Volume 2, Issue 6, June 2012 **ISSN: 2277 128X**

International Journal of Advanced Research in Computer Science and Software Engineering

Research Paper

Available online at: www.ijarcsse.com

Performance Evaluation of IEEE 802.11e QOS Enhancements

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Abstract—**This paper presents an evaluation of the QoS enhancements, named IEEE 802.11e, currently under specification. Both the Enhanced Distributed Coordination Function (EDCF) and Hybrid Coordination Function (HCF) modes of Medium Access Control (MAC) operation are analysed and compared with legacy Distributed Coordination Function (DCF) and Point Coordination Function (PCF). Performance evaluation is attained through computer simulation of a scenario of 802.11b/e access to an IP core network through an Access Point (AP) in an infrastructure WLAN.**

Keywords— **WLAN, IEEE 802.11e, QoS,*CWmin*,*CWmax*.**

I. INTRODUCTION

The need for mobile computing has launched a successful wireless LAN market, with WLANs promising to replace most wired LAN infrastructures in the near future. WLANs allow users to roam inside a building without interrupting their communication sessions and avoid the use of cables. The mostly commercialised WLAN products are nowadays based on the IEEE 802.11 standard [1]. However, the widespread use of multimedia networking applications has brought more requirements to the network, creating a need for end-to-end quality of service (QoS). The latter requires not only QoS support mechanisms in the core IP network, but also in the access network at the user premises. While the initial IEEE 802.11 standard has little QoS support, a set of QoS enhancements to the Medium Access Control (MAC) form the main part of IEEE 802.11e, which is currently being specified. The Enhanced Distributed Coordination Function (EDCF) adds transmission prioritisation to CSMA/CA. On the other hand, a new coordination function named Hybrid Coordination Function (HCF) allows a Hybrid Coordinator (HC) located at the Access Point (AP) to start polling based contention-free access at any time during the contention period as needed to conform to the QoS parameterisation.

This paper presents the main QoS support mechanisms of IEEE 802.11e and evaluates their performance through computer simulation of a scenario of 802.11b/e (802.11b PHY with 802.11e MAC) access to an IP core network through an AP in an infrastructure WLAN, comparing the results with those attained by legacy IEEE 802.11b.

II. THE LEGACY IEEE 802.11 MAC

The IEEE 802.11 MAC defines two transmission modes for data packets: the Distributed Coordination Function (DCF) based on CSMA/CA and the contention-free Point Coordination Function (PCF) where the Access Point controls all transmissions based on a polling mechanism. The DCF and PCF modes are time multiplexed in a superframe, which is formed by a PCF contention-free period (CFP) followed by a DCF contention period (CP), positioned at regular intervals (see Fig. 1). The AP transmits beacon frames periodically in order to deliver management information to terminals. The boundaries between CFPs and CPs are marked by beacons carrying the Delivery Traffic Indication Message (DTIM). Terminals can use the information present in the beacons in order to associate with the AP, which is performed during the CP. This association is mandatory if the terminal needs to have its transmissions scheduled by the PCF, which is usually required for QoS sensitive data.

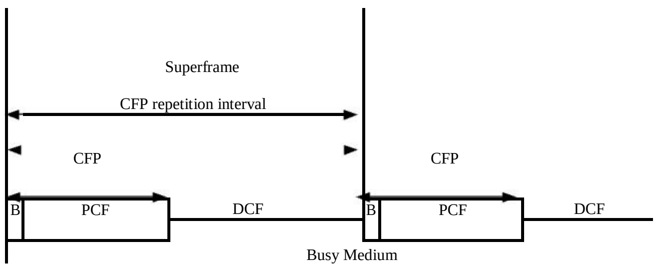

Fig. 1. Beacons and contention free periods.

Packet priorities are implemented defining three different length Interframe Spaces (IFSs):

- SIFS (Short IFS): This is the shortest IFS. It is used for transmission of high priority frames: acknowledgements of DATA frames, CTS frames, PCF frames and all DCF DATA frames except the first fragment of a burst.
- PIFS (PCF IFS): Greater than SIFS. After this interval expires, any PCF mode frames can be transmitted.

- DIFS (DCF IFS): Greater than PIFS. After this interval expires, any DCF mode frames can be transmitted asynchronously according to the CSMA backoff mechanism (see below).

The DCF mode is based on a CSMA/CA mechanism. The access control scheme is shown in Fig. 2.

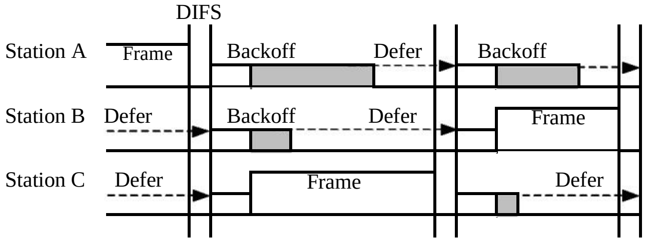

Fig. 2. Backoff mechanism in DCF.

A terminal that intends to transmit and senses the channel busy waits for the end of the ongoing transmission, then waits for a time period of DIFS length, and then randomly selects a time slot within the backoff window. The backoff length is calculated as follows:

$$Backoff\_length = Random(0, CW) \cdot aSlotTime \quad (1)$$

The slot duration, ***aSlotTime***, depends on the network round-trip propagation delay. The number of backoff slots is derived from a uniform distribution over the interval [0, *CW*], where the contention window (*CW*) parameter ranges from a minimum value of ***aCWmin*** up to a maximum value ***aCWmax***. Initially, the *CW* parameter is set to ***aCWmin*** and can be increased up to 255.

If no other terminal starts transmitting before the intended slot is reached, the transmission of a fragment with maximum size of **aFragmentationThreshold** is started. Collisions can only occur in the case that two terminals have selected the same slot. For each unsuccessful transmission the contention window is updated as follows:

$$CW = 2^{2+i} - 1 \quad (2)$$

where *i* is the number of transmission attempts.

If another terminal has selected an earlier slot, transmission is deferred and its backoff counter is frozen. Then, the terminal waits for the channel to become idle and then waits for the backoff slots remaining from the previous competition. After the successful transmission of the first fragment of a MAC SDU (MSDU), the remaining fragments are transmitted sequentially separated by a SIFS interval. Transmission ends when all fragments of the MSDU are transmitted or the maximum dwell time (**aMediumOccupancyLimit**) expires.

In order to guarantee undisturbed transmission even if hidden terminals are present, an RTS/CTS mechanism is used. When this mechanism is applied, the contention winner does not transmit the data immediately. Instead it sends an RTS frame to which the receiver answers with a CTS frame. This guarantees that all terminals in the range of both the sender and the receiver know that a packet will be transmitted, remaining silent during the entire transmission. Only then the sender transmits the data frames. While the two extra messages present additional overhead, the mechanism is particularly useful in the case of large data frames because the RTS and CTS frames are short.

The PCF mode is based on a polling mechanism controlled by the AP as depicted in Fig. 3.

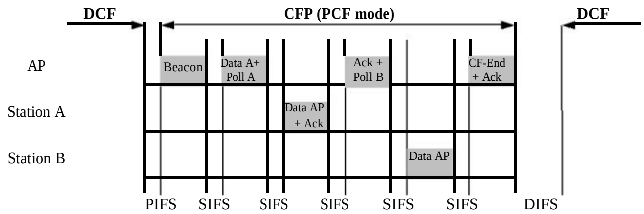

Fig. 3. Polling mechanism in PCF.

During the CFP, the AP polls the terminals registered in its polling list and allows them undisturbed contention-free access to the medium. As already said, in order to become registered in the polling list, the terminals have to associate with the AP during the CP. The maximum duration of a CFP is given by the 802.11 Management Information Base (MIB) variable **aCFPMaxDuration**, while the frequency is given by the variable **aCFPRate**.

During the CFP a frame can be a composite of control and data information. The following combinations are allowed for PCF frames: DATA, CF-ACK, CF-POLL, DATA+CF-ACK, DATA+CF_POLL, DATA+CF-ACK+CF-POLL and CF-ACK+CF-POLL. Only the AP has the capability to issue frames with CF-POLL. The terminals can answer with DATA, CF-ACK, DATA+CF-ACK or a NULL frame (the latter is sent when there is no data to transmit and no pending acknowledgement). Each DATA frame cannot be longer that the maximum size **aFragmentationThreshold**. If the terminal does not answer a polling request within an interval of PIFS, the AP concludes that an uplink frame was lost and may decide to poll the same terminal again.

DCF is the most appropriate in a context where there is no network infrastructure and users must form temporary ad hoc networks to communicate directly between each other. On the other hand, if the objective is to offer a permanent network infrastructure to provide access to the Intranet/Internet with guaranteed QoS bounds, PCF is the best choice.

III. ENHANCED DISTRIBUTED COORDINATION FUNCTION (EDCF)

It can be easily noticed that legacy DCF cannot fulfil the QoS requirements of multimedia applications such as telephony and videoconference, as it does not include prioritisation mechanisms. IEEE 802.11 Task Group E has elected Virtual DCF (VDCF) as the EDCF mechanism to be incorporated in the upcoming IEEE 802.11e standard [2]. EDCF introduces a prioritisation enhancement based on different Access Categories (ACs). One or more User Priorities (UPs) can be assigned to each AC [3]. In this case, each UP within each AC has a different queue, different IFS (Arbitration IFS – AIFS) and contention window parameters. Each AC contends for medium access with only one CSMA instance using the parameters that belong to its lowest UP. This corresponds to the priority of the AC as a whole.

The AIFS length of UP i is set according to the following formula:

$$AIFS_i = SIFS + aAIFS_i \cdot aSlotTime \qquad (3)$$

The default value for $\boldsymbol{aAIFS_i}$ is 2 slots, which makes $AIFS_i$ equal to DIFS time as in the legacy DCF. A terminal having several ACs maintains a separate backoff timer for each of those ACs, with each backoff timer independently counting down. The backoff of each AC j is chosen according to a uniform distribution over $[0, CW_i]$, where i is the lowest UP of the AC and CW_i is the corresponding contention window:

$$Backoff_j = Random\ (0\ ,\ CW_i) - {}_{a}SlotTime \qquad (4)$$

CW_i is an integer within the range $\boldsymbol{aCWmin_i}$ and, optionally **aCWmax**$_{\mathbf{i}}$ upon collision it is updated using the same gaenerator function. As for legacy DCF.When the backoff timer of an AC counts down to zero,the terminal transfer a frame from the queue with highest priority and an initiates a transmission opportunity (TXOP), when it is bounded duration time interval in which the station may transmit a sequence of SIFS-separated DATA frame exchanges.Internal conflicts between local ACs accurs when the corresponding Backoff time expire at the same time.In that case the ESTA transmits a frame from the AC of highest priority and then resets all expires backoff timers. During the TXOP, the terminal can send a burst of DATA frames separated by SIFS in the same way already explained for legacy DCF.The TXOP ends when there are no frames to be transmitted or when the TXOP maximum duration expires. The default TXOP maximum duration is given by the MIB variable **dot11DefaultCPTXOPLimit,** but the TXOP limit can be modified the AP in beacons or association response frames.

IV. HYBRID COORDINATION FUNCTION (HCF)

Besides enhancing DCF, IEEE 802.11e will also specify a new mode of operation named Hybrid Coordination Function (HCF) [4][5]. HCF is based on a polling mechanism similar to legacy PCF, but it allows the HC to start contention-free Controlled Access Periods (CAPs) at any time during a CP, after the medium remains idle for at least a PIFS interval (see Fig. 4). This more flexible contention-free mechanism renders PCF useless, although IEEE 802.11e terminals are still allowed to support PCF [6].

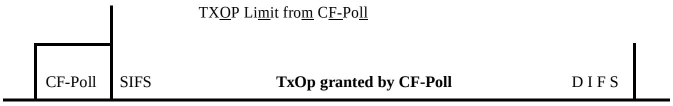

Fig. 4. Transmission opportunity in HCF.

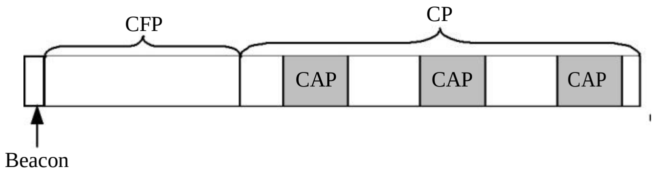

Fig. 5. Generation of CAPs during the CP.

A new set of frames is defined, which is similar to the legacy PCF frame set but with a QoS attribute added: QoS NULL, QoS DATA, QoS CF-ACK, QoS CF-POLL, QoS DATA+CF-ACK, QoS CF-ACK+CF-POLL, QoS DATA+CF_POLL and QoS DATA+CF-ACK+CF-POLL and. A CAP is a sequence of TXOPs initiated by the HC with the transmission of a QoS Data frame or QoS CF-POLL frame (see Fig. 5).

A TXOP ends when at least once one of the following condition is met:

Transmission of QoS DATA frames with the non-Final(NF) flag set to 0,which means that there are no further frames queued for transmission.

Expires of the TXOP duration implicitly given by the default MIB variable **dot11DefaultCPTXOPLimit** or explicitly set by the HC in beacons,associations response frames or QoS CF-POLL frames.

The polled stations allow the wireless medium to remain idle for PIFS.

The CAP ends when the wireless medium remains idle for a DIFS interval.

As already seen in legacy IEEE 802.11 the multiplexing of DCF and PCF in a superframe is fixed,the MIB variables **aCFPRate** and **aCFPMaxDuration** clearly definr the length and frequency of the CFPs and by extension the proportion between the CP and CFP,In HCF this proportion must also be specified in ordered to limit polling.as contension is still needed for important management tasks.

In HCF, the rate and proportion of CAPs are given by the MIB variables **dot11CAPRate** and **dot11CAPMax**. The meaning of **dot11CAPRate** is very different from the corresponding PCF variable **aCFPRate** and specifies the fraction of the CP that can be used for CAPs, expressed in units of microseconds per 64 microseconds (e.g. a **dot11CAPRate** value of 32 means that at most one half of the CP can be spent with CAP). On the other hand, the variable **dot11CAPMax** specifies the maximum duration of a CAP. Together, the two HCF MIB variables define a token bucket of time whose state is given by a CAP timer. The CAP timer is initialised to zero and counts upwards at a rate defined by **dot11CAPRate**, until it reaches the maximum value of **dot11CAPMax**. At any time, the AP can deduct from the CAP timer a number of units equal or less than its current value and start a CAP whose duration in microseconds corresponds to the number of deducted units.

HCF also defines Reservation Request (RR) frames that can be used by the stations to request TXOPs to the HC. Additionally, Controlled Contention (CC) frames can be used by the HC to initiate a controlled contention interval (CCI) in which the stations contend for transmission using the short RR frames, with actual data transmission being done after contention by the winning station. The HCF specification will also include appropriate signalling to negotiate QoS parameters for specific data streams [7]. These parameters are similar to the IP FlowSpecs defined in RFC 1363 [8]. These advanced features are left for future study and are not reflected in the simulations presented in this paper.

V. SIMULATION PARAMETERS AND TRAFFIC SOURCE MODELS

The simulations consider three types of traffic sources: bursty data (e.g. HTTP sessions), VoIP and video.

The model for bursty data sources is the Source Type 1 defined for 802.14 performance evaluation [9], which consists on a Poisson distribution, which generates the following message sizes and respective probabilities: (64, 0.6), (128, 0.06), (256, 0.04), (512, 0.02), (1024, 0.25) and (1518, 0.03). Each busty data source generates 200 Kbps.

The audio source model generates 60-byte messages periodically with an interval of 20 ms resulting in bitrate of 24 Kbps, a suitable model for G.729A [10] plus RTP/UDP/IP overhead.

For the video source model we have used a trace of a real H.263 video stream captured in an office environment encoded to provide an output bitrate of 256 Kbps, which is illustrative of H.323 videoconference traffic. For more information refer to the Office-Cam video stream presented in [11].

In the simulations, each wireless terminal runs only one session and all sessions are bi-directional, i.e. each terminal is the source of an uplink flow and the sink of a downlink flow for the session it runs. It is considered that VoIP and video have higher priority than bursty data.

The most relevant parameters that the IEEE 802.11 simulation considers are listed in Table 1, together with the respective configuration used in the simulations.

TABLE 1
SIMULATION PARAMETERS

IEEE 802.11 (1999 specification)	**aSlotTime**	20 ∝ s
	aFragmentationThreshold	1024 bytes
	aMediumOccupancyLimit	5040 ms
	aRTSThreshold	500 bytes
	SIFS	20 ∝ s
	PIFS	40 ∝ s
	DIFS	60 ∝ s
	Superframe length (PCF capable terminals only)	30 ms
	aCFPMaxDuration (PCF capable terminals only)	10 ms
	PLCP preamble and header length	192 bits
	PLCP preamble and header bit rate	1 Mbps
	PSDU bit rate	11 Mbps (maximum for 802.11b)
EDCF	**aCWmin (VoIP+video)**	7
	aCWmax (VoIP+video)	15
	AIFS (VoIP+video)	60∝ s
	aCWmin (data)	7
	aCWmax (data)	255
	AIFS (data)	60∝ s+ 15· **aSlotTime**
	dot11DefaultCPTXOPLimit	5040 ∝ s
HCF	**dot11CAPRate**	21∝ s
	dot11CAPMax	5040∝ s
	CAP timer update time	5120∝ s
Wireless Medium	**Bit Error Rate**	BER11=1.3E-5 BER1 ≈ 0

PCF capable terminals have the CFP configured as 10 ms (**aCFPMaxDuration**) in each superframe of 30 ms, which means that PCF is only used one third of the time. This restriction could happen in practice if low association latency was required for a highly mobile network. HCF has the same restriction proportion between contention and contention-free transmission.

A **dot11CAPRate** value of 21∝ s means that on average only one third of each unit of 64∝ s can be used for CAP. The maximum duration of a CAP is configured as 5040∝ s, i.e. the maximum duration of a TXOP or a frame burst in legacy DCF. The CAP timer update interval was configured as 5120∝ s, which follows the recommendation expressed in [6] that the CAP timer should be updated at uniform intervals that are multiples of 64∝ s, and no less than 1024∝ s.

Both PCF and HCF schedule contention-free transmission with SETT [12], a scheduling discipline that can differentiate between different service classes and optimise network utilisation. In EDCF, prioritisation is achieved by establishing different IFSs and contention window limits for the different priorities, which results in non-overlapping contention windows. The AIFS for high priority traffic has the same length as DIFS and the contention window varies between 7 and 15 time slots. On the other hand, the AIFS for low priority traffic is equal to DIFS plus an offset of 15 time slots. The contention window for low priority traffic varies between the standard 7 and 255 time slots after the AIFS. A contention window of 15 time slots for high priority traffic is acceptable when high priority flows are few. It would have to be enlarged if the number of high priority flows increased in order to keep a low collision probability. On the other hand the AIFS for low priority traffic must not be so large as to introduce a significant overhead and greatly affect its throughput.

A fixed Bit Error Rate of 1.3E-5 was considered for transmission at 11 Mbps (BER_{11}). In these conditions the bit error rate in the PLCP preamble and header transmitted at 1 Mbps (BER_1) becomes negligible.

VI. **SIMULATION RESULTS**

The simulations aim to evaluate the prioritisation capabilities of the several IEEE 802.11 MAC operation modes, as well as the cost of prioritisation in terms of network utilization. It considers a video session running simultaneously with one G.729A VoIP session and *n* bursty data sessions.

Fig. 6 shows the average packet transmission delay for the high priority VoIP and video sessions.

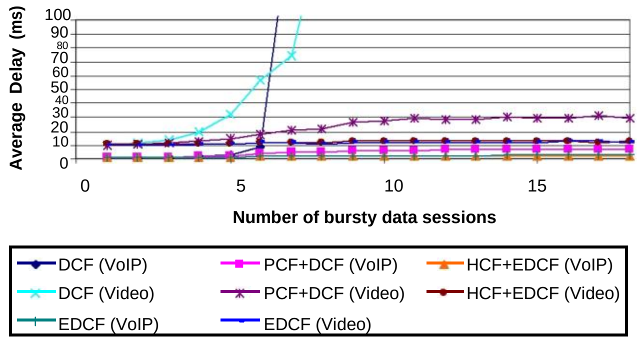

Fig. 6. Average packet delay for the VoIP and video streams.

Legacy DCF alone is not able to protect high priority traffic. Average delay rises above 70 ms after 6 data sessions for both VoIP and video. IEEE 802.11e presents good performance with or without HCF, as it keeps average delay below 2.6 ms and 13 ms for VoIP and video respectively. PCF+DCF definitely presents worse performance than IEEE 802.11e, with VoIP average delay rising above 6 ms with more than 9 bursty data sessions, and video average delay rising above 25 ms with more than 8 bursty data sessions. This is because the CFPs have fixed duration (10 ms) and fixed positions within the superframe of 30 ms, being separated by CPs of 20 ms. As DCF defines no prioritisation mechanism, VoIP and video have to contend with bursty data in equal terms during the CP and can only be transmitted with QoS guarantees during the CFP. As the number of bursty data sessions increases, it becomes more difficult for VoIP and video to win contention in the CP, and it becomes more probable that the VoIP and video packets have to wait in the queue during the 20 ms of the CP and be transmitted only during the CFP. Although the proportion of contention-free and contention based access in HCF+EDCF is configured to be the same, EDCF includes prioritisation while HCF is able to generate CAPs during the CP, which means that high priority data can obtain TXOPs at any time.

The confirmation of these results is provided by the maximum packet delay depicted in Fig. 7.

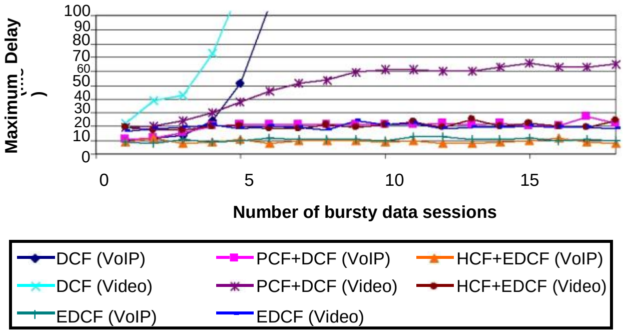

Fig. 7: Maximum packet delay for the VoIP and video streams.

IEEE 802.11e is always able to keep the maximum delay below 13 ms and 26 ms for VoIP and video respectively, independently of HCF being used or not. On the other hand, in PCF+DCF operation, some VoIP packets present delay above 20 ms with more than 3 bursty data sessions. Video performance is even worse as maximum delay goes above 60 ms with more than 9 bursty data sessions. These high delay values are likely to fall on large video frames. In MPEG video these are normally the intra-coded periodic 'I' frames that provide synchronization. In real-time video streams, high delay and jitter can cause the codec to discard those frames and greatly degrade the quality perceived by the user.

It should be noted that simulations were also conducted regarding coupling HCF with legacy DCF, in order to compare HCF and PCF in the same conditions. The results have shown that HCF still presents better performance than PCF and in special for VoIP. The difference tends to become more accentuated as the superframe size increases, increasing the separation of PCF periods.

As delay is usually not much relevant for bursty data, Fig. 8 depicts only the throughput for these sessions. Note that the represented throughput is the sum of uplink and downlink throughput. As can be seen, all configurations have similar performance. Maximum throughput is approximately 2.5 Mbps for PCF+DCF and HCF+EDCF, 2.6 Mbps for EDCF and DCF.

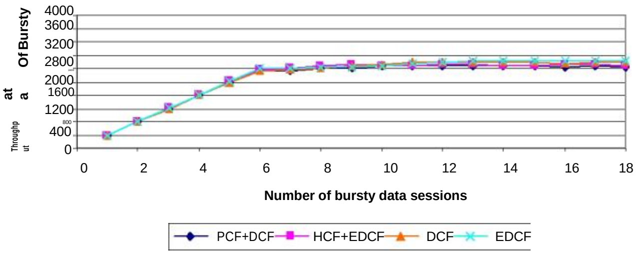

Fig. 8: Throughput of bursty data streams.

When the throughput is analysed separately in the uplink and downlink directions, the asymmetry of the contention based methods becomes evident (see Fig. 9 and Fig. 10 for the uplink and downlink throughput respectively).

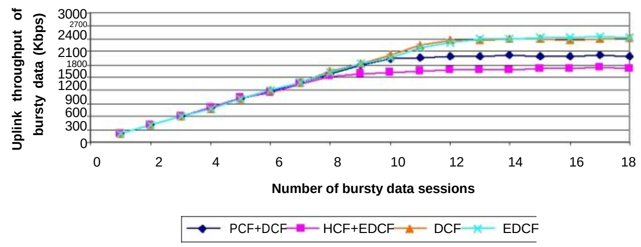

Fig. 9: Uplink throughput of bursty data streams.

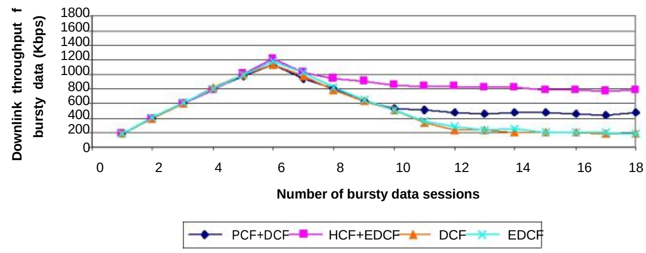

Fig. 10. Downlink throughput of bursty data streams.

With more than 6 data streams, uplink throughput is always higher than downlink throughput, and the later tends to decrease as the total amount of traffic increases. This is more accentuated when DCF and EDCF are used alone. The maximum difference between uplink and downlink throughput is approximately 2.2 Mbps for both DCF and EDCF. The asymmetry decreases when PCF or HCF is used, with the maximum difference being approximately 1.5 Mbps for PCF+DCF and 950 Kbps for HCF+EDCF.

This asymmetry of DCF and EDCF is easy to explain. As each terminal or AP has only one CSMA instance for each AC (we can consider that DCF is a special case of EDCF with only one AC), this means that with *n* terminals and for each AC there are *n* uplink CSMA instances contending with only 1 downlink CSMA instance. One way to solve the problem in EDCF could be to configure the AP with separate CSMA instances per associated terminal, though this would increase its complexity. A simpler solution is to increase the fraction of PCF/HCF as much as possible for the scenario at hand.

VII. CONCLUSIONS

This paper has presented a performance evaluation of the IEEE 802.11e QoS enhancements. Simulation results clearly show that IEEE802.11e improves QoS support in

IEEE802.11 networks, in special when both EDCF and HCF are used.

Enhanced DCF (EDCF) allows QoS differentiation, which is an important improvement over legacy DCF. Nevertheless it presents the same throughput asymmetry, giving advantage to uplink transmission as the number of terminals increases. This is because the aggregate downlink traffic sent by the AP must compete for resources in equal terms with all the terminals that want to transmit in the uplink direction. The way to solve the problem is to increase the amount of PCF or HCF contention-free transmission as much as possible.

Like PCF, HCF includes a polling mechanism controlled by the AP, which is used during Controlled Access Periods (CAPs). Nevertheless HCF is more flexible than PCF because CAPs can occur anytime during the superframe, with the ratio between contention-free and contention transmission being controlled by a token-bucket of time units. Due to the more flexible distribution of CAPs, HCF presents lower transmission delays as the size of the superframe increases and in special for traffic with low burstiness. This is in contrast with legacy IEEE 802.11 in which the PCF based contention-free period (CFP) has a fixed position in the superframe, forcing QoS sensitive traffic to wait for the entire DCF contention period (CP) before being polled. To make the CFPs more frequent would require shortening the DTIM period, which would increase protocol overhead and wake-up stations in power save mode more frequently [5].

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