

Research Article

Assessment of the IEEE 802.11e EDCA Protocol Limitations when Dealing with Real-Time Communication

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The timing behavior of the EDCA mechanism defined in the IEEE 802.11e standard is analyzed. More specifically, the target of this paper is to evaluate the limitations of the highest priority level of the EDCA mechanism (voice category) when supporting real-time (RT) communication. By RT communication, we mean small-sized packets generated in periodic intervals that must be delivered before the end of the message stream period. Otherwise, the message is considered to be delayed and a deadline loss occurs. We have assessed the EDCA mechanism considering an open communication environment, where both RT and non-RT stations share the same frequency band. Furthermore, a realistic error-prone model channel was used to measure the impact of interferences against an error-free channel. We show that, for most part of the evaluated scenarios, when using the default parameters of the EDCA mechanism both the number of packet losses and the average packet delays forecast an unacceptable number of deadline losses. However, if adequate Contention Windows (CW) parameters are configured in the set of RT stations, it becomes possible to adequately handle RT traffic. As a conclusion of this paper, we present some potential future directions toward improved QoS in wireless networks.

1. Introduction

In the last few years, there has been a growth in the use of wireless technologies in several application domains that require a trustworthy Quality of Service (QoS). Driving examples range from voice over IP (VoIP) to Networked Control System (NCS). For such type of application domains, the support of reliable communication is one of the major requirements. For instance, in automation systems, real-time (RT) control data must be periodically transferred between sensors, controllers and actuators according to strict transfer deadlines.

The IEEE 802.11e amendment, incorporated in the revised version of IEEE 802.11 standard [1], intends to provide differentiated levels of QoS to the supported applications. A significant number of studies have evaluated the behavior of this standard considering typical multimedia traffic requirements. That is, requirements usually applied

for transferring voice and video streams together with background traffic. However, when the communication services are used to support RT applications, specific communication requirements must also be considered, including RT and reliability constraints [2].

Besides, a relevant aspect that must be considered when addressing wireless networks, is that all the stations share the access to the same radio channel, as the medium is an *open communication environment*. Thus, any new participant can try to access the medium at any instant according to the MAC rules and establish its own communication channels. Furthermore, the wireless communication environment is susceptible to interferences, not only from devices using the same communication technology, but also from other technologies working in the same frequency band [3]. As a consequence, the system load cannot be predicted at system setup time, nor can it be effectively controlled during the system run-time. Therefore, when dealing with

RT communication in wireless networks, it is of utmost importance to consider that the communication medium is shared with external traffic, that may unpredictably overload the communication medium.

Most published works on the performance analysis of IEEE 802.11 networks are focused on typical metrics used in multimedia domain. Moreover, the impact of varying network conditions, which is one of the main challenges to ensure QoS support in IEEE 802.11 networks as identified in [4], are usually not considered. A relevant exception is [5], where the authors have experimentally assessed the impact of background traffic upon RT traffic in a wireless communication environment.

In this paper, we evaluate the timing behavior of the IEEE 802.11e EDCA mechanism by simulation. This paper is an extended version of [6]. Its main target is to assess the impact of external traffic upon the behavior of the voice category, when this access category is used to transfer RT traffic. By RT traffic we mean small sized packets, generated in periodic intervals. We also consider both the case of error-free and error-prone channels. That is, the effect of typical interferences within industrial environments (e.g., EMI) are taken into account and compared to the ideal environment behavior. Additionally, we investigate how the RT traffic can be more adequately handled by tuning four of the EDCA parameters: AIFSN, TXOP, aCW_{min}, and aCW_{max}.

The remainder of this paper is organized as follows. In Section 2, we describe the basics of the IEEE 802.11e MAC mechanism. In Section 3, we carefully analyze relevant RT simulation scenarios. These simulation scenarios target the analysis of the EDCA timing behavior when supporting RT traffic. In Section 4, state-of-the-art and future directions for the support of RT communication in open communication environments are drawn. Finally, the paper is concluded in Section 5.

2. IEEE 802.11 Medium Access Mechanisms

The Carrier Sense Multiple Access (CSMA) mechanism defines a probabilistic MAC family of protocols, where stations contending for the access to a shared medium must listen before transmitting. Basically, this family of protocols has the following behavior.

- (i) When a station wants to transmit, it listens to the transmission medium.
- (ii) If the medium is idle, the station will start the transmission (either immediately, or after a defined interval, depending on the specific protocol).
- (iii) If the medium is busy, that is, another station is transmitting, the station will defer its transmission to a later time instant that depends on the specific protocol.
- (iv) A collision will occur whenever two (or more) stations sense the medium idle and decide to simultaneously transmit.

The CSMA medium access mechanisms that are implemented by different communication protocols differ on

how the waiting time intervals before transmitting are evaluated, either after sensing the medium idle, or before re-transmitting after a collision.

The medium access mechanism of the IEEE 802.11 standard [1] is a CSMA with *Collision Avoidance* (CSMA/CA), also called *Distributed Coordination Function* (DCF). More precisely, the IEEE 802.11 MAC sublayer introduces two medium access coordination functions, the mandatory DCF and the optional Point Coordination Function (PCF) (an additional mechanism, RTS/CTS, is defined in the IEEE 802.11 standard to solve the hidden terminal problem and to adequately handle the transmission of long messages. For further details, please refer to [1]). DCF is the basic IEEE 802.11 mechanism, where stations perform a so-called *backoff procedure* before initiating a transmission. When a station wants to transmit, it first senses the medium (carrier sensing); if the medium remains idle during a specific time interval called DIFS (Distributed Interframe Space) it immediately starts the transmission. Otherwise, the station selects a random time called *backoff time*. The duration of this time interval is a multiple of the Slot Time (ST), which is a system parameter that depends on the characteristics of the physical layer (PHY). The number of slots is an integer in the range of $[0, CW]$, where CW (contention window) is initially assigned as aCW_{\min} . A *backoff counter* is used to maintain the current value of the backoff time.

In this case, stations keep sensing the medium (listening) for this additional time, after detecting the medium as idle for a DIFS interval. If the medium gets busy due to interference or other transmissions while a station is down-counting its backoff counter, the station stops down-counting and defers the medium access until the medium becomes idle for a DIFS interval again. A new independent random backoff value is selected for each new transmission attempt, where the CW value is increased by $(oldCW \times 2 + 1)$, with an upper bound given by aCW_{\max} . As soon as the backoff counter reaches zero, the station can retry its transmission (Figure 1).

The DCF access method imposes an idle interval between consecutive frames, which is called the Interframe Space (IFS). Different IFSs are defined in order to impose different priorities to multiple frame types as following: SIFS (Short Interframe Space), PIFS (PCF Interframe Space), DIFS (Distributed Interframe Space), and EIFS (Extended Interframe Space). SIFS is the shortest of the interframe spaces and it is used for ACK frames. Only stations operating under the Point Coordination Function (PCF) will use PIFS. DIFS is used by stations operating under the DCF mechanism to transmit data frames and management frames. EIFS is used in communication-error conditions.

The main goal of the IEEE 802.11e amendment is to support QoS. It incorporates an additional coordination function called Hybrid Coordination Function (HCF) that is only used in QoS network configurations (Figure 2). The HCF mechanism schedules the access to the channel by allocating transmission opportunities (TXOP) to each of the stations. Each TXOP interval is defined by a starting time and a maximum duration, that is, the time interval during which the station keeps the medium access control.

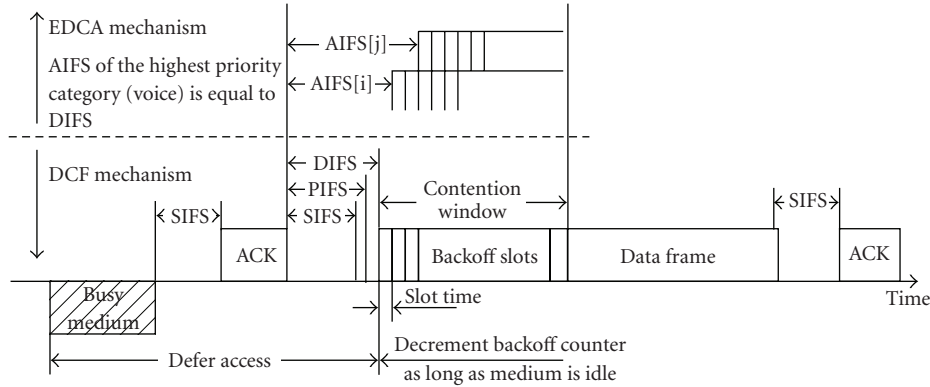


FIGURE 1: Interframe spaces in the DCF and EDCA mechanisms.

Consequently, multiple frames may be transmitted within an acquired TXOP. It may be allocated through one of two access mechanisms specified by the HCF: the *Enhanced Distributed Channel Access* (EDCA) and the *HCF Controlled Channel Access* (HCCA) [1].

The EDCA mechanism was designed to enhance the DCF mechanism providing differentiated transmission services with four access categories (AC). Each frame arriving at the MAC layer with a defined priority will be mapped into one of the four ACs. These ACs are based on the eight priority levels of IEEE 802.1D standard, as follows: priorities 1 and 2 for background traffic (BK); priorities 0 and 3 for best effort traffic (BE); priorities 4 and 5 for video traffic (VI); and, finally, priorities 6 and 7 to voice traffic (VO) that is the highest priority level.

Different levels of service are provided to each of the ACs, based on three independent mechanisms: the Arbitration Interframe Space (AIFS), the Transmission Opportunity time interval (TXOP), and the Contention Window (CW) size. For a station operating under EDCA, each frame will wait that the medium remains idle during an AIFS[AC] interval. The duration of the AIFS[AC] interval is given by

$$\text{AIFS[AC]} = \text{AIFSN[AC]} \times \text{aSlotTime} + \text{aSIFSTime} \quad (1)$$

where the AIFSN[AC] is a positive integer that must be greater than or equal to 2 for all stations, except for the QoS Access Points (QAPs), where it will be greater than or equal to 1. The default parameters defined for the EDCA mechanism are presented in Table 1. The aCWmin and aCWmax parameters depends on the characteristics of the physical (PHY) layer, for example, in the IEEE 802.11a PHY mode aCWmin = 15 and aCWmax = 1023. Figure 1 shows the relationships between the multiple AIFSs in the DCF and EDCA mechanisms. It is worth mentioning that default AIFSN value for the voice category is 2. Thus, AIFS[VO] = AIFS[VI] = DIFS.

3. The Simulation Analysis

Traditionally, the analysis of the RT communication behavior should be made for worst-case scenarios. A common specification for the delay requirement is through the use of a

deterministic delay bound, where $D_i \leq D_{\max} \forall i$, D_i being the delay of each message i , and D_{\max} the upper bounded delay specified by the application. Usually, this upper bounded delay is also called deadline. However, worst-case scenarios in probabilistic medium access networks, as it is the case of IEEE 802.11 networks, address only rarely occurring cases. The analysis of such cases may only be relevant for safety-critical applications.

Instead, this paper is focused on the timing analysis of the average communication behavior. The main reason for this option is that the target applications are usually loss tolerant in what concerns the loss of some message deadlines. For instance, the transfer of a video stream may be specified to tolerate a maximum of 10% deadline loss rate, if the lost frames are “adequately” spaced. Other examples of relevant loss tolerant applications are Networked Control Systems (NCSs) scheduled according to the (m,k)-firm model [7], or the support of VoIP applications, where an average packet delay below 150 ms and an average jitter <50 ms are acceptable for most user applications [8]. On the other hand, packet loss rates up to 3% are generally acceptable for VoIP applications.

In the case of loss tolerant applications, the RT behavior can be specified through the use of statistical metrics. For example, the statistical delay bound can be represented by $\text{Prob}(D_i \leq D_{\max}) \geq Z_{\min}$, where D_i and D_{\max} are defined as above, and Z_{\min} is the lower bound for the probability of successful and timely delivery. Adequate values for Z_{\min} within the industrial environment can be defined among 0.95 to 0.98, when dealing with soft RT applications.

The proposed scenarios analyze the behavior of the highest access category (*voice*), when this category is used to transmit RT data (small-sized packets in periodic intervals) from RT stations. The communication environment is shared with external traffic sources that are out of the sphere-of-control of the RT architecture. These RT periodic data exchanges are intended to model both sensor messages sent to plant controllers, and output messages sent from plant controllers to the actuators.

There are a number of analytical models to evaluate the EDCA mechanism [9–17]. However, the majority of these models assume simplifying approaches. Common examples

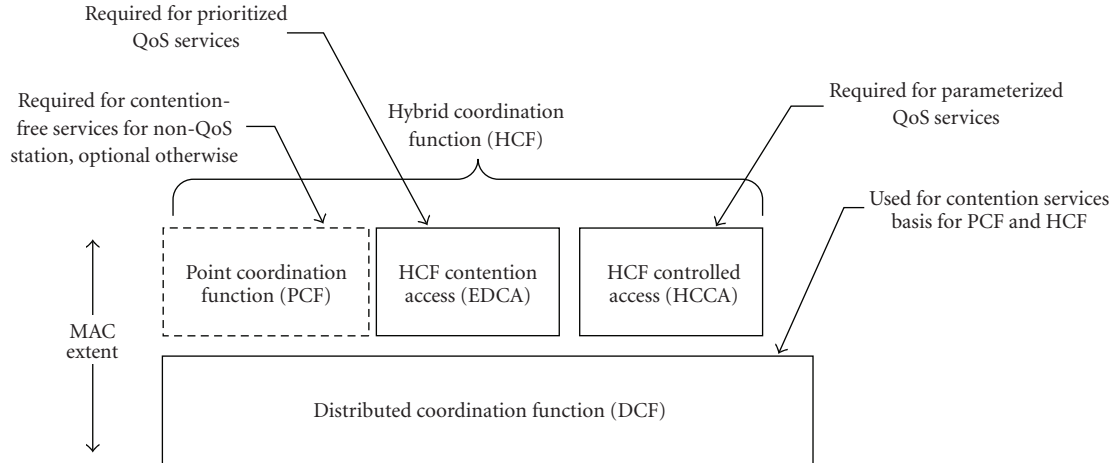


FIGURE 2: IEEE 802.11e MAC architecture.

TABLE 1: Default DCF and EDCA parameter set.

Parameters	CWmin	CWmax	DIFS/AIFSN	
DCF	aCWmin	aCWmax	2	
EDCA	AC.BK	aCWmin	7	
	AC.BE	aCWmin	3	
	AC.VI	$(aCWmin + 1)/2 - 1$	aCWmin	2
	AC.VO	$(aCWmin + 1)/4 - 1$	$(aCWmin + 1)/2 - 1$	2

of these simplifications are related to the modeling of the AIFS procedure, backoff counter, transmission opportunities (TXOP), virtual collisions, and retransmission limits. Moreover, most of the analytical models presented in the literature assume that the network operates in saturated traffic conditions.

Regarding the scenarios analyzed in this paper, we consider that the use of analytical models is not an adequate approach by the following reasons. First, the simplifications usually employed on these models cannot capture properly important aspects that are considered in our evaluation. Second, most part of these models assume Poisson traffic sources. Therefore, it would be difficult to accurately model real-time traffic. Third, in the considered scenarios the network is operating in a nonsaturation area, and therefore only a few of these models were useful. Therefore, we have used a simulation model built using a *Stochastic Petri Net* (SPN) tool [18], that describes the dynamics of the EDCA function of the IEEE 802.11e protocol. This simulation model has been validated against the results obtained from simulation models [19, 20]. The achieved results clearly highlight the validity of the employed simulation model (for a complete description of the SPN model, validation and related documentation, please refer to [21]). For all the simulations, it is used a Gilbert-Elliot variant error model, which is a Semi-Markov model where the channel is always in one of two states: good or bad. This model assumes an independent bit error rate (BER) in each state. For the parametrization of the model, the states holding times are log-normal distributed according to [22]. The

mean duration of good state is 65 ms, the mean duration of bad state is 10 ms and, the coefficient of variation (CoV) for the bad state holding times has been set to 10 and for the good state to 20. This set of values leads to a rather bad channel [23], where the steady-state probability for finding the channel in bad state is approximately 13.3%. Two sets of simulations are assessed, differing in their respective mean bit error rate. The first set defines a mean BER of $1.0E-4$, while the second set defines that no bit errors occur. Therefore, during the bad channel states for the first set, the BER is about $7.5E-4$ and, for the good state no errors will occur.

The simulation scenarios were built considering an *ad hoc* network topology, where multiple ST and RT stations are operating in the same frequency band. It is assumed that all stations are within the range of transmission of each other and there is no node mobility nor hidden stations.

RT stations only transfer RT messages, using the default set of parameters defined by the EDCA function for the voice (VO) access category. On the other hand, ST (standard IEEE 802.11e stations that are out of the sphere-of-control of the RT architecture) stations are intended to model external traffic sources, that are sharing the communication medium with the RT traffic sources. These ST stations transmit several types of traffic using the default set of parameters defined by the EDCA mechanism (Table 1).

The physical parameters used in the simulations are based on the IEEE 802.11a PHY mode $aCWmin = 15$ and $aCWmax = 1023$. Each station operates at OFDM (Orthogonal Frequency Division Multiplexing) PHY mode,

control frames are transmitted at a basic rate equal to 1 Mbps, while the MSDU (MAC service data units) are transmitted at 36 Mbps. The maximum number of transmission attempts is set to 7. The MAC queue size is set to 50 positions.

All the simulation results have been obtained with a 95% confidence interval with a half-width relative interval of 5%. The performance metrics analyzed in this scenario include: average delay, average queue size and packet loss rate. The *average delay* is the average delay required to transfer a packet, measured from the start of its generation at the application layer to the end of the packet transfer at the receiving station. The *average queue size* represents the average output buffer occupancy. The *packet loss* represents the percentage of packets that are lost for each traffic stream.

3.1. Simulation Scenario 1. In this scenario, two simulation cases are assessed, the first (*small population case*) considers 10 ST stations operating in the same frequency band with 10 or 20 RT stations. The second one (*large population case*) extends the number of ST stations to 40.

The generated data frames have a constant size. The RT traffic is characterized by periodic traffic sources with a small amount of imposed jitter (<1% of the message period). It is also guaranteed that the RT traffic is not correlated among RT stations. Each RT station generates packets with fixed Message Stream Periods (MSPs) of 2 ms, 10 ms, or 20 ms with 45 bytes of data payload, which is equivalent to generate 500, 100 or 50 packets/s. Therefore, each RT station imposes a constant network load of 180, 36 or 18 kbits/s that represents about 0.5%, 0.1% or 0.05% of the total network load (without considering the MAC and PHY headers).

On the other hand, ST stations transmit three types of traffic: voice (VO) (packet size of 160 bytes), video (VI) (packet size of 1280 bytes) and background (BK) (packet size of 1600 bytes), according to a Poisson distribution. The network load imposed by the set of ST stations ranges from 10% to 90%, by steps of 10%. In order to impose this set of network loads, each ST station generates a tuple of VO, VI and BK packets at a predefined data rate. For the case of the small population scenario, the predefined set of data rates were {14.80; 29.60; 44.40; 59.21; 74.01; 88.82; 103.62; 118.42, or 133.22} packets/s. Similarly, for the case of the large population scenario, the predefined set of data rates were {3.70; 7.40; 11.10; 14.80; 18.50; 22.20; 25.90; 29.60, or 33.31} packets/s. Therefore, as it is used the same data rates for all types of traffics (VO, VI, and BK), the percentage of each traffic category is directly related with their packet sizes (VO = 5.2%, VI = 42.1% and BK = 52.7%).

From this simulation setup, it is clear that the RT traffic is transferred at the same priority level as the VO (voice) traffic transferred by ST stations. The only distinguishing difference is the length of the transferred packets.

The target of these simulations is to assess the behavior of the RT traffic when the communication medium is shared with external traffic generated by a set of ST stations. It is assumed that RT messages have a production period that is equal to their deadline. If the message delay is higher than its deadline, when it arrives to the receiver its contents are

(temporally) invalid, which is equivalent to a deadline miss. If a RT message arrives to a nonempty transmission queue this means that the previous message was not yet transmitted and therefore it missed its deadline.

Based on the RT application characteristics presented above, a set of qualitative upper boundaries are used to assess *how well* the EDCA mechanism is able to handle RT traffic in those communication scenarios. Those boundaries include: (a) an average message delay that must be clearly smaller than the message stream period (MSP); (b) an average queue size that must be clearly smaller than 1 packet; (c) a packet loss rate that must be clearly smaller than 5%. Any values above these qualitative thresholds indicate an unacceptable rate of message deadlines loss, and therefore an unacceptable RT behavior.

3.2. Simulation Results for Scenario 1. For the sake of simplicity, only the values for RT traffic are plotted in the following figures. Ni [24] has already demonstrated that the EDCA mechanism can provide a service differentiation between different types of traffic, that is, the highest-priority voice message streams have lower delays than the lowest-priority ones (video, best-effort, and background). However, that service differentiation is reached by downgrading the service of the lower priority message streams. That is, the performance of video and data flows will be degraded, specially when the channel is heavily loaded. Thus, within the context of this work it is not relevant to further discuss the timing behavior of the ST traffic.

3.2.1. The Impact of External ST Traffic upon the Average Queue Size. A first simulation analysis concerns the assessment of the average queue size. Figure 3 shows the average queue size for MSPs of 2 ms, considering the case of 10 RT stations operating in both the small and large population scenarios (10ST-40ST).

When comparing the undisturbed scenario (by undisturbed scenario, we mean that there are no external ST stations trying to transfer its own messages, that is, there is an unrealistic *closed* communication environment) with the case of 10 RT stations operating together with 10/40 ST stations (MSP = 2 ms), the impact that external traffic has upon the average queue size of RT message streams is clear. Figure 3 shows that the average queue size for the RT stations is already larger than 1 packet when the network load imposed by ST stations is only slightly above 10%. This means that the EDCA mechanism is not able to provide a RT communication service for MSPs of 2 ms in open communication environments, when considering the use of default EDCA parameters.

Figure 3 also illustrates the impact of increasing the number of ST stations contending for the medium access, which highlights a degradation of the QoS for the large population scenarios. This is due to the effect of having multiple stations contending for the transmission medium, which leads to a large number of collisions. As a consequence, the medium access delay increases due to the backoff mechanism, and therefore there is also an increase of the average queue size.

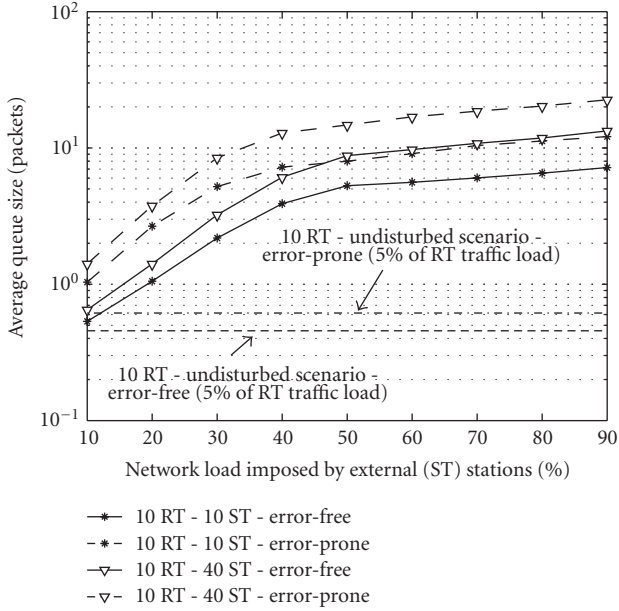


FIGURE 3: Average queue size (small and large pop.): error-free versus error-prone—MSP = 2 ms.

From Figure 3 we can also infer that one of the major sources of perturbation upon the RT communication is the increase of the external network load and not only the error-prone characteristics of the wireless medium. It is interesting to compare the impact upon the average queue size between an error-free and an error-prone channel in the following cases: (a) undisturbed versus disturbed scenarios; (b) increase of the network load from 10% to 50%. In both cases, the impact of an error-prone channel is smaller than the increase of the external network load.

Consider now the case of a set of RT stations with MSPs of 10 ms (Figure 4). It can be observed that whatever the external network load, the average number of queued packets is always kept under 1 packet. Similar results were obtained for MSPs of 20 ms, and also for the case of 10 RT stations with MSPs of both 10 ms or 20 ms. Therefore, in what concerns the average queue size of RT messages, it indicates that for MSPs larger or equal to 10 ms the EDCA mechanism can be suitable to support RT traffic in open communication environments.

3.2.2. The Impact of Uncontrolled External Traffic upon the Packet Loss Rate. A second simulation analysis concerns the assessment of the packet loss rate. Figure 5 shows the packet loss results for RT stations with MSPs of 10 ms, for both small (10 ST stations) and large (40 ST stations) population scenarios.

This simulation analysis shows that, for a number of RT stations above 10, the impact of the external traffic becomes clearly undesirable (similar results were obtained for MSPs of 20 ms). For instance, when considering 20 RT stations, the packet loss rate is already above 10%, when the external network load increases from just 10% to 30%. This value

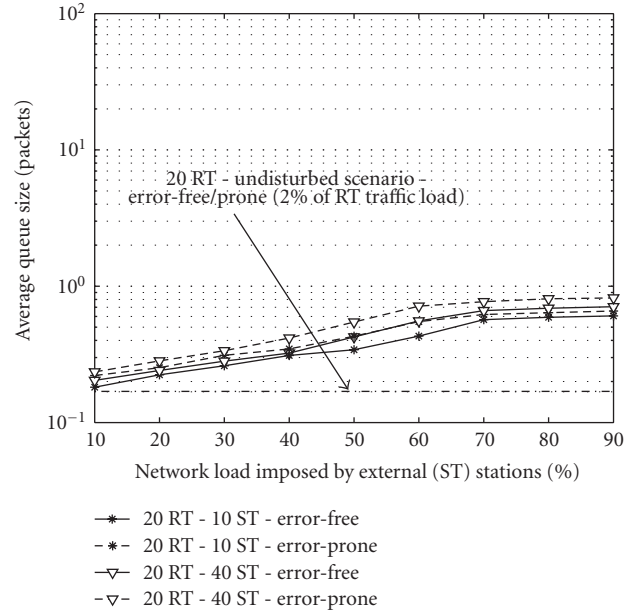


FIGURE 4: Average queue size (small and large pop.): error-free versus error-prone—MSP = 10 ms.

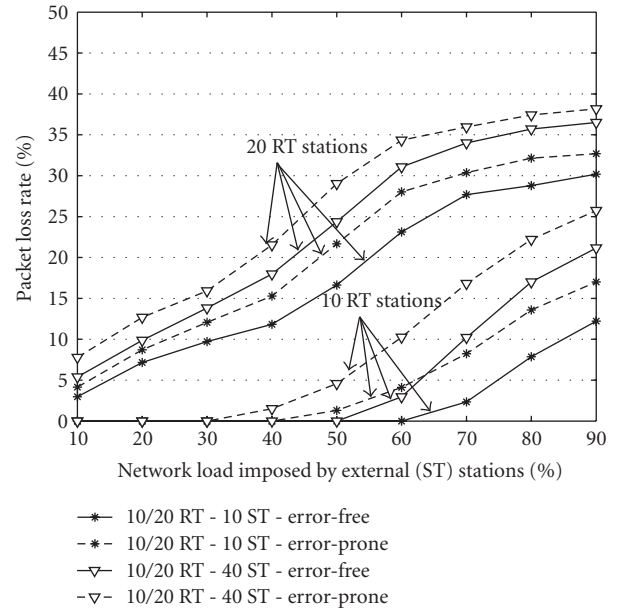


FIGURE 5: Packet loss rate (small and large pop.): error-free versus error-prone—MSP = 10 ms.

clearly exceeds the upper bound for the admissible packet loss rate, which means that EDCA is no longer able to provide a RT communication service to the supported applications. Comparing the results presented in Figures 4 and 5, it can be observed that while the average queue size is kept at a reduced value, the packet loss rate quickly increases. The reduced queue size is mainly a consequence of the reduced RT traffic load, the RT packets being losses due to the higher number

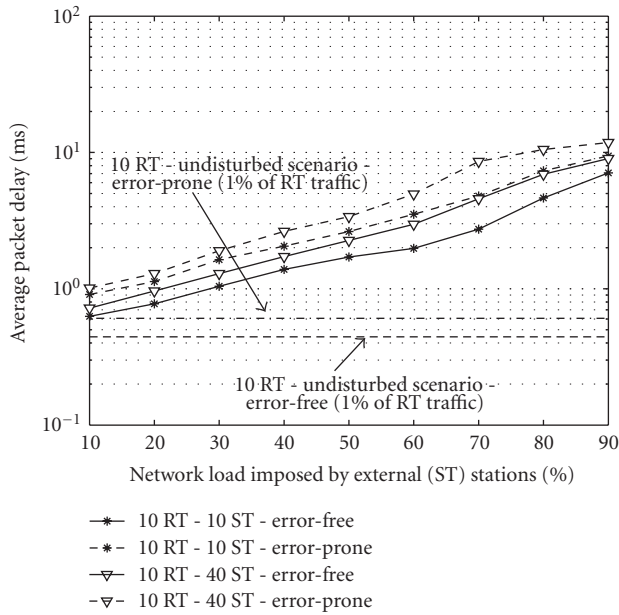


FIGURE 6: Average delay (small and large pop.): error-free versus error-prone—MSP = 10 ms.

of retransmissions due to collisions with the increasing ST traffic.

3.2.3. The Impact of External ST Traffic upon the Average Packet Delay. Finally, there is the need to assess the average packet delay for transferring RT packets, when considering both small and large population scenarios. It has been previously concluded that using the EDCA mechanism with default parameters does not allow to provide a RT communication service to more than 10 RT stations, even for MSPs of 10 ms or 20 ms. Therefore, the average packet delay will be assessed only for the case of 10 RT stations with MSPs of 10 ms (similar results were obtained with MSPs of 20 ms), which are represented in Figure 6.

This set of simulations shows that using the EDCA mechanism enables the support of RT traffic with MSPs of 10 ms or 20 ms generated by up to 10 RT stations (the average packet delay remains smaller than the respective MSP). The upper bound for external disturbances may be as high as 70% or 90% in the case of message stream periods of 10 ms or 20 ms, respectively. Additionally, it can be seen that the effect of the error-prone channel becomes more relevant when the external network load increases. However, the impact of the external traffic load upon the average packet delay is higher than the impact of the error-prone channel. The only exception occurs in scenarios with small loads, where the influence of both elements can be considered to be balanced.

3.3. Simulation Scenario 2. The main conclusion from the previous results is that default set of parameter values that are used for the EDCA mechanism are just able to guarantee RT communication requirements for a number of RT stations smaller than 10 with MSPs above 10 ms. Another important

result is that the major source of perturbation upon RT communication is caused by the increase of external network load and not by the error-prone characteristics of the wireless medium. Remark that the level of communication errors used in the simulations has been set to a realistic value when dealing with wireless transmission in harsh industrial environments [23].

One of the simplifications of simulation Scenario 1 is that the voice traffic has been modeled as a Poisson traffic. This can be seen as an acceptable modeling simplification, as the RT traffic is sharing the same AC with the voice traffic. However, there is the need to assess if using the voice AC to transfer VoIP traffic according to a realistic CODEC model is less damaging to the transfer of RT traffic. Additionally, it was also decided to consider RT traffic sources with MSPs of 5 ms, and 50 ms (and not only MSPs equal to 2 ms, 10 ms, and 20 ms), which are equivalent to generate 200 or 20 packets/s.

As in Simulation Scenario 1, it is considered a number of RT stations operating in the same frequency band with a set of ST stations. In Scenario 2, ST stations transmit two types of traffic: voice (VO) and background (BK). The voice traffic is characterized as a two state Markov ON/OFF to represent realistic voice calls [25]. The ITU-T G.711 recommendation has been selected, with 64 kb/s bit rate, packet size of 160 bytes generated every 20 ms during a talking (ON) period, and no packet generated in a listening (OFF) period. The mean value of ON/OFF period is 1.5 s and 1.35 s, respectively. The transmission rate of background traffic is 1024 kb/s (packet size of 1500 bytes).

Basically, a scenario that considers 2, 6, 10, 14, 18, 22, 26, and 30 external ST stations is analyzed, where each ST station imposes a network load of 2.84%. This means that the network load imposed by this set of external ST stations varies from 5.69% (case of 2 ST stations) to 85.33% (case of 30 ST stations). These external ST stations are operating in the same frequency band together with 10, 20, and 30 RT stations. The characteristics of the RT traffic pattern are the same as defined in Simulation Scenario 1. As in the previous scenario, it is assumed that all stations are within the range of transmission of each other and there is no node mobility nor hidden stations. Finally, it is only considered the case of an error-free channel.

3.4. Results Scenario 2. The main target of Scenario 2 is to further analyze which are the main sources for the poor quality results of the EDCA mechanism. A first simulation analysis concerns the assessment of the average packet delay. Figure 7 shows the average packet delay results for transferring RT packets in the case of 10/20/30 RT stations.

From this set of simulations (Figure 7), it can be observed for the case of MSPs equal to 50 ms that, whatever the external network load, the average packet delay is always kept below than deadline. In the case of MSPs equal to 5 ms, the average packet delay is kept below the deadline only up to 20 RT stations. On the other hand, the average packet delay is almost independent of the number of external ST stations. The main reason for this behavior is the higher priority traffic of ST not being always active (ON/OFF behavior) and therefore the level of interference with the RT traffic is much

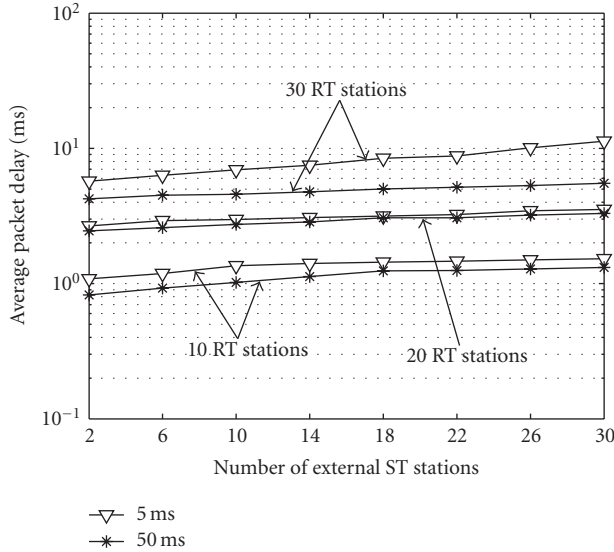


FIGURE 7: Average packet delay—MSP = 5 ms and 50 ms.

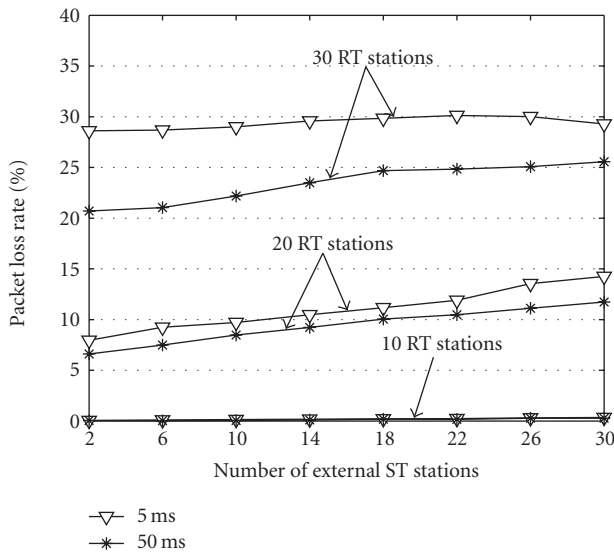


FIGURE 8: Packet loss rate—MSP = 5 ms and 50 ms.

smaller. Remember that for this simulation scenario the ST stations do not generate any voice packet during the listening (OFF) period (1.35 s), while in the previous scenario they generate Poisson distributed traffic in the voice AC category.

A second simulation analysis concerns the assessment of packet losses. Figure 8 shows the packet loss results for MSPs of 5 ms and 50 ms. This figure shows that, for 20 RT stations, the impact of the external traffic is clearly undesirable. This confirms that, when using default parameter values, the EDCA mechanism is not able to provide any acceptable RT guarantee for a number of RT stations above 10, as the percentage of packet losses quickly overpass the 5% qualitative upper bound.

As previously discussed, the EDCA mechanism provides traffic differentiation based on three independent

TABLE 2: Effects of varying AIFSN on the RT traffic, for 20 RT stations—MSP = 5 ms and 10 ms.

MSP	Metric	AIFSN				
		2	3	4	5	6
5 ms	loss (%)	10.48	10.03	12.42	14.91	16.98
	delay (ms)	3.07	3.29	3.75	4.32	5.18
10 ms	loss (%)	10.14	8.94	9.09	9.51	11.22
	delay (ms)	3.04	3.29	3.71	4.16	4.86

mechanisms: the Arbitration Interframe Space (AIFS), the Transmission Opportunity time interval (TXOP) and the Contention Window size (CW). For a reduced network load, small CW values can be an adequate choice. However, when the number of stations and the network load increase, small CW values means higher number of collisions. Throughout the previously presented results, it becomes clear that one of the main sources of weakness of the EDCA mechanism when dealing with uncontrolled traffic sources is related to the increasing number of RT stations. Therefore, considering that we have full control upon the set of RT stations, it is reasonable to propose the selection of adequate AIFS, TXOP, and CW parameter values according to the characteristics of the RT application. This proposal is consistent with the “*open communication environment*” concept, as we are not imposing any particular setup for the external ST stations. Therefore, in the next section we investigate the impact of the variation of these EDCA parameters upon the behavior of the RT traffic.

3.5. Tuning the AIFSN, TXOP, and CW Parameters. Considering the case of Scenario 2 with a specific number of ST stations (14 ST stations—external network load of about 40%). We firstly investigate the impact of the AIFSN parameter upon the behavior of the RT traffic. That is, we analyzed the average packet delay and packet loss rate of the RT traffic, when there are 20 RT stations transmitting messages with periodicity of 5 ms and 10 ms. For the sake of simplicity, we omit the values for MSPs of 2 ms, 20 ms and 50 ms and we only consider an error-free channel. In this experiment, the AIFSN parameter varies from 2 to 6 in the set of RT stations. Remember that, the default EDCA value for voice traffic is 2. The other parameters are the default values defined for the EDCA mechanism, where according to IEEE 802.11a PHY mode $aCW_{min}[VO] = 3$, $aCW_{max}[VO] = 7$; and $TXOP = 1.504$ ms.

Table 2 shows the results for both simulated cases. It is clear that the AIFSN value has some influence in the average packet delay and in the packet loss rate.

In a second experiment, we investigated the impact of the TXOP parameter upon the behavior of the RT traffic, also for 20 RT stations transmitting RT messages with MSPs of 5 ms and 10 ms, respectively. Similarly to the first set of experiments, we took into account a varying number of TXOP values (0.000; 0.512; 1.024; 1.512; 2.016; 4.000; and 8.192 ms). All the other parameters were set to the EDCA default values ($aCW_{min} = 3$, $aCW_{max} = 7$ and $AIFSN = 2$). We can observe from the simulation results (Table 3) that

TABLE 3: Effects of varying TXOP on the RT traffic, for 20 RT stations—MSP = 5 ms and 10 ms.

MSP	Metric	TXOP (ms)						
		0.000	0.512	1.024	1.512	2.016	4.000	8.192
5 ms	loss (%)	10.56	10.23	10.23	10.23	10.23	10.23	10.23
	delay (ms)	3.07	3.05	3.05	3.05	3.05	3.05	3.05
10 ms	loss (%)	10.14	10.14	10.14	10.14	10.14	10.14	10.14
	delay (ms)	3.04	3.04	3.04	3.04	3.04	3.04	3.04

TABLE 4: Effects of varying aCWmin, aCWmax on the RT traffic, for 20 RT stations—MSP = 5 ms.

aCWmin	aCWmax	Packet loss rate (%)	Average packet delay (ms)	Average queue size (packets)
1	3	34.27	4.28	0.56
1	7	12.37	3.18	1.11
<u>3</u>	<u>7</u>	<u>10.37</u>	<u>3.06</u>	<u>1.05</u>
1	15	3.85	2.97	1.35
3	15	2.57	2.83	1.40
7	15	1.72	2.70	1.48
1	31	1.55	3.06	1.50
3	31	0.73	2.89	1.42
7	31	0.28	2.68	1.41
15	31	0.11	2.53	1.28
⋮				
511	1023	0.56	49.44	16.17

the TXOP parameter does not have any influence upon the behavior of the RT traffic in the evaluated scenarios. This was an expected result, as small values for the average queue size means that there will be no bursts of messages during an acquired TXOP. Therefore, it will be irrelevant if the TXOP parameter is adjusted. Remember that the EDCA default value for TXOP is 1.504 ms.

Finally, we assessed the impact of the aCWmin and aCWmax parameters used to transmit RT traffic. In this experiment, we examined all the possible settings for aCWmin and aCWmax parameters between 1 and 1023. Tables 4 and 5 show the main results obtained from these experiments.

It can be observed a significant improvement in the RT traffic when comparing the CW default parameter values (underlined values: aCWmin = 3; aCWmax = 7) with the best CW values (bold values) for this specific scenario. It indicates that, depending on the type of supported RT communication, the EDCA mechanism may be considered suitable for RT communication purposes.

3.6. *Tuning the Best CW Parameters.* Based on the best CW parameters extracted from the previous simulation scenarios (aCWmin = 15 and aCWmax = 31), we repeated the simulations from Scenario 2, considering 20 RT stations transmitting messages with MSPs of 5 ms and 10 ms,

TABLE 5: Effects of varying aCWmin, aCWmax on the RT traffic, for 20 RT stations—MSP = 10 ms.

aCWmin	aCWmax	Packet loss rate (%)	Average packet delay (ms)	Average queue size (packets)
1	3	33.57	4.18	0.28
1	7	11.79	3.16	0.28
<u>3</u>	<u>7</u>	<u>10.14</u>	<u>3.04</u>	<u>0.28</u>
1	15	2.38	2.89	0.28
3	15	1.40	2.77	0.28
7	15	0.90	2.66	0.26
1	31	0.36	2.90	0.28
3	31	0.11	2.77	0.28
7	31	0.034	2.60	0.26
15	31	0.014	2.50	0.25
⋮				
511	1023	0.007	30.63	3.06

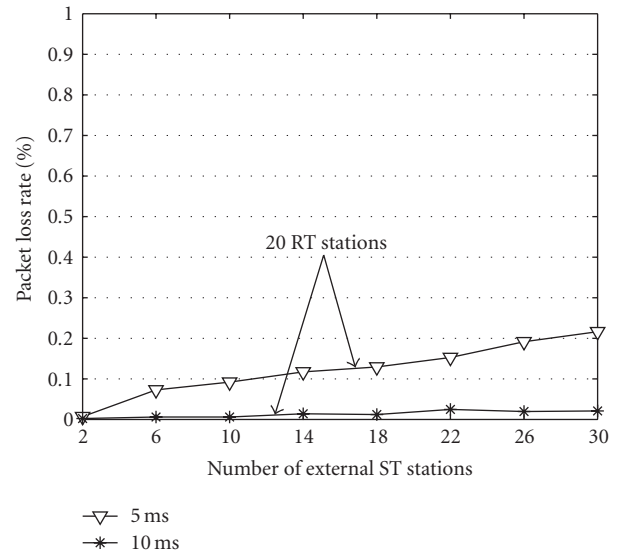


FIGURE 9: Packet loss rate—MSP = 5 ms and 10 ms.

operating together with 2, 6, 10, 14, 18, 22, 26, and 30 ST stations.

Figure 9 illustrates the packet loss rate for this scenario. It can be observed that there is a much smaller rate of RT packet losses, in the case of the new selected values for the CW parameters, when comparing with the results represented in Figure 8. This percentage of packet losses is clearly adequate for a vast number of RT applications.

Figure 10 shows the average packet delay when considering this scenario. As it would be expected, the use of longer contention windows (larger values for aCWmin and aCWmax) results in higher values for the average packet delay, as it can be remarked when comparing these results with those represented in Figure 7. Additionally, the average queue size increases at the same time. Comparing the results from Table 4 (MSPs = 5 ms) and Table 5 (MSPs = 10 ms) it

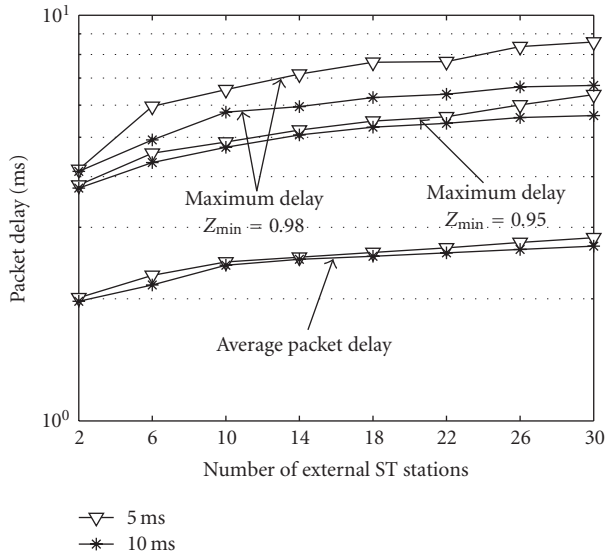


FIGURE 10: Packet delay—MSP = 5 ms and 10 ms.

can be seen that, for MSPs = 5 ms, the average queue size is always slightly greater than one. This means that when a new packet arrives to the queue, the queue is not always empty (there was a deadline loss). For these cases, having an average packet delay smaller than 5 ms just indicates that, in average, most part of the packets are served well before the deadline. It should also be noted that whenever there is a high packet loss rate, the average packet delay values are not really meaningful (i.e., the case of the first 3 lines of both Tables 4 and 5). The main reason is that the packet delay just takes into account the delay of the delivered packets, excluding all the lost packets.

Finally, an important metric when dealing with RT communication is the maximum transmission delay. In the case of Figure 10, it is also plotted the maximum delay when considering 95% and 98% of the successful message transmissions. These values are relevant for the case of loss tolerant applications, where up to 5% and 2% of deadline misses can be encompassed. The previous simulation analysis (average packet delays) are only indicative of the average communication behavior, but may be valuable enough to provide an indication of the QoS that can be supported by the EDCA mechanism.

4. State-of-the-Art and Future Directions

Traditionally, when supporting RT communication in CSMA wired networks, the timing behavior has been guaranteed through the tight control of every communicating device. The coexistence of RT controlled stations with non-RT stations has been made possible by constraining the traffic behavior of the latter. Unfortunately, this approach cannot be enforced in wireless environments as, at any instant, external traffic sources may start sharing the same radio channel. In this section, we briefly describe the state-of-the-art approaches intended to provide QoS support in IEEE 802.11 networks.

The Point Coordination Function (PCF) is one of the main solutions intended to support QoS guarantees in IEEE 802.11 wireless networks. It has been proposed in the original IEEE 802.11 standard as an optional access mechanism. It implements a centralized polling scheme to support synchronous data transmissions, where the Point Coordinator (PC) performs the role of polling master. Usually, the PC resides in the Access Point (AP).

The HCCA mechanism was proposed in the IEEE 802.11e amendment [1] to improve the PCF scheme. It is based on a round-robin scheme and it is intended to guarantee bounded delay requirements. Like the PCF scheme, the PC also polls *all* the stations in the polling list, even though some stations may not have messages to transmit. When the PC polls a station that has no packets to transfer, the station will transmit a null frame. As a consequence, the polling overhead is roughly equal to the time interval from sending the polling frame till the end of the ACK frame [26]. A number of improvements have been recently proposed to reduce the HCCA polling overhead. For instance, Gao et al. [27] proposed a new admission control framework to replace the traditional CSMA mechanism. It uses the mean data rate and the mean packet-size values to calculate the resource needed by each message flow. Thus, this new mechanism is particularly adequate when dealing with variable resource bit rate (VBR) traffic over HCCA.

Rashid et al. [28] have pointed out the performance deficiencies of the HCCA scheduler and proposed a new scheduling scheme. The prediction and optimization-based HCCA (PRO-HCCA) uses a prediction mechanism to account for the dynamic intensity of VBR traffic. This mechanism tries to find an optimal allocation of available transmission time among the competing traffic streams.

Solutions based on token passing mechanisms have also been proposed to support QoS communication in wireless networks. In [29], Ergen et al. proposed the WTRP (Wireless Token Ring Protocol), which is a MAC protocol that exchanges special tokens and uses multiple timers to maintain the nodes synchronized. The token is rotated around the ring. Each station transmits during a specified time and if enough time is left, the station invites nodes outside the ring to join. In [30], Cheng et al. presents a wireless token-passing protocol, named *Ripple*. Basically, Ripple modifies the data transmission procedure of 802.11 DCF and employs request-to-send (RTS) and ready-to-receive (RTR) frames as tokens. A node that has the right to send a data frame will send a RTS frame, and a node which has the right to receive a data frame will send a RTR frame to the sender if the expected RTS frame has not been received. Summing up, a station can only send a DATA frame if it holds the token.

Based on the EDCA mechanism, Villalón et al. [31] designed the B-EDCA mechanism that intends to provide QoS support to multimedia communication, and it is able to coexist with legacy DCF-based stations. Basically, it changes the AIFS value of the highest AC to SIFS + aSlotTime when stations are in the Backoff state. Moreover, in order to keep the compatibility with the HCCA mechanism, a station implementing the B-EDCA mechanism must wait for an

additional SIFS time when the backoff counter reaches zero, that is, $2 \times \text{SIFS} + \text{aSlotTime}$. Therefore, it is adequate when operating in the IEEE 802.11b PHY mode, as the waiting time will be larger than the one used by the HCCA mechanism and smaller than the time used by standard EDCA stations.

In [32], it is also proposed a modified EDCA mechanism, where soft real-time guarantees are provided by dynamically adjusting the priority level of a traffic flow based on the estimated per-hop delay, and generating a nonuniformly distributed backoff timer for retransmitted frames according to their individual end-to-end delay requirements.

Hamidian and Körner [33] presented an interesting solution that provides QoS guarantees to the EDCA mechanism. The proposed solution allows stations with higher priority traffic to reserve time for collision-free access to the medium. Basically, it proposes to transfer the HCCA admission control and scheduling algorithms from the HCCA controller to the contending stations.

Wang et al. [34] designed a new collision resolution mechanism, referred as *gentle* DCF or GDCF. The difference between GDCF and DCF is that GDCF takes a more conservative measure by halving the CW value only if there are c consecutive successful transmissions. Conversely, DCF resets its CW to the minimum value once there is a successful transmission. The GDCF needs to maintain a continuous successful transmission counter that is reset to zero after each collision. Then, when a collision occurs GDCF works similarly to DCF.

Yang and Vaidya [35] proposed the Busy Tone Priority Scheduling (BTPS) protocol. This scheme makes the following assumptions: (i) each station is capable to monitor the carrier status of the data channel; (ii) each station in idle state is capable to monitor two busy tone channels (BT1 and BT2) and lock onto the signal in the data channel as desired; (iii) it is assumed that the two busy tone signals (BT1 or BT2) sent by a station can be sensed by other stations within the interference range of the former station. In BTPS, busy tone serves as the indication of backlogged high priority packets.

BTPS works similarly to the IEEE 802.11 DCF, with the difference that high priority and low priority stations behave differently during IFS and *backoff* stages. The BTPS protocol uses DIFS as the IFS for high priority stations. However, during DIFS and backoff stages, high priority stations with queued packets send an energy pulse every M slots, where M is a constant. Between two consecutive busy tone pulse transmissions, there should be at least one empty Slot Time interval, as the station must have a chance to listen to the data channel. Therefore, M could be any value larger than or equal to 2 and, the IFS of low priority stations should be larger than M slots, in order to enable sensing the busy tone signal.

In [36], it was proposed a distributed algorithm intended to provide fair scheduling in a WLAN, referred as DFS (Distributed Fair Scheduling). The DFS protocol behaves quite similarly to IEEE 802.11 DCF, except in what concerns the backoff interval initially calculated, which is chosen proportional to the finish tag of the packet to be transmitted. The finish tag is calculated similarly to the SCFQ (Self-Clocked Fair Queueing) algorithm [37]. In [38], the authors

modify the distributed SCFQ algorithm combined with the prioritization schemes proposed in the EDCA mechanism and specify the RT-FCR (real-time fast collision resolution), where the priorities are implemented by assigning different backoff ranges based on the type of traffic.

In [39], Lopez-Aguilera et al. evaluated the performance of the IEEE 802.11e EDCA when its working procedure is unsynchronized. The authors proposed the use of AIFS values whose differences are not multiple of the slot time. As a consequence, it would become possible to avoid collisions between frames from different access categories.

Lo Bello et al. [40] proposed a wireless traffic smoother (WTS) to support soft RT traffic over IEEE 802.11 WLANs. The presented solution is similar to the traffic smoother scheme previously proposed for Ethernet networks [41]. However, its main drawback is that it requires the smoothing strategy to be implemented in all the communicating devices.

Another approach to support QoS guarantees are those based on *forcing* the collision resolution schemes in favor of the RT stations. A relevant proposal has been made by Sobrinho and Krishnakumar [42], who adapted the EquB mechanism (black burst) [43] to ad hoc CSMA wireless networks. This scheme requires the shutdown of the standard retransmission scheme. Real-time stations implementing the EquB approach contend for the channel access after a medium interframe spacing t_{med} , rather than after the long interframe spacing t_{long} , used by standard stations. Thus, RT stations have priority over standard stations. When a RT station wants to transmit, it sorts its access rights by jamming the channel with Black Bursts (BB's), that is, energy pulses immediately after an idle period of length t_{med} . The length of the BB transmitted by a RT node is an increasing function of the contention delay experienced by the node.

A similar scheme is presented in [44], where voice nodes (RT stations) use energy-burst (EB) (that are similar to BB) periods to prioritize RT packets over data packets. The AP (Access Point) can transmit a VoIP packet after PIFS without backoff or contention. On the other hand, each voice station has its own address (ID), referred as VID (virtual identification). The VID can be assigned during the traffic stream (TS) setup procedure. The VID is expressed as a binary value, which is determined by the voice packet resolution period (VPRP). The station with the highest VID wins the contention.

In [45], Sheu et al. proposed a priority MAC protocol based on Sobrinho's approach, complemented by a binary tree referred as *contention tree*. Basically, the black-burst scheme is adopted to distinguish the priorities of stations. Stations with the same priority send messages in a round robin manner. The basic idea is that a station can obtain a unique ID number, which depends on its position in the contention tree.

In [46], a RT-communication approach (VTP-CSMA) has been proposed based on a traffic separation mechanism. Such mechanisms are able to prioritize RT-traffic over other traffic, without directly controlling the latter. The proposed architecture is based on a Virtual Token Passing procedure that circulates a virtual token among a number of RT devices. This virtual token is complemented by an underlying traffic

separation mechanism that prioritizes the RT traffic over the non-RT traffic.

In [47], it was presented preliminary results of a new mechanism called Contention Window Adapter. It dynamically changes the contention window size of the different ACs of the EDCA mechanism, according to the workload conditions of the wireless network. It represents one step more forward the use of wireless communication in industrial environments.

A common characteristic of almost all these previous solutions is that an enhanced (QoS-enabled) station is not able to support RT communication in the presence of unconstrained IEEE 802.11 stations, unless these standard stations do not initiate any communication. That is, the majority of the RT communication approaches are not able to provide QoS guarantees whenever unconstrained stations try to access the shared communication medium. Relevant exceptions are the improvements included in the HCF (PCF and HCCA) mechanism of the IEEE 802.11e amendment. However, despite PCF being well suited to handle delay-sensitive applications, it is rather complex and almost none WLAN network card ever implemented the PCF scheme [48]. Preliminary studies [28, 49] have also shown that the HCCA mechanism may not be suitable to guarantee the special requirements of RT applications. Furthermore, it is still not clear whether the HCCA mechanism will be implemented in next generation WLAN network cards, overcoming the unavailability problem of the PCF mechanism.

We entirely agree with Bianchi et al. [50] that in wireless architectures, the service differentiation mechanism must be compulsory introduced as medium access control (MAC) layer extension. Within this context, other relevant exceptions are those based on the black burst (BB) scheme [42] and VTP-CSMA approach [46]. BB solutions are technically very interesting, but they are in practice not viable [51], since they require an extensive modification of the WLAN cards, which prevents the use of commercial-of-the-shelf (COTS) hardware. Although the VTP-CSMA mechanism also needs to modify parts of the MAC layer, it could be easily implemented in COTS hardware (e.g., Field Programmable Gate Array—FPGA) upon standard IEEE 802.11e hardware.

Besides these approaches, all other mechanisms require the strict control of the communication environment. Therefore, they are not suitable to handle communication environments where a set of non-RT stations can be operating in the same frequency band as the RT stations. For next generation communication environments, allowing the coexistence of both RT and non-RT stations in the same communication domain is likely to become a major requirement. Furthermore, solutions that enable the prioritization of RT traffic only when considering closed communication environment are unacceptable. Specifically, the underlying wireless communication protocol must be able to guarantee the timing constraints of the RT traffic in a communication environment shared with timing unconstrained devices. Therefore, the most promising solutions to provide RT communication in IEEE 802.11 are those that force the collision resolution in favor of the RT stations, compelling

all the other contending stations to postpone the medium access.

5. Conclusion

We have evaluated the timing behavior of the EDCA function when supporting RT traffic, similar to the traffic that is usually found in industrial environments. The simulated scenarios consider a set of RT stations operating in the same frequency band of a set of timing unconstrained ST stations. Basically, we have assessed the impact of external traffic (generated by ST stations) upon the behavior of the highest access category (voice), when this access category is used to transfer RT traffic (small-sized packets in periodic intervals). Moreover, both error-free and error-prone channels are considered. The obtained results show the following:

- (i) both increasing the number of stations and the network load strongly influence the RT traffic, by means of higher packet delays, higher percentage of packet losses, higher average queue sizes;
- (ii) the packet loss rate for transferring RT packets is highly influenced in an error-prone channel, mainly for high rates of network load imposed by external (ST) stations.

The main conclusion of the first set of simulated scenarios is that default parameter values of the EDCA mechanism are just able to guarantee industrial communication (timing) requirements for a number of RT stations smaller than 10 stations with message stream periods above 10 ms. From a second set of simulated scenarios it can be concluded that the RT behavior of the EDCA mechanism can be significantly improved by adequately tuning the EDCA parameters. More specifically, by enlarging the default values of the CW parameters, the packet loss rate can be reduced down to an acceptable range values. This improvement in the packet loss rate is achieved at the cost of slightly increasing the average packet delay. Therefore, by carefully tuning the CW parameters, it is possible to significantly improve the RT behavior of the EDCA mechanism.

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