QOS ENHANCEMENT WITH DYNAMIC TXOP ALLOCATION IN IEEE 802.11E

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ABSTRACT
The transmission opportunity (TXOP) mechanism defined in the IEEE 802.11e Hybrid Coordination Function (HCF) is not optimized to meet the QoS requirements of heterogeneous applications, since it is usually allocated by default according to the different Access Categories (AC).

In this paper we propose a new algorithm, named DTXOP, for the dynamic assignment of the TXOP maximum duration at the Access Point (AP) of an IEEE 802.11e WLAN. DTXOP is periodically updated according to the current traffic conditions of each specific AC. Simulation experiments show that DTXOP allows to enhance delay and throughput performance and to maintain fairness between upstream and downstream channel access times.

I. INTRODUCTION

The development of high rate wireless networks has favoured nomadic access to data communications anytime and anywhere and with different type of terminals, ranging from common laptops to PDAs and smart phones. Nomadic access is often provided through hotspots or university campuses and a very popular solution is the adoption of IEEE 802.11 standard [1]. Most terminals are now capable of multimedia communications not limited to email or web browsing, but extended to VoIP and video applications. Such applications impose stringent QoS requirements which cannot be met by the 802.11 standard itself and this is the reason why the 802.11e Task Group has recently defined an emerging standard for QoS support to multimedia traffics [2].

With respect to the DCF and PCF functions of the legacy 802.11 technology, the 802.11e standard introduces some extensions of the medium access control protocol for efficient bandwidth sharing, namely, the Hybrid Coordination Function (HCF). HCF envisages two access schemes: the Enhanced Distributed Channel Access (EDCA) and the HCF Controlled Channel Access (HCCA).

Several studies [3], [4], [5] have shown the QoS enhancements provided by the adoption of EDCA, since this is the function supported by all 802.11e devices, while HCCA is rarely implemented [6], just as it happens with DCF and PCF respectively. Recent experiments [7], [8], [9], have also shown the significant impact of EDCA parameters on QoS performance.

An important parameter is the “transmission opportunity” (TXOP), allowing stations to transmit multiple packets on a single channel access until the expiration of a maximum TXOP interval. A suitable allocation of TXOP duration results in efficient bandwidth sharing and positive influence on delay and throughput of multimedia traffics.

In this paper we evaluate how QoS of multimedia applications can be enhanced through a dynamic tuning of the TXOP EDCA parameter. We propose a dynamic algorithm, named DTXOP, for adjusting TXOP according to the current traffic conditions, in order to privilege real-time applications with respect to best-effort and background traffics.

The paper is organised as follows: section II provides a general description of EDCA; in section III we define our DTXOP allocation scheme; sections IV is dedicated to simulations and results, before coming to the final remarks in section V.

II. IEEE 802.11e EDCA SCHEME

With respect to the DCF of legacy IEEE802.11b MAC, the 802.11 Task Group E has defined an enhanced MAC scheme for wireless QoS deployment, namely the Hybrid Coordination Function (HCF). HCF incorporates two access modes: the Contention Period (CP), called EDCA, and the Contention Free Period (CFP) called HCCA [2], [10].

Our work focuses on the EDCA scheme, according to which differentiated channel access probabilities are provided to frames contending for the channel resources. Depending on traffic QoS requirements, packets are assigned to suitable Access Categories (AC) implemented in different queues at the QoS enabled stations (QSTA). AC-based prioritization is realized through independent backoff entities. Specifically, each AC is characterized by different contention parameters regulating channel access and frame transmission timings. Such parameters are:

- \( AIFS_D \) (Arbitration Inter Frame Space Duration) - it is the minimum interval of time after which a station detecting the idle channel can start its backoff timer, thus entering into its contention window. It is greater or equal to DIFS of the legacy MAC; smaller \( AIFS_D \) values correspond to higher priority frames.

\[
AIFS_D[AC] = SIFS + AIFS_N[AC] \times SlotTime
\] (1)

where \( AIFS_N \) is an integer greater than zero and \( SlotTime \) is fixed to 20 µs for the 802.11b PHY.

- \( CW_{\text{min}} \) and \( CW_{\text{max}} \) (Contention Window parameters) - they represent the minimum and maximum values controlling the size of the random backoff; first it is initialized at \( CW_{\text{min}} \), but after each unsuccessful channel access, the random backoff window size is doubled up, with an upper bound of \( CW_{\text{max}} \).

- \( TXOP \) (transmission opportunity) - it is the time a station has the right to transmit a burst of data frames; it is defined by a starting time and a maximum duration, and it is assigned by the AP on the basis of traffic classification and requirements.

Instead of \( DIFS, CW_{\text{min}} \) and \( CW_{\text{max}} \) of the legacy DCF, in EDCA each AC uses its own \( AIFS_D[AC], CW_{\text{min}}[AC] \) and \( CW_{\text{max}}[AC] \) respectively. Smaller values of these parameters
provide higher priority. EDCA values are announced periodically by the AP via the beacon frame.

According to this mechanism, a QSTA implements internally several virtual stations, one for each AC. Channel access is thus based on the same principles of DCF. If an AC detects the channel idle for its AIFS, it starts its backoff counter. When the counter reaches zero, transmission can start. An internal scheduler resolves virtual collisions inside the QSTA, by choosing the highest priority frame for transmission. In such a case, the other colliding frames perform a backoff, doubling up their contention window size.

The maximum number of permitted ACs is fixed to \( n = 4 \) transmission queues for managing background, best-effort, video and voice traffics.

### A. TXOP Parameters

TXOP is an important parameter in HCF, since it allows a station to transmit a burst of frames back-to-back without re-entering the contention phase for the channel (figure 1).

![Figure 1: Bursty transmission within a TXOP limit.](image)

A key parameter regulating the maximum duration of the transmission opportunity is \( \text{TXOPlimit} \). Its value is delivered by the AP to the wireless stations via the beacon frame, together with the other EDCA parameters (i.e., \( CW_{\text{min}}, CW_{\text{max}}, AIFS \)) for each AC [11] and its value is expressed in units of 32 \( \mu \text{s} \).

Usually the EDCA parameters for the PHY in use are set by default according to table 1.

**Table 1: Default EDCA parameters setting.**

<table>
<thead>
<tr>
<th>Traffic Type</th>
<th>AC</th>
<th>AIFS ( \mu \text{s} )</th>
<th>( CW_{\text{min}} )</th>
<th>( CW_{\text{max}} )</th>
<th>( \text{TXOP limit} )</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>0</td>
<td>2</td>
<td>7</td>
<td>15</td>
<td>3,264 ms</td>
</tr>
<tr>
<td>Video</td>
<td>1</td>
<td>2</td>
<td>15</td>
<td>31</td>
<td>6,016 ms</td>
</tr>
<tr>
<td>Best effort</td>
<td>2</td>
<td>3</td>
<td>31</td>
<td>1023</td>
<td>0</td>
</tr>
<tr>
<td>Background</td>
<td>3</td>
<td>7</td>
<td>31</td>
<td>1023</td>
<td>0</td>
</tr>
</tbody>
</table>

A \( \text{TXOPlimit} \) equal to zero indicates that the station can transmit only one frame, after which it has to enter a new contention phase.

### III. DYNAMIC TXOP SCHEME

Unfortunately, the adoption of a unique value of \( \text{TXOPlimit} \) for all stations, included the AP, results in an unfair behavior. In fact, with \( n \) wireless stations and for each AC there are \( n \) upstream entities contending the channel with just one downstream entity at the AP. This creates an unbalance between upstream and downstream throughputs, because the AP has to deliver to the wireless stations the whole traffic coming from the wired network section [3].

Consequently, different algorithms have been studied for providing the AP with a greater \( \text{TXOPlimit} \), in order to increase the throughput of downstream traffics, otherwise penalized by wireless stations contending the channel for upstream transmissions. Previous efforts [8], [9] assume that downstream traffic volumes are greater than upstream ones, and this is true in general, but it is not a rule. There are also situations in which the wireless stations need equal or even more resources than the AP, because upstream traffic is more demanding than downstream.

The study of \( \text{TXOPlimit} \) is important mainly in critical cases, when the network is loaded and is close to saturation; consequently a careful allocation of transmission resources inside the CP can enhance the WLAN performance in terms of throughput and delay of multimedia traffics.

To solve this problem we propose an algorithm capable of tuning the \( \text{TXOPlimit} \) value at the AP, in order to dynamically change its channel resources, according to the current traffic conditions. In contrast with previous approaches, our algorithm can also reduce the AP channel occupation time when upstream demand is greater than downstream demand. In such an asymmetric situation the \( \text{TXOPlimit} \) of the AP can be smaller than the \( \text{TXOPlimit} \) of the other stations.

Our algorithm counts the number of lost packets during the \( i \)-th observing interval that we have set equal to the beacon interval (i.e., 100ms). The term “lost packets” here refers to packets transmitted but not yet acknowledged (at the MAC level) during the observing time by the destination station; we assume that each packet corresponds to one frame (no fragmentation). We chose this unique indicator because it is the main symptom of transmission problems, regardless of the events that caused such problems. Moreover, it can be easily obtained from the standard itself and no substantial modification to the IEEE802.11e standard is required. However, several other indicators could be considered (e.g., average \( CW \), free channel time histograms, etc…), but the algorithm at the AP would be much more complex in comparison to the performance gain.

We define \( L(i) \) as:

\[
L(i) = \text{Lost_down}(i) - \text{Lost_up}(i)
\]

(2)

where \( \text{Lost_down}(i) \) and \( \text{Lost_up}(i) \) are the total number of lost packets in downstream and upstream direction respectively, calculated during the \( i \)-th interval (\( i=1, 2, ... \)) for a specific AC.

As regards the AC0 class, assigned to VoIP sessions which are intrinsically symmetric, we have:

\[
\text{TXOPlimitAP}(i) = N(i) \times \text{TXOPlimitQSTA}
\]

(3)

where \( \text{TXOPlimitAP}(i) \) represents the TXOP limit value at the AP during the \( i \)-th observing interval, \( \text{TXOPlimitQSTA} \) is the TXOP limit at the other QSTA and it is set equal to 3,264 ms by default for AC0 (see table 1), and \( N(i) \) represents the number of wireless stations involved in VoIP sessions during the \( i \)-th interval. According to this equation, if the BSS has \( N \) transmitting/receiving QSTAs, its AP receives half of the channel resources instead of \( 1/(N+1) \) that it would receive
without dynamic adjustment. This fact allows to balance the transmission of VoIP frames between upstream and downstream directions, as requested by the symmetric nature of VoIP sessions.

As regards the AC1 class, generally assigned to video flows which can be symmetric (e.g., videoconference) or asymmetric (e.g., streaming video), we have the following algorithm:

\[
\begin{align*}
\text{If} & \quad |L(i-1)| < \alpha \\
\text{TXOPlimitAP}(i) &= \text{TXOPlimitAP}(i-1) \times \frac{N(i)}{N(i-1)}
\end{align*}
\]

\[
\begin{align*}
\text{else} \\
\text{TXOPlimitAP}(i) &= \text{TXOPlimitAP}(i-1) + \\
&\quad [\text{TXOPlimitQSTA} \times L(i-1)/\alpha] \times \frac{N(i)}{N(i-1)} \\
&\quad \times \frac{N(i)}{N(i-1)}
\end{align*}
\]

(4)

where the algorithm is initialized as below:

\[
\text{TXOPlimitAP}(0) = N(0) \times \text{TXOPlimitQSTA}
\]

(5)

and \(\text{TXOPlimitQSTA}\) is set equal to 6,016 ms by default for AC1 (see table 1). The threshold \(\alpha\) is an integer and it is a tuning parameter that allows us to regulate the aggressiveness of the algorithm. The increase of \(\text{TXOPlimitAP}\) is slower when \(\alpha\) increases. Our experimental results have shown that \(\alpha=10\) is a good choice. Moreover, the factor \(N(i)/N(i-1)\) appearing in (4) takes into account that the number of contending entities of a given AC can dynamically change and it has been introduced in order to provide a prompt reaction to a new traffic load situation.

Through the proposed algorithm, the starting situation is characterized by fairness between AP (transmitting downstream) and the other stations (transmitting upstream); this situation can evolve during time and consequently \(\text{TXOPlimitAP}\) is periodically tuned according to the difference between lost frames in the two directions during the previous beacon interval. In fact, the increase of this difference is the symptom of a growing unbalance and hence more channel time is to be allocated to upstream or downstream flows of that AC.

The values of the variables needed for solving the above equations are all available without any modifications to the IEEE802.11e standard; they can be directly read inside the “dot11QosCounterTable” of the MIB standard [2].

IV. SIMULATIONS AND RESULTS

In our simulations we have used a scenario based on a single IEEE 802.11b BSS, with one AP and six wireless stations sharing the same channel. We assume that all the stations are at a short distance from the AP (maximum 20 m) and they transmit at the maximum bit rate of 11 Mb/s. The AP is then connected with a server through a 10BaseT switch (figure 2).

The BSS implements QoS support as defined in IEEE802.11e.

We have adopted ns2 [12] for performing simulations, with the additional module for IEEE802.11e QoS support developed by TKN [13]. In order to reflect a real WLAN implementation inside a building, we have used the parameters (i.e., TX power, RX sensitivity, Range-Closed, Frequency) based on the specification of Orinoco 802.11b 11Mbps PC card in closed environments [14].

The WLAN is characterized by heterogeneous traffics with different QoS requirements, in particular:
- VoIP sessions, associated with AC0, reflecting ITU G.726 audio codec bit rate of 32 Kb/s (CBR source at 64 Kb/s PCM coding rate with 50% on/off activity periods);
- video traffic, associated to AC1, at medium quality (bit rate of 300 Kb/s);
- data, intended as FTP background traffic, with no particular requirements in terms of delay, associated to AC4.

All traffics are established between the server and the wireless stations and their start/stop times are varied during simulations (see table 2). We have chosen a balanced scenario (not usually studied in literature), where each traffic typology is mainly symmetric with similar upstream/downstream flows. In some simulations we also tested an unbalanced scenario and even better results were found (not reported here for a matter of space).

The main characteristics of heterogeneous traffics and their respective packet size are reported in table 3.

<table>
<thead>
<tr>
<th>QSTA Upstream traffics</th>
<th>Downstream traffics</th>
<th>Start time</th>
<th>Stop time</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>10</td>
<td>160</td>
</tr>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>10</td>
<td>160</td>
</tr>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>10</td>
<td>160</td>
</tr>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>10</td>
<td>160</td>
</tr>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>60</td>
<td>160</td>
</tr>
<tr>
<td>VoIP,Video,FTP</td>
<td>VoIP,Video,FTP</td>
<td>110</td>
<td>160</td>
</tr>
</tbody>
</table>

Figure 2: Simulation scenario.
Table 3: Traffic types.

<table>
<thead>
<tr>
<th>Traffic</th>
<th>Protocol</th>
<th>Bit rate</th>
<th>Packet Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>UDP/CBR</td>
<td>32 Kbit/s</td>
<td>80 bytes</td>
</tr>
<tr>
<td>Video</td>
<td>UDP/CBR</td>
<td>300 Kbit/s</td>
<td>1464 bytes</td>
</tr>
<tr>
<td>FTP</td>
<td>TCP/FTP</td>
<td>---</td>
<td>1500 bytes</td>
</tr>
</tbody>
</table>

We have run many simulations, under the same scenario in two cases, without and with the adoption of our DTXOP algorithm.

In the first simulation set we have used the standard values of TXOPlimit for all traffic categories, as indicated in table 1. A TXOPlimit equal to zero in AC2 and AC3 indicates that it is allowed the transmission of one single frame per each channel access (i.e., no transmission of bursts).

In the second set we have implemented our dynamic TXOPlimit adjustment described in section IV.

The results of both simulations are reported in the following figures, in terms of goodput and delay performance.

Figures 3 and 4 show the goodput performance for VoIP and video respectively.

In the second temporal window (60-110s) new flows are activated and the network is entering in a critical phase of saturation of channel resources; here we can see the benefits introduced by the dynamic adjustment of the TXOPlimit parameter for both ACs. This fact is very evident especially in the downstream direction, which is penalized with a static TXOPlimit allocation. Moreover, with our algorithm the goodput resources are much more balanced between downstream/upstream, as it is clearly depicted in figures 3 and 4.

In the third temporal window (110-160s), the network is overloaded and channel resources are no more sufficient to satisfy all the different traffic flows; thus there is a significant decrease in goodput performance, that is more evident for video traffics. However, even in this case the dynamic TXOPlimit approach shows much better results, allowing applications to resist for a longer time before ceasing.

In such a heavily loaded situation the network is not able to guarantee acceptable QoS performance and an admission control mechanism is needed [15], [16], [17].

In figures 5 and 6 we reported the results in terms of packet delay for VoIP and video traffics. Here the benefits introduced by the dynamic adjustment of TXOPlimit are even more evident.

In fact, with static TXOPlimit, only the upstream flows meet good QoS requirements, keeping their packet delay values constantly under 50ms for VoIP and 100ms for video. On the contrary, downstream VoIP packets experience a sudden increase, exceeding the value 200ms when the network is approaching saturation (around the time value of 85s in fig.5). As regards downstream video, delays show a similar behaviour, reaching values above 700ms.

With dynamic TXOPlimit, delay performance are hugely increased especially for downstream. In both directions VoIP and video exhibit very good delay values, even if QoS degradation appears evident when too many sources are contending the channel. However the network is saturated later than with static TXOPlimit, as the critical point is now around the time value of 120s, after the ingress of more traffic.
sources in the network. Moreover it is to be noted that after a relatively long transitory period, due to the growth of traffic load, the network seems to recover and delays show a rapid decrease to more acceptable values; this behavior is not shown with static TXOP limit allocation. The same observation applies to goodput performance.

Finally, figure 7 shows the ratio between the average upstream and downstream goodputs. The ideal case of equal allocation of channel resources in the two directions should follow a horizontal line around the unitary value. This condition is satisfied only when the WLAN is not heavily loaded. When traffic volumes increase, balance between upstream and downstream goodputs is constantly maintained only with the dynamic adjustment of TXOP limit.

\[ \text{Average goodput uplink/downlink} \]

![Average goodput uplink/downlink](image)

Figure 7: Average upstream/downstream goodput ratio.

V. FINAL REMARKS

This work proposes a new algorithm for the dynamic allocation of TXOP duration to each AC of a IEEE 802.11e WLAN. As demonstrated by simulation results, the proposed DTOP scheme is capable of enhancing QoS performance of heterogeneous multimedia applications and of achieving fairness between the channel access time in upstream and downstream directions. However, even if DTOP allows to defer the attainment of network saturation, when traffic load is too high, an admission control mechanism is needed in order to meet QoS requirements of high priority traffics.

REFERENCES


