated after an exponentially distributed random time, with average value 1 min. Each terminated flow is withdrawn from the polling list by the HC after that no more packets from that flow are received for an interval of time equal to the Inactivity Interval reported in Table 1. Each simulation lasts 1 hour. In the sequel, we report some significant comparisons of the proposed dynamic bandwidth allocation and CAC algorithms with respect to the simple scheduler and the CAC scheme described in the 802.11e draft [4]. We have tested the proposed algorithm for several values of α , herein we report simulation results obtained for $\alpha = 5$, $\alpha = 10$, $\alpha = 12$, $\alpha = 15$.

Fig. 2 shows the ratios of admitted flows when the simple scheduler or the FBDS algorithm is used. When the simple scheduler

is used, a higher ratio of flows is admitted with respect to the case when the FBDS is employed. The reason is that the simple scheduler allocates TXOPs by taking into account only the average data rate declared into the TSPECs, in this way the allocated bandwidth can be very smaller than the actual load and a very high number of flow is admitted. On the other side, when FBDS is used, the allocated bandwidth tracks the actual load being the allocation algorithm dynamic, and, as a consequence, a smaller number of TS is admitted. Figs. 3, 4, and 5 show the average

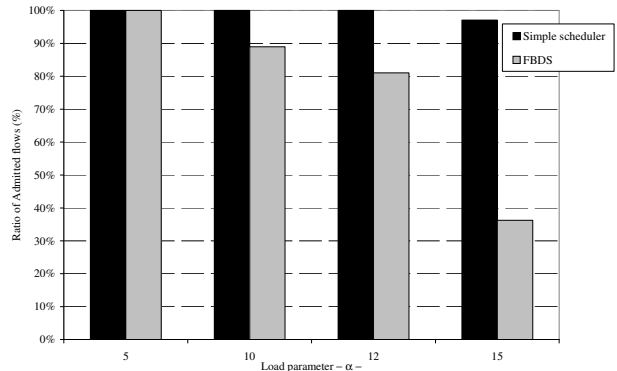


Fig. 2. Ratio of admitted Traffic Streams as a function of the load parameter α .

one-way packet delay of the MPEG4, H263 and G729 flows as a function of the load parameter α . It is straightforward to note that, when the simple scheduler is used, the average delays critically increase with α . The reason is that the CAC is not able to protect the network from overloads and, as a consequence, the transmission queues and the queuing delays build up with increasing α . On the other side, when FBDS is used, delays are kept bounded as expected. Finally Figs. 6 and 7 show the average and the peak

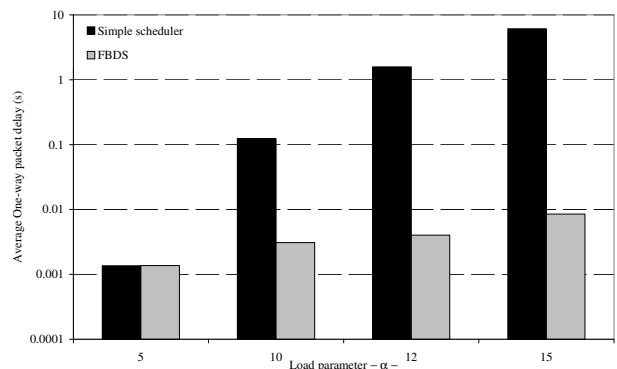


Fig. 3. Average one-way packet delay of the MPEG4 flows as a function of the load parameter α .

superframe utilization, respectively, as a function of α . They show that the simple scheduler provides higher(smaller) average(peak) superframe utilizations with respect to FBDS. The reason is that the simple scheduler is a static bandwidth allocation algorithm, which allocates always the same amount of bandwidth to the flows, also when this is not due. On the other hand, FBDS tracks the bandwidth requirements of the traffic streams by giving them the bandwidth they need during each superframe.

V. CONCLUSIONS

In this paper an innovative dynamic bandwidth allocation algorithm associated with a measurement based CAC algorithm for providing delay guarantees to real-time media flows in IEEE 802.11e networks has been proposed. The performance of the

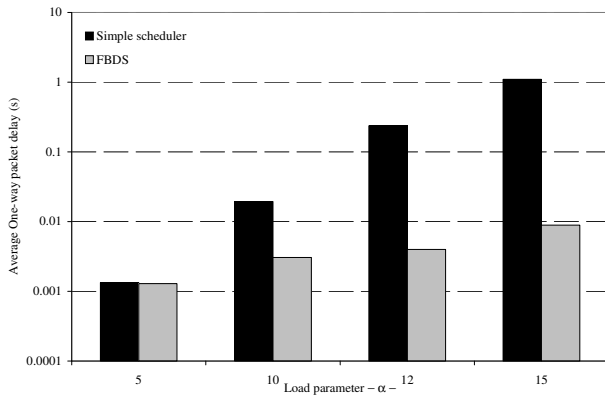


Fig. 4. Average one-way packet delay of the H.263 flows as a function of the load parameter α .

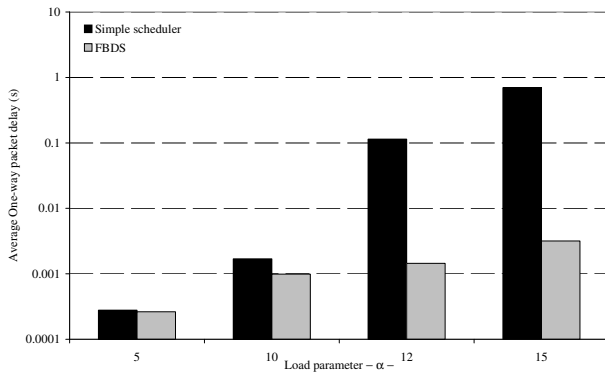


Fig. 5. Average one-way packet delay of the G.729 flows as a function of the load parameter α .

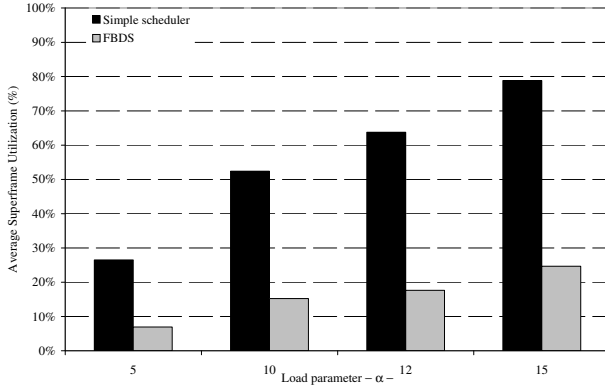


Fig. 6. Average Superframe utilization as a function of the load parameter α .

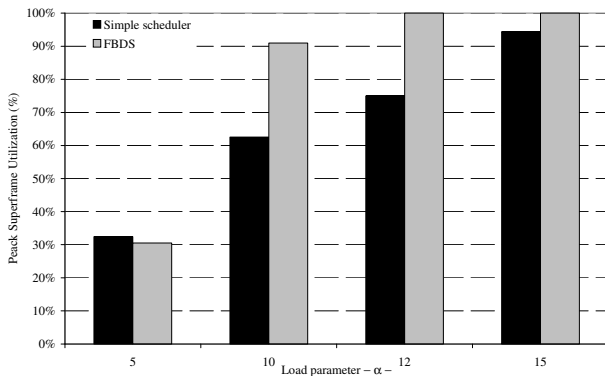


Fig. 7. Peak Superframe utilization as a function of the load parameter α .

proposed schemes have been compared with the ones obtained using the standard algorithm proposed in the 802.11e draft using the *ns-2* simulator. Simulation results have shown that the proposed algorithms are able to provide bounded delays to multimedia flows also in high traffic load conditions.

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