Distributed Reservation-based QoS in Ad Hoc Networks with Internet Access Connectivity

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***Abstract*—Real-time applications introduce new requirements on wireless networks and impose quality thresholds on parameters like delay, jitter, throughput, and packet loss in order to run smoothly. This paper addresses this issue by presenting a MAC scheme that offers real-time applications the opportunity to reserve transmission time based on their QoS requirements for contention-free medium access. Our scheme, which is called EDCA with Resource Reservation (EDCA/RR), operates in a fully distributed manner, is compatible with IEEE 802.11, and provides both prioritized and parameterized QoS. In this study, we have extended EDCA/RR to handle reservation collisions and, through extensive simulations, we show that our proposal can handle multiple reservations as well as uninformed stations that lie outside the transmission range of both the transmitter and the receiver while providing QoS guarantees. We compare EDCA/RR with EDCA and our results show that, as the traffic in the network increases, EDCA/RR succeeds providing the required service to QoS-demanding applications whereas EDCA fails in this task. In addition, when the medium is lossy we show that, not only does EDCA/RR give better service to real-time traffic, but also to contending non-real-time traffic.**

I. Introduction

The IEEE 802.11-2007 [1] standard specifies the Hybrid Coordination Function (HCF) with two Medium Access Control (MAC) protocols: the contention-based Enhanced Distributed Channel Access (EDCA) and the contention-free HCF Controlled Channel Access (HCCA). EDCA is a distributed scheme so it can be used in both infrastructure and ad hoc networks. However, it cannot provide any Quality of Service (QoS) guarantees; only service differentiation. On the other hand, HCCA can provide QoS guarantees through resource reservation but it is a centralized and more complex scheme, which is useful in infrastructure networks only.

Since the standardization of HCF, we have seen Wi-Fi MultiMedia (WMM), which is a subset of EDCA, replacing the older Distributed Coordination Function (DCF) as the dominant medium access scheme for IEEE 802.11-based wireless networks. At the same time, WMM Scheduled Access (WMM-SA), which is a subset of HCCA, has been ignored by the Wi-Fi Alliance. Thus, we believe that any realistic QoS proposal for IEEE 802.11 networks should be distributed and compatible with EDCA. However, since EDCA can provide service differentiation only, the motivation of our work is to find a *distributed* QoS solution that offers both *contention-based* and *contention-free* medium access. To achieve this, we incorporated the favorable features of HCCA into EDCA, resulting in EDCA with Resource Reservation (EDCA/RR). EDCA/RR provides all existing features of EDCA, and in addition, gives applications with hard QoS requirements, the possibility to reserve transmission time for guaranteed medium access. In other words, EDCA/RR provides both *prioritized and parameterized QoS*.

Even though there are many proposals extending EDCA, most of them provide service differentiation only. Among the few studies that have the potential to provide QoS guarantees, are based on IEEE 802.11, and offer distributed solutions, we can mention the Distributed Reservation Request Protocol (DRRP) [2], which is a decentralized MAC scheme based on EDCA. The scheme is similar to EDCA/RR, allowing stations to reserve access to the medium. However, as opposed to EDCA/RR, DRRP has no distributed admission control mechanism, cannot handle reservation collisions caused by uninformed stations[1] that lie outside the transmission range of both the transmitter and the receiver, and requires applications to specify in advance how many reservation slots they need. Finally, although multi-hopping is one of the advantages of DRRP, since the routing protocol does not consider QoS requirements during the route discovery process, it might very well find a route that cannot support the requested service. On the other hand, a multi-hop extension of EDCA/RR for IEEE 802.11s [3] mesh networks, which are based on EDCA as well, could easily collaborate with the mesh routing protocol also operating at the MAC sublayer.

The Distributed end-to-end Allocation of time slots for REal-time traffic (DARE) [4] is another distributed MAC scheme that allows stations to reserve periodic time slots. As opposed to EDCA/RR, DARE is based on DCF instead of EDCA, has no distributed admission control mechanism, cannot handle uninformed stations and reservation collisions but relies on real-time applications being robust to packet loss, wastes resources by transmitting dummy packets during silent periods to prevent a false reservation release instead of having an explicit reservation deletion process, and last but not least, has a very complex and inefficient method for multiple reservations. Similar to DRRP, DARE supports multi-hopping

[1] An uninformed station is a station that is not informed of the existing reservations in the network.

but since the routing protocol does not take into account the QoS requirements, the discovered route might very well not support the requested service.

In this paper, we have extended EDCA/RR to handle reservation collisions due to stations unaware of the existing reservations in the network. Moreover, as opposed to our previous study [5] mainly focusing on describing the operation of the protocol, this study comes with a very extensive performance evaluation studying multiple reservations, uninformed stations, and the impact of increased traffic load on different types of traffic such as Voice over IP (VoIP), File Transfer Protocol (FTP), Hypertext Transfer Protocol (HTTP), and video streaming. In addition, the considered scenario is a hybrid wired/wireless network, i.e., an ad hoc network with Internet connectivity.

II. Internet Access Support for Ad Hoc Networks

Since Internet plays an important role in the daily life of many people by offering a broad range of services, a practical ad hoc network should provide access to the World Wide Web. Therefore, we have considered a scenario where wireless stations have access to the Internet. This network interconnection is achieved by using our AODV+ code [6], which extends the widely used Ad hoc On-Demand Distance Vector (AODV) routing protocol to route packets between the wireless ad hoc network and the wired Internet, through a gateway. In addition, our extended routing protocol implements three different methods for gateway discovery: reactive, proactive, and hybrid gateway discovery.

In AODV+, whenever a wireless station in the ad hoc network needs to communicate with a wired station on the Internet, the wireless station starts a route discovery process if it does not find a route to the wired station in its routing table. The route discovery process is started by broadcasting a Route Request (RREQ). If the RREQ is received by a neighboring wireless station it will be retransmitted, whereas a gateway will send a reply to indicate its presence as a gateway with a route to the wired network. Since the destination is a wired station, no wireless station will ever send a RREP. When the timer of the RREQ expires, it will be retransmitted. However, the wireless station will still not receive any RREP. After a network-wide search without any RREP, the wireless station assumes that the destination is a wired station and sends its data packets to the gateway, which in turn forwards them to the destination. As an alternative approach to waiting for a network-wide search, the gateway could respond to incoming RREQs on behalf of wired stations on the Internet.

III. QoS in 802.11-based Wireless Networks

The key idea of EDCA/RR is to combine the advantages of HCCA (allowing for resource reservation) with those of EDCA (being distributed). Since EDCA/RR is based on EDCA and keeps all of its advantageous features while extending it with resource reservation capability, EDCA/RR can be considered as a realistic scheme that can be implemented into existing systems without too much effort, requiring software modifications only. With the great void left behind by HCCA/WMM-SA when being shelved, EDCA/RR can play an important role in filling the gap between EDCA and HCCA. The detailed operation of EDCA/RR can be found in [5].

A. EDCA vs. EDCA/RR

As mentioned earlier, EDCA/RR has all the existing functionalities of EDCA plus the capability to reserve resources for high-priority traffic with strict QoS needs. Hence, if we prevent stations to reserve Transmission Opportunities (TXOPs), EDCA and EDCA/RR work exactly the same - except for delivering important control messages. While studying EDCA, we noticed that connections between two stations could be heavily delayed because control messages such as management frames, Address Resolution Protocol (ARP) frames, routing packets, and other important higher-layer packets, shared their transmission queue with data frames. This could result in long connection setup times due to control frames being transmitted after all enqueued data frames having been serviced. In order to avoid such situations, one could either use one of the four existing Access Categories (ACs), or add a new AC, especially configured for important control messages. In our EDCA/RR implementation we chose the second alternative, adding a new AC with the same high-priority access parameters as AC_VO except for TXOP limit, which was set to zero. This is necessary in order to remove the possibility of sending more than one frame during each successful medium access, as control messages are usually transmitted one at a time.

B. Enhancing EDCA/RR to Handle TXOP Collisions Caused by Uninformed Stations

In our previous work, EDCA/RR handled uninformed stations by extending the Add Traffic Stream (ADDTS) Response frame with a Traffic Specification (TSPEC) element, which contains information about the reservation in progress, and letting all stations overhear the ADDTS Response to store the included TSPEC information. This solution works fine as long as all uninformed stations lie in the transmission range of either the source or destination. However, this might not always be the case in reality. Thus, even though the goal is to inform all 2-hop neighbors of the existing reservations in the network, it is unavoidable that sometimes a station might not be able to overhear an ADDTS Response; as a consequence, there is a chance that its transmissions collide with reserved TXOPs. Here, we present a solution to this problem, which is not necessarily specific to EDCA/RR, but could also be applied to other reservation-based MAC protocols for wireless networks based on IEEE 802.11.

A TXOP owner that senses the medium busy at the start of its TXOP realizes that the ongoing transmission causing the busy medium must belong to a station that has not yet been informed of the reservation of the TXOP owner. To prevent a transmission collision, the TXOP owner may choose to skip the collided TXOP and wait until the next TXOP. Meanwhile, the station causing the collision must be notified of the reservation of the TXOP owner. In order to do that, we use those neighbors of the colliding station that are informed of the existing reservations. If the informed neighbors sense the medium busy just before a TXOP is about to start, just like the TXOP owner they realize that the transmitting station cannot be aware of the upcoming TXOP reservation. However, just by sensing a busy medium, the informed neighbors cannot know the address of the colliding station; but once the colliding transmission is finished, the neighbors can decode the frame header to get the address of the colliding station. Then, the neighbors wait until the end of the ongoing TXOP and send a *Schedule frame* to notify the colliding station about the existing reservations in the network.

The Schedule frame contains a Schedule element. Both the Schedule frame and the Schedule element are defined in IEEE 802.11e. Here, we have extended the schedule element with three fields from the TSPEC element, namely, *nominal MAC Service Data Unit (MSDU) size*, *mean data rate*, and *minimum PHY rate*. These fields are necessary as they contain information about a TXOP reservation. Moreover, the Schedule element has been extended with another field, called *TXOP Owner Address*, containing the MAC address of the station whose TXOP was corrupted. This information is needed by the destination of the Schedule frame, i.e., the colliding station, so it can update its TSPEC information for the correct TXOP owner. Thus, extended with these four fields, the Schedule frame is used to reactively notify colliding stations of the TXOP reservation that was corrupted.

IV. Evaluation

In order to evaluate the performance of EDCA and EDCA/RR, we used the popular network simulator ns-2 [7]. Although many EDCA studies using ns-2 are based on the TKN EDCA model [8], our implementation of EDCA/RR is based on an accurate and detailed EDCA implementation by Mike Moreton [9] for ns-2.26. The main reason for this choice is that the TKN model is based on the legacy 802.11 implementation in ns-2, which is reported to contain many errors, some of which are still remaining in the TKN model; instead, Moreton's model is reported to be correct [10].

The presented results are averages over 100 simulation runs, each ran for 300 simulated seconds. We were interested in studying the behavior of the network in steady state, i.e., after the transient state during which the connections are set up. After some testing, we concluded that it took somewhat less than 30 seconds until all connections had been set up, so in our simulations, the first 30 seconds are ignored.

A. Simulation Setup

The simulated scenario, illustrated in Fig. 1, consists of 25 wireless stations, a gateway, a router, an FTP server, an HTTP server, and a video streaming server. The gateway is placed in the middle of the scene and connected to the three servers via a router on the wired network. All wireless stations can communicate directly with the gateway. The placement of the wireless stations is chosen such that they are within the interference range of each other.

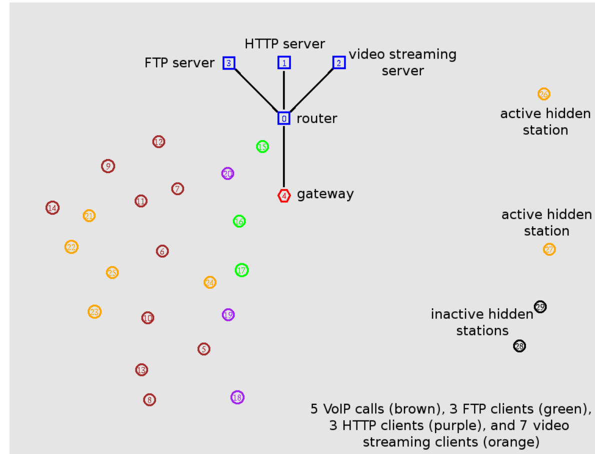

Fig. 1. A snapshot of the scenario showing the case with 5 VoIP calls

In order to evaluate the effectiveness of EDCA and EDCA/RR in protecting real-time VoIP traffic from interfering transmissions by uninformed stations, we deliberately placed four stations outside the transmission range of all wireless stations involved in VoIP communication, i.e., outside the transmission range of *both transmitters and receivers*. Moreover, to disturb the VoIP traffic even further, we deliberately chose to use applications with different characteristics; these are modelled as follows:

- VoIP: The bi-directional VoIP traffic is modelled according to a G.711 voice codec generating 160 bytes every 20 ms, resulting in 64 kbps. This kind of traffic is given high priority and its frames are sent using AC_VO. When EDCA/RR is used, these streams reserve TXOPs for contention-free medium access.
- FTP: The FTP application represents a bulk data transfer of large size, sending TCP segments equal to 1000 bytes. The application has always something to send and runs throughout the whole simulation. FTP is given low priority and its frames are sent using AC_BE.
- HTTP: The HTTP traffic is modelled according to NSWEB/SURGE [11]. HTTP is given low priority and its frames are sent using AC_BE.
- Video: The video traffic is modelled as an H.261 video codec generating 30 frames per second; each with a size equal to 1600 bytes, resulting in 384 kbps. Video is given high priority and its frames are sent using AC_VI, but these streams do not reserve TXOPs. The fragmentation threshold for UDP packets, equal to 1000 bytes by default in ns-2, is increased to prevent video packets becoming fragmented.

In our simulations, between 0 and 12 of the 25 stations are involved in VoIP communication, that is, there are 0-6 VoIP calls[2]. The choice of varying the number of real-time VoIP

[2]The notion of a VoIP call refers to two VoIP streams, one going from station A to station B, and the other going in the opposite direction.

TABLE I
SIMULATION PARAMETERS

Parameter	Value
Topology area	1000 m × 1000 m
Number of stations	25
Number of VoIP calls	0-6 (variable)
Number of FTP clients	3
Number of HTTP clients	3
Number of video streaming clients	7
VoIP packet size and data rate	160 bytes, 64 kbps
Video packet size and data rate	1600 bytes, 384 kbps
FTP packet size	1000 bytes
Transmission range	250 m
Carrier sense range	550 m
Simulation time	300 s
Warmup time	30 s

calls was made to demonstrate the ability of EDCA/RR to handle *multiple reservations*. Three stations are downloading files from the FTP server on the Internet, whereas three others are surfing the Web; i.e., they communicate with the HTTP server. Finally, seven stations are involved in downlink video streaming sessions with the video streaming server. When there are five or six VoIP calls, the uninformed stations are active and involved in video streaming transmission. Fig. 1 illustrates the case with five VoIP calls (ten VoIP stations), three FTP clients, three HTTP clients, and seven video streaming clients. The simulation parameters are summarized in Table I.

Since the operation of EDCA/RR is basically the same as that of EDCA when there are no TXOP reservations, we deliberately configured the admission control unit to allow for many reservations in order to be able to see the differences between the two schemes. The scheduler and admission control unit calculate a Service Interval (SI) equal to 25 ms and TXOPs equal to 1.23 ms for the VoIP calls under EDCA/RR.

The VoIP and video messages are encapsulated in RTP/UDP/IP packets, while the FTP messages are encapsulated in TCP/IP packets. At the network layer, AODV+ is used as the ad hoc routing protocol, using reactive gateway discovery to access the fixed network via the gateway. At the MAC sublayer the stations use either EDCA or EDCA/RR, depending on the MAC scheme under evaluation. Finally, at the physical layer they use IEEE 802.11b, or more specifically High Rate Direct Sequence Spread Spectrum using the short preamble and header mode (HR/DSSS/short).

The traffic sources are started randomly between 1.0 and 1.5 s from each other, according to a uniform distribution. There is no mobility, i.e., we simulate a common scenario where users sit in a café, university campus, conference hall, airport, or train station and access the Internet using their laptops. The VoIP calls are made within the ad hoc network, whereas the FTP, HTTP and video streaming clients communicate with the corresponding wired server.

As discussed in the IEEE 802.11e standard amendment, the unpredictable and error-prone nature of wireless media in general, and unlicensed spectra in particular, may make it impossible to provide absolute QoS guarantees. However, in a controlled environment free of external interference, it is possible to provide techniques that can provide guaranteed medium access and thus, QoS guarantees [1]. Studying EDCA/RR in both error-free and lossy media, allows us to see whether it really is capable of providing true QoS guarantees in controlled environments free of external interference and how well it fulfills the task in error-prone media. Therefore, we use the error model provided by ns-2 to simulate error-free as well as lossy media with 5% link-level packet error.

B. Performance Metrics

In comparing the ability of EDCA and EDCA/RR to provide QoS, the evaluation is done according to the following metrics:

- **The average end-to-end delay** together with its **99% confidence interval** and **Complementary Cumulative Distribution Function (CCDF)**: the end-to-end delay is calculated as the time when a packet is received at the destination's application layer minus the time when the packet was generated at the source's application layer.
- **The jitter**: calculated as the variance of the end-to-end delay.
- **The packet delivery ratio**: calculated as the number of data packets received at the destination's application layer divided by the number of data packets generated at the application layer of the source. The amount of data packets not delivered to the application (that is, one minus packet delivery ratio) is referred to as packet loss.
- **The average throughput**: calculated as the number of data bits received at the destination's application layer divided by the time the considered traffic type (VoIP, FTP/HTTP, or video) is active.

C. Simulation Results

In this section, we sometimes use the term "contending traffic" when referring to FTP, HTTP, and video traffic since these always contend for medium access as opposed to VoIP traffic, which gets contention-free medium access under EDCA/RR. Also, the results of the TCP-based traffic, i.e., FTP and HTTP, are presented together.

1) Average End-to-End Delay Analysis:
Table IIa and IIb show the average end-to-end delay and its 99% confidence interval experienced by the VoIP calls. Regarding EDCA, both tables show that the average end-to-end delay increases to very high levels as the traffic load increases. This is a typical behavior for contention-based medium access schemes like EDCA and it is this kind of behavior that we would like to avoid. Another typical, but more advantageous, behavior for random-access schemes is that they have very low medium access delays when the network load is light. This is also shown in the tables.

An interesting observation that needs to be commented is the sharp increase of the average end-to-end delay as the number of VoIP calls increases from four to five. This is because the uninformed stations are active in video streaming transmissions when there are five or six VoIP calls. Moreover, we recall that these stations lie outside the transmission range of all wireless stations involved in VoIP communication,

TABLE II
THE AVERAGE END-TO-END DELAY AND ITS 99% CONFIDENCE INTERVAL OF THE VOIP TRAFFIC

(a) 0% packet error

Number of VoIP calls	Delay (ms)		Confidence interval (ms)	
	EDCA	EDCA/RR	EDCA	EDCA/RR
1	5.16	12.59	(5.11, 5.21)	(12.33,12.84)
2	7.07	12.64	(6.99, 7.14)	(12.44,12.84)
3	11.33	12.61	(11.20,11.47)	(12.44,12.77)
4	17.93	12.63	(17.69,18.16)	(12.49,12.78)
5	88.81	12.60	(87.27,90.35)	(12.47,12.72)
6	93.51	12.60	(91.80,95.21)	(12.49,12.71)

(b) 5% packet error

Number of VoIP calls	Delay (ms)		Confidence interval (ms)	
	EDCA	EDCA/RR	EDCA	EDCA/RR
1	9.26	16.84	(9.01, 9.50)	(16.60,17.08)
2	14.30	17.00	(13.89,14.70)	(16.81,17.19)
3	22.41	17.16	(21.89,22.92)	(16.98,17.34)
4	32.00	17.21	(31.06,32.93)	(17.06,17.36)
5	60.98	17.53	(59.57,62.39)	(17.38,17.69)
6	77.01	18.03	(75.56,78.47)	(17.84,18.23)

i.e., outside the transmission range of both transmitters and receivers. Hence, the results show that these stations have a great negative impact on EDCA.

For the contention-free EDCA/RR, on the other hand, the average end-to-end delay is rather constant when the medium is error-free, whereas we see a slight increase when the medium is lossy. Moreover, we can see that EDCA/RR can handle uninformed stations since the average end-to-end delay does not increase sharply when the number of VoIP calls increases from four to five. As the results show, EDCA/RR is clearly a technique that achieves the goal of providing guaranteed medium access within the limitations of error-prone wireless media.

Let us continue to analyze the results, focusing on the average end-to-end delay for EDCA when the medium is error-free compared to when it is lossy. It is interesting to note that, when the traffic load is high (5-6 VoIP calls), the average end-to-end delay for EDCA is lower when the medium is lossy compared to when it is error-free. The main reasons for this behavior are:

a) Dropped frames are not considered in the end-to-end delay calculations. As the packet delivery ratio analysis will show in Sect. IV-C5, more VoIP frames are dropped in lossy compared to error-free media, as one would expect. This effect becomes more notable when the traffic load is high, causing the difference of the delivery ratio between error-free and lossy media to increase significantly from below 1% to 6-7%. The low packet delivery ratio for VoIP in lossy media with high traffic load shows that the combination of lossy media and high traffic load results in, not only retransmissions, but also packet drops (UDP-based VoIP frames are dropped after four transmission attempts at the MAC sublayer). The dropped VoIP frames would likely have the highest end-to-end delays if they would have been received successfully. However, since they are not considered in the delay calculations, when the medium is lossy and the traffic load is high, the average end-to-end delay becomes lower for those frames that are successfully transmitted.

b) Transmission failures (as a consequence of lossy media and high traffic load) have a greater negative impact on low-priority ACs (AC_BK and AC_BE) than on high-priority ACs (AC_VI and AC_VO), thereby making the network appear less loaded to VoIP traffic when the medium is lossy and especially when the traffic load is high. This is because the Contention Window (CW), which is doubled after each transmission failure, becomes much larger for low-priority ACs than for high-priority ACs. The default CWmin and CWmax values are 31 and 1023 for AC_BE (used by FTP and HTTP), and 7 and 15 for AC_VO (used by VoIP). For example, after three transmission failures, CW is equal to 127 for AC_BE and 15 for AC_VO, resulting in much higher medium access delays for FTP and HTTP traffic compared to VoIP traffic. Since high traffic load, in combination with lossy media, increases the probability of transmission failures, the result is longer and longer medium access delays for low-priority traffic, and in effect, decreased low-priority traffic load. The throughput analysis in Sect. IV-C4 will support this claim, showing that the throughput of low-priority FTP/HTTP traffic falls much more than that of high-priority VoIP traffic, when comparing EDCA in error-free and lossy media. For example, with six VoIP calls, the VoIP throughput falls from 593 kbps in error-free medium to 543 kbps in lossy medium giving 8% throughput fall, compared to the 98% throughput fall for FTP/HTTP falling from 306 kbps to 6 kbps!

To sum up, since the frames with largest end-to-end delays are dropped and not considered in the calculations, and since the backoff mechanism in IEEE 802.11e disfavors low-priority traffic after transmission failures, the average end-to-end delay is reported to be lower for EDCA when the medium is lossy and the traffic load is high. This behavior is not seen in EDCA/RR thanks to periodic and contention-free medium access for the VoIP calls.

2) CCDF Analysis:

Figure 2 shows the CCDF of the end-to-end delay experienced by the VoIP calls in lossy media (the case with error-free media is left out due to space limitations). Although the average value, confidence interval and variance of the end-to-end delay reveal useful information to us, its CCDF will add to our understanding about the delay characteristics of the two MAC schemes under investigation. For example, we can see that, more than 8% of the VoIP frames have an end-to-end delay over 150 ms under EDCA, resulting in a significant impact on the voice quality [12]; the corresponding value for EDCA/RR is less than 0.3%. We should also comment on the EDCA/RR curve showing a "knee point" at 25 ms, which is the value of the SI calculated by the scheduler.

Again, we see that EDCA/RR can deliver a good service independent of the traffic load. For EDCA, on the contrary,

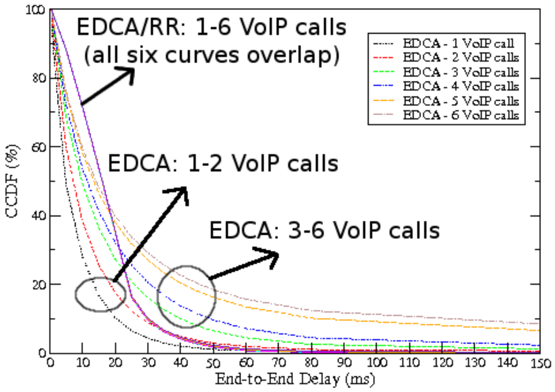

Fig. 2. The CCDF of VoIP End-to-End Delay - 5% packet error

TABLE III
THE JITTER OF THE VOIP TRAFFIC

(a) 0% packet error

Number of VoIP calls	Jitter ($10^{-4}s^2$)	
	EDCA	EDCA/RR
1	0.7	0.50
2	1.5	0.51
3	4.8	0.51
4	14	0.51
5	390	0.51
6	560	0.51

(b) 5% packet error

Number of VoIP calls	Jitter ($10^{-4}s^2$)	
	EDCA	EDCA/RR
1	6.0	1.4
2	24	1.7
3	53	2.7
4	139	2.4
5	339	3.8
6	561	5.5

the situation is totally different with an increasing amount of VoIP frames with very high end-to-end delays as the traffic load increases.

3) Jitter Analysis:
Table IIIa and IIIb show the jitter experienced by the VoIP calls. Regarding EDCA, the tables show that the jitter starts from very low values and increases by 2-3 orders of magnitude as the number of VoIP calls increases. Considering EDCA/RR, the jitter is constant low in error-free media, whereas it increases very slowly in lossy media. Moreover, we can see that high traffic load (that is, when there are five or six VoIP calls) have a great negative impact on EDCA, whereas their impact on EDCA/RR is very limited. To sum up, again the results show that EDCA/RR is able to provide QoS to high-priority traffic even during high traffic load.

4) Average Throughput Analysis:
Table IVa and IVb show the average throughput experienced by the VoIP, FTP/HTTP and video traffic. First, let us concentrate on whether the two schemes are able to provide the required throughput to the VoIP calls. Since we consider bi-directional VoIP communication, each VoIP call requires 2×64 kbps = 128 kbps. As the results show, EDCA/RR fully manages to give the required throughput to the VoIP applications both in error-free and lossy media, whereas EDCA fails to do so when the traffic load increases. With the given traffic load (three FTP clients, three HTTP clients and seven video streaming clients), EDCA can provide the required throughput to one VoIP call only in lossy media. Two VoIP calls require 256 kbps, which EDCA is close to fulfill, whereas six VoIP calls require 768 kbps, which EDCA is far from being able to provide. In fact, EDCA fails to fulfill the throughput requirements, not only in lossy, but also in error-free media when the traffic load is high. This will of course have consequences on the voice quality that the users experience.

Let us now move on to the analysis of the FTP/HTTP and video traffic. The general view revealed by the results is expected: the throughput of the FTP/HTTP and video traffic decreases with increasing traffic and error rate for both EDCA and EDCA/RR. Leaving the general view, to focus on the FTP/HTTP throughput in lossy media, it is worth to note that the FTP/HTTP throughput drops to extremely low levels for EDCA when the traffic load is high. Obviously, the TCP-based FTP/HTTP flows are starved by the UDP-based VoIP and video streams. For EDCA/RR, on the other hand, the throughput does not decrease as dramatically as for EDCA (64 kbps compared to 6 kbps when there are six VoIP calls). The reason for this is that, due to the contention-free medium access for VoIP traffic in EDCA/RR, the TCP-based flows have to contend for medium access with UDP-based video streams only; whereas in EDCA, they have to contend with UDP-based VoIP streams as well. The more streams contending for medium access, the higher is the probability for collisions and retransmissions resulting in lower throughput.

Next, we notice that EDCA/RR performs better than EDCA even though there are no VoIP calls, that is, there are no reserved TXOPs. One might have expected a comparable or similar performance since both schemes have contending traffic only. However, from Section III-A we recall that, except for the resource reservation capability of EDCA/RR, the two schemes differ in another way: EDCA/RR has an extra AC used for important control messages. Thanks to this extra AC, ARP frames, AODV packets, and ADDTS Request/Response frames are delivered faster to their destinations, resulting in faster address resolution, route discovery and connection setup and thus higher throughput. The observant reader might now wonder why the impact of this enhancement is seen in the throughput of FTP/HTTP only, and not in that much in the throughput of the video traffic. For example, when the medium is error-free, the video throughput is exactly the same (2684 kbps) for both EDCA and EDCA/RR, whereas there is a significant difference in the FTP/HTTP throughput of the two schemes: 1358 kbps compared to 1283 kbps. To explain this behavior, we recall that each video stream requires 384 kbps so seven video streams require 2688 kbps, which is basically what they receive[3]. The TCP-based FTP and HTTP flows, on the other hand, adapt their data rate according to the flow and congestion control mechanism of TCP and depending on the condition of the network, they try to transmit as fast as possible while avoiding congestion. To sum up, EDCA/RR performs better than EDCA, even when there are no TXOP

[3]The reason why the video streams do not get exactly 2688 kbps, but "only" 2684 kbps, is a small amount of packet loss due to collisions on the wireless medium. The packet delivery ratio analysis in the next section confirms this claim.

TABLE IV
THE AVERAGE THROUGHPUT OF THE VOIP, FTP/HTTP, AND VIDEO TRAFFIC

(a) 0% packet error

Number of VoIP calls	Throughput (kbps) VoIP EDCA	VoIP EDCA/RR	FTP/HTTP EDCA	FTP/HTTP EDCA/RR	Video EDCA	Video EDCA/RR
0	0	0	1283	1358	2684	2684
1	128	128	1105	1102	2683	2685
2	256	256	836	841	2681	2685
3	384	384	592	574	2672	2685
4	512	512	414	358	2658	2606
5	576	640	399	300	2507	2211
6	593	768	306	204	2453	1829

(b) 5% packet error

Number of VoIP calls	Throughput (kbps) VoIP EDCA	VoIP EDCA/RR	FTP/HTTP EDCA	FTP/HTTP EDCA/RR	Video EDCA	Video EDCA/RR
0	0	0	386	612	2670	2678
1	128	128	255	306	2651	2680
2	255	256	106	171	2591	2680
3	381	384	32	115	2274	2459
4	500	512	14	89	1748	2083
5	531	640	8	77	1247	1702
6	543	768	6	64	1138	1326

reservations, thanks to the introduction of an extra AC used for control messages.

Another point worth to comment is that, comparing the throughput of FTP/HTTP with that of video, the throughput fall is much larger for FTP/HTTP than that for video. For example, in the case of EDCA in error-free media, the throughput of FTP/HTTP falls with 76% from 1283 kbps to 306 kbps as the number of VoIP calls increases, whereas the video throughput decreases with only 9% from 2684 kbps to 2453 kbps. One reason for this is of course the higher priority given to video traffic compared to FTP/HTTP traffic. Another reason is the flow and congestion control mechanism of TCP slowing down the sending rate of FTP and HTTP traffic when the traffic load is high, whereas UDP continues to aggressively send packets at the same rate without caring about the condition of the network.

Yet another interesting observation is made by studying the throughput of the contending traffic, i.e., FTP/HTTP and video, when the medium is error-free compared to when it is lossy. With 0% packet error, the throughput of the contending traffic becomes higher for EDCA compared to EDCA/RR as the number of VoIP calls increases. On the other hand, with 5% packet error the throughput is higher for EDCA/RR. In the case of EDCA/RR and error-free medium, the throughput of the contending traffic is negatively affected by the increasing part of the medium being reserved by VoIP applications. However, in the case of lossy medium, the effect of the capacity reservation by EDCA/RR is not that negative anymore. This is because at the same time as decreasing the available resources for contending traffic, capacity reservation results in fewer traffic streams contending for medium access as the VoIP streams in EDCA/RR are not allowed to transmit at time instants other than during their reserved TXOPs. Thus, only when the medium is error-free and the amount of reserved capacity starts becoming significant (after 2-3 VoIP calls), the negative impact of not having access to the whole capacity affects the performance of the contending traffic more than the positive impact of less number of contending traffic streams. In all other cases, the contending traffic benefits from the fact that, in EDCA/RR, VoIP applications with reserved TXOPs do not contend for medium access causing collisions, backoff and retransmissions.

5) Packet Delivery Ratio Analysis:

Table Va and Vb show the packet delivery ratio (or equivalently, one minus the packet loss) experienced by the VoIP, FTP/HTTP and video traffic. Let us start the analysis by studying the packet delivery ratio for VoIP in error-free and lossy media. The results show that, the delivery ratio decreases when EDCA is used. For the contention-free EDCA/RR, on the other hand, the packet loss is negligible. Once more we see that EDCA suffers from high traffic load, with up to 29% of the VoIP frames being lost when the medium is lossy.

Let us now study the delivery ratio of the video traffic when the medium is error-free compared to when it is lossy. Here we observe that, in the case of error-free medium, the packet loss becomes higher for EDCA/RR compared to EDCA as the number of VoIP calls increases, whereas the opposite behavior is seen in lossy media. To explain this, once more we look at the delivery ratio for VoIP traffic, and note that its delivery ratio is equal or very close to 100% for EDCA/RR, whereas it was significantly lower in EDCA. Thus, it is clear that EDCA/RR takes capacity from video and gives it to VoIP; in other words, the price of nearly loss-free VoIP transmission is lower performance for the video traffic. However, despite this cost, we can see that EDCA/RR has lower packet loss than EDCA when the medium is lossy. The reason for this behavior is the same as for EDCA/RR reporting higher video throughput than EDCA in lossy media, but lower video throughput in error-free media with increasing traffic load. This was discussed in the previous section analyzing the throughput: in error-free media, where there is no external source of error and, consequently, a very low probability of packet loss, the performance of the video traffic is negatively affected by an increasing part of the medium being reserved by VoIP traffic. In lossy media with higher packet loss probability, on the contrary, reserving TXOPs for VoIP transmission actually helps improving the performance of the video traffic. This is because the consequence of TXOP reservations is, not only less time available for contending traffic, but also less traffic contending for medium access, and thus, lower probability of collisions and packet loss. The less traffic contending for medium access is a result of the policy that traffic streams with TXOP reservations are allowed to transmit only during their reserved TXOPs.

Studying the packet delivery ratio of FTP/HTTP in error-free and lossy media, we see a very high delivery ratio in error-free media for both EDCA and EDCA/RR. This is due to the reliable delivery service provided by TCP. In lossy media, on

TABLE V

THE PACKET DELIVERY RATIO OF THE VOIP, FTP/HTTP, AND VIDEO TRAFFIC

(a) 0% packet error

	Packet delivery ratio (%)					
Number of VoIP calls	VoIP		FTP/HTTP		Video	
	EDCA	EDCA/RR	EDCA	EDCA/RR	EDCA	EDCA/RR
0	-	-	99.03	99.08	99.87	99.87
1	100	100	98.78	98.80	99.85	99.88
2	99.99	100	98.59	98.65	99.77	99.89
3	99.98	100	98.59	98.41	99.51	99.91
4	99.95	100	98.76	98.17	98.97	97.01
5	90.48	100	99.07	99.37	93.71	82.32
6	77.91	100	99.09	99.84	91.90	68.12

(b) 5% packet error

	Packet delivery ratio (%)					
Number of VoIP calls	VoIP		FTP/HTTP		Video	
	EDCA	EDCA/RR	EDCA	EDCA/RR	EDCA	EDCA/RR
0	-	-	99.28	99.25	99.36	99.64
1	99.92	99.98	99.03	98.87	98.81	99.69
2	99.79	99.97	98.66	98.56	96.73	99.71
3	99.35	99.97	97.25	98.35	85.49	91.54
4	99.01	99.97	94.37	98.10	65.58	77.56
5	83.23	99.96	83.37	98.12	47.19	63.40
6	71.35	99.96	69.94	97.77	43.42	49.40

the other hand, the delivery ratio decreases for EDCA, whereas it remains rather high for EDCA/RR. This effect has a common reason as for the dramatic throughput fall of FTP/HTTP when EDCA is used in lossy media: whereas the TCP-based traffic in EDCA/RR has to contend for medium access with UDP-based video traffic only, in EDCA it has to contend with UDP-based VoIP streams as well. Hence, using EDCA/RR, the less number of contending stations in lossy media results in lower probability of collisions and retransmissions and, consequently, better performance for the contending traffic.

Moreover, the results show that, as the traffic load increases, the packet loss becomes much larger for video than for FTP/HTTP, in spite of higher priority given to video traffic. This behavior is expected and explained by the reliable delivery service offered by TCP. FTP and HTTP use the connection-oriented TCP, which retransmits dropped packets, whereas video uses the connection-less UDP, which provides an unreliable delivery service. Thus, when video packets are dropped by 802.11 MAC after four transmission attempts, FTP and HTTP packets will continue to be retransmitted by TCP, resulting in higher packet delivery ratio. Also note that the higher priority for video traffic has a slightly noticeable impact on the results: when the medium is rather reliable (low traffic load, no packet error), the higher priority makes up for the unreliable service provided by UDP, resulting in a slightly higher packet delivery ratio for the video traffic compared to that of the FTP/HTTP traffic. However, as soon as the medium becomes unreliable, this positive impact of higher priority can no longer match the positive impact of reliable delivery service of TCP.

V. CONCLUSION

In this paper we evaluate the ability of EDCA and EDCA/RR to provide QoS guarantees. Our results show that, whereas EDCA suffers from severe performance degradation with increased network load, EDCA/RR succeeds providing low and controlled end-to-end delay and jitter, the required throughput, and negligible packet loss to real-time applications. In addition, we would like to stress that, not only does EDCA/RR provide better service than EDCA in lossy wireless media regarding real-time traffic, but also when it comes to contending non-real-time traffic.

To sum up, the advantages of EDCA/RR are that i) it is based on EDCA and is *compatible* with IEEE 802.11; ii) it operates in a *fully distributed* manner offering distributed admission control, scheduling, and medium access; iii) it provides *QoS guarantees* by allowing applications with strict QoS requirements to reserve TXOPs for contention-free medium access; iv) it provides all the existing favorable features of EDCA (which performs very well during light traffic load), i.e., in addition to *contention-free* medium access and *parameterized QoS*, it also provides *contention-based* medium access and *prioritized QoS*; and v) it offers a solution that *handles uninformed stations* that lie outside the transmission range of both the transmitter and the receiver. In other words, EDCA/RR is a *realistic* approach that can be implemented into existing wireless systems to *fill the gap* between the distributed but contention-based EDCA/WMM and the contention-free but centralized and ignored HCCA/WMM-SA.

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