Performance Evaluation of a Feedback Based Dynamic Scheduler for 802.11e MAC*

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Abstract. IEEE 802.11 Wireless Local Area Networks (WLANs) are a fundamental tool for enabling ubiquitous wireless networking. Their use has been essentially focused on best effort data transfer because the basic access methods defined in the 802.11 standard cannot provide delay guarantees to real-time flows. To overcome this limitation, the 802.11e working group has recently proposed the Hybrid Coordination Function (HCF), which is an enhanced access method that allows service differentiation within WLAN's. While the 802.11e proposal gives a flexible framework to address the issue of service differentiation in WLAN's, it does not specifies effective algorithms to really achieve it. This paper compares the performance of standard access methods proposed by the 802.11e working group with a novel Feedback Based Dynamic Scheduler (FBDS), which is compliant to 802.11e specifications. Simulation results, obtained using the *ns*-2 simulator, have shown that FBDS guarantees bounded delays to real-time flows for a very broad set of network loads and packet loss probabilities, whereas, analogous algorithms proposed by the 802.11e working group fail in presence of high network load.

1 Introduction

The continuous growth of the Wireless LAN (WLAN) market is essentially due to the leading standard IEEE 802.11 [1], which is characterized by easy installation, flexibility and robustness against failures. At present, IEEE 802.11 has been essentially focused on best effort data transfer because its radio access methods cannot provide delay guarantees to real-time multimedia flows [2–5]. In particular, 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) and an optional centrally controlled channel access scheme called Point Coordination Function (PCF) [1, 6]. 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.

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Recently, in order to support also delay-sensitive multimedia applications, such as real-time voice and video, the 802.11e working group has enhanced the 802.11 MAC with improved functionalities. Four Access Categories (ACs), with different priorities, have been introduced and an enhanced access function, which is responsible for service differentiation among different ACs and is referred to as Hybrid Coordination Function (HCF), has been proposed [7]. 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. Similarly to the PCF, the time is partitioned in a infinite series of superframes. Stations operating under 802.11e specifications are usually known as enhanced stations or QoS Stations (QSTAs).

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 [8].

EDCA parameters have to be properly set to provide prioritization of ACs. Tuning them in order to meet specific QoS needs is a current research topic. A method to set these parameters to guarantee throughput of each TC has been described in [9]. Regarding the goal of providing delay guarantees, several papers have pointed out that the EDCA behavior is very sensitive to the value of the contention parameters [10, 11] and that the Interframe-space-based priority scheme used by the EDCA mechanism can provide only a relative differentiation among service classes, but not absolute guarantees on throughput/delay performance [8]. Finally, in the paper [12] it has been shown that that EDCA can starve lower priority flows. To overcome these limitations, adaptive algorithms that dynamically tune EDCA parameters have been recently proposed in [13, 14], however, the effectiveness of these heuristic schemes have been proved only using simulations and no theoretical bounds on their performance in a general scenario have been derived.

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. 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. 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 [7].

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

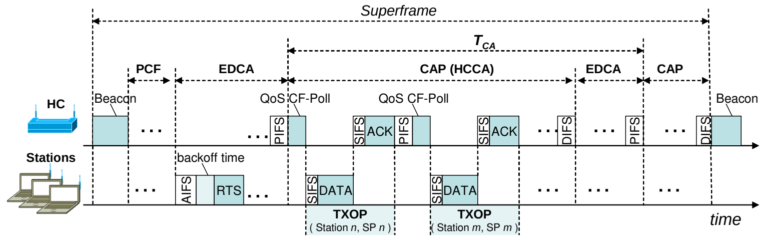

Fig. 1. Scheme of a superframe using the HCF controlled access method.

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 [15, 16], 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 [7] and [17]). In particular, the simple scheduler designed in the draft [7] states that the $TXOP_i$ assigned to the i^{th} queue should be computed as follows:

$$TXOP_i = max\left(\frac{N_i \cdot L_i}{C_i} + O, \frac{M}{C_i} + O\right) \tag{1}$$

where L_i is the nominal MAC Service Data Unit (MSDU) size associated with the i^{th} queue; C_i is the physical data rate at which the data of the i^{th} queue are transmitted over the WLAN; O is the overhead due to ACK packets, SIFS and PIFS time intervals; M is the maximum MSDU size; and $N_i = \left\lceil \frac{T_{SI} \cdot \rho_i}{L_i} \right\rceil$, where ρ_i is the Mean Data Rate associated with the i^{th} queue and T_{SI} is the Service Interval. This scheduler does not exploit any feedback information from mobile stations in order to dynamically assign TXOPs. Thus, it is not well suited for bursty media flows [17]. An improved bandwidth allocation algorithm has been proposed in [18], 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 the multimedia flows. An adaptive version of the simple scheduler, which is based on the Delay-Earliest Due-Date algorithm, has been proposed in [17]. 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.

Recently, a Feedback Based Dynamic Scheduler (FBDS) has been proposed in [19] in order to address the first-hop bandwidth allocation issue in WLAN's. It is based on a closed-loop control scheme based on feeding back queue levels [20].

Preliminary investigations reported in [19, 21] have shown that FBDS is able to provided bounded delays to real-time multimedia flows over an ideal loss-free WLAN channel. The present paper will evaluate the performance of FBDS also in the presence of a noisy WLAN channel affected by bursty frame losses. The rest of the paper is organized as follows: Section 2 proposes and analyzes the dynamic bandwidth allocation algorithm, and illustrates the proposed CAC scheme; Section 3 shows simulation results; finally, the last Section draws the conclusions.

2 FBDS: Theoretical Background

FBDS distributes the WLAN bandwidth among all the multimedia flows by taking into account the queue levels fed back by the QSTAs [19]. Bandwidth allocation is pursued by exploiting the HCCA functionalities, which allows the HC to assign TXOPs to the ACs by taking into account the specific time constraints of each AC.

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. In 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 queue levels $q_i,\ i = 1, \ldots, M$ at the beginning of the previous CAP, where M is the total number of traffic queues in the WLAN system. The latter is a worst case assumption, in fact, queue levels are fed back using frame headers, as a consequence, if the i^{th} queue length has been fed at the beginning of the previous CAP, then the feedback signal might be delayed up to T_{CA} seconds.

The dynamics of the i^{th} queue can be described by the following discrete time linear model:

$$q_i(k+1) = q_i(k) + d_i(k) \cdot T_{CA} + u_i(k) \cdot T_{CA}, \qquad i = 1, \ldots, M, \tag{2}$$

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); $d_i(k) = d_i^s(k) + d_i^{RTX}(k) - d_i^{CP}(k)$ where $d_i^s(k) \geq 0$ is the average input rate at the i^{th} queue during the k^{th} CAP, $d_i^{RTX}(k) \geq 0$ is the amount of data retransmitted by the i^{th} queue during the k^{th} CAP divided by T_{CA}, and $d_i^{CP}(k) \geq 0$ is the amount of data transmitted by the i^{th} queue during the k^{th} CP 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 piece-wise constant model for the disturbance $d_i(k)$ can be assumed [15]:

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

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, the feedback control law can be designed by considering only a step disturbance: $d_i(k) = d_0 \cdot 1(k)$.

2.1 The control law

The goal of FBDS is to drive 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:

$$u_i(k+1) = -k_i \cdot q_i(k) \tag{4}$$

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

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

whit $D_i(z) = \mathcal{Z}[d_i(k)]$. From eq. (5)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:

$$0 < k_i < \frac{1}{T_{CA}}. \tag{6}$$

In the sequel, we will always assume that k_i satisfies this asymptotic stability condition stated by (6).

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 Eq. (5). 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:

$$u_i(+\infty) = \lim_{k \to +\infty} u_i(k) = \lim_{z \to 1} (z-1) U_i(z) = -d_0; \qquad q_i(+\infty) = \frac{d_0}{k_i},$$

which implies that the the steady state queueing delay is:

$$\tau_i(+\infty) = \left| \frac{q_i(+\infty)}{u_i(+\infty)} \right| = \frac{1}{k_i}. \tag{7}$$

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

$$k_i \geq \frac{1}{\tau_i^T}. \tag{8}$$

By considering inequalities (6) and (8) we obtain that the T_{CA} parameter has to fulfill the following constraint:

$$T_{CA} < \min_{i=1..M} \tau_i^T. \tag{9}$$

2.2 TXOP assignment

We have seen in Sec. 1 that every time interval T_{CA} the HC allocates TXOPs to mobile stations in order to meet the QoS constraints. This sub-section shows how to transform the bandwidth $|u_i(k)|$, defined above, into a $TXOP_i$ assignment. In particular, if the i^{th} queue is drained at data rate C_i, the following relation holds:

$$TXOP_i(k) = \frac{|u_i(k) \cdot T_{CA}|}{C_i} + O \tag{10}$$

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, SIFS and PIFS 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.

2.3 Channel saturation

The above bandwidth allocation algorithm is based on the implicit assumption that the sum of the TXOPs assigned to each queue is smaller than the maximum CAP duration, which is the *dot11CAPLimit*; this value can be violated when the network is saturated. When a network overload happens, it is necessary to reallocate the TXOPs to avoid exceeding the CAP limit. Thus, when $\sum_{i=1}^{M} TXOP_i(k_0) > dot11CAPLimit$, each computed $TXOP_i(k_0)$ is decreased by an amount $\Delta TXOP_i(k_0)$ to satisfy the following capacity constraints:

$$\sum_{i=1}^{M} [TXOP_i(k_0) - \Delta TXOP_i(k_0)] = dot11CAPLimit. \tag{11}$$

In particular, the generic amount $\Delta TXOP_i(k_0)$ is evaluated as a fraction of the total amount $\sum_{i=1}^{M} TXOP_i(k_0) - dot11CAPLimit$, as follows:

$$\Delta TXOP_i(k_0) = \frac{TXOP_i(k_0)C_i}{\sum_{i=1}^{M} TXOP_i(k_0)C_i} \left(\sum_{i=1}^{M} TXOP_i(k_0) - dot11CAPLimit \right). \tag{12}$$

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.

3 Performance Evaluation

In order to test FBDS in realistic scenarios, computer simulations involving voice, video and FTP data transfers have been run using the *ns*-2 simulator [22]. We have considered the reference scenario shown in Fig. 2 where a 802.11a WLAN network is shared by a mix of 3α voice flows encoded with the G.729 standard [23], α video flows encoded with the MPEG-4 standard [24], α videos encoded with the H.263 standard [25], and α FTP best effort flows. For the video flows, we have used traffic traces available from the video trace library [26]. For voice flows, we have modelled G.729 sources using Markov ON/OFF sources [27], where the ON period is exponentially distributed with mean value 350 ms, and the OFF period is exponentially distributed with mean value 350 ms [28]. During the ON period, the source sends 20Bytes sized packets every 20ms (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 ON periods. 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 in [7].

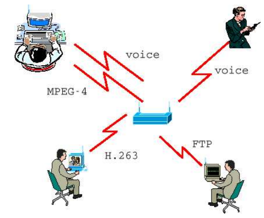

Fig. 2. Scenario with multimedia flows.

In the *ns*-2 implementation, the T_{CA} is expressed in Time Unit (TU), which in the 802.11 standard [1] is equal to 1024μs. 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.

To take into account also the effect of random bursty losses, we have considered a Gilbert two state discrete Markov chain to model the loss process affecting

Table 1. Main features of the considered multimedia flows.

Type of flow	*Nominal (Maximum) MSDU Size*	*Mean (Maximum) Data Rate*	*Target Delay*
MPEG-4 HQ	1536(2304) byte	770 (3300) kbps	40 ms
H.263 VBR	1536(2304) byte	450 (3400) kbps	40 ms
G.729 VAD	60(60) byte	8.4 (24) kbps	30 ms

the WLAN channel [29]. In particular, we assume a frame loss probability equal to 0, when the channel is in the Good state, and ranging from 0 to 0.1 when the channel is in the Bad state. The permanence times in the Good and Bad states are geometrically distributed with mean values equal to 0.1 s and 0.01 s, respectively. The transition probabilities between the two states have been set equal to 0.1. When a MAC frame gets lost due to interference or channel unreliability, it is retransmitted up to a maximum number of times equal to 7.

We have evaluated the performance of FBDS, Simple scheduler, EDCA, and DCF for a very broad set of values for the load parameter α. Herein we report significant comparisons obtained for $\alpha = 5$ ($\alpha = 10$), which represents the case of a low(high) loaded WLAN. Fig. 3 shows the CDF of the one-way-packet delay experienced by the MPEG flows for all the considered medium access schemes when $\alpha = 5$. In particular, it shows that all the considered schemes are able to ensure a bounded delay, which is less than 30ms for the 90% of the packets. This result is due to the low load offered to the WLAN, which is able to drain all the transmittng queues also with the basic DCF access function. For the same reason, the impact of the loss rate on the one-way packet delay is negligible.

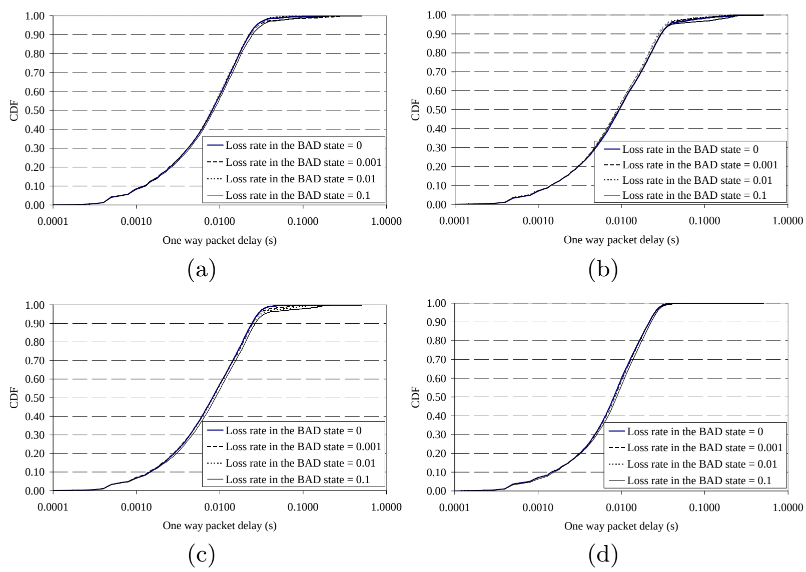

Fig. 3. CDF's of the one way packet delay experienced by the MPEG4 flows when $\alpha = 5$, obtained using the (a) DCF; (b) EDCA; (c) Simple scheduler; (d) FBDS.

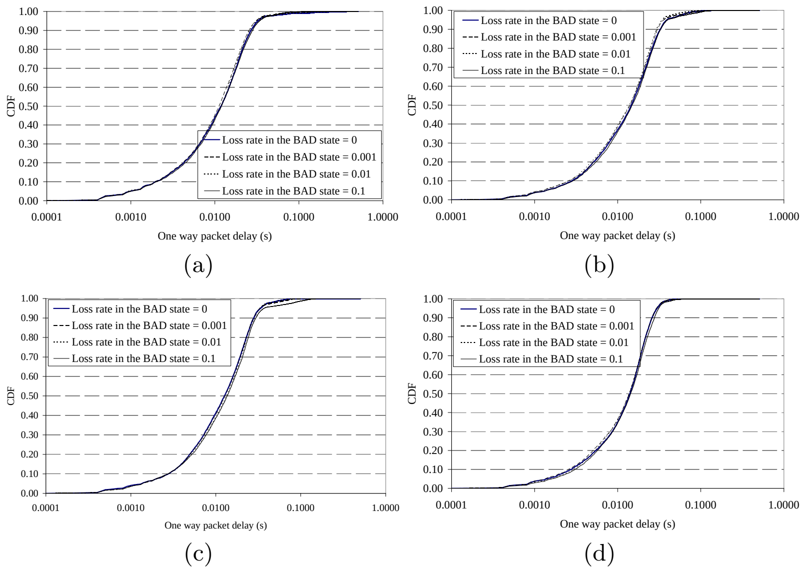

Fig. 4. CDF's of the one way packet delay experienced by the H.263 flows when $\alpha = 5$, obtained using the (a) DCF; (b) EDCA; (c) Simple scheduler; (d) FBDS.

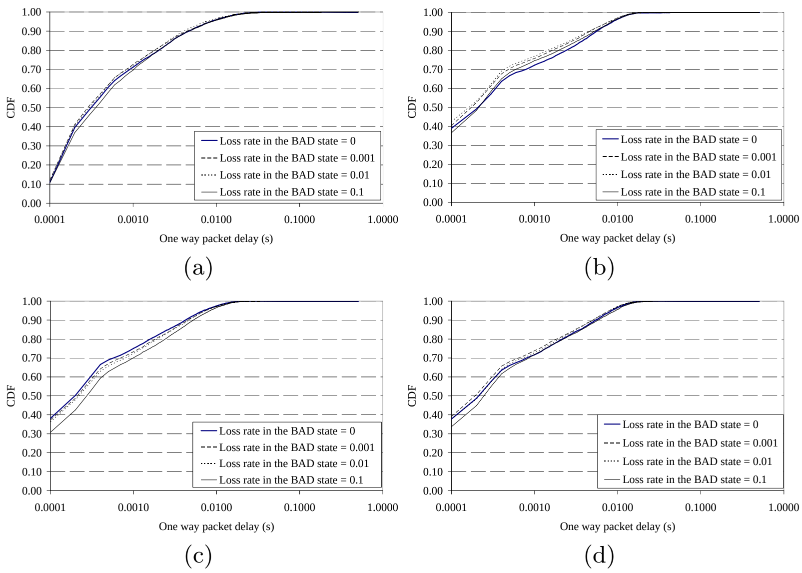

Fig. 5. CDF's of the one way packet delay experienced by the G.729 flows when $\alpha = 5$, obtained using the (a) DCF; (b) EDCA access function; (c) Simple scheduler; (d) FBDS.

Similar results have been obtained for the H.263 flows (see Fig. 4). Results for the G.729 flows are reported in Fig. 5, which shows that the one-way packet delay is less than 6 ms for the 90% of the packets. In this case, we observe delays smaller than those obtained for the video flows because G.729 packets are smaller than those generated by the video streams.

Results are very different when $\alpha = 10$ due to the higher load. In this case, only the FBDS scheduler ensures bounded delays for all kinds of flow. In fact, Fig. 6 shows that, when DCF or ECDA or the Simple scheduler are used, the MPEG flows experience delays larger than hundreds of milliseconds for more than a half of the received packets. On the other hand, FBDS provides bounded delays. It is worth to note that delays provided by FBDS increase with the loss rate because a fraction of the channel capacity is used for retransmissions at the MAC layer. Similar results have been obtained for the H.263 flows as shown in Fig. 7. Delays experienced by the G.729 flows exhibit a different behavior with respect to the video flows (see Fig. 8). The reason is that when the EDCA access method is used, a higher priority is given to the voice flows [7]. This effect was not evident in the previous simulations with $\alpha = 5$ because the network load was small, so that also flows with a lower priority got enough bandwidth to drain their respective transmission queues. It is worth to point out that with EDCA the performance improvement for the G.729 flows is obtained at the expense of the video flows; on the other hand, with FBDS the WLAN bandwidth is properly distributed among all streams, thus obtaining bounded delays for all kind of flows.

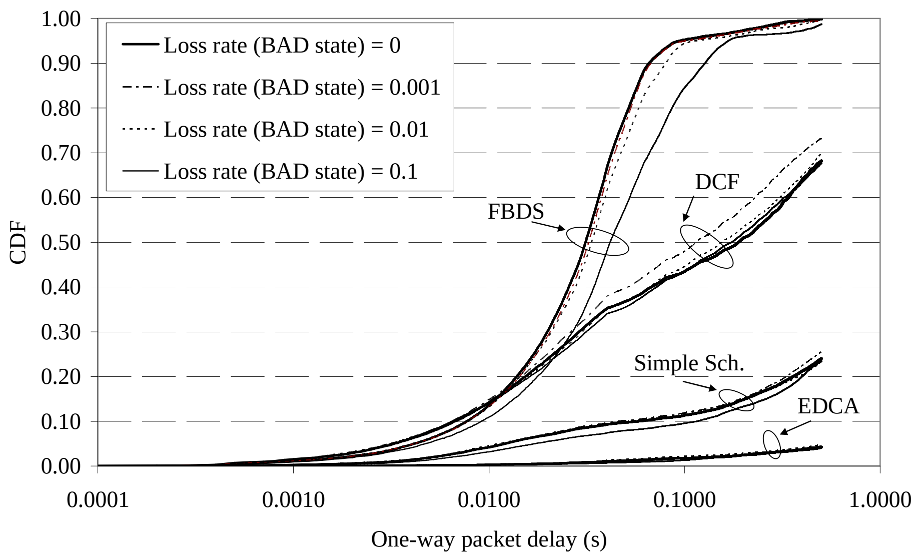

Fig. 6. CDF's of the one way packet delay experienced by the MPEG4 flows when $\alpha = 10$, obtained using the DCF, EDCA, Simple scheduler, and FBDS.

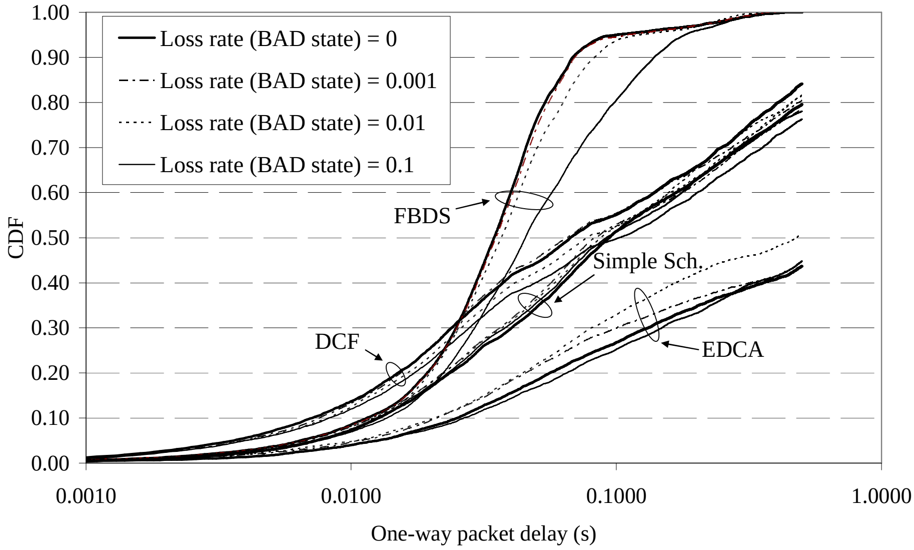

Fig. 7. CDF's of the one way packet delay experienced by the H.263 flows when $\alpha = 10$, obtained using the DCF, EDCA, Simple scheduler, and FBDS.

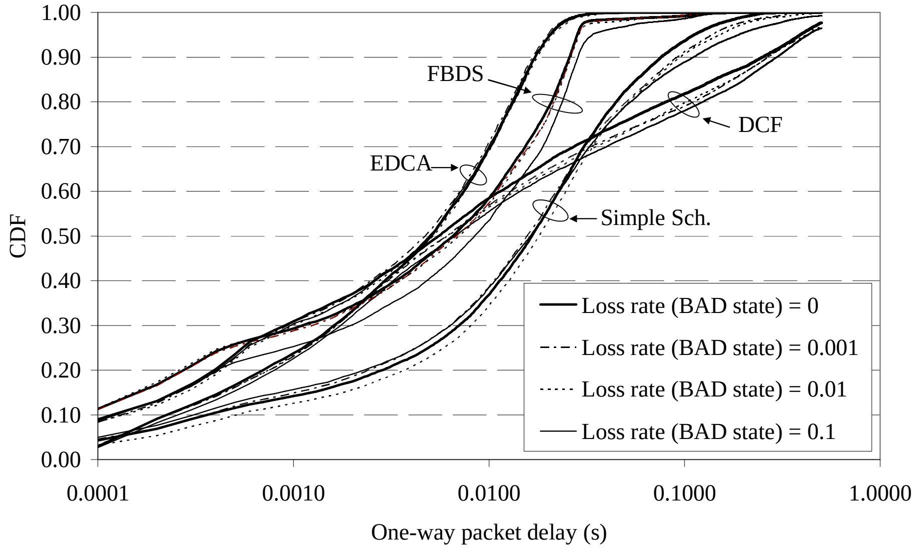

Fig. 8. CDF's of the one way packet delay experienced by the G.729 flows when $\alpha = 10$, obtained using the DCF, EDCA, Simple scheduler, and FBDS.

4 Conclusion

In this paper, the performance of a novel Feedback Based Dynamic Scheduler have been evaluated in comparison to the simple scheduler proposed by the IEEE 802.11e working group, the EDCA and the DCF access methods, in a wide range of traffic load conditions and frame loss probabilities. Results have shown that the FBDS allows a more cautious usage of the WLAN bandwidth with respect to the others schemes, giving better results also in the presence of high network load.

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