Admission Region of Multimedia Services for EDCA in IEEE 802.11e Access Networks

Rosario G. Garroppo, Stefano Giordano, Stefano Lucetti, and Luca Tavanti

Dept. of Information Engineering, University of Pisa,
Via Caruso, I-56122, Pisa, Italy,
{r.garroppo, s.giordano. s.lucetti, luca.tavanti}@iet.unipi.it,
WWW home page: http://netgroup.iet.unipi.it

Abstract. This paper presents a simulation analysis for the evaluation of the admission region of a IEEE 802.11e network adopting the EDCA (Enhanced Distributed Channel Access) mechanism. In particular, this study gives an estimate of the number of QoS-aware applications, namely videoconference and Voice over IP (VoIP), that can be admitted to the transport service offered by the EDCA while satisfying their QoS requirements. The traffic sources adopted for the simulation are obtained from measurement campaigns led on the emulation of VoIP and videoconference services based respectively on the G.723.1 and the H.263 codecs. The results emphasize the bottleneck role played by the Access Point when services producing symmetrical traffic are conveyed over an 802.11e access network. Furthermore, the QoS parameters experienced in a mix of VoIP, videoconference and TCP traffic under EDCA are compared with those obtained when the DCF mechanism is adopted. This comparison clearly highlights the efficiency in traffic differentiation of the EDCA algorithm.

1 Introduction

While the IEEE 802.11 technology [1] is gaining wide popularity, particularly in its 11 Mbps extension known as 802.11b [2], it is also becoming clear that it can hardly be adopted to face the growth of multimedia services over wireless LANs. It does not provide any means to differentiate the transport service offered to various applications and cannot provide any guarantee on the timely delivery of frames[1].

The IEEE 802.11 working group has therefore created a new Task Group (TG11e) whose target is the definition of mechanisms for differentiating the radio channel access depending on the requirements of the supported traffic types. One of the most recent 802.11e draft standards [3] introduces two new modes of operation: an enhanced version of the legacy DCF, called EDCA, and an hybrid access method, called HCCA. EDCA, which we have focused our work on, defines four different transport modes, indicated as Access Categories (AC),

[1] The PCF mode partly accomplishes these tasks, but it is seldom implemented.

each having its own queue and MAC parameters (a detailed description is given in Section 2). The basic philosophy of this scheme is to give quicker access to medium to high priority traffic, i.e. traffic which is more sensitive to delay.

The activity carried out in the standardization body has been complemented by frenetic activity from research centers and universities that has produced a number of papers describing the behavior of the 802.11e service differentiation mechanism in the most diverse contexts[2]. While there is broad consensus about its fair capability to support real-time applications with a reasonable quality of service, it has also been shown that this does not come without pains. This relatively simple protocol provides less predictable performance than a reservation-based method and also suffers from network congestion. Scarce reliability of QoS guarantees, starvation of low priority traffic and unbalanced uplink/downlink bandwidths are the most serious drawbacks hampering the use of a distributed access mechanism such as EDCF or EDCA for multimedia services.

Just to cite a few examples, the authors of [4] remark that the EDCF could be optimized by adapting the access parameters at run-time, depending on network load and applications, and, for acceptable QoS provisioning, there should be an admission control process in place. The same conclusions are confirmed by [5], which, after performing several simulations under heavy load conditions, is able to show that low priority traffic rapidly experiences bandwidth shortage.

For these reasons, it can be argued that the support for service differentiation in wireless LAN cannot be easily achieved if disjoined from the relevant issue of admission control. In detail, it assumes a paramount relevance the determination of the number of users for each class that can be admitted to the service while satisfying the respective QoS requirements.

Works on the topic of the identification of the admission region have been recently proposed considering the plain 802.11 access method, which, however, does not provide service differentiation. In [6] the authors estimate an upper bound to the number of VoIP users that can be admitted in a 802.11(a/b) coverage area while maintaining the service level of already active voice traffic. The bound, calculated as a function of VoIP codec and length of the audio payload, proves the inadequacy of base-stations to handle a large number of VoIP calls and the inherent channel inefficiency of 802.11b at small frames sizes, as pointed out in several papers, such as [7][8][9]. The maximum number of VoIP calls is very low, and is further reduced by spatial distribution of the clients. A similar analysis, supported by extensive simulations, has been recently presented in [10], where the authors evaluate the capacity of an 802.11b system for a variety of scenarios. The capacity is found to be very sensitive to delay constraints and packet sizes, while almost independent of channel conditions. Furthermore, the impact on network capacity of different MAC-layer parameters has been discussed.

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[2] Most of the works actually refers to previous versions of the 802.11e draft, thus taking into consideration the precursor of EDCA, called EDCF (Enhanced Distributed Coordination Function).

At present, to the best of our knowledge, the only work on admission control specifically designed for 802.11e networks is exposed in [11]. The authors propose and evaluate, through simulations, the behavior of a new admission control mechanism, but do not care of estimating the capacity of the system. The 802.11e draft itself suggests a distributed admission control algorithm in which the Access Point can control the traffic load from each AC as well as each station by periodically announcing the available bandwidth for each AC. This algorithm, however, is rather complex and of difficult implementation and has therefore received scarce attention from both the research and the industrial communities.

The main contribution of our activity is the determination of an admission region for videoconference and VoIP sources multiplexed with TCP traffic in a WLAN system supporting the EDCA mechanism, thus conforming to one of the most recent available versions of the draft. The number of videoconference and VoIP sources that can be accepted in the 802.11e coverage area has been evaluated by means of simulation, considering the actual QoS requirements that can be assumed for these services. In the study, a mapping between these services and the Access Categories defined in the draft standard is assumed.

2 Description of IEEE 802.11e EDCA

In this section we present an overview of the enhancements introduced by the EDCA access mechanism to guarantee service differentiation in IEEE 802.11 wireless LANs. Note that a Basic Service Set (BSS) supporting the new priority schemes of the 802.11e is now called QoS-capable BSS (QBSS). Stations operating under the 802.11e are called QoS-capable STAtions (QSTAs) and a QoS station which works as the centralized coordinator is called QoS-capable Access Point (QAP) [3].

EDCA maintains the distributed approach of the CSMA/CA protocol as in legacy DCF, but introduces four Access Categories (ACs), each one defining a priority level for channel access and having a corresponding transmission queue at the MAC layer. Payload from higher levels is assigned to a specific AC following its QoS requirements; according to the IEEE 802.1D standard [12], eight User Priority (UP) classes are defined. The mapping from UPs to ACs is defined by the 802.11e draft standard [3].

The medium contention rules for EDCA are the same as of legacy DCF, i.e. wait until the channel is idle for a given amount of time, then access/retry following exponential backoff rules in case of collision or transmission failures. In legacy DCF, each station (or STA) must sense the channel idle for Distributed Inter Frame Spacing (DIFS) time, which is equal for all the STAs. EDCA differentiates this value, as well as the other involved in the backoff procedure, in order to perform prioritization of the medium access for the different ACs.

Each one of the four ACs in a QSTA behaves like a virtual STA, independently contending with the others to gain access to the medium. Each AC having a frame to transmit listens on the medium and, when it senses the channel idle for a period of time equal to its Arbitration IFS (i.e. AIFS[AC]), starts decrementing its backoff timer, suspending it as soon as it detects that some other STA has started transmitting. A simplified time diagram of the mechanism is reported in Fig. 1. The AIFS for a given class is equal to a DIFS plus a certain number (*AIFSN*) of time-slots (whose duration is $20\mu s$ for DSSS PHY layer), depending on the considered AC; the higher the priority class, the lower the value of *AIFSN*. The exact values are reported in Table 1, along with the other relevant parameters used in 802.11e EDCA to differentiate the channel access procedure, which will be described further on. The corresponding values for legacy DCF stations are also reported (last row of the table).

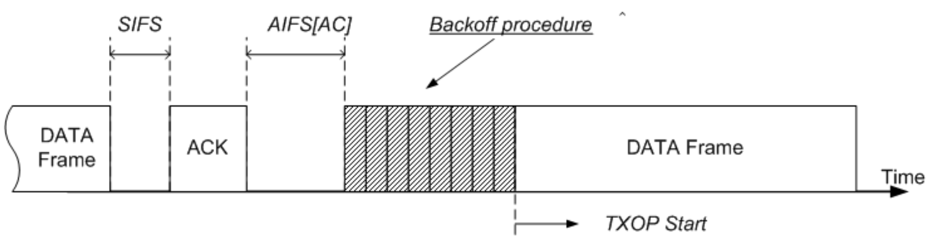

Fig. 1. The channel access procedure

Table 1. EDCA parameters

AC	*AIFSN*	CW_{min}	CW_{max}	*TXOPLimit*
0	2	7	15	3.008 ms
1	2	15	31	6.016 ms
2	3	31	1023	-
3	7	31	1023	-
DCF	2	31	1023	-

The backoff time depends on a random integer number drawn from a uniform distribution between 1 and CW[AC]+1, where CW indicates the current value of the Contention Window. The values assumed by CW are in the interval [CW_{min}[AC], CW_{max}[AC]], where CW_{min}[AC] and CW_{max}[AC] are further parameters used to differentiate the traffic treatment. Clearly, lower values of CW_{min}[AC] and CW_{max}[AC] will be assigned to high priority classes, as results from Table 1.

CW is updated after every transmission, being it successful or not. If the previous transmission was successful, CW assumes the value CW_{min} for the related AC; otherwise, CW assumes the next value in the series $2^n - 1$, up to CW_{max} value. After a predefined number of retransmission attempts, as defined in the standard, the frame is dropped.

Should two or more ACs in the same QSTA (or QAP) terminate their backoff period simultaneously, the contention is solved by an internal scheduler which assigns the grant to transmit (or Transmission Opportunity, TXOP) to the highest priority AC. Other virtual stations behave as if an external collision occurred. As in legacy DCF, when the medium is determined busy before the backoff counter is zero, the QSTA freezes its count down until the medium is sensed idle for an AIFS again.

One of the most noticeable differences between DCF and EDCA is the adoption of packet bursting, introduced in IEEE 802.11e to achieve better medium utilization. Once a station (to be precise, an AC) has gained access to the medium, i.e. it has acquired a TXOP, it may transmit more than one frame without contending for the medium again: the station (AC) is allowed to send as many frames as it wishes, provided that the total medium access time does not exceed the *TXOPLimit*[AC] bound, another parameter used to differentiate the service offered to the various ACs (higher priority ACs have higher *TXOPLimit* values). To ensure that no other station interrupts the packet bursting, a Short IFS (SIFS) is used between the burst frames[3]. Should a collision occur during this period, packet bursting terminates. Note that IEEE 802.11e draft standard establishes that the QAP may dynamically adjust the contention window parameters as well as the *TXOPLimit* for each AC by advertising the access parameters associated to each AC; this is done by setting the QoS Parameter Set Element in the beacon frame.

Summarizing, the transport service differentiation with EDCA is obtained giving different values to the CW_{min}, CW_{max}, the *AIFSN* and the *TXOPLimit* for each AC. Furthermore, these parameters may be dynamically adjusted by the QAP, and the adopted values communicated to every QSTA by means of the beacon frame. In our simulation study, the QoS Parameter Set Element is not set, thus all the QSTAs adopt the standard values of the EDCA parameters.

3 Simulation Model

The simulation study has been carried out using Network Simulator 2 (NS2) version v2.26 [13]. In particular, we used a patch developed by the Technical University of Berlin that models the EDCA mechanism [14]. The MAC layer of the patch has been updated in order to consider the four different AC parameters defined in the EDCA. This means that for each queue we can configure the values of the channel access parameters characterizing the AC, i.e. *AIFSN*, CW_{min}, CW_{max} and *TXOPLimit*. In our simulation we have considered the AC and the channel access function parameters as defined in the Draft Standard 5.0 of 802.11e [3]. The model permits to establish a mapping among traffic flows and

[3] The SIFS is the shortest allowed IFS, therefore no other STA can sense the medium idle long enough to decrement its backoff counter or to start a transmission.

the AC by means of an appropriate field in the IP header of NS2. A relevant feature of the patch is the implementation of the Contention Free Burst (CFB), which permits a station to transmit frames for a time equal to *TXOPLimit* after winning the contention.

3.1 Validation of the Simulation Model

We performed a few tests in order to validate the simulation model. The tests consisted in comparing simulation and theoretical results in a simple scenario. In particular, we focused on the ACs having the *TXOPLimit* equal to zero. This implied considering the channel access function parameters defined for AC 2 and AC 3. In these conditions, when using a simple point-to-point connection, we could evaluate the maximum goodput[4] by analyzing the time budget necessary for the transmission of a frame, as presented in [9]. The obtained results are reported in Table 2, which details the simulation and theoretical maximum goodput for two different packet sizes (the worst case, 64 bytes, and the best case, 1472 bytes). It is relevant to emphasize that the maximum goodput has been evaluated at the 11 Mbps data rate considering a single point-to-point connection, hence, without collisions. The table shows a good accordance among theoretical and simulation results, with differences within the 1% in the worst cases.

Table 2. Comparison of theoretical and simulation results for the maximum goodput

	Packet Size	Theoretical goodput	Simulation goodput	Difference (%)
AC 3	64	0.511	0.506	0.98
AC 3	1472	5.81	5.79	0.34
AC 2	64	0.555	0.549	1.08
AC 2	1472	6.05	6.03	0.33

4 Simulation Study

The simulation study is mainly aimed at evaluating the admission region for two relevant QoS-aware services such as VoIP and videoconference. The region is defined in a cartesian plane, where abscissa and ordinate components represent, respectively, the number of VoIP and videoconference sources that can be admitted in the system while guaranteeing their QoS requirements. The definition

[4] The maximum goodput is defined as the maximum rate at which user data, without considering the protocol overhead, is transferred.

of the QoS parameters is outlined in subsection 4.2, while the adopted traffic sources are presented in the next subsection. Then, the following subsections describe the simulation scenario and the obtained results.

4.1 Traffic Sources

The traffic data acquisition has been carried out in an experimental testbed, by means of the software protocol analyzer Ethereal [17]. From the acquired data, through the use of an ad hoc post processing routine, we have obtained the trace file, which holds the inter-arrival times and the sizes of the IP packets belonging to a particular traffic flow.

The voice call employed the G.723.1 codec with VAD (Voice Activity Detection), which produces information at 6.3 Kbps when the voice activity is detected. The codec fills every packet with 24 bytes (obviously, to obtain the packet size at IP or lower levels we must consider the protocol overhead). For the videoconference service, that employed the H.263 codec of Microsoft NetMeeting, we have taken two different traffic data sets, each referring to audio and video packets transmitted by one of the two involved users. The statistics on the acquired traffic are summarized in Table 3.

Table 3. Voice and Videoconference traffic characteristics

	Voice	Videoconference
Mean packet rate (pps)	21.58	58.82
Mean IP packet size (bit)	512	5664
Mean throughput (Kbps)	11.05	333.16

The length of the traffic data acquisition has been set long enough to permit an accurate estimation of the QoS parameters. Considering a packet loss rate of 10^{-3}, we have assumed that each source should transmit at least 10^4 packets. This means that, being the VoIP source the most critical (it has the lower mean packet rate, 21.58 pps), our acquisition period is about 1000 seconds (actually, 1000 seconds enable the transmission of about $2 \cdot 10^4$ packets).

4.2 QoS parameters

For the definition of the QoS parameters associated to the considered services, we refer to the ITU-T recommendations Y.1540 [15] and Y.1541 [16]: the first defines the QoS parameters and how to measure them, the second introduces the Class of Service (CoS) concept and defines six different classes. For each CoS, the Y.1541 recommendation indicates the maximum values that the QoS parameters should not exceed. Table 4 summarizes these values (NS indicates that the value for the parameter is Not Specified). The significance of the parameters considered in the table is described in the following:

- IPTD (IP Packet Transfer Delay) is the time necessary to transfer a packet from the network interface of a measurement point (e.g a transmitter) to that of the companion measurement point (e.g. the receiver).
- The Mean IP Packet Transfer Delay is the arithmetic average of IPTD for a population of interest.
- IPDV (IP Packet Delay Variation) is the difference between the IPTD and a fixed reference IPTD value, which can be assumed equal to the Mean IPTD.
- IPLR (IP Packet Loss Ratio) represents the ratio of the total lost IP packets to the total transmitted IP packets.
- IPER (IP Packet Error Ratio) can be estimated as the ratio of the total errored IP packets to the total of successful and errored IP packets.

The performance parameters are estimated considering the single packet flow, which can be identified by all or some of the following fields: the destination IP address, the source IP address, the transport protocol port and the CoS. As can be observed from the table, in Class 0 and Class 1 the upper bounds of the four considered QoS parameters are well defined. This feature let us deduce that Classes 0 and 1 have been thought for real time services. Classes 2 and 3 differ from Classes 0 and 1 only in terms of IPDV, which is not specified. Hence, Classes 2 and 3 can be adopted for data transfer requiring only IPTD constraints. Finally, Class 4 is defined for services with no strict delay constraints, such as videostreaming, while Class 5 permits to support best effort services. Features and examples of the services that can be supported are the following:

- Class 0 is for real-time, sensitive to delay jitter, high interactivity services such as Voice over IP and videoconference;
- Class 1 is like Class 0, but for a medium-low interactivity service such as some kind of Voice over IP or videoconference services;
- Class 2 is for data transfer with high interactivity, such as signaling services;
- Class 3 is like Class 2 but for low interactivity services, such as some kind of signaling;
- Class 4 is for only data loss sensitive services, such as data transfer and video streaming;
- Class 5 is for best effort services.

It is worth mentioning that user traffic can be dropped by the network when it exceeds the figures specified in the traffic contract between the user and the network operator. The packets dropped by this preventive action of congestion control performed by the network are not considered in the estimation of the IPLR.

4.3 Simulation Scenario

The simulation scenario is depicted in Fig. 2. In order to have ideal channel conditions, we set a very low distance between QAP and QSTAs. Furthermore,

Table 4. Objective values (upper bounds) for different QoS parameters for each CoS

Class	0	1	2	3	4	5
Mean IPTD $[ms]$	100	400	100	400	1000	NS
IPDV $[ms]$	50	50	NS	NS	NS	NS
IPLR	10^{-3}	10^{-3}	10^{-3}	10^{-3}	10^{-3}	NS
IPER	10^{-4}	10^{-4}	10^{-4}	10^{-4}	10^{-4}	NS

we considered high transmission rate and low latency for the wired link in order to have in the EDCA mechanism the only bottleneck of the system. The scenario is then composed by a variable number of VoIP and Videoconference users, whose traffic is generated from the data acquired in our experimental session as previously described. Each QSTA supports the EDCA mechanism and has a VoIP or videoconference session active with a companion station on the wired network. During each active session, traffic is produced in both directions. Among the sources, a single couple of stations simulates a data transfer, based on the TCP protocol, by setting up a greedy file transfer. The use of a single client-server connection for data transfer can be backed by two reasons: first, TCP sources can modify their data rate according to the congestion state of the network and, second, the issues associated to guarantee a minimum bandwidth to TCP sources are outside of the scope of this study.

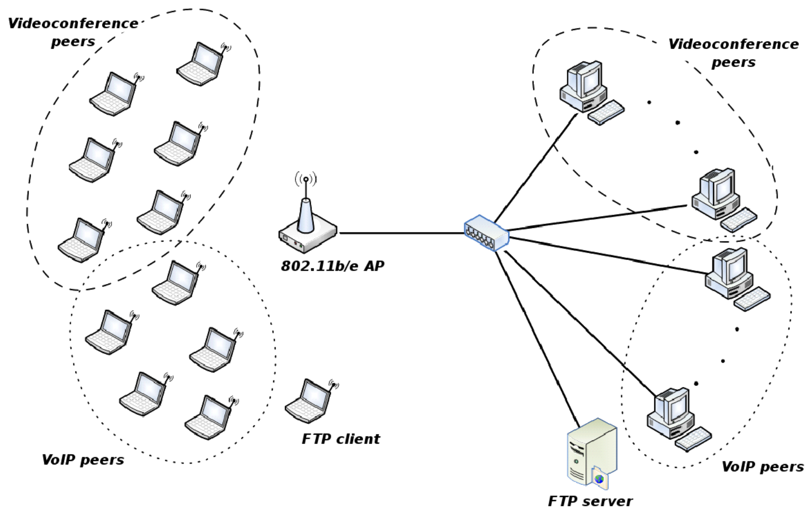

Fig. 2. Simulation scenario

For the estimation of the admission region, we assume that packets exceeding the Mean IPTD and the IPDV are dropped by the application. Then, we compute a Virtual IPLR (vIPLR) as the ratio between the total number of discarded packets and the transmitted packets of the considered traffic flow. The total number of discarded packets is obtained as the sum of the number of lost packets (due, for example, to buffer overflow or to reaching the maximum number of retransmissions at MAC layer) and of the packets exceeding the Mean IPDT and IPDV limits. Finally, we suppose that a new user service cannot be accepted by the system if its activation implies that the estimated vIPLR of at least one of the active CoS exceeds the IPLR upper bound.

The constraints for the determination of the admission region are obtained from Table 4 after associating a Class of Service to VoIP and videoconference services. In particular, we reckoned VoIP services to require the QoS parameter constraints defined for Class 0, while the videoconference those for Class 1. Then, in the mapping of EDCA's AC to the CoS, we have assumed that VoIP traffic can be transported with AC 0, while AC 1 is used by the videoconference traffic; the traffic of the TCP connection has been transmitted using AC 2.

Several sets of simulation have been carried out using different number of VoIP and Videoconference sessions simultaneously active. For each set we carried out at least 10 different simulations having diverse seeds for the random number generators in order to estimate mean values and 95% Confidence Interval for the vIPLR parameter.

The admission region has been obtained as the set of scenarios, each one characterized by a different number of VoIP and videoconference users, where the constraints on the upper value of IPLR are satisfied (i.e. a value lower than 10^{-3}, found for the 95%-CI of vIPLR). The analysis of the QoS parameters has been carried out in both directions of the traffic flows exchanged in a single session, i.e. from the mobile station to the wired one and vice versa.

4.4 Simulation Results

We start the discussion of the simulation results considering the simple scenario where only VoIP sources are active. We recall that in our simulation the TCP connection is always present. The obtained performance parameters, reported in Table 5, refer to two scenarios with zero videoconference and 21 and 22 VoIP sources (columns named "0-21" and "0-22") and are the arithmetic averages over simulations carried out with different seeds. Each parameter has been observed for both the upstream (from mobile towards wired stations) and the downstream flows, in order to point out if there is a different behavior in the two directions of traffic. For both flows we observed about 20000 packets.

From the table, we can deduce that the system is unable to satisfy the QoS requirements for 22 VoIP sources: the constraint on vIPLR is not satisfied in the downlink stream[5] (see Table 4, where the constraints are reported). Hence, a first point delimiting the admission region is represented by the scenario where

[5] The value exceeding the constraint is reported in bold.

Table 5. Simulation results for the 0-21 and 0-22 scenarios

	0-21		0-22	
	Uplink	Downlink	Uplink	Downlink
Mean IPTD (ms)	1.72	2.87	1.82	3.06
Lost packets	1.55	5.95	2.66	9.09
Discarded packets for IPTD	0.08	2.23	0.21	5.46
Discarded packets for IPDV	0.6	4.31	0.86	8.72
vIPLR (10^{-4})	1.11	6.29	1.85	**11.7**
95%-CI (10^{-4})	0.789	2.06	1.14	4.03

Table 6. Simulation results for the 5-0 scenario

	Uplink	Downlink
Mean IPTD (ms)	3.59	9.05
Lost packets	0.12	3.78
Discarded packets for IPTD	0	0
Discarded packets for IPDV	0.06	**206.38**
vIPLR (10^{-4})	0.03	**35.5**
95%-CI (10^{-4})	0.18	27.2

zero videoconference and 21 audio sources are active. It should be noted, however, that in the uplink the vIPLR is always under the 10^{-3} upper bound, even in the 0-22 scenario. This remark highlights the bottleneck role played by the QAP: considering all the four ACs, the QAP has the same probability to acquire the right to transmit a frame as whatever QSTA, while it is expected to transmit a traffic that is about N times the traffic generated by a single mobile station (with N being the number of mobile stations having an active symmetrical and bidirectional session in the coverage area of the AP). For completeness, we report Fig. 3 and Fig. 4 showing the complementary probability of IPTD and IPDV parameters observed for upstream and downstream traffic in the 0-21 and 0-22 scenarios. From these figures, we can note the different behavior of the downstream link with respect to the uplink.

A second set of simulations shows that the considered system is unable to satisfy the QoS requirements of 5 videoconference users (scenario 5-0). The details on the observed performance parameters are summarized in Table 6. In this case the number of transmitted packets for each direction is about 59000. In this case too, the maximum number of active sources satisfying their QoS requirements is imposed by the IPDV parameter in the downlink. In addition, the large packet sizes produced by videoconference sources lead to higher performance differences between uplink and downlink: the vIPLR experimented by the downlink flow is three order of magnitude higher than that observed in the

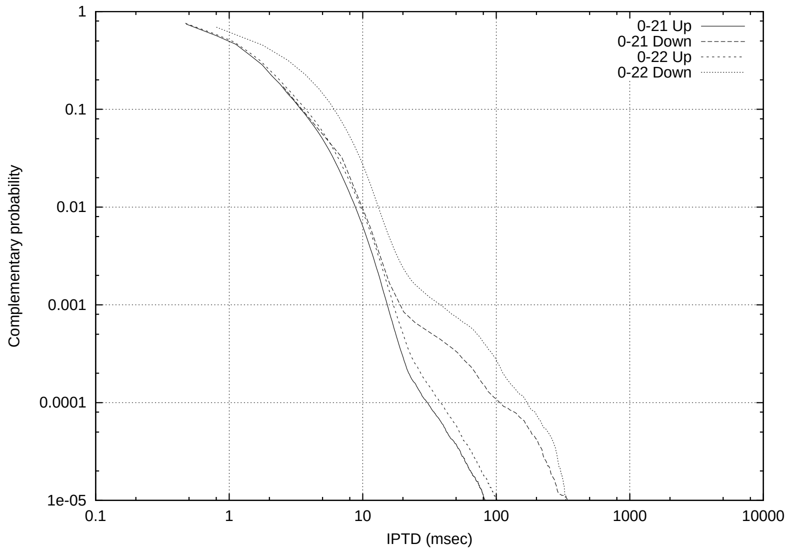

Fig. 3. Complementary probability of audio IPTD for the 0-21 and 0-22 scenarios

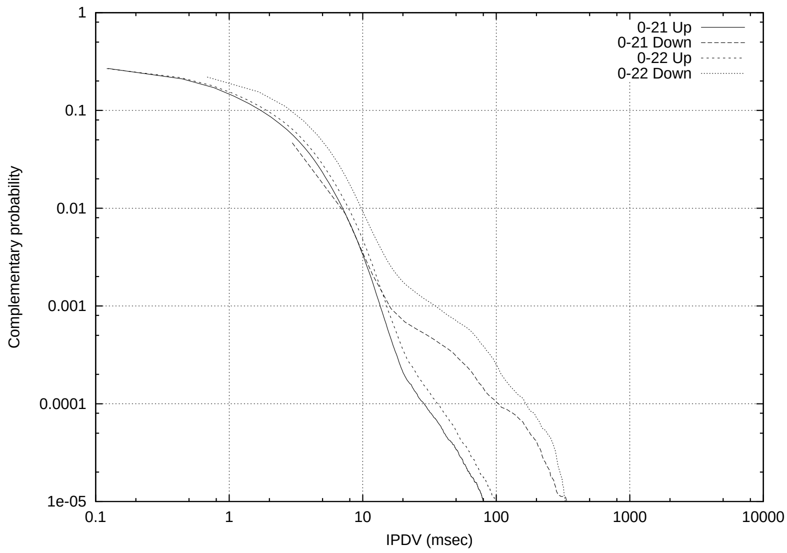

Fig. 4. Complementary probability of audio IPDV for the 0-21 and 0-22 scenarios

uplink flow ($3.55 \cdot 10^{-3}$ vs. $3 \cdot 10^{-6}$). This is a further proof of the congesting behavior of the AP. It is also worth noting that the main contribution to the vIPLR is given by the packets discarded for overcoming the IPDV upper bound.

This time, for the sake of simplicity, we report just the complementary probability of the IPTD parameter observed for upstream and downstream traffic, see Fig. 5. Similar behavior has been observed for the complementary probability of IPDV parameter.

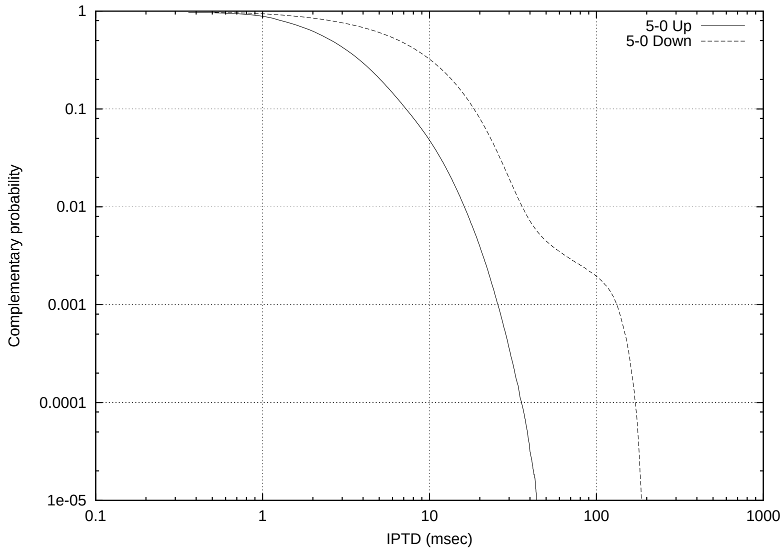

Fig. 5. Complementary probability of videoconference IPTD for the 5-0 scenario

After the evaluation of the boundaries of the admission region in case of homogeneous QoS aware sources (i.e. only VoIP or videoconference sources), we have taken into account all the other points of the admission region mixing active VoIP and videoconference sources. The results of this work are depicted in Fig. 6 where the x-axis represents the number of VoIP sources and the y-axis represents the number of videoconference sources. The grayed area indicates the admission region, in terms of joint number of voice and videoconference applications which can be activated concurrently in the QBSS satisfying the QoS requirements. The results show the low number of simultaneous videoconferences that can be admitted. A similar statement holds for the VoIP services: in spite of the low traffic produced by a single source (a VoIP service requires about 11 Kbps for each direction), only 21 VoIP sessions can be simultaneous active if we want to satisfy their QoS requirements. The number of QoS-aware services that can be simultaneously supported by the 802.11e technology is therefore surprisingly low. This is due to the constraints of QoS imposed to the transport service,

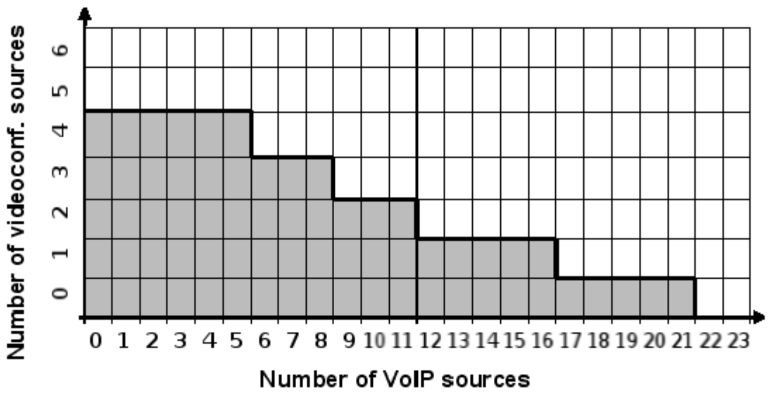

Fig. 6. The admission region

to the distributed EDCA mechanism and to the presence of just a single TCP traffic session.

4.5 Comparison between 802.11b and 802.11e

To verify the actual performance improvement of the EDCA, we carried out the same tests on the admission region with the 802.11b version [2], keeping the same QoS limits and the same traffic parameters to provide for a fair comparison. The "b" version of IEEE 802.11 does not allow to split the different streams into separate queues, but they are handled together by the same queue with no priority levels. Therefore we should expect highly variable delivery times for all the streams.

The results, reported in Table 7, have been obtained for the scenario with 0 videoconference and 1 VoIP sessions. We can easily deduce that the DCF does not allow any VoIP communication when a TCP connection is active. Moreover, we can observe that this is once again due to the bottleneck represented by the AP in the downlink: for the uplink traffic the vIPLR limit is satisfied.

The reason for the non-admission of the real-time streams is the presence of the TCP traffic, which is kept active. To confirm this statement, we ran a simulation where the data traffic has been turned off and where 10 VoIP calls have been inserted. Under this configuration the obtained values were well below the imposed limits. The figures for the considered QoS parameters in the downlink are shown in Table 8.

The difficulties in maintaining the real-time communications come from the excessive delay that the packets experience, resulting in a high drop rate. We can put this fact in relation with the presence of the TCP traffic if we consider that the TCP protocol expects a confirmation for the correct reception of each

Table 7. Statistics for the audio streams in the 0-1 bidirectional configuration

	Uplink	Downlink
Mean IPTD (ms)	3.02	3.29
Lost packets	0	0.2
Discarded packets for IPTD	0.6	17.6
Discarded packets for IPDV	4	9
vIPLR (10^{-4})	2.27	**13.1**
95%-CI (10^{-4})	1.39	34.4

Table 8. Downlink statistics for the 10 audio streams without TCP traffic

	Downlink
Mean IPTD (ms)	0.58
Lost packets	0.06
Discarded packets for IPTD	0
Discarded packets for IPDV	0
vIPLR (10^{-4})	0.03
95%-CI (10^{-4})	0.06

packet. This confirmation translates into another packet, the ACK packet, that is a further burden on the AP's queue. Hence all the packets, including multimedia ones, are subject to increased delivery times.

5 Conclusions

The aim of this work was determining, via simulation, the admission region for multimedia streams for the 802.11e EDCA protocol. In particular, we focused on VoIP and videoconference services, adopting G.723.1 and H.263 codecs respectively. The simulator has been fed with values drawn from measurements of real VoIP and videoconference sessions.

The results highlights on one hand the efficiency of transport differentiation offered by the EDCA's ACs, while, on the other hand, it has emerged the presence of a bottleneck at the AP's transmission queue (towards the mobile nodes). Hence the admission region turned out to be dependent on the overcoming of the limits imposed by the ITU-T Y1541 recommendation by the streams originated at the fixed stations.

Simulations have also been run to quantify the improvements with respect to the legacy 802.11 DCF mode, whose inability to guarantee the QoS requirements has been proven in the presence of TCP traffic.

References

1. IEEE Std for Wireless LAN - Medium Access Control and Physical Layer Specification, P802.11, November 1997.
2. IEEE Std 802.11b-1999, Part11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: Higher-Speed Physical Layer Extension in the2.4 GHz Band, September 1999.
3. IEEE Std 802.11e/D5.0, Draft Supplement to Part 11: Wireless Medium Access Control (MAC) and Physical Layer (PHY) specifications: Medium Access Control (MAC) Enhancements for Quality of Service (QoS), July 2003.
4. S. Choi, J. del Prado, S. Shankar, S. Mangold, IEEE 802.11e Contention-Based Channel Access (EDCF) Performance Evaluation, in *Proc. of IEEE International Conference on Communications*, 11-15 May 2003, Volume 2, Pages 1151-1156.
5. D. He, C. Q. Shen, Simulation study of IEEE 802.11e EDCF, in *Proc. of the 57th IEEE Semiannual Vehicular Technology Conference*, 22-25 April 2003, Volume 1, Pages 685-689.
6. S. Garg, M. Kappes, Can I add a VoIP call?, in *Proc. of the IEEE International Conference on Communications*, 11-15 May 2003, Volume 2, Pages 779-783.
7. M.G. Arranz, R. Aguero, L. Munoz, P. Mahonen, Behavior of UDP-based applications over IEEE 802.11 wireless networks, in *Proc. of the 12th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, 30 Sept.-3 Oct. 2001, Volume 2, Pages F72-F77.
8. G. Anastasi, E. Borgia, M. Conti, E. Gregori, IEEE 802.11 ad hoc networks: performance measurements, in *Proc. of the 23rd International Conference on Distributed Computing Systems Workshops*, 2003, Pages 758-763.
9. R. G. Garroppo, S. Giordano, S. Lucetti, IEEE 802.11b performance evaluation: convergence of theoretical, simulation and experimental results, in *Proc. of Networks 2004*, Volume 1, Pages 405-410.
10. D. P. Hole, F. A. Tobagi, Capacity of an IEEE 802.11b Wireless LAN supporting VoIP, to appear in *Proc. of the IEEE International Conference on Communications*, 2004.
11. D. Gu, J. Zhang, A new measurement-based admission control method for IEEE802.11 wireless local area networks, in *Proc. of the 14th IEEE Personal, Indoor and Mobile Radio Communications*, 7-10 September 2003, Volume 3, Pages 2009-2013.
12. ANSI/IEEE Std 802.1D, Information technology - telecommunications and information exchange between systems - local and metropolitan area networks - common specifications. Part 3: Media Access Control (MAC) bridges, 1998 Edition.
13. The Network Simulator - ns-2, http://www.isi.edu/nsnam/ns.
14. An IEEE 802.11e EDCF and CFB Simulation Model for ns-2, http://www.tkn.tu-berlin.de/research/802.11e_ns2.
15. ITU-T Rec. Y.1540, IP Packet Transfer and Availability Performance Parameters, December 2002.
16. ITU-T Rec. Y.1541, Network Performance Objectives for IP-Based services, May 2002.
17. Ethereal network protocol analyzer, http://www.ethereal.com.