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Dynamic and distributed packet aggregation to solve the performance anomaly in 802.11 wireless networks ☆

Tahiry Razafindralambo [a,*], Isabelle Guérin-lassous [b], Luigi Iannone [c,*], Serge Fdida [d]

[a] CITI Lab., Project INRIA ARES, Bât L. De Vinci, 21 av. J. Capelle, 69621 Villeurbanne, France
[b] LIP/ENS Lyon, Université Lyon I, 46, allée d'Italie, 69364 Lyon Cedex 07, France
[c] INL, Université Catholique de Louvain, 2, Place St. Barbe, 1348 Louvain la neuve, Belgium
[d] LIP6/CNRS, Université Pierre et Marie Curie, Paris 6, 104 Avenue du Président Kennedy, 75016 Paris, France

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Abstract

In the widely used IEEE 802.11 standard, the so-called *performance anomaly* is a well-known issue. Several works have tried to solve this problem by introducing mechanisms such as packet fragmentation, backoff adaptation, or packet aggregation during a fixed time interval. In this article, we present and thoroughly analyze PAS, a dynamic and distributed approach solving the performance anomaly problem. PAS is based on packets' aggregation using a dynamic time interval, which depends on the wireless channel occupation time perceived by each node. Since each station senses the medium independently, this makes PAS a totally distributed solution. Even more, PAS may coexist with standard IEEE 802.11 nodes without any particular adaptation, yet being able to improve performance. Our solution differs from other propositions in the literature because of its dynamic and distributed nature, which makes it suitable in the context of multi-hop networks. Furthermore, it allows increasing fairness, reactivity, and in some cases efficiency. In this article, we thoroughly analyze and emphasize the performance evaluation of our proposal.

Keywords: IEEE 802.11; MAC protocol; Anomaly; Algorithm

1. Introduction

Performance anomaly is a key issue in IEEE 802.11 multi-rate wireless networks. It decreases the network global performance because of bad time sharing between stations transmitting at high bit rate (fast stations) and stations transmitting at low bit rate (slow stations). This bad time sharing causes an unfair throughput, with slow stations throttling fast stations' traffic [4].

Several solutions have been proposed in the literature to solve this problem. Some of them are based on a static predefined time sharing between slow and fast stations, by shaping the MTU (maximum transmission unit) on a transmission rate basis.

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* Corresponding authors.
E-mail addresses: tahiry.razafindralambo@insa-lyon.fr (T. Razafindralambo), isabelle.guerin-lassous@ens-lyon.fr (I. Guérin-lassous), luigi.iannone@uclouvain.be (L. Iannone), serge.fdida@lip6.fr (S. Fdida).

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Other approaches set a maximum amount of time a station can hold the medium, like with the TXOP (transmit opportunity) introduced in the IEEE 802.11e standard. Finally, other approaches try to adapt the contention window size of IEEE 802.11, according to the transmission rate of the station.

The main problem of existing solutions is that they are static or centralized. In this paper, we tackle both issues, solving the performance anomaly with a dynamic and distributed approach. Our solution is dynamic because it introduces a transmission time, as the TXOP, that changes depending on the perceived channel occupancy, which evolves with the traffic load of the network. Our solution is a distributed approach because each node computes locally the maximal channel occupancy time. The carrier sensing mechanism provided by IEEE 802.11 natively allows this computation. Once a node gains access to the medium, it can send a burst of packets. The number of transmitted packets is limited by the computed transmission time, which depends on the maximal occupancy time perceived by the station.

In this article, besides presenting our solution, which we called PAS, we also propose an analytical evaluation in a classical scenario where all stations are within communication range. A detailed and careful simulation-based evaluation is also given. We evaluate our protocol both in terms of efficiency and of fairness on many configurations, not limited to single-hop networks. We also analyze the improvements provided by our approach on both UDP and TCP traffic. Furthermore, we compare our solution to three different approaches that belong to the three main classes of solutions solving the performance anomaly. This article is an extended version of [12], which includes more performance evaluation results, protocol details, evaluation using TCP traffic, and the impact of the RTS/CTS mechanism.

The remainder of the paper is organized as follow. We give a short overview on the IEEE 802.11 access function, and describe the performance anomaly in Section 2. In Section 3 we propose a review of the existing modifications of the IEEE 802.11 solving the performance anomaly. In Section 4 we describe PAS, our proposal. In Section 5 we propose an analytical evaluation for a specific topology while in Section 6 we carry out extended simulations to evaluate the performance of our approach. We also study the impact of the different parameters of PAS on various scenarios. Finally, we conclude the paper with the perspectives raised by this work in Section 7.

2. The performance anomaly

The IEEE 802.11 standard [11] provides a fully distributed medium access protocol, called the distributed coordination function (DCF). The DCF belongs to the carrier sense multiple access with collision avoidance (CSMA/CA) family of MAC algorithms. Basically, the emitters have to wait for the channel to become idle before sending a frame. This frame is also protected by a collision avoidance mechanism.

In particular, when a frame is ready to be emitted, the emitter first waits until it senses the medium idle for a fixed amount of time called DIFS (distributed inter frame space). Once this condition has been achieved, the emitter generates a random number called *backoff* within an interval called *contention window* (CW). This number indicates the amount of time to be waited before really transmitting the frame. The backoff is a simple collision avoidance mechanism, since it strongly reduces the probability of colliding transmissions for synchronized emitters. If, while the backoff is decreased, the medium becomes busy, the decrementing process is stopped. When the medium becomes idle again, the station waits for a DIFS time before restarting to decrease its remaining backoff. As soon as the backoff reaches 0, the frame is emitted. Since collision detection is not possible, each unicast frame has to be acknowledged. When a receiver successfully receives a frame, it waits for a SIFS (short inter frame space) time and then emits the acknowledgment. The SIFS is shorter than the DIFS in order to give priority to acknowledgments over data frames. The lack of the reception of an acknowledgment is considered as a collision. In that case, the CW size is doubled and the same frame is emitted again with the same process described previously. If another collision happens, the CW size is doubled again if it has not yet reached the maximum value defined by the standard. After a fixed number of retransmissions, the frame is dropped. The CW size is reset when a frame is dropped or after a successful transmission.

Heusse et al. [4] have shown that the presence of slow stations in a multi-rate wireless network slows down all other stations. During the transmission of a slow station the medium is busy for a longer period than during the transmission of a fast station,

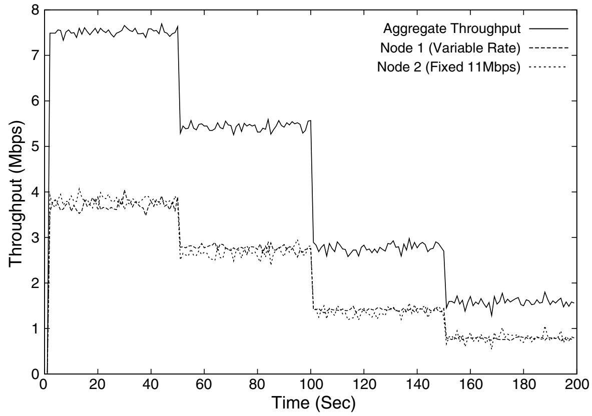

Fig. 1. Effects of the anomaly in the case of IEEE 802.11b.

assuming the same packet size. Since IEEE 802.11 provides simple per-packet fairness in a single-hop network, this means that in a long period each emitter has statistically sent the same number of frames. On a time basis, however, slow stations have occupied the channel for a longer period of time. This time unfairness that arise as soon as multiple rates are present, leads to a loss of performance due to the existence of slow transmissions. Fig. 1 shows a simple but clear example of the consequences of the anomaly in the case of IEEE 802.11b. The figure shows the simulation of the throughput that two stations can obtain when one station transmits always at 11 Mbps, while the other station starts at 11 Mbps, but lowers the transmission rate each 50 s. We can observe that the slow station throttles the throughput of the fast station, resulting in a lower aggregate throughput.

3. Related works

By letting both the fast and slow stations capture the channel for the same amount of time, the performance of IEEE 802.11 should be improved. The issue has been tackled in several different ways, with solutions placed at different levels of the protocols stack. Here we present the most relevant works that try to solve the performance anomaly by introducing tiny modifications in the IEEE 802.11 standard itself, as we do in our solution.

In this context, there are three main approaches: (i) packet fragmentation; (ii) contention window adaptation and (iii) packet aggregation. In the following subsections, we describe briefly each approach and we give few relevant examples to illustrate the state of the art.

3.1. Packet fragmentation approach

Packet fragmentation is the first and simplest approach. Iannone et al. [6] propose a solution based on a virtual time division scheme that reduces the performance anomaly of IEEE 802.11. In this solution packets of higher layers are fragmented according to the transmission rate at which they are sent at the 802.11 MAC level. The packet fragment size is fixed and computed offline. Simulation results, presented in that work, show that this solution reduces the performance anomaly while increasing global throughput. Nevertheless, the static nature of the proposed solution is efficient only for stations transmitting on the network at high bit rate and with a packet size equal to the MTU (maximum transmission unit). Furthermore, when only slow hosts are present in the network the performance decreases considerably, due to the overhead introduced by the high level of fragmentation in small packets. A similar approach is proposed by Dunn et al. [3], but at a higher level. The MTU discovery process is used to determine the packet size according to the transmission rate. This solution has the same poor performance as the previous one when only slow hosts are present in the network.

3.2. Contention window adaptation approach

The second category of solution is based on the modification of the backoff mechanism, in particular by changing the contention window (CW) size. Heusse et al. [5] propose a two-step mechanism scheme based on the station transmission rate. The first step is a protocol that tries to reach an optimal CW size. This optimal value (CW_{opt}) is computed according to the number of idle slots perceived on the medium by the station. Then, in a second step, this CW_{opt} is modified according to the transmission rate of the station and the maximum available transmission rate of the network. The proposed protocol reduces the performance anomaly while improving the throughput. The authors show that the main issue of the protocol is the way to compute the optimal windows. The optimal windows values are computed offline according to a fixed transmission rate. Another problem that can be encountered with this protocol is the long convergence time, especially when the stations' rates are changing frequently.

3.3. Packet aggregation approach

The third and last category is the packet aggregation approach, in which our solution is also included. This type of solution was first introduced by Sadeghi et al. [13]. The authors propose an opportunistic media access for multi-rate ad hoc networks. The solution is based on the fact that a station that is transmitting at a high rate is likely to have good channel conditions. It is thus allowed to send more than one packet to take advantage of this favorable situation. The number of successive packets to transmit is computed according to the basic rate of the network. For example if the basic rate is 2 Mbps and the channel conditions allow transmissions at 11 Mbps, the sender is granted a channel access time sufficient to send $11 \div 2 = 5$ packets. With this solution, the performance anomaly can be solved. However, if there are only fast stations on the network, short-term unfairness issues arise.

The packet aggregation solution is also proposed in the IEEE 802.11e standard [8]. In IEEE 802.11e, a transmission opportunity (TXOP), *i.e.*, a maximum channel occupation time, is granted to every station. This transmission opportunity is broadcasted by the base station to every node. The computation of TXOP is not really clear in the standard. As far as we know, it is computed according to the time needed to send the MTU at the lowest data rate. Thus, during a TXOP fast stations can aggregate their packets, while slow stations can only send one packet. The main problem of IEEE 802.11e is that it is centralized. Another problem with a static packet aggregation is that the performance anomaly is solved on one hand, but short time unfairness may arise on the other hand.

To solve the performance anomaly and at the same time this possible short time unfairness issue, we propose a dynamic packet aggregation policy. Our solution is different from the other aggregation solutions because it is not centralized but totally distributed, and because it is not static but totally dynamic. The transmission time is computed dynamically at each node, according to simple information perceived on the medium as described in the next section. Our approach does not need any additional information except that already provided by the IEEE 802.11 standard.

4. PAS: a dynamic packet aggregation

The idea of our protocol, called PAS (performance anomaly solution), is based on the fact that each station should have the same transmission time on the radio channel. Therefore, if an emitter senses a channel occupancy time that is longer than the transmission time of the current packet to be emitted, then it can aggregate more packets in order to get a better channel occupancy time. The aggregation is realized by spacing the reception of the previous packet's acknowledgment and the emission of the next packet with a SIFS. There are two main mechanisms in PAS: the first one is the medium sensing mechanism that computes the transmission time; the second one is the packets' sending, based on the transmission time computed previously.

4.1. Computing the transmission time

The first mechanism for the computation of the allowed transmission time is given in Algorithm 1. A station always senses the radio medium, and maintains the channel occupancy time. This time is the channel busy time due to a transmission, including transmission that can be only sensed but not decoded (*i.e.*, in the carrier sensing area). Each station maintains the maximum channel occupancy time in a variable called *t_p_max*. This parameter is set to 0 after each successful transmission of the station.[1] This avoids the stations monopolizing the channel after a transmission and improves the reactivity of the protocol. Furthermore, this mechanism reduces the short time unfairness that can arise when the same node successively accesses the radio channel.

It is worth noting that with this approach, the computed transmission time will never correspond to the time required for an exchange of packets like Data-ACK or RTS-CTS-Data-ACK. Indeed, this time is deduced from a continuous signal and will be recomputed as soon as there is a silence period. Moreover, it is very difficult to determine these exchange times since our computation takes into account signals in the carrier sensing area. It is

[1] Here the algorithm is described for point-to-point communications but the same process holds for broadcasted communications.

not always possible to distinguish a control packet (RTS, CTS or ACK) from a data packet with the same transmission time.

Algorithm 1. Performance Anomaly Solution – Sensing Phase

```
t_p_max := 0;
repeat
  if (a signal is sensed at the physical layer) then
    t_p_current := signal's channel occupancy time;
    if (t_p_current > t_p_max) then
      t_p_max := t_p_current;
    end if
    if (packet type == ACK) and (Destination == me) then
      t_p_max := 0;
    end if
  end if
until 1;
```

4.2. Packet emission

The second mechanism concerns the emission phase and is given in Algorithm 2. The station can either transmit its packet classically by using the medium access mode of IEEE 802.11 or aggregate some of its packets. To know whether it can aggregate or not, it uses the parameter *t_p_max*: if its channel occupancy time is smaller than the value of this variable, then it can aggregate. In Algorithm 2, *t_my_packet* is the time required to send the current packet, while *t_my_left* corresponds to the remaining allowed transmission time. The value of this last parameter evolves with time and with the packets previously emitted. When this value becomes too small, no more aggregation is possible. Otherwise the medium occupancy time of this station would become larger than the maximum transmission time sensed on the channel, which is not fair.

The boolean variable *sending* indicates whether the current packet to send is the first packet to be emitted or not. If it is the first (*sending* set to *false*), the packet has to be emitted with the classical medium access of IEEE 802.11. If it belongs to an aggregated packets burst (*sending* set to *true*), in this case the packet is sent after a *SIFS*.

Algorithm 2. Performance Anomaly Solution – Emission Phase

```
sending := false;
t_my_left := 0;
for (each packet to send) do
  if (t_my_left ⩽ 0) then
    t_my_left := t_p_max;
  end if
  α = (⌈t_my_left / t_my_packet⌉ − t_my_left / t_my_packet) ∗ t_my_packet;
  t_my_left := t_my_left − t_my_packet;
  if (sending==true) then
    if (t_my_left + α > 0) then
      aggregated_sending();
    else
      t_my_left := 0;
      sending := false;
      classical_sending();
    end if
  else
    if (t_my_left + α > 0) then
      sending := true;
      classical_sending();
    else
      t_my_left := 0;
      classical_sending();
    end if
  end if
end for
```

The parameter α is used to maintain a good overall throughput. Indeed, let us consider a scenario with two emitters, one at 11 Mbps and one at 5.5 Mbps, both sending packets of the same size. Due to the physical header overhead (the physical header is sent at the same basic rate whatever the data transmission rate), the time for transmitting two packets at 11 Mbps is a little bit longer than the time for transmitting one packet at 5.5 Mbps. Therefore, without the use of the variable α, the fast station will never aggregate and the performance anomaly will still remain present. By choosing:

$$\alpha = \left(\left\lceil \frac{t\_my\_left}{t\_my\_packet} \right\rceil - \frac{t\_my\_left}{t\_my\_packet} \right) * t\_my\_packet \tag{1}$$

packet aggregation and good aggregated throughput are ensured, due to the over-approximation of the transmission time. Note that this parameter is the smallest over-approximation of the transmission time. A new value of α is computed at each new packet arrival at the MAC layer. Thus, we have a real dynamic approach adapted to the current traffic. Furthermore, such an approach does not require a specific assumption on the packet size.

If a collision occurs on a packet sent on an aggregated packets burst, then the transmission is deferred after a SIFS if *t_my_left* is large enough to send the packet once again. Otherwise if *t_my_left* is too small, the backoff window size is increased according to the binary exponential backoff scheme and *sending* is set to *false*, while *t_my_left* is set to 0. For the sake of clarity this process is not described in the algorithm sketch.

4.3. Further improvements

The transmission time is determined by computing on line the number of packets that can be emitted and whose total time corresponds to the maximum channel occupancy perceived on the channel. The transmission time of one packet includes the time to transmit the packet header. Therefore, if a fast station aggregates many small packets, a lot of time is lost due to overhead. Thus, the overall throughput of network may not be very good. To improve the overall throughput, it is possible to penalize the stations that send small packets. An easy way to do it is to compute the ratio between packet payload and packet header (including acknowledgment), we call this ratio *t_rate*, and we use this parameter to limit the aggregation. In our proposition (PAS), the computation of the next value of *t_my_left* is conditioned by the value of *t_rate*. In particular, the instruction 8 of Algorithm 2 is changed to the following:

if ($t\_rate < 1$) **then**
  $t\_my\_left := t\_my\_left - ((1/t\_rate) * t\_my\_packet)$;
else
  $t\_my\_left := t\_my\_left - t\_my\_packet$;
end if

At each step the remaining time for the aggregation is reduced for a station that sends small packets. If at the next step, the packet does not satisfy this condition $t\_rate < 1$, then *t_my_left* is computed normally.

In order to be compatible with all the IEEE 802.11 features, the protocol must also handle the RTS/CTS mechanism. In this case, PAS uses the duration time given in RTS and CTS frames to update its maximum occupancy time if this duration time is greater than the maximum occupancy time computed previously. The parameter *t_my_left* is still computed like in Algorithm 2. When $t\_p\_max \geqslant t\_my\_packet$ and $packet_{length} \geqslant RTS_{thresh}$, then the exchange is as follow: RTS-CTS-DATA-ACK-SIFS-DATA-ACK... and so on. The duration time in the RTS and CTS is the duration for only one packet transmission. There are two reasons to not put the value of *t_p_max* (aggregation time) in the duration field of the RTS and CTS frames: (i) since the number of packets in the LL (Link Layer) queue is not known *a priori* when a RTS is sent, it is possible that the emitter will not use its whole transmission time, which will unnecessarily stop some potential emitters; (ii) reactivity is improved. If, for example, we assume two fast stations and one slow station, the two fast stations may aggregate their packets based on the transmission time of the slow station. If the slow station stops emitting and *t_p_max* was announced in RTS and CTS frames, the two fast stations will maintain their aggregation, which is not fair from a short-term point of view. This happens because the duration field remains the same for these two stations.

With PAS, the collisions, when RTS/CTS mechanism is used, are solved in the following way. If a collision occurs on a RTS, the RTS is retransmitted according to the IEEE 802.11 protocol, *i.e.*, after a backoff window increment. When a collision occurs on the data, the data packet is sent after a SIFS, if *t_my_left* is large enough to send the packet again. If *t_my_left* is not large enough, then a RTS is sent after a backoff window increment.

5. Theoretical analysis

In this section, we investigate the efficiency and the fairness achieved by PAS. Tan et al. [14] have proposed the notion of time-based fairness that gives to each node an approximately equal occupancy of the channel. They show that providing a time-based fairness is more efficient than a mechanism that is fair in the medium access. The solution they propose takes into account the time required for the exchange DATA-ACK for the computation of the transmission time.[2] In PAS, the computation of the transmission time is instead based on the maximum channel occupancy. In the following, we

[2] The work has not been described in Section 3, since the solution is also considered at upper layers and not only at the MAC layer.

show that PAS is more efficient than solutions based on DATA-ACK exchanges and we study the time-based fairness of PAS.

5.1. Efficiency

The transmission time in our protocol is based on packet time and not on the time required for an exchange. An exchange time can be defined as $T\_ex = t\_my\_packet + T\_SIFS + T\_PHY + T\_ACK$, where T_SFIS is the duration of a SIFS, T_PHY is the duration of the PHY header and T_ACK is the time duration of an ACK. We denote by t_p_max the maximum channel occupancy time; by t_my_packet the time required to transmit the packet;[3] and by T_ack the sum of $T\_SIFS + T\_PHY + T\_ACK$. We assume that T_ack is independent from the data rate at which a node transmits and is a constant. We also assume a scenario with two stations within the communication range (one fast station and one slow station). The stations use the same packet length. The number of packets sent by the fast station with PAS is given by

$$n_a = \frac{t\_p\_max}{t\_my\_packet}. \quad (2)$$

While the number of packets sent by the fast station using the exchange time for the aggregation, like in the work of Tan et al. [14], is given by

$$n_{et} = \frac{t\_p\_max + T\_ack}{t\_my\_packet + T\_ack}. \quad (3)$$

We have $t\_my\_packet \leqslant t\_p\_max$. Thus, with these assumptions:

$$n_a \geqslant n_{et}. \quad (4)$$

Therefore, each time the slow station sends a packet, the fast station, in its next transmission, will aggregate more packets with PAS than with the solution proposed by Tan et al. [14], showing the higher efficiency of PAS. Furthermore, the maximum aggregate throughput is obtained when the fast station aggregates as many packets as possible, on the basis of the transmission time of the slow station.

5.2. Fairness

In this section, we investigate the time-based fairness as discussed by Tan et al.. For the sake of simplicity, in this analysis we assume that each node uses the same packet length L (expressed in bytes). We also assume that T_i with $i = 1, 2, 5.5, 11$ is the time needed to transmit a packet at rate i Mbps. T_i includes the transport layer header, the network layer header, the MAC layer header and PHY layer header. We can easily compute the aggregated time T_Agg_i used by a station transmitting at rate i as

$$T\_Agg_i = n_{a_i} \times (T_i + T\_ack) + (n_{a_i} - 1) \times T\_SIFS, \quad (5)$$

where $n_{a_i} = t\_p\_max/T_i$. From the medium point of view, the time proportion used for an aggregated transmission of one node is

$$T\_Occ_i = \frac{T\_Agg_i}{\sum_j (T\_Agg_j \times N_j) + N \times DIFS}, \quad (6)$$

where N_j is the number of stations transmitting at a data rate j, with $\sum_j N_j = N$. We assume here that the probability to access the medium is the same for all the nodes and that during a time interval; each node has accessed the medium exactly once. The number of packets sent by a node transmitting at a data rate i, in a time interval t, is

$$NBp_i = \frac{n_{a_i}}{\sum_j (T\_Agg_j \times N_j) + N \times (DIFS + T\_Avg_{bckf})} \times t, \quad (7)$$

where T_Avg_{bckf} is the average backoff time. We can thus derive the average throughput in bps of a station transmitting at a data rate i with the following equation:

$$TH_i = NBp_i \times L \times 8. \quad (8)$$

All the above results can be applied with different packet sizes, the main parameter to know is t_p_max. In this analysis, we assume that stations access the medium in a TDMA mode, *i.e.*, one station after the other. This assumption is legitimate due the fair access provided by the backoff scheme implemented in the DCF of IEEE 802.11. Nevertheless, as we will see in Section 6, due to this assumption there are some small differences that arise between the analytical results and the simulation results. Indeed, IEEE 802.11 does not provide a perfect TDMA scheduling over a short-term time scale.

Fig. 2 shows the proportion of medium occupancy time for two stations obtained analytically. One of the two stations transmits always at 11 Mbps

[3] Note that this time includes the time of the PHY header.

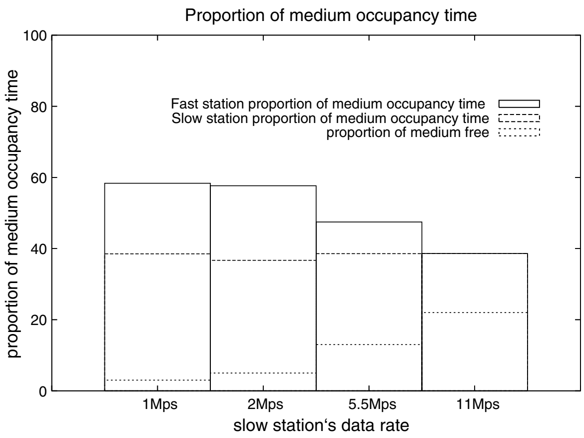

Fig. 2. Proportion of medium occupancy time for two stations.

while the other transmits at 1, 2, 5.5, or 11 Mbps (on the x-axis; i Mbps indicates that one station emits at i Mbps while the other emits at 11 Mbps). Packet size is equal to 1000 bytes. For each i, this figure gives the proportions of medium occupancy time of the fast station (11 Mbps) and of the slow station (i Mbps) and the time proportion when the medium is free. We can see that the fast station gets a larger proportion of medium occupancy than the slow station and that the proportion of each station is not 50% as it should be with a perfect time-based fairness. This difference may be easily explained by the fact that the allowed transmission time computed with PAS does not take into account the acknowledgments that consume transmission time. We can also see from this figure that the higher the data rate of the slow station, the higher the proportion of medium free time. This is due to the proportion between the backoff time and the medium occupancy time that increases.

Table 1 shows the throughput obtained by Eq. 8. We included the Jain fairness index [9] to evaluate the fairness of our solution. The Jain index is defined as

Table 1
PAS: analytical results

	Throughput (kbps)	Number of packets (1/s)	Index
5.5 Mbps	1547.2	193.4	0.98
11 Mbps	3095.2	386.9	
2 Mbps	624.8	78.1	0.93
11 Mbps	3749.6	468.7	
1 Mbps	344.8	43.1	0.92
11 Mbps	3791.2	473.9	

$$FI = \frac{\left(\sum_i r_i/r_i^*\right)^2}{n\sum_i (r_i/r_i^*)^2}, \tag{9}$$

where r_i is the rate achieved on flow i, n is the number of flows, and r_i^* is the reference rate on flow i. As reference rate we use the one defined by Tan et al. This rate r_i^* is computed as if all the flows in the wireless networks were emitted at the same rate as flow i. For example, if we consider two nodes transmitting at 11 (flow 1) and 1 Mbps (flow 2), then r_1^* will be the throughput of flow 1 if flow 2 is transmitted at 11 Mbps. In the same way, r_2^* will be the throughput of flow 2 if flow 1 is transmitted at 1 Mbps. The value of r_i^* is the throughput value when the medium occupancy time is equal for all nodes. This is the reason why the indices computed in Table 1 are not equal to 1.

6. Simulations results

In order to evaluate PAS, we implemented it on the NS-2 simulator [7] as an independent MAC. Furthermore, we also added multi-rate features to NS-2, in order to reflect the IEEE 802.11b modulations. All the studies listed below are done in steady state condition (*i.e.*, nodes do not move). In order to reduce the simulation time and to better evaluate the protocol, ARP and routing protocol exchanges are disabled. In all simulations, if not differently stated, UDP saturated traffic is used. Nevertheless, we will also describe tests performed using TCP traffic. If not differently specified, each packet contains 1000 bytes of data. Nevertheless, we also performed tests with different packet sizes. For this purpose, we used a specific module developed to generate packets of a random size, uniformly distributed in a specific interval. This module is able to vary the time spacing between two successive packets in order to maintain a constant data rate. Note that the fairness index of each simulation is given in brackets in the x-axis of some figures.

6.1. Model validation

In order to validate the improvements to NS-2 and the code of our proposal, we first simulate two pairs of station transmitting at 11 Mbps with 1000 bytes of data. In this simulation, no aggregation is done because the maximum occupancy time perceived by each node is equal to the time required to send a packet. In this specific case, the throughput of IEEE 802.11 and PAS should be the same.

Table 2
Model validation

		Throughput (kbps)	Conf. Int. (0.05)
802.11	11 Mbps	2747.04	[2731.35; 2762.72]
	11 Mbps	2752.80	[2736.80; 2768.81]
	Total	5499.84	[5491.02; 5508.66]
	Index	0.99999	
PAS	11 Mbps	2740.61	[2726.91; 2754.30]
	11 Mbps	2753.71	[2740.51; 2766.92]
	Total	5494.32	[5485.78; 5502.86]
	Index	0.99999	
Theoretical	11 Mbps	2802.5919 (kbps)	
	11 Mbps	2802.5919 (kbps)	
	Total	5605.1839 (kbps)	

This is confirmed by the results presented in Table 2, which includes the theoretical throughput derived from Section 5, in order to show the accuracy of our model.

6.2. Basic simulations

This section contains the first simulation results of PAS. The simulation carried out is based on the classical scenario where two stations transmit packets of 1000 bytes, one at x Mbps (x equal to 1, 2, 5.5 or 11) and the other at 11 Mbps.

Fig. 3 gives the simulation results in this scenario. The x-axis gives the tested scenario, where in brackets the fairness index is given. The y-axis gives the achieved throughputs (in kbps): the striped block gives the throughput of the 11 Mbps station, the white block gives the throughput of the x Mbps station, while the top of the two blocks gives the overall throughput. This figure clearly shows that the aggregate throughput of PAS is always greater than the IEEE 802.11 one, proving that PAS is more efficient. It can also be observed that when using PAS, the throughput of the fast station remains almost the same, independent of the rate used by the slow station. This is because the time occupation is roughly equally shared between the fast station and the slow station. Thus, since the total simulated time is always the same, the fast station has always the same occupation time and sends the same amount of packets. The fairness index (given in brackets) shows that PAS achieves a very good fairness in terms of medium occupancy in these scenarios.

Fig. 4 shows the simulation results for four transmitting stations. The transmission rates of each scenario are respectively at $\{1, 2, 5.5, 11\}$ Mbps, at $\{1, 1, 1, 11\}$ Mbps and at $\{1, 1, 5.5, 11\}$ Mbps. The results clearly show that the aggregate throughput of PAS is always greater than the aggregate throughput of IEEE 802.11. The throughput for the fast station at 11 Mbps with PAS remains almost the same in the different scenarios. This is because the time accorded to each station to send its packets is based on the slowest packet time transmission. The fairness index also shows that PAS is fair in terms of medium occupancy.

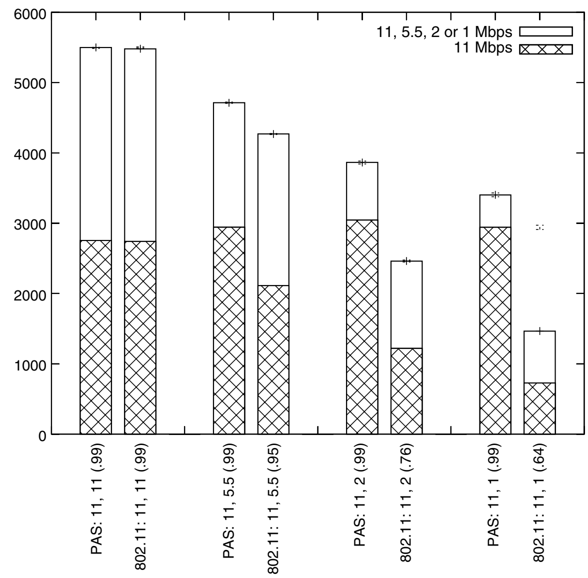

Fig. 3. PAS vs. IEEE 802.11 in a two stations scenario.

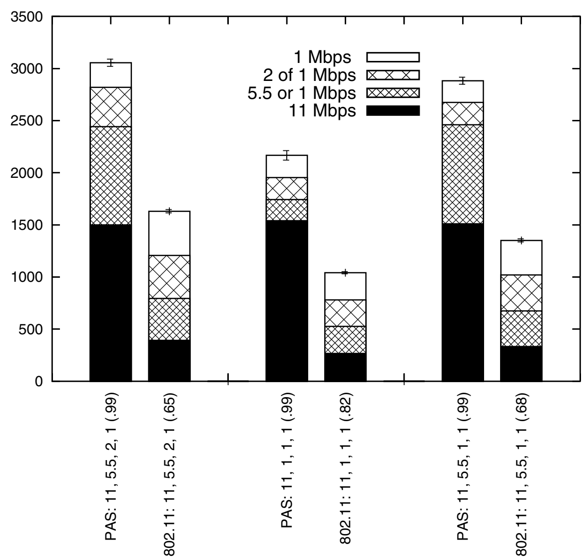

Fig. 4. PAS vs. IEEE 802.11 in a four stations scenario.

The difference between the theoretical results (Table 1) and the simulation results can be explained by the backoff algorithm present in the IEEE 802.11 MAC. Indeed, the backoff algorithm does not provide a TDMA-like access to the medium. When there are only two stations, due to the randomness of the backoff, each station can successively access the medium. In the case of PAS, the fast station will first aggregate packets during its transmission time. Nevertheless, if it accesses successively the medium when its transmission time is elapsed, it will send its packets using normal IEEE 802.11 access. Therefore, this feature of PAS reduces the throughput of the fast station because it does not always aggregate its packets. This reduction can be worsened when the slow station also sends successive packets. The difference between the analytical results and the simulation results increases when the difference in the rates of the two stations increases.

6.3. Dense single-hop cell

In this section we present the simulation results run in an IEEE 802.11 (resp. PAS) cell with an increasing number of nodes. The simulations are carried out with a fixed number of stations, with half of them using 11 Mbps data rate and half of them transmitting at x Mbps (x equal to 5.5, 2, 1). The packets size at each station is randomly and uniformly distributed within [550; 1450] bytes.

Fig. 5 gives the total throughput in such configurations with an increasing number of transmitters. We can see that PAS outperforms IEEE 802.11, especially when the slow stations transmit at 2 or 1 Mbps. The larger confidence interval of PAS is due to the aggregation scheme which can allow more throughput variation than IEEE 802.11. We can also note that the overall throughput is a little bit higher with PAS than with IEEE 802.11 when all the stations transmit at 11 Mbps. This is due to the fact that the packet size is not fixed and chosen randomly in an interval. Therefore, PAS may sometimes aggregate some packets. Nonetheless, the overlapping of the confidence intervals shows that this increase is limited.

6.4. Reactivity

A way to test the reactivity of PAS is to introduce the well-known auto-rate fallback (ARF) mechanism used by wireless stations to adapt their transmission rate to the channel conditions. We have implemented the ARF mechanism to see the behavior of PAS when the transmission rates of stations vary in time. The simulation is done using two emitters transmitting to a third station (*i.e.*, an Access Point), with one station moving away and the other remaining static. Note that the moving station follows a simple linear direction at constant speed, no particular mobility model has been used. Note that the mobility model should only impact the medium occupancy time. Since, in this simulation, we only consider a one-hop cell, then the medium occupancy time depends on the rate selected by the ARF mechanism. Therefore, the point here is to evaluate the reactivity to such a mechanism.

At the beginning all nodes are sufficiently close such that 11 Mbps can be used, then the mobile station will decrease gradually the transmission rate due to the increasing distance from the AP. Fig. 6 shows the simulation results with PAS and IEEE 802.11. We can see from this figure that when using PAS, the throughput of the fast station remains constant, while the throughput of the moving station decreases. With IEEE 802.11, the throughput of both emitters (static and mobile) decreases.

6.5. Delay

In this section we present a simulation of 20 s with two emitters: one with a data rate of 11 Mbps

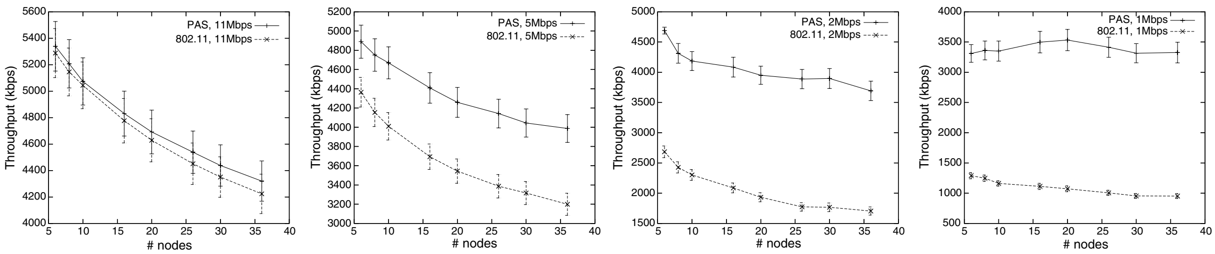

Fig. 5. PAS vs. IEEE 802.11 with different cell densities.

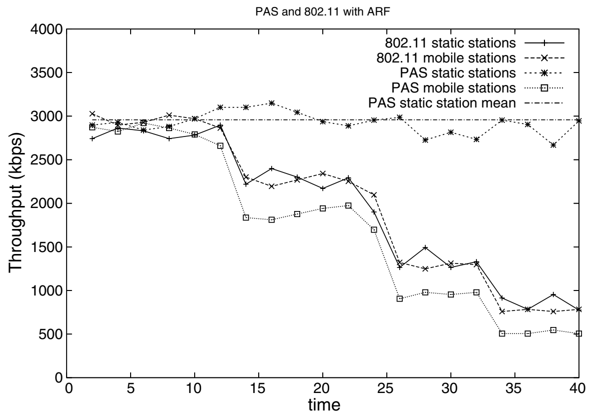

Fig. 6. PAS vs. IEEE 802.11 in presence of the ARF mechanism.

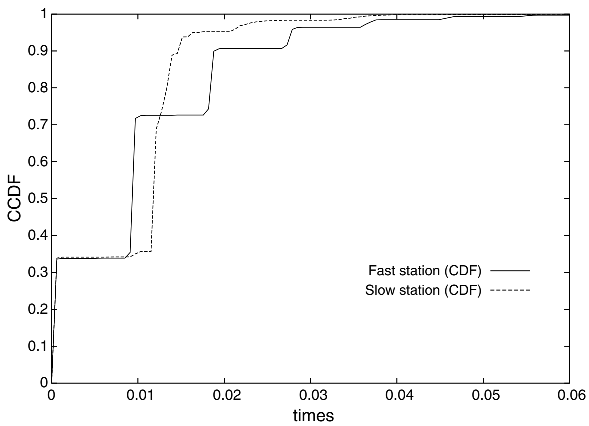

Fig. 7. Cumulative inter-burt time distribution for the fast and slow stations.

and the other with a data rate of 1 Mbps. During this simulation we compute the inter-burst time. An inter-burst time is defined as the time between the end of a burst and the beginning of another burst from the same station. For the station transmitting at the lower data rate a burst consists always of a single packet. For the station transmitting at the higher data rate, a burst can be either a real packet burst (several aggregated packets) or a single packet if the wireless station accesses the medium immediately after a burst.

Table 3 gives the number of the bursts sent and the average inter-burst time for the two stations. One can observe that IEEE 802.11 (and in turns PAS) provides a fair access to the medium, since the number of bursts for the slow and the fast stations is nearly the same. The table also shows that the average inter-burst time is close to the packet transmission time of the slow station (8576 μs).

Fig. 7 shows the cumulative inter-burst time distribution for the fast and slow stations. This distribution has the shape of a stair function with very big steps for very small values of time. One can easily see that the medium access provided by the backoff algorithm is not really a TDMA-like access due to the two big steps close to 0 in the figure. This means that there are many packets that are sent successively with the backoff algorithm of IEEE 802.11. Such behavior reduces the performance of PAS. In this figure, the difference (in time) between two steps for the fast station is close to the packet duration of the slow station. So the presence of successive steps shows that the slow station can send many successive packets. This confirms what we claim in Section 5 about the difference between simulation and analytical results. Concerning the slow station, this figure clearly shows that the average inter-burst time is close to the time needed by the fast station to transmit aggregated packets. We can also see that the distribution for the slow station is different from the distribution of the fast station. The reason is that even if the fast station can send successive packets, it is just for the transmission of a single packet and not for a burst, which explains the main step around 0.012 s. This also explains that the average inter-burst time of the slow station is smaller than the one of the fast station.

Table 3
PAS: burstiness

	Nb bursts	Avg inter-burst
Fast	5911	9867.70 μs
Slow	6004	8776.46 μs

6.6. Effect of α

In this section, we investigate the effect of the α parameter on the performance of PAS. We simulate two emitters transmitting 1000 bytes of data at 11 Mbps and at 5.5 Mbps. The simulation is carried out with and without the use of α. The results shown in Fig. 8 show that in this specific simulation, when α is not used, there is no aggregation. Indeed, in this case the condition $t\_my\_left - t\_my\_packet > 0$ never holds for the fast station, thus it does not perform any aggregation.

We have also simulated a scenario with four emitters, respectively at 1, 2, 5.5 and 11 Mbps. The results of this second simulation set are resumed in Fig. 9. From both Figs. 8 and 9, we

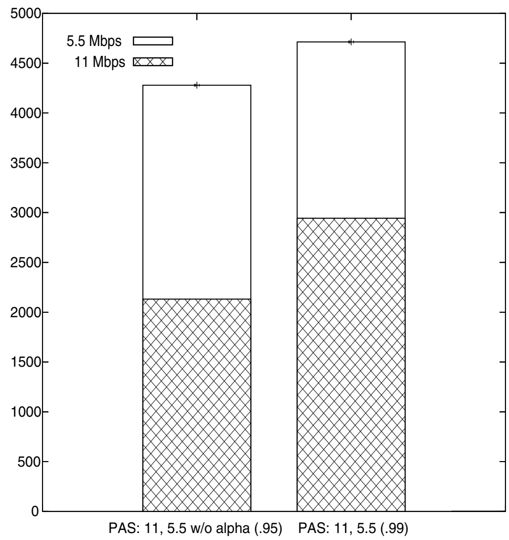

Fig. 8. The influence of α on performance in a two stations scenario.

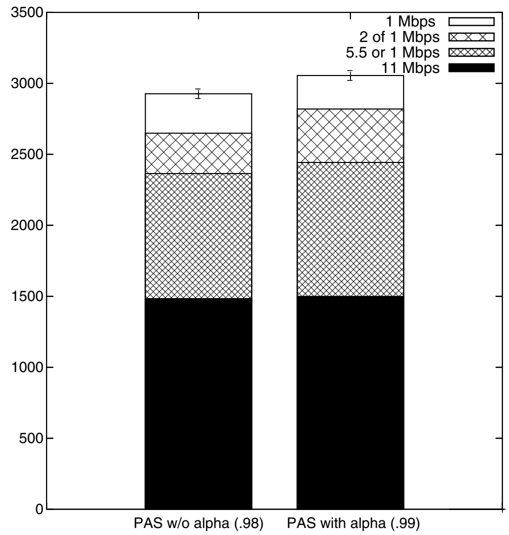

Fig. 9. The influence of α on performance in a four stations scenario.

can see that α increases fairness and efficiency. Indeed, when α is used, the proportion of medium occupancy for the fast stations is increased.

6.7. Effect of *t_rate*

Another important parameter of PAS is *t_rate*, introduced in Section 4.3. This parameter controls the time left for an aggregated transmission. It

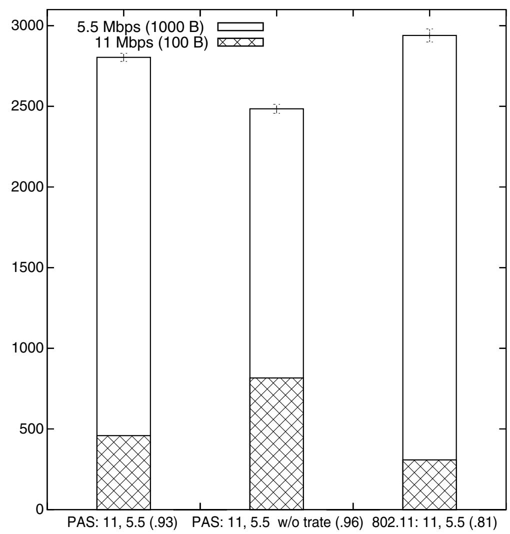

Fig. 10. The influence of *t_rate* on performances in a two stations scenario.

increases or reduces the aggregated transmission time, depending on the ratio between payload and the header. Fig. 10 gives the results of simulation runs with two emitters, one transmitting at 11 Mbps with packets of 100 bytes length, the other transmitting at 5.5 Mbps with packets of 1000 bytes length. From this figure we can observe how *t_rate* improves the global throughput of the network, but this overall throughput is smaller than in the case of IEEE 802.11. There are several possibilities to improve the use of *t_rate*. For instance, if *t_rate* $\leqslant 1$, setting *t_my_left* to 0 will stop the aggregated sending if a small packet was sent. The problem by using this scheme is that when a small packet from upper layer arrives (such as ACK from TCP protocol), it always penalizes the wireless station when it gains the access to the medium.

Fig. 10 also shows how *t_rate* has a negative impact on fairness. This is because *t_rate* is used to reduce the aggregation time. In this particular scenario, it appears that there is a trade-off between fairness and efficiency. We argue that PAS provides this good trade-off, as Figs. 11 and 12 confirm. One can see from these figures that when using the *t_rate*, PAS is not as efficient as IEEE 802.11 for small values of *t_rate*, however, the aggregated throughput of the two solutions are close (Fig. 11). Furthermore, for small values of *t_rate*, the fairness index of PAS using *t_rate* is lower than the fairness index of PAS not using *t_rate*, however, they are very close (Fig. 12).

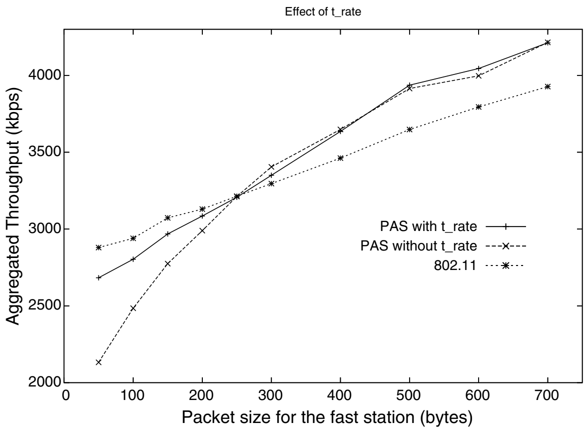

Fig. 11. Aggregated throughput depending on the packet size.

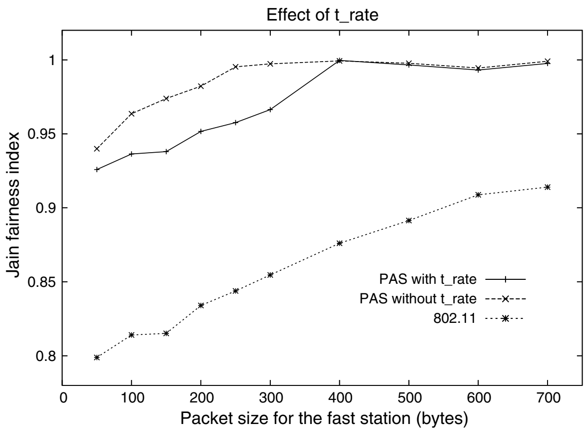

Fig. 12. Fairness index depending on the packet size.

6.8. Comparison with some other solutions

We have also compared PAS, our proposal, to other solution. The results we obtained are presented hereafter.

6.8.1. A simple backoff-based approach

We have developed a simple backoff-based approach to solve the performance anomaly. This approach is based on the solution proposed by Heusse et al. [5]. The size of the contention window (CW) is adapted in the following way:

$$\mathrm{CW} = \mathrm{CW} * \frac{11e6}{dataRate}. \tag{10}$$

In the simulations, the size of packets is uniformly distributed in the interval [550; 1450] bytes and there are two emitters, one transmitting at 5.5 Mbps and the other at 11 Mbps. Fig. 13 gives the different

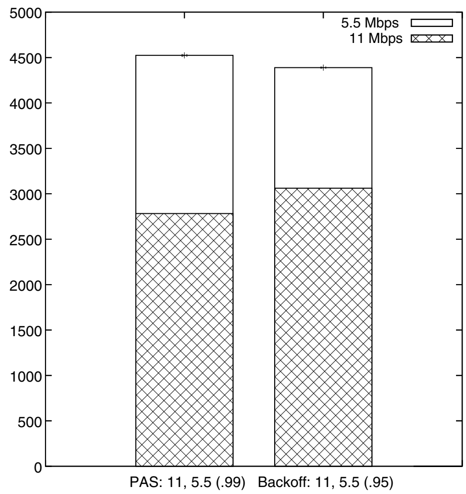

Fig. 13. Performance of the backoff-based approach.

average throughputs for the two protocols as well as the fairness index. It can be observed that this approach is efficient, but not as efficient as our solution. This is due to the overhead introduced for each packet by the backoff algorithm. Another problem of this approach is when small packets are sent by the fast station. In this case, the performance of the backoff-based approach decreases even more.

6.8.2. Packet division approach

We have also tested the packet division approach proposed by Iannone et al. [6]. The simulations are carried out with two emitters, one of them transmitting at 11 Mbps and the other at 5.5 Mbps. The packet size of the fast station is set to 1500 bytes, while the packet size of the slow station is set to 727 bytes due to the fragmentation required in this solution. In the simulation, the two packet sizes are set to 1500 bytes with PAS. Fig. 14 shows the results of these simulations. It can be observed from this figure that the packet division approach is less efficient, due to the increased overhead introduced by the backoff and the header when fragmenting packets. It would also be trivial to show that when all wireless stations in the network use a small data rate, the network performance is reduced because the packet fragmentation increases the payload/header ratio.

6.8.3. Fixed time aggregation approach

To carry out this simulation we have modified our implementation of PAS, introducing a fixed

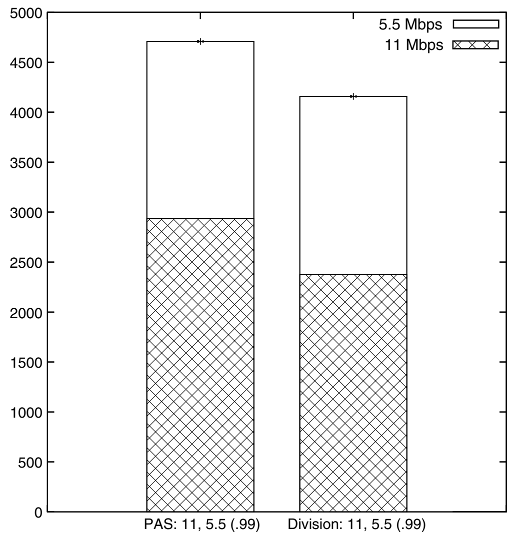

Fig. 14. Performance of the packet division approach.

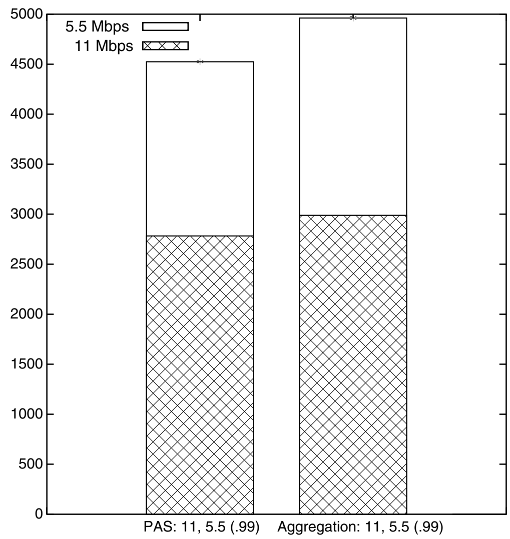

Fig. 15. Fixed aggregation time.

$t\_p\_max = 8000$ µs. With this value, a node transmitting a 1500 bytes data at 1 Mbps can send only one packet. From Fig. 15, it can be observed that the aggregation using fixed time is more efficient than our approach. This is due to the fact that, differently from PAS, the aggregation is always used. On the other hand, this permanent aggregation implies longer delays between bursts. Table 4 shows the number of bursts and the average time between two bursts emitted by the same station. The results in this table clearly show that the delay induced by PAS is much smaller compared to the other approach.

Table 4
PAS vs. fixed aggregation burstiness

		Number of bursts	Avg. inter-access
Fixed	5.5 Mbps	7123	11,230.07 µs
	11 Mbps	6666	12,000.80 µs
PAS	5.5 Mbps	19,570	4087.80 µs
	11 Mbps	19,346	4135.11 µs

6.9. Hidden terminals

In Section 4, we proposed a RTS/CTS mechanism for PAS. Here we evaluate this mechanism by simulating the case of two hidden nodes. The RTS/CTS threshold is set to 200 bytes and packet size to 1000 bytes. In order to evaluate the performance of PAS in this scenario, one of the hidden nodes uses a data rate of x, where $x \in \{1, 2, 5.5\}$ Mbps, while the other sends at 11 Mbps. Fig. 16 shows the results of these simulations. We can see that PAS is more efficient and fairer than IEEE 802.11, when one of the pairs has a data rate of 1 or 2 Mbps. This is because more aggregated packets can be sent by the fast station. On the other hand, we see that the results of PAS at 11 and 5.5 Mbps are very close to the ones of IEEE 802.11. Since the time duration in the RTS corresponds to the transmission time of the packet to send, a collision is only likely to occur on the second

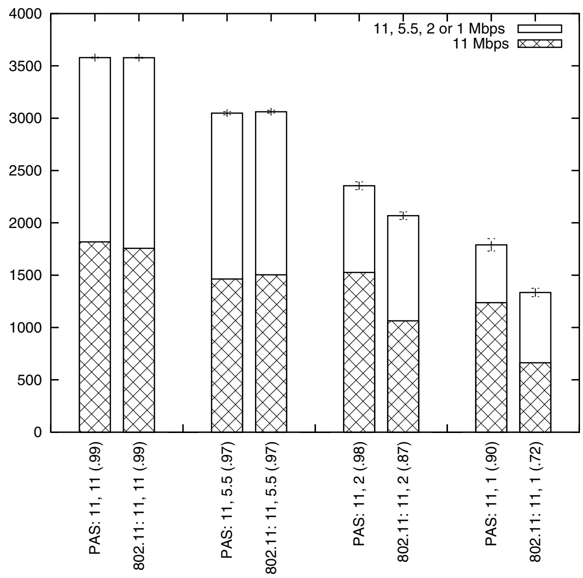

Fig. 16. PAS with RTS/CTS mechanism.

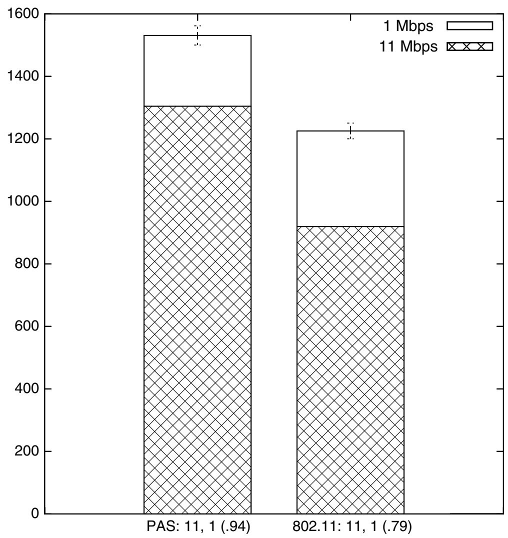

Fig. 17. The RTS/CTS mechanism with 1 and 11 Mbps nodes and dynamic packets' size.

packet of the aggregated series. With 11 and 5.5 Mbps, *t_my_left* is not large enough to aggregate the packet again, whereas with 11 and 2 Mbps or 11 and 1 Mbps, *t_my_left* is large enough to aggregate the retransmission of the packet that has collided. In these two latter configurations, after some collisions, the contention window of the slow station is large enough to allow the aggregated sending of the fast station.

Fig. 17 shows the simulation results for two hidden nodes transmitting at 1 and 11 Mbps, with a packet size dynamically changed and uniformly distributed between [550; 1450] bytes. In this simulation we set the RTS threshold to 1000 bytes. We can observe from these results that, even with different packet sizes, thus with a different RTS/CTS policy for each packet (*i.e.*, the RTS/CTS is not always activated), PAS is more efficient and fairer than IEEE 802.11. Note that in this simulation, the value of *t_p_max* when RTS/CTS is not used corresponds to the transmission time of the acknowledgment.

6.10. Asymmetric TCP flows

In order to evaluate PAS when running TCP [2] flows, we run a set of simulation based on a well-known TCP issue: the asymmetric bandwidth problem [10]. Fig. 18 describes the scenario we have run. In this scenario, A is the fast station (11 Mbps) and B the slow one (2 Mbps). Each station has a saturating TCP flow TCP_{ab} from station A to station B and

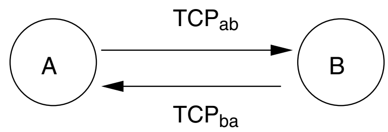

Fig. 18. Scenario used for TCP flow simulation.

TCP_{ba} for the flow from B to A. Note that TCP uses the same Link-Layer queue for TCP data and TCP acknowledgment. Thus, the throughput of the fast station will degrade due to the delay and loss induced by the packet transmission of the slow station. Actually, the contention window (at TCP level) of the fast station will remain small and thus reduce its throughput.

Fig. 19 shows the throughput of the two flows when IEEE 802.11 and PAS are used. PAS is much more efficient than IEEE 802.11 because the slow station can also aggregate the ACKs (from TCP level), which increases the throughput of the fast station. Here computing the fairness index is not a relevant indication because it is very difficult to obtain the ideal time sharing due to the acknowledgment flows generated by the data flows at TCP level. These results show that a simple dynamic packet aggregation in the MAC layer can improve the performance of upper layer protocols. This improvement can be seen in Figs 20 and 21. These figures plot the contention windows size evolution during 40 s. They show that with PAS the contention window of the fast station is greatly increased. This is due to the possible aggregation of acknowledgment

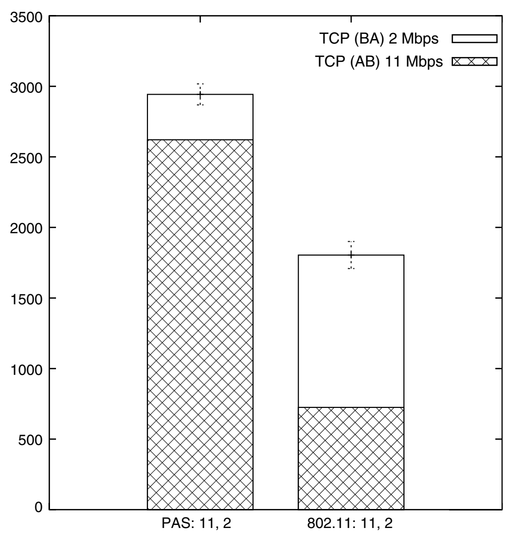

Fig. 19. Results on TCP flows.

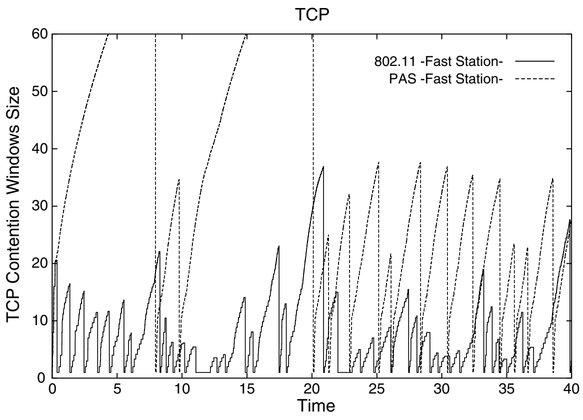

Fig. 20. TCP contention windows for the fast stations.

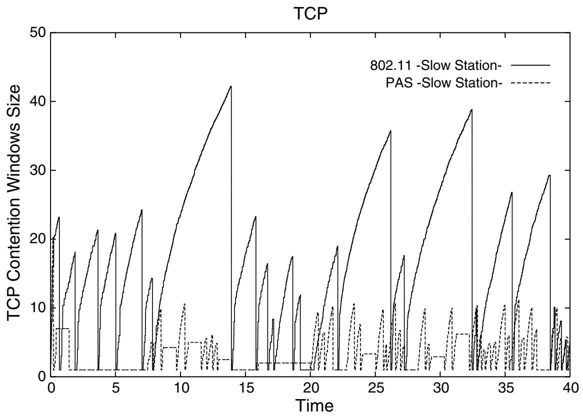

Fig. 21. TCP contention windows for the slow stations.

of the slow station, even if sending short packets is penalized in our scheme.

6.11. PAS in heterogeneous context

Our protocol is *based on* and *compliant with* the original IEEE 802.11 standard. This means that PAS and the original standard may coexist and that it is not mandatory (but recommended in order to increase performance) to have PAS on all stations of the network. As a proof of this claim, here we show the results of a simulation run with four stations. Two stations use standard IEEE 802.11, and the two others use PAS. For the stations running the same protocol, there is a fast (11 Mbps) station and a slow (2 Mbps) station. The four stations send saturated UDP traffic to a base station. Table 5 shows the throughput of the four stations in this heterogeneous context. Furthermore, in the same table can be found the results of the case where IEEE 802.11 is the MAC protocol of the four stations and of the case where PAS is the MAC protocol of all stations. The results in the table clearly show that even in heterogeneous scenarios the presence of nodes using PAS allows to increase the throughput.

Table 5
Performance of PAS and IEEE 802.11 in heterogeneous context

		Throughput (kbps)	Conf. Int. (0.05)
Heterogeneous	Fast (802.11)	445.62	[431.30; 459.94]
	Fast (PAS)	1815.40	[1777.37; 1853.44]
	Slow (802.11)	477.73	[465.98; 489.48]
	Slow (PAS)	478.76	[468.64; 488.87]
	Total	3217.51	[3185.84; 3249.17]
All 802.11	Fast	566.04	[548.81; 583.27]
	Fast	581.13	[566.05; 596.21]
	Slow	583.42	[570.67; 596.18]
	Slow	611.37	[599.62; 623.12]
	Total	2341.97	[2321.05; 2362.89]
All PAS	Fast	1484.23	[1435.37; 1533.09]
	Fast	1511.53	[1471.98; 1551.07]
	Slow	403.10	[394.02; 412.18]
	Slow	395.38	[386.01; 404.75]
	Total	3794.24	[3759.19; 3829.29]

6.12. Limitations of PAS

6.12.1. Three pairs scenario

Since all the mechanisms in PAS are fully distributed, PAS can also work in a multi-hop context, where the wireless stations do not perceive the same medium occupancy. If we consider the scenario depicted in Fig. 22 we can see that the external pairs are fully independent. In this scenario, the central pair accesses the medium 95% less than the external pairs, as demonstrated by Chaudet et al. [1]. The medium occupancy perceived by the central pair is given in Fig. 23. It is easy to observe that the value of *t_p_max* for the central pair can be at most equal

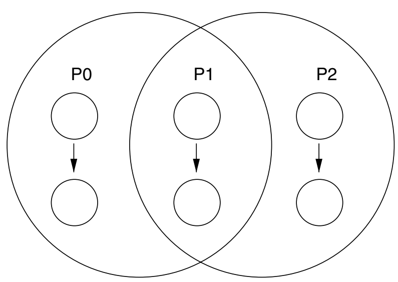

Fig. 22. The three pairs scenario.

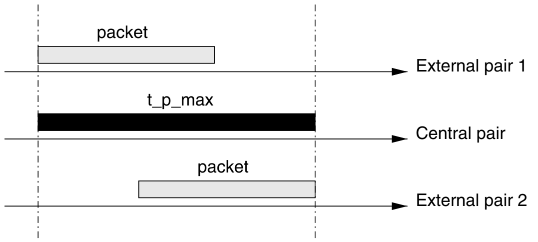

Fig. 23. The medium occupancy perceived by the central pair.

Table 6
Results of the three pairs scenario

		Throughput (kbps)	Conf. Int.
PAS	P0	1592.49	[1584.16; 1600.82]
	P1	102.21	[68.28; 136.15]
	P2	1592.49	[1584.09; 1600.89]
802.11	P0	1634.15	[1632.03; 1636.27]
	P1	6.44	[1.78; 11.11]
	P2	1632.86	[1630.23; 1635.49]

to $t\_p0 + t\_p2$, where $t\_pi_{i \in \{0,2\}}$ is the time needed for the pair i to transmit its packet. It is important to remark that here the maximum medium occupancy time does not specifically correspond to a packet transmission time. Table 6 shows the results on the three pairs scenario where the external pairs send 1000 bytes of data at 2 Mbps and the central pair sends 1000 bytes of data at 11 Mbps.

The results in this table show that, even if PAS does not solve the problem (the unfairness issue remains), the throughput of the central pair is highly improved. Nevertheless, in this scenario a temporal fairness cannot solve the problem and it seems necessary to modify the IEEE 802.11 medium access control in order to provide each node the same probability to access the medium.

6.12.2. Multi-hop flows

In this section we present some simulation results run with multi-hop flows. In this scenario, the source and the destination are separated by 4 hops. Let us assume that node 0 has to send a TCP flow to node 5 through nodes 1, 2, 3, 4. The data rate of all nodes is 11 Mbps except for node 2 that is different in each simulation. Table 7 shows the results for different values of data rate at node 2. We can see that even if PAS outperforms IEEE 802.11, the performance of the two protocols is roughly the same. This is due to the increasing number of collisions in this context. To increase the performance of PAS in this kind of scenario, we have to provide a more efficient scheduling that can avoid collisions. Table 8 gives the same results for an UDP flow. It is quite surprising to see how the performance of PAS are lower compared to the performance of IEEE 802.11. When UDP is used, the queues of each node are full, which is not the case with TCP traffic, thanks to the feedback provided by TCP. In the case of UDP, at each node (except node 2), packets can be aggregated and therefore the expected collision duration is equal to t_p_max, which reduces the performance of PAS. In the case of IEEE 802.11, the expected collision duration is smaller and thus the performance is increased.

Table 7
Performance of PAS and IEEE 802.11 in multihop TCP context

	Node 2	Throughput (kbps)	Conf. Int. (0.05)
PAS	1 Mbps	422.69	[409.07; 436.30]
802.11	1 Mbps	396.04	[379.87; 412.20]
PAS	2 Mbps	537.28	[507.19; 567.37]
802.11	2 Mbps	524.59	[504.66; 544.51]
PAS	5.5 Mbps	700.20	[679.57; 720.83]
802.11	5.5 Mbps	676.97	[666.34; 687.61]
PAS	11 Mbps	759.48	[748.96; 770.01]
802.11	11 Mbps	660.75	[620.71; 700.79]

Table 8
Performance of PAS and IEEE 802.11 in multihop UDP context

	Node 2	Throughput (kbps)	Conf. Int. (0.05)
PAS	1 Mbps	348.78	[335.46; 362.11]
802.11	1 Mbps	533.431	[514.56; 552.30]
PAS	2 Mbps	600.44	[578.47; 622.40]
802.11	2 Mbps	746.79	[720.29; 773.28]
PAS	5.5 Mbps	1061.53	[1023.75; 1099.31]
802.11	5.5 Mbps	1007.09	[971.59; 1042.58]
PAS	11 Mbps	1155.29	[1114.39; 1196.20]
802.11	11 Mbps	1113.00	[1073.64; 1152.36]

These different scenarios (including single-hop flow in a multi-hop configuration and multi-hop flows) show that the use of PAS is not as efficient as in single-hop cells. Providing a fair time-sharing based on medium sensing is not enough to integrate the complex radio medium sharing that arises in ad hoc networks.

7. Conclusion

In this paper, we propose and thoroughly analyze PAS, a dynamic and distributed packet aggregation mechanism to solve the performance anomaly of

IEEE 802.11. Our solution is based on the fact that the same transmission time is given to each station. This transmission time is computed dynamically and is equal to the maximum occupancy time perceived on the medium. When a node has the opportunity to use the channel, it sends as many packets as the previously perceived transmission time allows. The aggregation is done by waiting only for a SIFS period between the reception of an ACK and the beginning of the next transmission. To increase the dynamics and to reduce the convergence time, the transmission time is set to 0 after each successful transmission (or burst of aggregate transmissions).

We have shown, through both analytical analysis and simulation, that our protocol solves the performance anomaly in many scenarios for both UDP and TCP traffic. The aggregate throughput can be increased and the time-based fairness is almost reached in nearly every of the tested configurations. We have also shown that our approach does not need extra information, than that already provided by the IEEE 802.11 standard thus it can be easily implemented. Even more, we show how nodes using PAS can coexist and communicate with nodes using standard IEEE 802.11. An important characteristic of our proposal is the fact that it is totally distributed. Such a feature allows the use of PAS in multi-hop networks. Nonetheless, a fair time-sharing based on carrier sensing, like the one designed in PAS, is not enough to provide a time-based fairness in ad hoc networks.

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Tahiry Razafindralambo is a Ph.D. student at Institut Nationale de Recherche en Informatique et en Automatique (INRIA) in the ARES project. His research interests cover several areas of wireless networking research. He is particularly interested in the design, analysis, and implementation of wireless protocols and algorithms. His work is mainly focused on MAC protocols for wireless networks.

Isabelle Guérin Lassous is a Professor at Université Claude Bernard Lyon 1 (UCBL) and Ecole Normale Supérieure (ENS) Lyon from September 2006. Before, she was a full researcher at Institut Nationale de Recherche en Informatique et en Automatique (INRIA). Her research area concerns wireless networks and distributed algorithms.

Luigi Iannone is currently Research Assistant at Université Catholique de Louvain (UCL) in the IP Networking Lab (INL). He received his Ph.D. in 2006 at Université Pierre et Marie Curie-Paris VI as member of the Networking and Performance Analysis group of the LIP6 Laboratory. His research interests focus on wireless networks architecture, including MAC protocols, routing protocols, and cross-layer optimization. He is also interested in future Internet network architecture, in particular in Internet routing based on locators/identifiers separation.

Serge Fdida is a full professor at the Université Pierre et Marie Curie-Paris VI and head of the Networking and Performance Analysis group of the LIP6 Laboratory. His research interests are in the areas of content networking, pervasive communication, resource management, and performance analysis. He was the chairman (or cochair) of many events, including IFIP Networking 2000 and Conext 2005. He was the editor of the proceedings of these conferences and is the author of a book on performance evaluation and a book on networking. He is on the editorial boards of many journals: Computer Communication, SIGCOMM CCR, Computer Networks Journals, Journal of Wireless and Optical Communications, and Annals of Telecommunications.