On the Effectiveness of IEEE 802.11e QoS Support in Wireless LAN: A Performance Analysis*

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Abstract. The IEEE 802.11e draft is a proposal defining the mechanisms for wireless LANs aiming to provide QoS support to time-sensitive applications, such as, voice and video communications. The 802.11e group is currently working hardly, but the ratification of the standard has a long way to go. In this paper we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) upcoming standard. We show that the defaults parameters setting recommended in the EDCA draft standard by 802.11e group do not fulfill the requirements of time-sensitive services, such as, voice and video. We show that the performance of EDCA can be improved by properly tuning its parameters.

Keywords: WLAN, IEEE 802.11, IEEE 802.11e, EDCA, QoS, Multimedia Communications and Performance Evaluation.

1 Introduction

The IEEE 802.11 Working Group is in the process of defining the IEEE 802.11e Standard: the QoS-aware standard for IEEE 802.11 WLANs [1][2]. In this paper, we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) to provide QoS guarantees. Our main results show that the default parameters setting recommended by the standard do not meet the QoS requirements of time-sensitive applications. We show that the performance of EDCA can be improved by properly tuning its system parameters. Previous studies reported in the literature have evaluated the performance of the IEEE 802.11 standard [3][4][5]. However, they have not undertaken an in-depth analysis when delay bounds to the time-sensitive applications. Furthermore, we consider a multiservice scenario, i.e., a WLAN supporting four different services: voice, video, best-effort and background traffic applications.

This paper is organized as follows. Section 2 describes the upcoming IEEE 802.11e QoS enhancement standard. In Section 3, we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) upcoming standard, when

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supporting different services, such as voice, video, best-effort and background traffic. Finally, Section 4 concludes the paper.

2 The Upcomming IEEE 802.11e Standard

The IEEE 802.11e draft standard [2] is a proposal defining the QoS mechanisms for wireless LANs for to supporting time-sensitive applications such as voice and video communications. 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 802.11 WLAN, a third coordination function is 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 the 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. On the other hand, 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

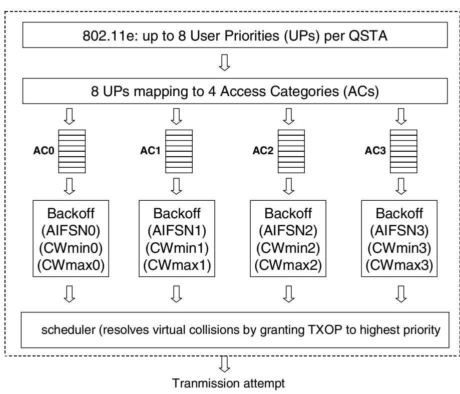


Fig. 1. EDCA

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 DCF with its own *Arbitration Inter-Frame Space* ($AIFS[AC] = SIFS + AIFSN[AC] \times SlotTime$) and its own CW[AC] (CWmin[AC] ≤ CW[AC] ≤ CWmax[AC]) (Fig. 1). If an internal collision arises among the queues within the same QSTA, the one having higher priority obtains the right to transmit. It is said that the queue that is able to gain access to the channel obtains a transmission opportunity. Each TXOP has a limited duration (*TXOPLimit)* during which an AC can send all the frames it wants**.**

3 Performance Evaluation

In this section, we carry out a performance analysis on the effectiveness of the IEEE 802.11e (EDCA) upcoming standard. We demonstrate that the defaults parameters setting recommended in the EDCA draft standard [2] by IEEE 802.11e group are not the best, when the system is supporting different services, such as voice, video, best-effort and background traffic applications. We show that the performance of EDCA can considerable be improved by properly tuning its parameters.

3.1 Scenario

In our simulations, we model an IEEE 802.11b wireless LAN (using OPNET Modeler tool 10.0 [6]) supporting four types of services: Voice(Vo), Video(Vi), Best-effort(BE) and Background(BK). This classification is on line with the IEEE802.1D standard specifications. 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 a transmission from any other station. The parameters for the wired link were chosen to ensure that the bandwidth bottle-neck of the system is within the wireless LAN.

Each wireless station operates at 11 Mbit/s IEEE 802.11b mode and transmits a single traffic type (Vo, Vi, BE or BK) to the access point. We assume the use of constant bit-rate voice sources encoded at a rate of 16 kbits/s according to the G.728 standard[7]. The voice packet size is equal to 168 bytes including the RTP/UDP/IP headers. The voice sources are randomly activated within of the interval [1,1.5] seconds from the starting time of simulation. For the video applications, we have made use of the traces generated from a variable bit-rate H.264 video encoder[8]. We have used the sequence mobile calendar encoded on CIF format at a video frame rate of 25 frames/sec. It is clear that these types of sources exhibit a high degree of burstiness characterized by a periodic traffic pattern and a high variance bit rates. The average video transmission rate is around 480 kbits/s with a packet size equal to 1064 bytes (including RTP/UDP/IP headers). Each video application begins transmitting within a random period given by t = uniform(1; 1+12/f) being *f* the frame rate. In this way, the peak periods of the source rates are randomly distributed along a GOP (*Group of Pictures*) period, a situation most likely to arise in an actual system setup. The transmission of a video frame is uniformly distributed along the time interval of a frame

(1/f). The best-effort and background traffics have been created using a *Pareto* distribution traffic model. The average sending rate of best-effort traffic is 128 kbit/s, using a 552 bytes packet size (including TCP/IP headers). The average sending rate of background traffic is 256 kbit/s, using a 552 bytes packet size (including TCP/IP headers). The traffic sources of these two latter traffic types 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.

Table 1. Parameter settings evaluated

	Vo	Vi	BE	BK
AIFSN	**2**	**2**	**3**	**7**
	2	2	4	7
	2	2	5	7
	2	3	5	7
	1	2	3	7
CW_{min}	**7**	**15**	**31**	**31**
	7	31	31	31
	7	31	63	63
	15	31	31	31
	15	31	63	63
CW_{max}	**15**	**31**	**1023**	**1023**
	15	63	1023	1023
	15	127	1023	1023
	31	63	1023	1023
	31	127	1023	1023
TXOP Limit	0	0	-	-
	3	4	-	-
	3	5	-	-
	3	6	-	-
	3	7	-	-

For all the scenarios, we have assumed that one fourth of the stations support one of the four kinds of services: voice, video, BE and BK applications. We start by simulating a WLAN consisting of four wireless stations (each one supporting a different type of traffic). We then gradually increase the *Total Offered Load* of the wireless LAN by increasing the number of stations by four. In this way, the stations are always incorporated into the system in a ratio of 1:1:1:1 for voice, video, BE and BK, respectively. We increase the number of stations 4 by 4 starting from 4 and up to 36. In this way, the normalized offered load is increased from 0.12 up to 1.12. We have preferred to evaluate a normalized offered load, rather than the absolute value. The normalized 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 I, boldface values). This will allow us to set up a base point for comparison purposes as well as to tune up the system parameters.

3.2 Metrics

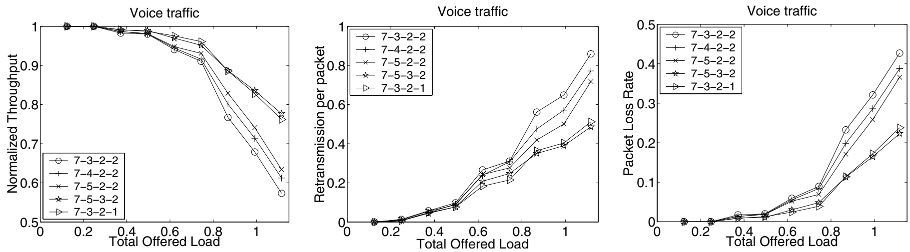
For the purpose of our performance study, the four metrics of interest are: throughput, collision rate, delay distribution and packet loss rate. To be able to compare the graphs from different levels of load (traffic patterns of different applications), we have preferred plotting the normalized throughput rather than the absolute throughput. The normalized throughput is calculated as the percentage of the offered data that is actually delivered to the destination. In order to limit the delay experienced by the video and voice applications, the maximum time that video packet and voice packet may remain in the transmission buffer has been set to 100ms and 10ms, respectively. These time limits are on-line with the values specified by the standards and in the literature. Whenever a video or voice packet exceeds these upper bounds, it is dropped. The loss rate due to this mechanism is given by the *packet loss rate due to deadline*. Our measurements started after a warm-up period allowing us to collect the statistics under steady-state conditions.

3.3 Results

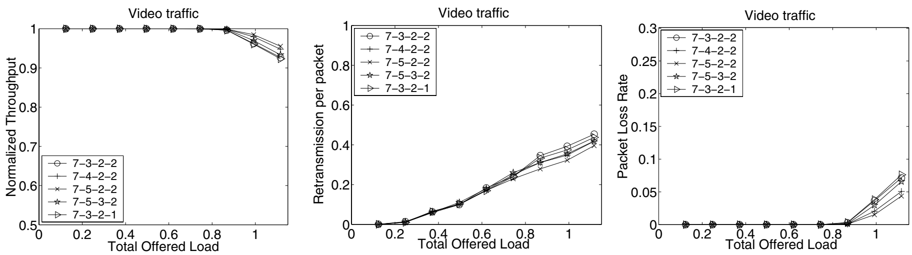
EDCA makes use of different waiting intervals as a mean to provide various priority levels. These time intervals are defined by the system parameters AIFS, CWmin and CWmax. Furthermore, the use of the extra parameter TXOP can further enhance the priority-handling scheme. We start our study by setting up the different system parameters under study (see Table I) to set up a base point for comparison purposes with respect to the system parameters recommended in the draft standard [2].

Figure 2 shows the performance results as a function of the network load and for various combinations of the waiting interval, AIFS. The metrics reported in this figure are throughput, number of collisions and the number of discarded voice and video packets as well as the overall network throughput and number of packet retransmissions. The AIFS´s 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 2.a shows the results obtained for the voice traffic. The voice performance starts to degrade for loads as low as 0.4. The worst results are obtained for the combination 7-3-2-2, i.e., the recommended value in the draft standard. The best results correspond to the combinations assigning a different numerical value to each one of the AIFS´s. 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 AIFS´s are used. Figure 2.b depicts the results for the video traffic. Contrary to the results for the voice traffic, the performance results obtained for the video traffic are similar for all the AIFS settings being considered. Figure 2.c shows the overall network throughput and number of packet retransmissions for all traffic types. The figure clearly shows similar results for all AIFS combinations under study. It is therefore possible to provide a better QoS to the real-time traffic without penalizing the overall network performance. This is confirmed by the fact that the setting 7 (BK) - 5 (BE) - 3 (Vi) - 2 (Vo) has provided the best results.

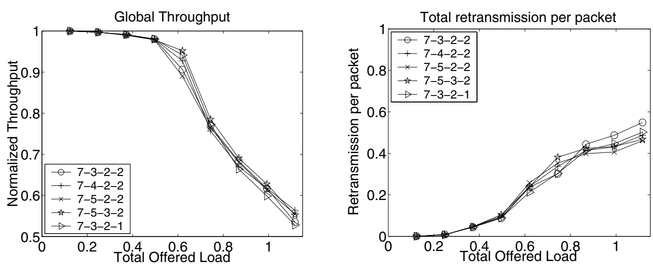
Figure 3 shows the performance for the voice and video traffic as well as for the overall network as a function of the network loads and for various values of the CWmin parameter. Recall that this parameter defines the initial (minimum) Backoff window size. The window size is increased after each collision. 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 Figure 2.a, the performance of the voice traffic



(a) Voice



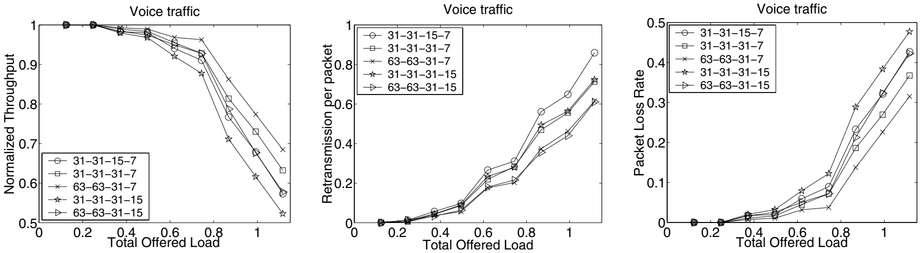
(b) Video



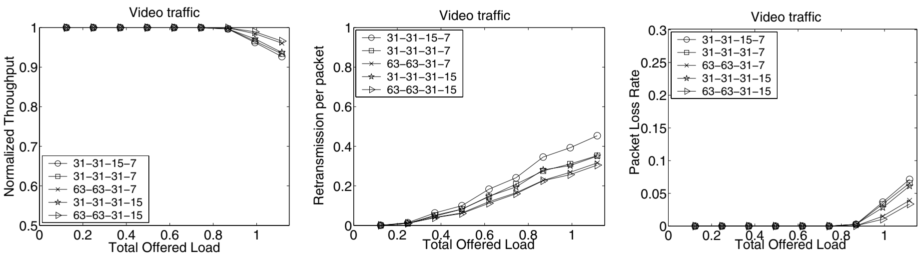
(c) Total

Fig. 2. Performance evaluation of EDCA using different AIFS values

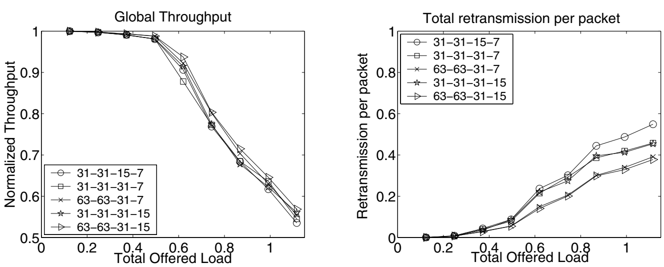
heavily depends on the parameter settings. The use of a larger CWmin improves the throughput of the voice traffic as well as a significant reduction on the number of collisions experienced by the voice traffic. The results also show that it is better to use small values for the voice traffic. The results for the video traffic are depicted in Figure 3.b. The performance results for this type of traffic are very similar for all settings under consideration. It is clear from the results that the video throughput could be improved by penalizing the background and best-effort traffic, i.e., by using larger values for the CWmin used for these two other types of traffic. In this way, the collision probability for the video traffic can be reduced. The worst results are obtained for the values recommended by the standard. From the results, it is clear that it is better to spread out the values for the CWmin for each type of traffic. This is confirmed by the fact that the setting 63-63-31-7 has provided the best results.



(a) Voice



(b) Video

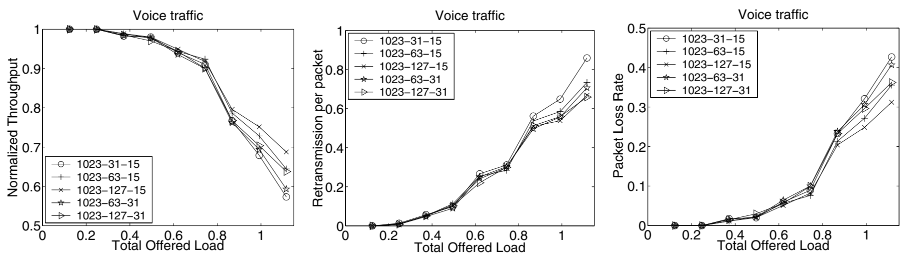


(c) Total

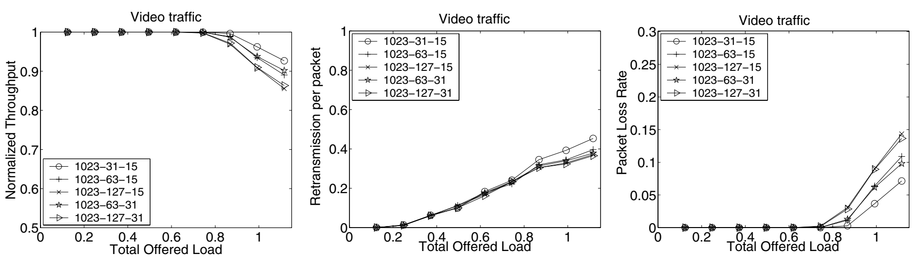
Fig. 3. Performance evaluation of EDCA using different CWmin values

Figure 4 shows the results when varying the system parameter: CWmax. The results are shown as a function of the network load and for various CWmax setting denoted as (BK y 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. It is at loads of 80% that this parameter plays an important role over the network performance. However, Figure 4.a shows that this parameter does not affect the performance of the voice traffic. This is due to the deadline defined for the voice traffic, i.e., the voice packets are discarded before reaching the CWmax. The results for the video traffic are given in Figure 4.b. Even though that by increasing this parameter, the number of video packet collisions is reduced, the number of video packets discarded increases resulting on a reduction on the video throughput. The overall network performance reported in Figure 4.c shows similar trend to the results obtained when the AIFS and CWmin have been varied.

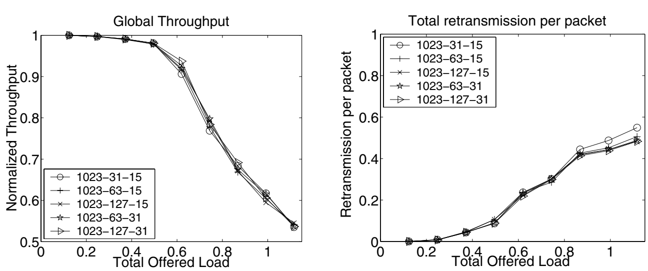
Figure 5 shows the results for various values of the TXOPLimit parameter. The use of this performance parameter has only been activated for the real-time traffic, i.e., voice and video traffic. However, the voice traffic can not benefit of this scheme. Recall that the voice packets are dropped as soon as they exceed the prescribed deadline, i.e., 10 ms. Moreover, the packetized scheme under consideration generates a voice packet every 80 ms. Therefore, no more than two voice packets will ever be ready to be transmitted. In other words, as soon as a station has sent a packet, the station will switch to the idle state. Figure 5.a shows this situation. Even more, it is clear from the results that the best results are obtained when the TXOPLimit is not used. The figure also shows that the number of collisions encountered by the voice



(a) Voice



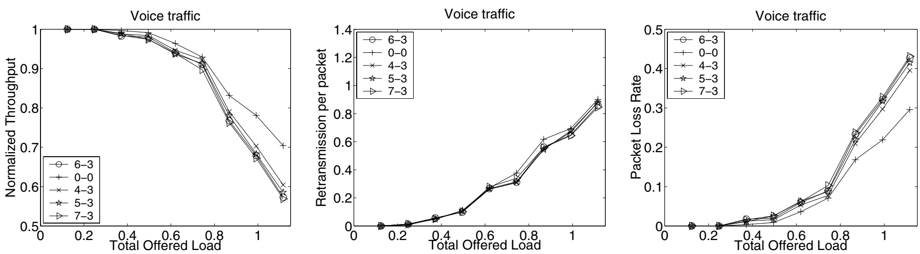
(b) Video



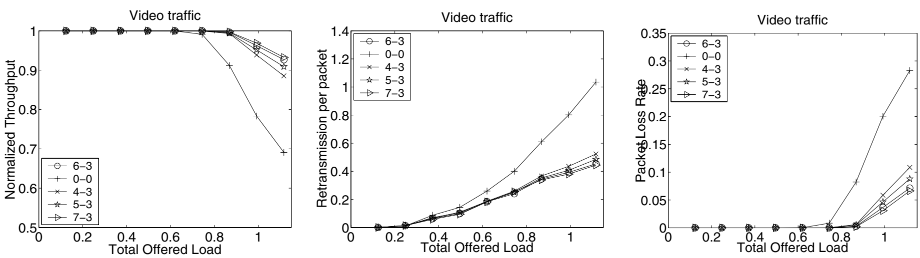
(c) Total

Fig. 4. Performance evaluation of EDCA using different CWmax values

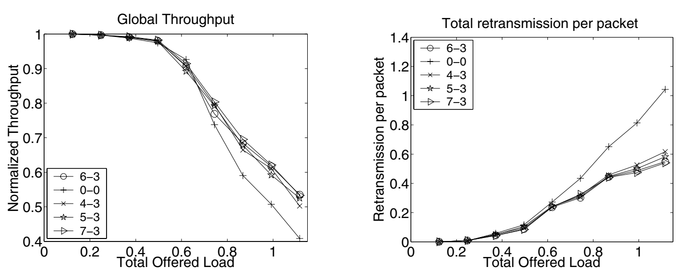
packets is independent of the TXOPLimit being used. In the case of the video traffic, the use of this facility clearly improves the performance of the video applications. This is particularly useful when transmitting video frames of the I type. The results depicted in Figure 5.b clearly show that the use of the TXOPLimit parameter reduces the number of collisions encountered by the video packets. From the results, it is also clear that the value of TXOPLimit should not be set higher than 6 ms. Figure 5.c shows that the network performance is affected by the use of the TXOPLimit. It proves a useful mechanism in managing the channel access mechanism.



(a) Voice



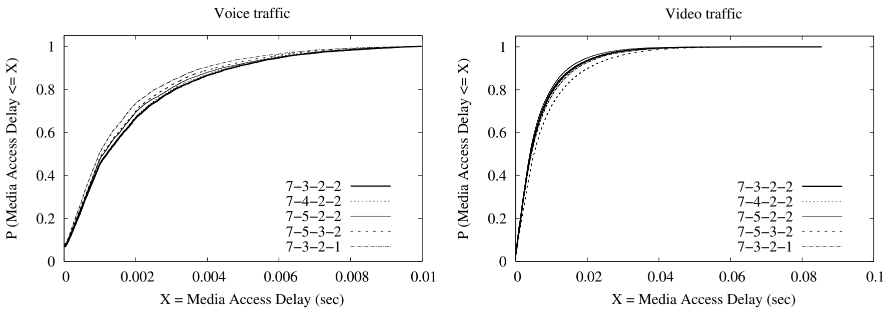
(b) Video



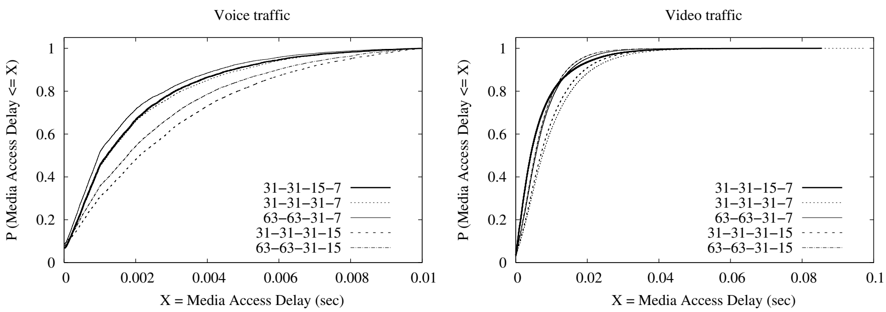
(c) Total

Fig. 5. Performance evaluation of EDCA using different TXOPLimit values

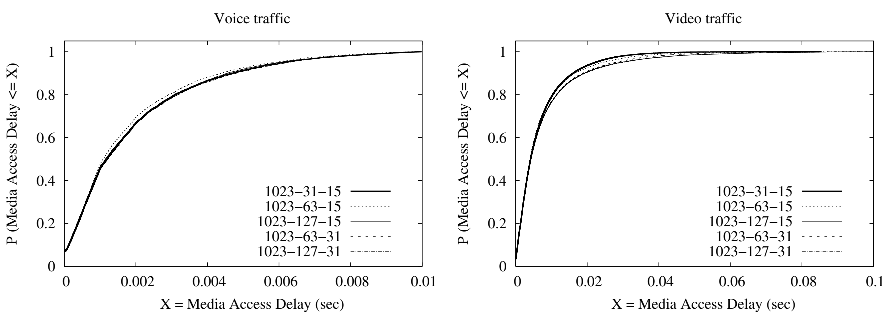
Another important metric to be reported is the cumulative distribution function for the delay experienced by the real-time applications. Figure 6 depicts this important metric for both real-time services for a network load of 0.75. From the results obtained by varying the IFS parameter, both services, voice and video, exhibit similar



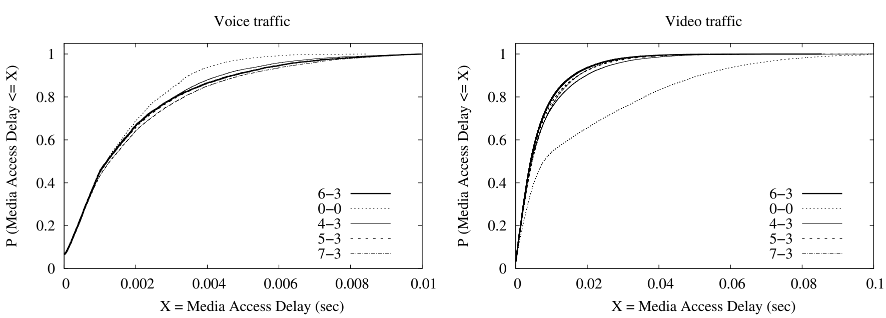
(a) AIFS



(b) CWmin



(c) CWmax



(d) TXOPLimit

Fig. 6. Performance evaluation of EDCA: CDF of Mean Access Delay

results. For the case of the voice traffic, the delays encountered are lower when different values for the AIFS are used. The setting of the CWmin has a more significant impact over the voice performance in particular when the values used for the CWmin parameter are significantly different from one another. This is expected, since the use of a shorter AIFS allows the voice traffic to promptly access the channel. Similar results are obtained when increasing the CWmin used by the BE and BK traffics. Figure 6.c confirms once again that the CWmax does not have a clear impact over the waiting time. Finally, Figure 6.d shows the results for various values of the TXOP-Limit parameter. In the case of the video traffic, it is clear that the use of this parameter can effectively reduce its waiting time. It is also clear from the figure, that the system performance is not very sensitive to the actual setting of this parameter.

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 0.75. 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 do not provide the best possible results under heavy load conditions. The performance of EDCA can considerable improved by properly tuning its parameters.

We have also shown that the AIFS 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 traffic by using a different value for the video traffic. In the same way, the video traffic can benefit from using a longer AIFS period for the BE traffic. We should point out that the overall network performance remains unchanged. It is therefore recommended to make use of the following setup: 7 (BK) - 5 (BE) - 3 (Vi) - 2 (Vo). Regarding the CWmin parameter, our results also show that the network performance can be greatly improved by properly setting this parameter. The voice traffic can benefit by increasing the length of this parameter for the other traffic types. It has further been shown that the video traffic can also benefit from the proper setting of this parameter. From our overall results, we recommend the use of the following set of values: 63 (BK) - 63 (BE) - 31 (Vi) - 7 (Vo). 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. Finally, we have examined the system sensitivity to the TXOPLimit parameter. Similar to the results obtained for the CWmax, the voice traffic does not benefit from this facility: the voice packets are discarded before the source generates the following packet. In the case of the video, it has been found that this parameter should be set to 5 or 6 ms.

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