Telecommun Syst (2007) 36: 27–38
DOI 10.1007/s11235-007-9059-8

On the capabilities of IEEE 802.11e for multimedia communications over heterogeneous 802.11/802.11e WLANs

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Published online: 20 November 2007

Abstract Multimedia communications over WLAN is widely acknowledged as one of the key, emerging applications for wireless LANs. As with any multi-service network, there is the need to provision the WLANs with the QoS mechanisms capable of guaranteeing the requirements of various services. The upcoming IEEE 802.11e (EDCA) standard is a proposal defining the mechanisms for wireless LANs aiming to provide QoS support to time-sensitive applications such as voice and video communications. Due to the fact that the IEEE 802.11e interface cards will take over the WLAN market, replacing the use of legacy IEEE 802.11 interface cards in most WLAN applications, an important number of networking scenarios will consist of a hybrid configuration comprising legacy IEEE 802.11-based stations and IEEE 802.11e-based stations. For this reason, in this paper we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) upcoming standard when supporting different services, such as, voice, video, best-effort, background and in the presence of traffic generated by legacy 802.11 (DCF) based stations.

This work was supported by the Ministry of Science and Technology of Spain under CONSOLIDER CSD2006-46 and CICYT TIN2006-15516-C04-02, the Council of Science and Technology of Castilla-La Mancha under project PAI06-0106 and FEDER.

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Keywords WLAN · 802.11 · 802.11e · EDCA · QoS · Multimedia communications · Performance evaluation · Video quality

1 Introduction

WLANs are gaining popularity at an unprecedented rate, at home, at work, and in public hot spot locations mainly due to their low cost, their ease of deployment and, above all, by allowing the end users to freely move around within the area they cover. Another influential factor is the appearance in 1997 of the standard IEEE 802.11, with its subsequent revision in 1999 [1], and its subsequent amendments that nowadays enable transmission speeds of up to 54 Mbps, allowing the use of multimedia applications. However, the multimedia applications are not only characterized by their high bandwidth requirements, but also impose severe restrictions on delay, jitter and packet loss rate. In others words, multimedia applications require *Quality of Service* (QoS) support. Guaranteeing those QoS requirements in IEEE 802.11 is a very challenging task due to the QoS-unaware functions of its MAC layer. This layer uses the wireless media that is very unpredictable due to the high risk of collisions and the difficulties faced by the signal propagation. Thus providing QoS to IEEE 802.11 has become an area of active research giving rise to several new service differentiation schemes.

In the last years, the IEEE 802.11 Working Group has worked on the definition of the IEEE 802.11e standard [2]. The IEEE 802.11e is a proposal defining the mechanisms for wireless LANs aiming to provide QoS support to time-sensitive applications, such as, voice and video communications. Many studies have shown that the IEEE 802.11e (EDCA) scheme performs poorly under heavy load conditions. The severe degradation is mainly due to high collision rates. This reason has led many researchers to design new techniques aiming to address the shortcomings of the IEEE 802.11e standard. However, many of the proposed techniques have overlooked two main implementation and operation issues: first, the implementation of the proposed mechanisms implies important and incompatible modifications to the IEEE 802.11e specifications, and second, the main deficiency of these mechanisms comes from its inability to provide the QoS for video flows when legacy DCF based stations are present in the same scenario.

The rest of the paper is organized as follows. In Sect. 2, we describe the upcoming IEEE 802.11e QoS standard and some of the most relevant proposals recently reported in the literature aiming to improve the performance of the IEEE 802.11e (EDCA) standard. In Sect. 3, we carry out a comparative performance evaluation when supporting different services, such as, voice, video, best-effort, background and in the presence of traffic generated by legacy DCF based stations. Finally, Sect. 4 concludes the paper.

2 The IEEE 802.11e standard

The IEEE 802.11e standard [2] aims to specify the mechanisms enabling the provisioning of QoS guarantees in IEEE 802.11 WLANs. In the IEEE 802.11e standard, distinction is made among those stations not requiring QoS support, known as nQSTA, and those requiring it, QSTA. In order to support both Intserv and DiffServ QoS approaches in an IEEE 802.11 WLAN, a third coordination function is being added: the *Hybrid Coordination Function* (HCF). The use of this new coordination function is mandatory for the QSTAs. HCF incorporates two new access mechanisms: the contention-based *Enhanced Distributed Channel Access* (EDCA), known in the previous drafts as the *Enhanced DCF* (EDCF) and the *HCF Controlled Channel Access* (HCCA).

One main feature of HCF is the definition of four *Access Categories* (AC) queues and eight *Traffic Stream* (TS) queues at MAC layer. When a frame arrives at the MAC layer, it is tagged with a *Traffic Priority Identifier* (TID) according to its QoS requirements, which can take values from 0 to 15. The frames with TID values from 0 to 7 are mapped into four AC queues using the EDCA access rules. The frames with TID values from 8 to 15 are mapped into the eight TS queues using the HCF controlled channel access rules. The TS queues provide a strict parameterized QoS control while the AC queues enable the provisioning of multiple priorities. Another main feature of the HCF is the concept of *Transmission Opportunity* (TXOP), which defines the transmission holding time for each station.

2.1 Enhanced distributed channel access (EDCA)

EDCA has been designed to be used with the contention-based prioritized QoS support mechanisms. In EDCA, two main methods are introduced to support service differentiation. The first one is to use different IFS values for different ACs. The second method consists in allocating different CW sizes to the different ACs. Each AC forms an EDCA independent entity with its own queue and its own access mechanism based on an DCF-like mechanism with its own *Arbitration Inter-Frame Space* defined by $AIFS[AC] = SIFS + AIFSN[AC] \times SlotTime$ and its own CW[AC] ($CWmin[AC] \leq CW[AC] \leq CWmax[AC]$) (see Fig. 1), where AIFSN[AC] is the *Arbitration Inter-Frame Space Number*. If an internal collision arises among the queues within the same QSTA, the one having higher priority obtains the right to transmit. The queue getting the right to access to the channel obtains a transmission opportunity (TXOP). The winning queue can then transmit during a time interval whose length is given by *TXOPLimit*.

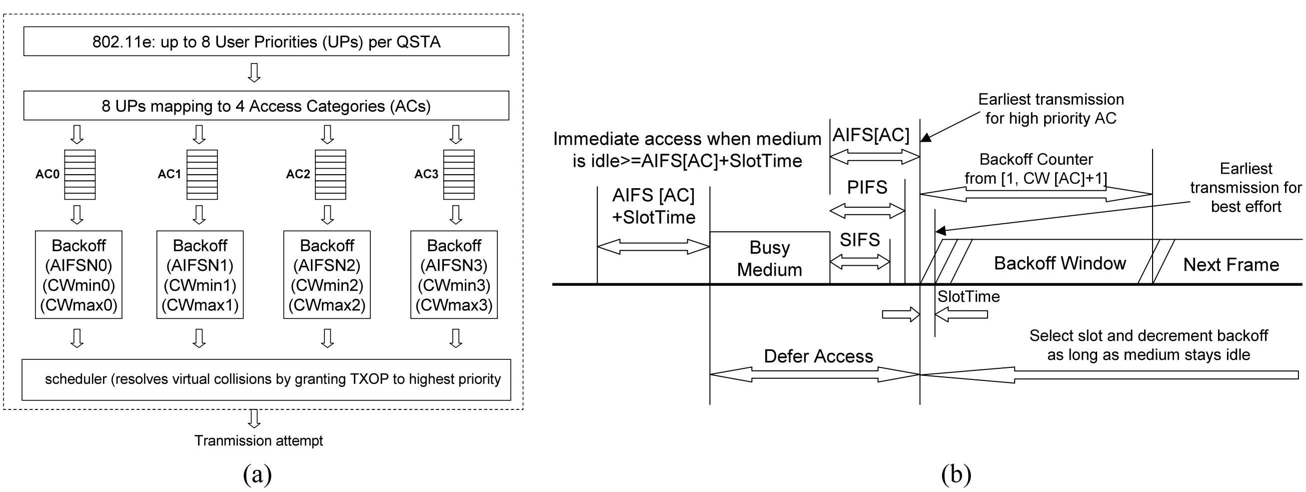


Fig. 1 EDCA. (**a**) Access categories. (**b**) IFS relationships

A closer look to EDCA and DCF shows an important issue; EDCA slightly differs from DCF on the way the backoff counter is managed. In EDCA, the backoff counter is also decremented at every idle slot time and frozen during channel activity periods. But it is resumed one slot time before the AIFS expiration. This means that when the AIFS timer elapses, the backoff counter will already be decremented by one unit. Moreover, since a single MAC operation per slot is permitted (backoff decrement or packet transmission ([2], clause 9.9.1.3)), when the counter decrements to 0, the station cannot transmit immediately, but has to wait for a further backoff slot if the medium is idle, or a further AIFS expiration in the medium is busy. Such apparently minor difference (which might perhaps appear as a technicality) has not been taken into consideration in previous studies in the literature, however it has important consequences in terms of performance of the EDCA access categories, especially when they compete with legacy DCF stations [3]. The performance of EDCA with AIFSN = 3 are similar to the DCF performance with AIFSN = 2. In our proposal, we take into consideration this important issue.

2.2 HCF controlled channel access (HCCA)

In the HCCA mechanism a central node is used for coordinating the access to the channel: the Hybrid Coordinator (HC). The main difference with the PC is that HC acts in both, the CFP and the CP intervals. When the HC takes control over the channel during the CP, it is said that a Controlled Access Phase (CAP) has been generated (see Fig. 2). In each CFP and in each CAP, the HC is responsible for polling those stations having made a TXOP request previously informing them of the time allocated for its transmissions. QSTA should reply to this poll in a time interval equal to SIFS. The polling scheduler and the implementation of the admission control do not come imposed by the IEEE 802.11e. It is worth noting that the HC should at all times hold the highest priority allowing it to initiate the CAP.

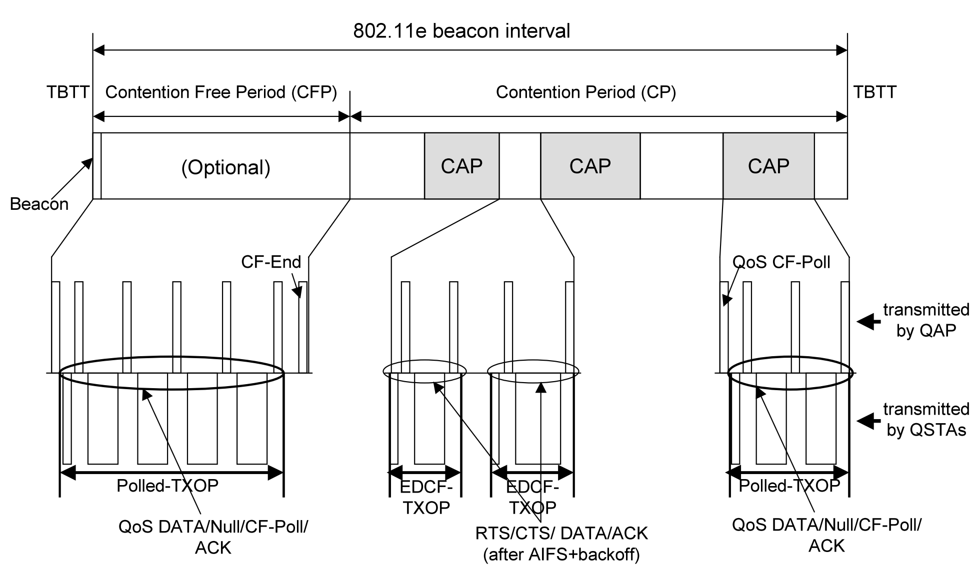
2.3 QoS enhancements to the IEEE 802.11e

Many on-going research efforts are focusing on the evaluation of the IEEE 802.11e standard. Many studies have revealed that the poor performance exhibited by the standard is mainly due to the high collision rates encountered when a large number of stations attempt to access the channel. Numerous proposals have been reported in the literature aiming to overcome this main drawback. In the following, we undertake the analysis of two of the most prominent ones.

The *Fast Collision Resolution Mechanism* FCR [4] aims to shorten the collision rate by increasing the contention window sizes of all active stations during the contention resolution period. To reduce the number of wasted (idle) slots, the FCR algorithm assigns the shortest window size and idle backoff timer to the station having successfully transmitted a packet. Moreover, when a station detects a number of idle slots (static backoff threshold), it starts reducing the backoff timer exponentially, instead of linearly as specified by the EDCA standard. To address the provisioning of QoS mechanisms, the authors further introduce an enhanced version of the FCR algorithm, namely, the *Real Time Fast Collision Resolution* (RT-FCR) [5] algorithm. In this algorithm, the priorities are implemented by assigning different backoff ranges based on the type of traffic. In their study, the authors have considered three main traffic types: voice, video, and best-effort (data) traffic.

Under this scheme, voice packets hold the highest priority to access the channel by setting $CW = CW_{min}$. All the other flows have to wait, at least, eight backoff slots before

Fig. 2 HCCA. CFP, CP and CAP periods



being allowed to gain access to the channel. The video traffic is assigned the second highest priority by using a smaller maximum contention window size than the one assigned to the best-effort data traffic.

The *Adaptive EDCF Mechanism* (AEDCF) [6] is another relevant mechanism recently reported in the literature. In [6], the authors state that the probability of collision increases is due to the re-setting of CW[AC] to CWmin[AC] after a successful transmission in the presence of multiple stations contending for the channel. Taking this fact into account, they have proposed decreasing the CW[AC] by multiplying by a factor lower than 0.8 after a successful transmission; the actual value of the factor will depend on the collision rate suffered by the AC. In [7], the same authors go a step further by introducing a new scheme called *Adaptive Fair EDCF* (AFEDCF) that improves AEDCF and FCR mechanisms. This mechanism uses an adaptive fast collision resolution mechanism (similar to the FCR mechanism) when the channel is sensed idle. In contrast with the FCR mechanism, AFEDCF computes an adaptive backoff threshold for each priority level by taking into account the channel load.

However, the main deficiency of these mechanisms comes from its inability to provide the proper QoS to the video service in scenarios comprising legacy DCF-based and IEEE 802.11e stations. This is due to the fact that, under theses schemes a station has always to wait for a minimum of eight backoff slots in order to comply with the highest priority assigned to the voice traffic. Under these schemes, the presence of voice and DCF stations may even result in starvation to the video flows. Moreover, the implementation of these mechanisms implies that the stations have to monitor the channel conditions in order to dynamically tune up the actual values of the key system parameters, such as the threshold and window size.

3 Performance evaluation

In this section, we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) upcoming standard in the presence of traffic generated by legacy 802.11 (DCF) based stations. We focus on assessing the behavior of the upcoming IEEE 802.11e (EDCA) standard with different parameter settings. Throughout our studies, we have made use of the OPNET Modeler tool 11.0 [8], which has IEEE 802.11 DCF functionality. We extended the simulator by implementing EDCA mechanism. We demonstrate that the defaults parameters setting recommended in the EDCA standard [2] by IEEE 802.11e group are set to assign higher priority to the voice and video traffic in the presence of DCF stations.

3.1 Scenarios

In our simulations, we model an IEEE 802.11b wireless LAN cell (using OPNET Modeler tool 11.0 [8]) comprising legacy DCF-based stations and EDCA-based stations. The EDCA-based stations support four different types of services: voice (Vo), video (Vi), best-effort (BE) and background (BK). This classification is in line with the IEEE802.1D standard specifications [9]. The DCF based stations support data traffic. We assume the use of a wireless LAN consisting of several wireless stations and an access point connected to a wired node that serves as sink for the flows from the wireless domain. All the stations are located within a *Basic Service Set* (BSS), i.e., every station is able to detect the transmission from any other station. The parameters for the wired link were chosen to ensure that the bandwidth bottleneck of the system is within the wireless LAN (see Fig. 3).

Each wireless station operates at 11 Mbit/s IEEE 802.11b mode and transmits a single traffic type to the access point. We assume the use of constant bit-rate voice sources encoded at a rate of 16 kbit/s according to the G.728 standard [10]. The voice packet size has been set to 168 bytes including the RTP/UDP/IP headers. For the video applications, we have made use of the traces generated from a variable bit-rate H.264 video encoder [11]. We have used the sequence mobile calendar encoded on CIF format at a video frame rate of 25 frames/s. The average video transmission rate is around 450 kbit/s with a packet size equal to 1064 bytes (including RTP/UDP/IP headers). The best-effort, background and DCF traffics have been created using a *Pareto* distribution traffic model. The average sending rate of best-effort and background traffic is 128 kbit/s, using a 552 bytes packet size (including TCP/IP headers). The average sending rate of DCF traffic is 256 kbit/s, using a 552 bytes packet size (including TCP/IP headers). All the traffic sources are randomly activated within of the interval [1, 1.5] seconds from the start of the simulation. Throughout our study, we have simulated the two minutes of operation of each particular scenario.

For all the scenarios, we have assumed that one fifth of the stations support one of the four kinds of services: voice, video, BE, BK and DCF applications. We start by simulating a WLAN consisting of five wireless stations (each one supporting a different type of traffic, see Fig. 3). We then gradually increase the *Total Offered Load* of the wireless LAN by increasing the number of stations by five. 1:1:1:1:1 for voice, video, BE, BK and DCF, respectively. We increase the number of stations 5 by 5 starting from 5 and up to 40. In this way, the normalized offered load is increased from 0.14 up to 1.12. We have preferred to evaluate a normalized offered load, rather than the absolute value. The normalized

Fig. 3 Scenario under study

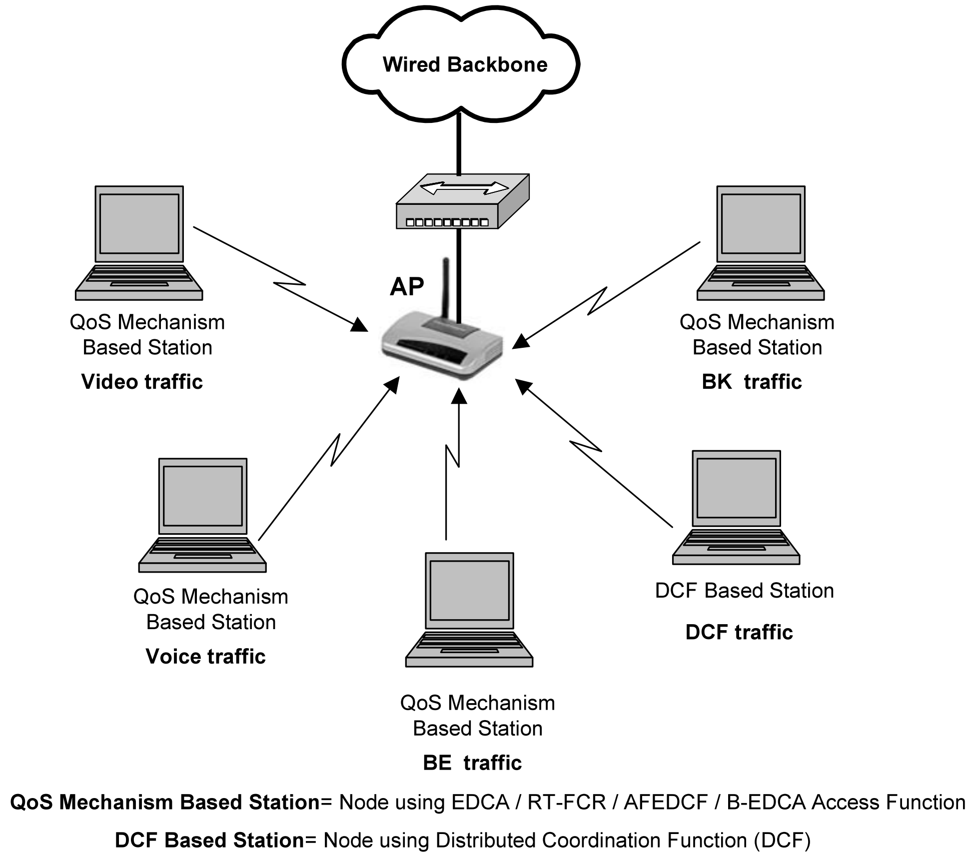


Table 1 Parameter settings evaluated

	BK	BE	Vi	Vo
AIFSN	**7**	**3**	**2**	**2**
	7	3	1	1
	7	3	2	1
	7	4	3	2
	7	4	3	2
CWmin	**31**	**31**	**15**	**7**
	31	31	31	7
	63	63	31	7
	31	31	31	15
	63	63	31	15
CWmax	**1023**	**1023**	**31**	**15**
	1023	1023	63	15
	1023	1023	127	15
	1023	1023	63	31
	1023	1023	127	31

offered load is determined with respect to the theoretical maximum capacity of the 11 Mbit/s IEEE 802.11b mode, i.e. 7.1 Mbit/s (corresponding to the use of the maximum packet size used by the MAC layer and in the presence of a single active station).

We start our study by setting up the parameters to the values recommended by the standards (see Table 1, boldface values). This will allow us to set up a base point for comparison purposes as well as to tune up the system parameters. We will show that the performance of EDCA can be greatly improved by properly tuning its parameters. The new parameter settings under study in this work are also shown in Table 1.

3.2 Metrics

In our study, we have been interested in assessing the performance in terms of the following metrics: Normalized throughput, collision rate, mean access delay, packet loss rate, and video quality. In the following, we provide the definitions of all the metrics being considered.

Normalized Throughput$_{\mathrm{TOTAL}}$ is the ratio between the traffic having been effectively sent through the channel over the overall traffic having been submitted by all types of sources. This metric can be simply defined as follows:

$$Throughput_{\mathrm{TOTAL}} = \frac{\sum_{type\in\Gamma}\sum_{i\in\Theta_{type}} Traf_{out_{type,i}}}{\sum_{type\in\Gamma}\sum_{i\in\Theta_{type}} Traf_{in_{type,i}}} \quad (1)$$

where $Traf_{in_{type,i}}$ is the traffic submitted by the i connection of type *type* and $Traf_{out_{type,i}}$ is the traffic of the connection i of type *type* having been effectively been transmitted through the channel. Moreover, $\Gamma = \{$voice, video, best-effort, background, DCF$\}$ and Θ_{type} is the set of connections of a given type.

Normalized Throughput$_{type}$ is the ratio between the traffic having been effectively sent by a given type of traffic through the channel over the overall traffic having been submitted by this type. This metric can be simply defined as follows:

$$Throughput_{type} = \frac{\sum_{i\in\Theta_{type}} Traf_{out_{type,i}}}{\sum_{i\in\Theta_{type}} Traf_{in_{type,i}}} \quad (2)$$

where $type \in \Gamma = \{$voice, video, best-effort, background, DCF$\}$ and Θ_{type} is the set of connections of a given type.

Collision_Rate$_{\mathrm{TOTAL}}$ is the ratio between the total number of collisions having been detected and the total number of packets sent through the channel. This metric can be simply defined as follows:

$$Collision\_Rate_{\mathrm{TOTAL}} = \frac{\sum_{type\in\Gamma}\sum_{i\in\Theta_{type}} |Z_{type,i}|}{\sum_{type\in\Gamma}\sum_{i\in\Theta_{type}} |\Psi_{type,i}|} \quad (3)$$

where $Z_{type,i}$ is the set of collisions packets of connection i of type *type* and $\Psi_{type,i}$ the set of all packets of connection i type *type* having been transmitted through the channel. Moreover, $\Gamma = \{$voice, video, best-effort, background, DCF$\}$ and Θ_{type} is the set of connections of a given type.

Collision_Rate$_{type}$ is the ratio between the number of collisions having been detected and the total number of packets being submitted to the network by a given connection of type $type'$, where $type' \in \Gamma' = \{$voice, video best-effort, background, DCF$\}$. The metric is given by

$$Collision\_Rate_{type} = \frac{\sum_{i\in\Theta_{type}} |Z_{type,i}|}{\sum_{i\in\Theta_{type}} |\Psi_{type,i}|} \quad (4)$$

where $\Theta_{type'}$ is the set of connections whose traffic belongs to type $type'$, $Z_{type,i}$ is the set of collisions packets of connection i of traffic type $type'$ and $\Psi_{type,i}$ is the set of all packets of connection i type $type'$.

Access_Delay$_{type}$ is the average access delay to the channel having been experienced by a given type of traffic. Let $access\_delay_{type,i,k}$ be the channel access delay experienced by packet k of the i connection of type *type*. Then, we can simply define $Access\_Delay_{type}$ as follows:

$$Access\_Delay_{type} = \frac{\sum_{i\in\Theta_{type}}\sum_{k\in\Psi_{type,i}} access\_delay_{type,i,k}}{\sum_{i\in\Theta_{type}} |\Psi_{type,i}|} \quad (5)$$

where Θ_{type} is the set of connections of a given type *type*, and $\Psi_{type,i}$ is the set of packets of connection i of a given *type* having been sent over the channel.

In order to limit the delay experienced by the voice and video applications, an essential condition to guarantee the QoS required by both applications, the maximum time that a unit of voice and video may remain in the transmission buffer has been set to 10 ms and 100 ms, respectively. These time limits are in line with the values specified by standards and in the literature. A packet exceeding this upper bound is dropped. The loss rate due to this mechanism is given by the *Packet Loss Rate* (PLR).

PLR_{type} is the ratio between the lost packets and the total number of packets being submitted to the network by a given connection of type $type'$, where $type' \in \Gamma' = \{$voice, video$\}$. The metric is given by

$$\mathrm{PLR}_{type'} = \frac{\sum_{i\in\Theta_{type'}} |Y^{lost}_{type',i}|}{\sum_{i\in\Theta_{type'}} |Y^{sent}_{type',i}|} \quad (6)$$

where $\Theta_{type'}$ is the set of connections whose traffic belongs to type $type'$, $Y^{lost}_{type',i}$ is the set of lost packets of connection i of traffic type $type'$ and $Y^{sent}_{type',i}$ is the set of all packets of connection i type $type'$.

Finally, one of the most important metrics in video communications is the quality of the received video sequence as

Table 2 Video quality scale

Rating	Impairment	Quality
5	Imperceptible	Excellent
4	Perceptible, not annoying	Good
3	Slightly	Fair
2	Annoying	Poor
1	Very annoying	Bad

perceived by the end user. This has been evaluated using the *Video Quality Metric* (VQM) [12, 13]. This metric has been proved to behave consistently with the human judgments according to the quality scale that is often used for subjective testing in the engineering community (see Table 2).

Video Quality (VQM) is the average of the VQM metric whose formal definition is given by

$$\mathrm{VQM} = \frac{\sum_{i \in \Theta_{video}} \mathrm{VQM}_i}{|\Theta_{video}|} \tag{7}$$

where Θ_{video} is the set of all video connections and VQM_i is the value of the VQM metric having been obtained for the video connection i.

For all the metrics described in this section, all our measurements started after a warm-up period allowing us to collect the statistics under steady-state conditions. Each point in our plots is an average over thirty simulation runs, and the error bars indicate the 95% confidence interval.

3.3 Results

EDCA makes use of different waiting intervals as a means to provide various priority levels. These time intervals are defined by the system parameters AIFS, $\mathrm{CW}_{\min}$ and $\mathrm{CW}_{\max}$. We start our study by setting up the different system parameters under study (see Table 1). This set up allows us to define a reference point for comparison purposes with respect to the system parameters recommended by the standard [2].

Figure 4 shows the normalized throughput and the collision rate as a function of the network load and for various combinations of the waiting interval, AIFS. The AIFSs used by the various types are denoted by BK-BE-Vi-Vo, corresponding to the AIFS used for the background, best-effort, voice and video traffic, respectively. Figure 4(a) shows the throughput obtained for the voice traffic. The voice performance starts to degrade for loads as low as 0.6. The worst results are obtained for the combination 7-3-2-2, i.e., the recommended value in the standard. The best results correspond to the combinations assigning a different numerical value to each one of the AIFSs. By assigning different values; the various traffic streams do not compete simultaneously for the channel access. This is clearly demonstrated by the fact that the number of collisions reduces significantly when different values of AIFSs are used (see Fig. 4(d)). Figure 4(b) depicts the throughput for the video traffic. Contrary to the results for the voice traffic, the performance results obtained for the video traffic are worse when the AIFS used in the video flows are higher of 2. With this AIFS, the video flows have a smaller priority, and lost the protected slot with respect to the DCF-based stations. This results on an increasing number of collisions produced in the video traffic (see Fig. 4(e)). Figures 4(c) and 4(f) show the overall network throughput and number of packet retransmissions for all traffic types. The figures show a relation between the global throughput and the number of collisions. The best results are obtained for the combination 7-3-1-1. This combination introduces a second protected slot in the voice and video applications, and reduces the number of collisions. However, setting up AIFSN = 1 to these two services is incompatible with the HCCA. As already explained, the HC should be able to take the control of the channel at any time. This is to say, the HCCA should hold the highest priority over all the services to be supported by the standard.

Figure 5 shows the mean access delay and the packet loss rate for the voice and video traffic. Figure 5(a) shows the mean queuing time for the voice traffic for the various AIFS values under consideration.

The delay experienced by the voice packets is lower for the case when the AIFS assigned to the voice traffic is shorter than the one used by the video traffic. It is clear that by assigning a shorter value, the number of collisions reduces. This lower delay in the voice packet for these combinations (different AIFS for the voice and video packets) reduces the packet loss rate experienced by the voice traffic. Figure 5(b) shows the mean access delay for the video packet. The figure shows an increase in the delay when the AIFSN used in these applications are higher of 2. On the contrary, the smaller delay is obtained by the combinations 7-3-1-1. With this combination, the packet loss rate experienced by the video traffic is reduced.

Figure 6 shows the performance for the voice and video traffic as well as for the overall network as a function of the network load and for various values of the CWmin parameter. Recall that this parameter defines the initial (minimum) Backoff window size. The window size is increased after each collision. Following the same convention as above, the CWmin used for each traffic type is denoted as BK-BE-Vi-Vo. Similar to the results shown in Fig. 4(a), the performance of the voice traffic depends on the parameter settings. The use of a larger CWmin for the other applications improves the throughput of the voice traffic as well as a significant reduction on the number of collisions experienced by the voice traffic (see Fig. 6(d)). The results also show

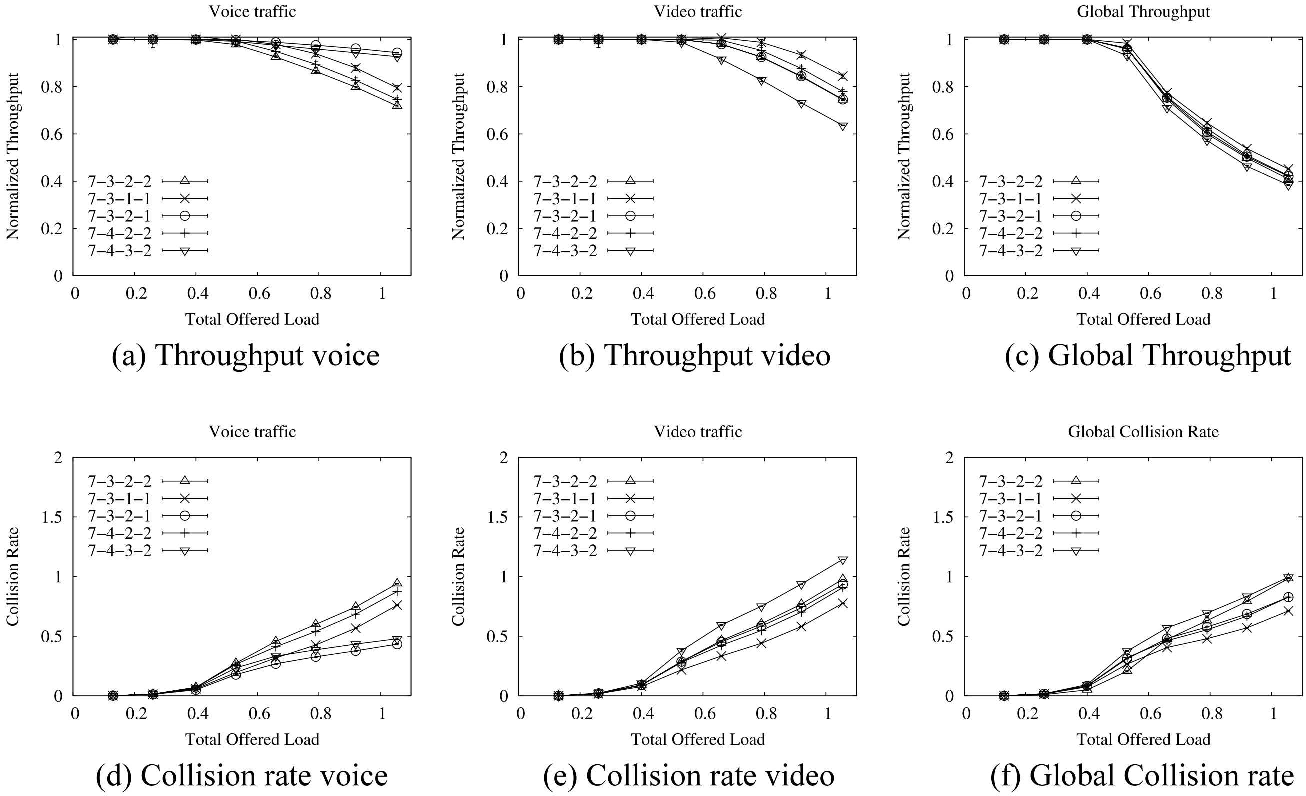
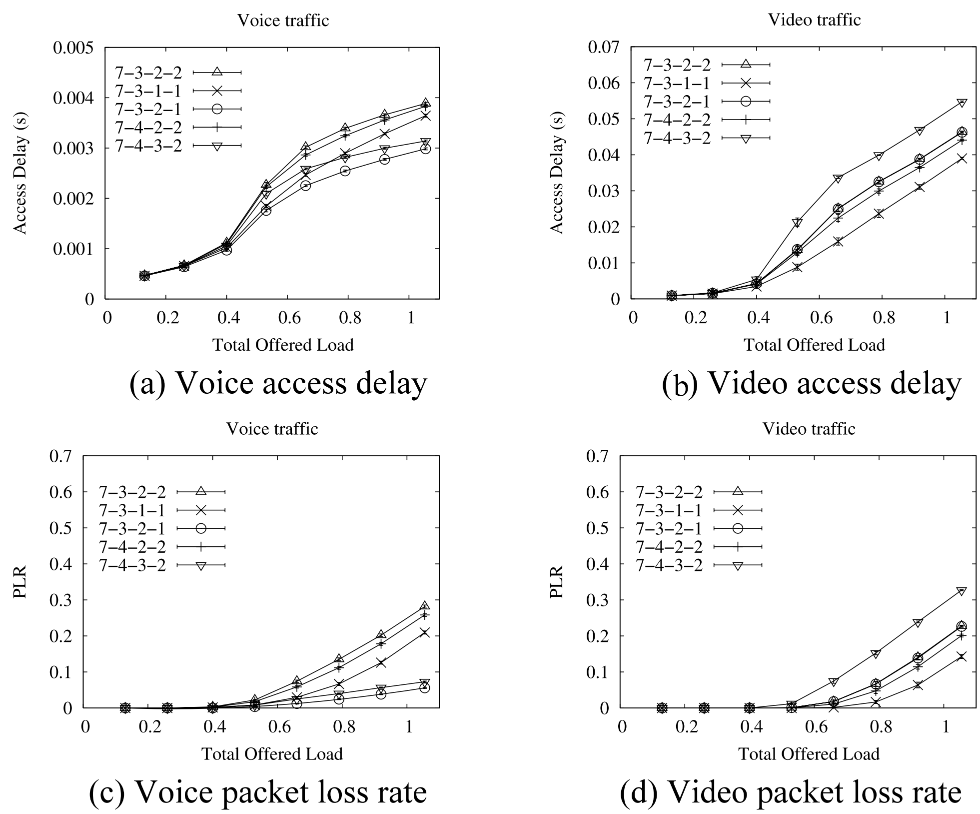
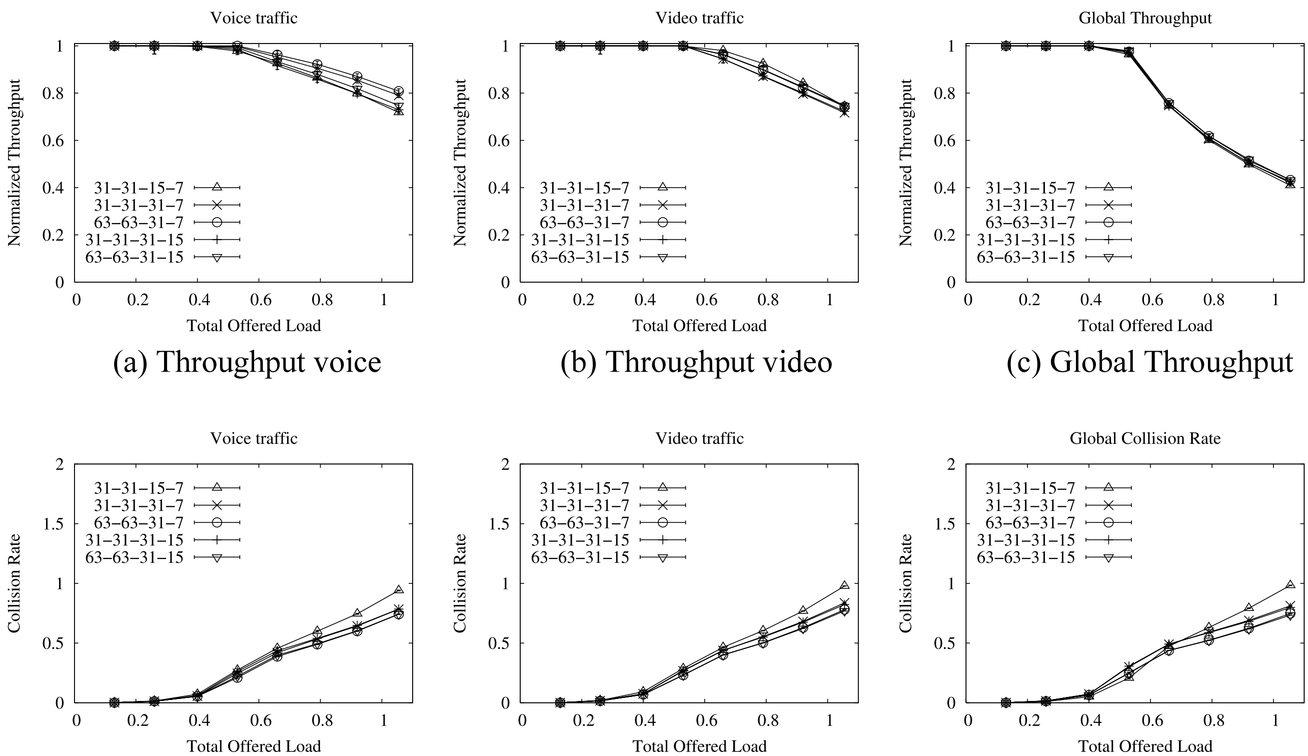


Fig. 4 Average normalized throughput and collision rate using different AIFS values

Fig. 5 Average access delay and packet loss rate using different AIFS values



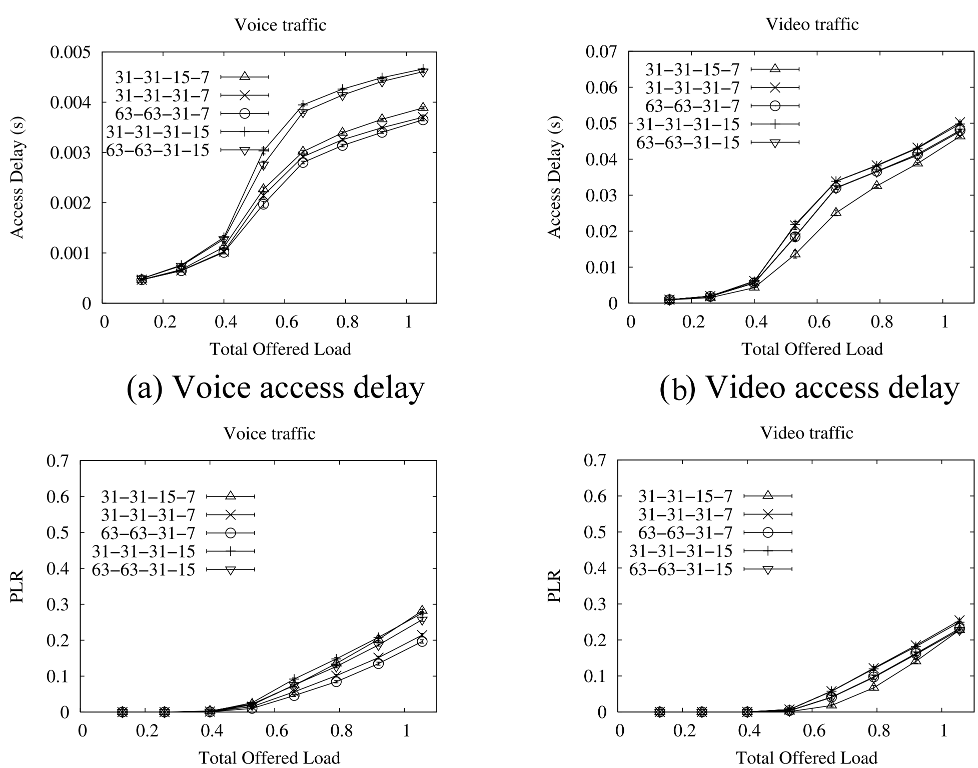


(a) Throughput voice (b) Throughput video (c) Global Throughput

(d) Collision rate voice (e) Collision rate video (f) Global Collision rate

Fig. 6 Average normalized throughput and collision rate using different CWmin values

Fig. 7 Average access delay and packet loss rate using different CWmin values



(a) Voice access delay (b) Video access delay

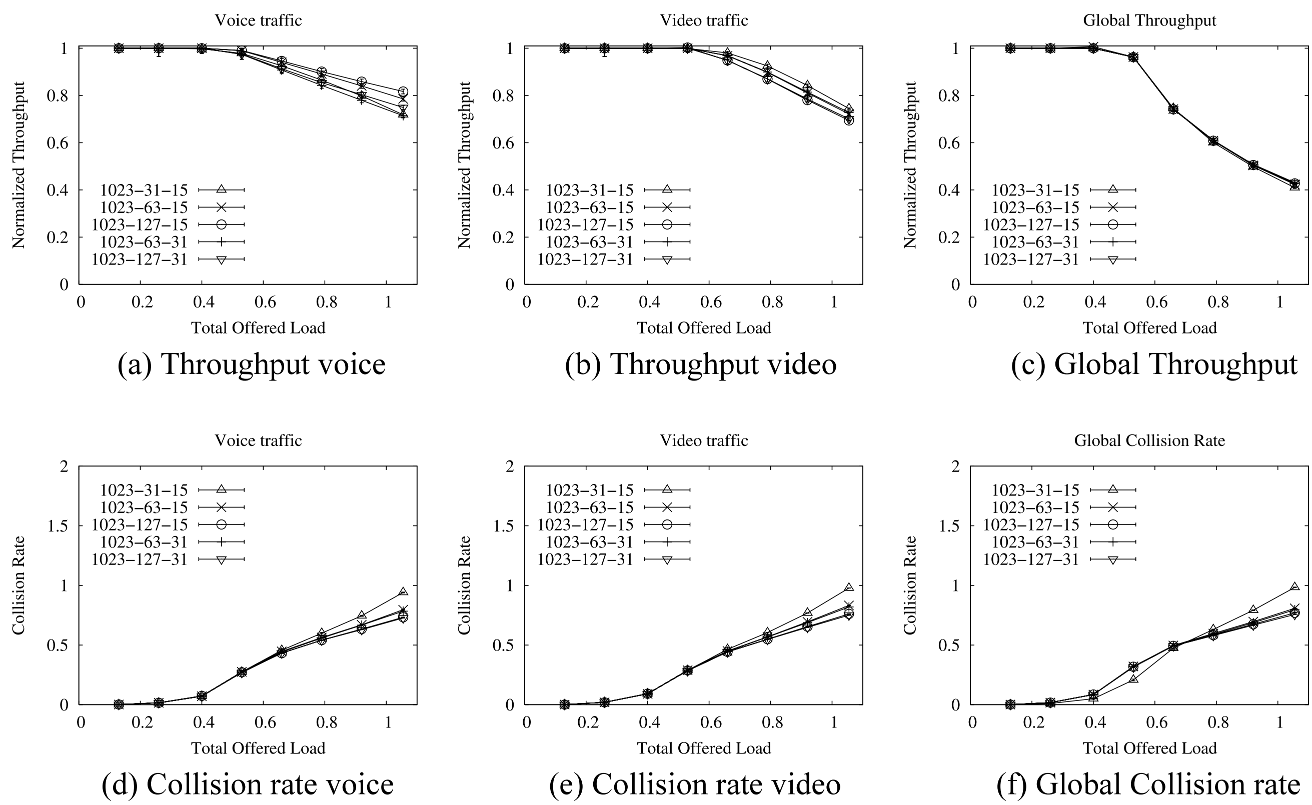
(c) Voice packet loss rate (d) Video packet loss rate that it is better to use small values for the voice traffic. The throughput for the video traffic is depicted in Fig. 6(b). The performance results for this type of traffic are very similar for all settings under consideration reducing the throughput when EDCA use larger values for the CWmin. The best results for the video traffic are obtained for the values recommended by the standard. Finally, Fig. 6(c) shows the overall network throughput. The figure shows a similar throughput for the all the studied combinations.

The mean access delay and packet loss rate for the voice and video packet using different CWmin values are shown in Fig. 7. Figures 7(a) and 7(b) show an increase in the mean access delay for the voice and video packets when EDCA uses higher sizes of CWmin. This increase in the mean access delay causes a higher packet loss rate by the video traffic.

Figure 8 shows the results when varying the CWmax parameter. The results are shown as a function of the network load and for various CWmax setting denoted as (BK and BE)-Vi-Vo. Given that this value is only used when a packet requires to be retransmitted several times, the results obtained under low loads are very similar for all combinations. Similar to the result shown in Fig. 6, Fig. 8(a) shows an increase in the voice throughput when the CWmax used in the others applications is greater. In these combinations the priority of the voice traffic is increased. However, this increase in the CWmax penalizes the other applications. Figure 8(b) shows the throughput for the video traffic. The figure shows that the best results for the video traffic are obtained for the values recommended by the standard. When the CWmax used in the video packet is increased, the throughput for this traffic is reduced. The overall network performance reported in Fig. 8(c) shows similar trend to the results obtained for all the combinations.

Figures 9(a) and 9(b) show the mean waiting time for various values of the CWmax parameter. Recall that this time is the time elapsed between the time the packet is generated by the source and the first transmission attempt. It is for this reason that the value of the CWmax does not have any effect over this metric. Respect to the packet loss rate, Fig. 9(c) shows a reduction in the number of voice packet discarded when the CWmax used in the other applications is greater. However, with these combinations, the packet loss rate for video packet is increased.

Finally, Fig. 10 shows the video quality using the VQM metric for all scenarios. Figure 10(a) shows that the 7-3-1-1 combination ensures a better quality of the video delivered to the end user. This is mainly to the lower packet loss rates produced in this combination. Also, Figs. 10(b) and 10(c) show a lower quality when the CWmin and CWmax parameters used by the video flows are increased. This worst



(a) Throughput voice (b) Throughput video (c) Global Throughput

(d) Collision rate voice (e) Collision rate video (f) Global Collision rate

Fig. 8 Average normalized throughput and collision rate using different CWmax values

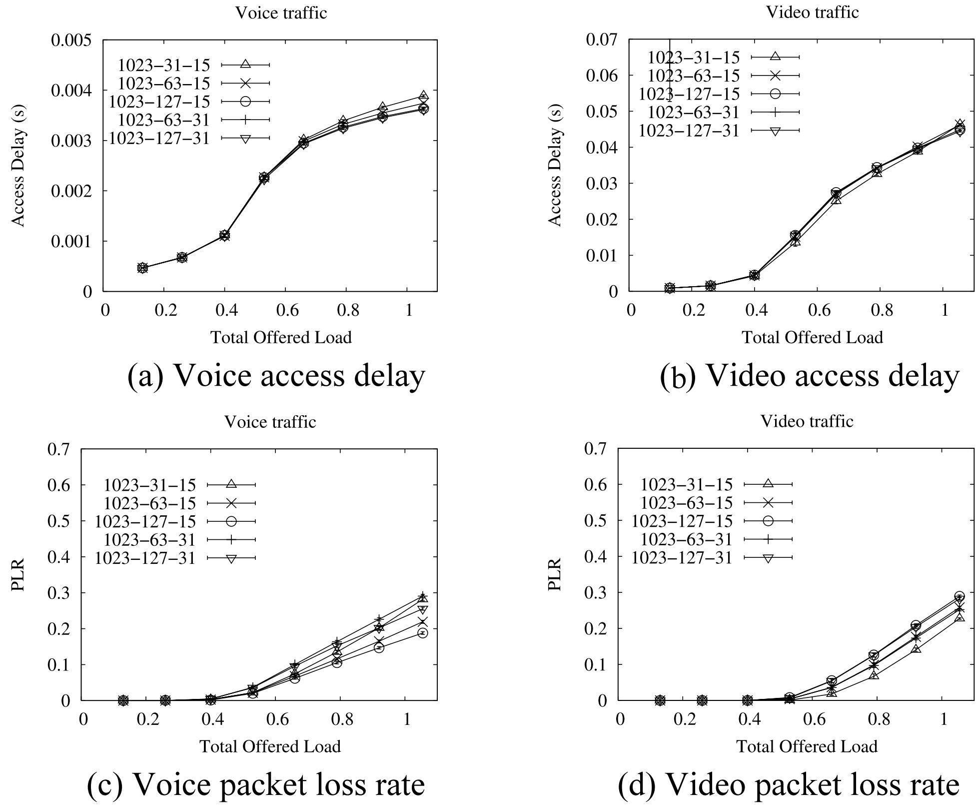


Fig. 9 Average access delay and packet loss rate using different CWmax values

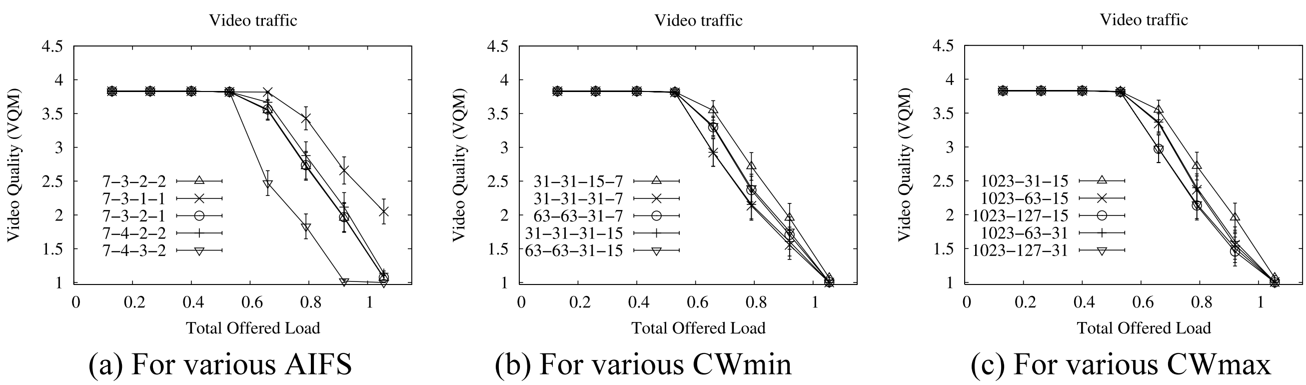


Fig. 10 Video quality using different AIFS, CWmin and CWmax values

quality is mainly due to the higher packet loss rate produced in theses cases.

4 Conclusions

In this paper, we have evaluated the IEEE802.11e. Our results show that by limiting the number of collisions, the network performance and QoS provisioning can be effectively achieved. The EDCA is unable to guarantee a good performance for loads beyond 0.6. In this latter scheme, the steeply performance drop is mainly due to the excessive number of collisions. The collisions are in turn mainly due to the fact that the AIFS parameter has been fixed to the same value for the video and voice services. Furthermore, the values used for CWmax are too short, 15 and 31 contributing to a higher collision probability. From our results, we can conclude that the values recommended by the standard have been selected to assign higher priority to the voice and video traffic in the presence of DCF stations. The compatibility with the IEEE 802.11 restrict the EDCA method to provide QoS support to time-sensitive applications.

We have also shown that the AIFS parameter plays an important role for differentiating the various traffic types. Our results suggest that it is possible to provide a better service to the voice and video traffic by using a AIFSN $= 1$.

However, setting up AIFSN = 1 to these two services is incompatible with the HCCA. As already explained, the HC should be able to take the control of the channel at any time. Regarding the CWmin parameter, our results also show that the voice performance can be improved by properly setting this parameter. The voice traffic can benefit by increasing the length of this parameter for the other traffic types. However, under this set-up the video performance are reduced. Regarding the CWmax parameter, this parameter has little effect over the voice and video performance. This is mainly due to the deadlines set up for these two traffic types, i.e., the voice and video packets are not kept for long on the buffer of the sending stations.

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