Packet Frame Grouping

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This paper presents a scheme to improve the performance of radio MAC protocols in the case of small packets. First, the performance problems with small packets and their consequences are analysed. Then the Packet Frame Grouping scheme is presented, which is shown to improve the throughput and reduce the latency over MAC protocols based on CSMA/CA. Finally, the new protocol is simulated in various scenarios to show its impact on different traffic types.

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1 Introduction

The development of radio LANs has been mostly done by adapting wired LAN techniques to radio. The channel access mechanism used in most radio LANs (CSMA/CA) is closely derived from Ethernet (CSMA/CD) and is based on packet contention. As the radio medium is different from the wired medium, the MAC protocol has to be adapted to take these differences into account, and radio MAC protocols continue to evolve to map more closely the phenomena observed on the radio channel.

One of the main constraints of the radio medium is that bandwidth is scarce. 100 Mb/s shared medium is mainstream in the wired world, but the bit rate of common radio LANs is still below 2 Mb/s [1]. Because the resource is limited, the medium is very likely to be used to its full capacity, and the efficiency and the fairness of the protocol is a real concern.

2 Small packets

Modern radio MAC protocols like 802.11 [1] include some techniques to improve the performance in the case of large and medium packets. For example, RTS/CTS decreases the collision penalty and fragmentation reduces the error penalty.

Because these techniques introduce some overhead, they are not worth using for small packets. And, if any network stack tries as much as possible to use large packets to increase the efficiency, in many cases it has to use small packets.

2.1 Management packets

To provide the services required by the upper layers, network stacks use some management packets. The most frequent one is the network layer acknowledgment, but there are usually many others to establish, maintain and close connections.

These packets carry a limited information and are therefore quite small (a few tens of bytes - 40 B for a TCP ack). On a typical large network, they can account for one third of the number of packets.

2.2 Network layer coalescence limitations

To avoid sending small data packets over the network, network protocols like TCP use coalescence to group small amounts of data in the same packet. TCP also uses the same kind of mechanism to reduce the number of TCP acks by acknowledging many packets at once.

The main risk with such algorithms is that they tend to increase the latency : the network stack has to wait for enough "chunks" of data to arrive before sending the packet to the network adapter. To avoid this increase of latency, the waiting time is bounded. So, if the data doesn't arrive fast enough, the network stack will time out and generate small data packets.

Network cards provide multiple transmit buffers to compensate for the system latencies (Typical modern networking cards include 64 kB of memory for packet buffers). This reduces the efficiency of coalescence, because the protocol stack can't use it with packets already buffered on the card or in the process of being transmitted.

2.3 High priority traffic

High priority traffic on a network may be a telnet session, a phone connection, gaming or other multimedia information. One of its main characteristics is that it very often uses small data packets (to reduce the latency). Typical sizes range from a few bytes to a few hundred bytes.

Of course, MAC designers could offer a high priority service at the MAC level optimised for that kind of traffic (802.11 point controller [2], some ad-hoc mechanism [3] or HiperLan [5] for example).

But, most networking stacks (like TCP/IP) don't include any priority management. So, for applications running on top of the networking stack, there is no way to communicate their priority requirements to the MAC level. And the MAC has no way to distinguish between the different kinds of traffic.

This is why very often the high priority service offered at the MAC level (if any) is not used and multimedia applications still use standard small UDP packets, despite the fact that some MAC protocols such as 100VG (802.12) have included dual priority for years.

3 Performance problem

The fact that many applications (especially performance sensitive multimedia applications) use small packets means that the performance of MAC protocols for small packets is quite important.

The performance of most radio MAC protocols with small packets is not very good, as their optimisations apply only to medium and large packets. The main concerns are the high overhead and the unfairness.

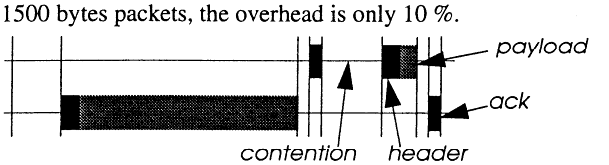
3.1 Overhead

Radio MAC protocols usually have a much higher overhead than their wired counterpart :

- Each packet is followed by a short acknowledgment message to detect collisions and errors.
- The headers have to include more information due to the shared nature of the medium (network ID, encryption information...).
- The synchronisation fields (for clock recovery & antenna selection) are larger.
- The CSMA contention is usually slotted because of the radio turnaround.
- These contention slots are large because this radio turnaround is slow.

This overhead is fixed for each packet (independent of the packet size). For large packets, it represents only a few percent or less, but for small packets its size may be of the same order as the payload itself. The result is that it is even more costly to use small packets on radio than on a wire.

For example, we can calculate the overhead for a TCP ack packet over an 802.11 network at 2 Mb/s. A 3 slot contention is 278 μs (typical for a 15 slots contention window). The header is 400 bits, i.e. 200 μs. The payload is 320 bits, i.e. 160 μs. An ack is 240 bits, after a SIFS, i.e. 148 μs. The total overhead is 4 times the payload ! For a 1500 bytes packets, the overhead is only 10 %.



3.2 Fairness

The goal of CSMA/CA is to provide each node with an equal opportunity to access the medium. This is done via contention, giving every node the same probability to send a packet on the network.

The fairness of this scheme is based on packets : over a period of time, each node can send an equal number of packets. The result is that nodes using large packets can send more data in that period of time than nodes using small packets (when the medium is saturated).

So, this scheme is fair in term of packets sent, but unfair in terms of amount of information.

4 Packet Frame Grouping

Packet Frame Grouping is a scheme to improve the performance with small packets. Packet Frame Grouping is optimised for "802.11-like" MAC protocols and, as it is implemented in the radio device itself, designed to be very simple and to use as few resources as possible.

4.1 The principle

The principle of Packet Frame Grouping is to group small packets together and to share the contention overhead between the grouped packets (instead of having it for each packet). If we send small packets as a train (a burst) without releasing the medium, we are creating a contention free frame. We therefore avoid the contention overhead between packets of the train. This allows the protocol to use the medium more efficiently and increases the throughput. This mechanism is totally transparent to any receiving node, even when not implementing Packet Frame Grouping, because the structure of the packets is unchanged.

The problem of most schemes for increasing the throughput of a protocol is that they also tend to increase the latency (and latency is quite important for multimedia applications). In fact, the latency is mainly dependent on the longest transmission time on the medium. To avoid increasing the latency, Packet Frame Grouping must limit the number of packets in the contention free frame.

Setting the limit of the frame to a certain number of bytes (instead of a number of packets) introduces a different kind of fairness which is no more biased towards nodes using large packets : the contention gives each node the right to send the same amount of data.

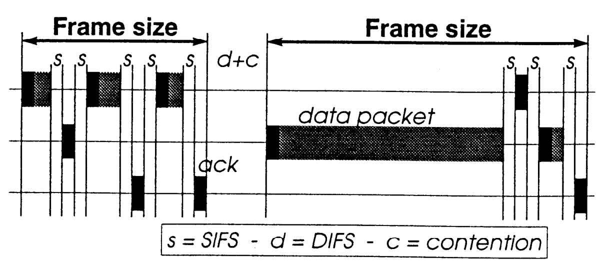
The next step is to choose the frame size. If it is set slightly larger than the maximum packet size (MTU - maximum transfer unit), the scheme can group a small packet with a big one (or many small ones) and the impact on latency should be limited. Larger frame sizes would allow to increase the throughput even more, to the detriment of the latency.

The concatenation of small upper layer packets in the same MAC packet could allow us to go even further (sharing a MAC header between two packets or more - a bit like coalescence in TCP), but is not an option because of its complexity. Compared to Packet Frame Grouping, concatenation requires data copy, more complex MAC mechanisms, is limited to packets destined to the same address and doesn't work with packets in the process of being transmitted.

4.2 Implementation

The implementation of Packet Frame Grouping in an 802.11 system would use a scheme very similar to fragmentation [1] and be compatible with it : Packet Frame Grouping sends a frame of small packets in the same way fragmentation sends fragments of a large packet.

The 802.11 channel access mechanism defines 2 main types of InterFrame Space (IFS) on the medium (this is the amount of time separating two transmissions). The DIFS is the largest and is used to start the contention : if the medium has been free (idle) for more than a DIFS, all nodes enter the contention phase. The SIFS is the shortest and is used between a packet and the ack and also between fragments. As the SIFS is shorter than a DIFS, between two packets separated by a SIFS no other node may go into contention and start transmitting a packet.



Fragmentation uses the SIFS to retain control over the medium and to send all fragments of a packet as a burst, each transmission of the burst being separated by a SIFS. Each fragment is individually acknowledged by the receiver by sending an ack message (after a SIFS), and upon reception of this ack the sender sends the next fragment (after a SIFS).

Packet Frame Grouping uses exactly the same mechanism, the contention free frame (the burst) containing many small independent packets (instead of fragments of a larger packet), which may have a different destination address.

To implement Packet Frame Grouping, the MAC needs only to include a counter to count the number of bytes sent in the current frame. Each time the MAC successfully sends a packet (i.e. receives the ack), it adds the size of the next packet in the transmit queue to the current frame size and compares it to the maximum frame size. If it is below, it sends this next packet after a SIFS, otherwise it resets the counter and goes into contention.

5 Simulation model

The models used for these simulations have been carefully chosen to be simple and realistic, to illustrate the scheme and to avoid side effects leading to invalid results.

5.1 MAC model

The MAC model includes a fairly complete 802.11 channel access mechanism. This model is based on an 802.11 backoff (slotted exponential contention). All management functionalities have been removed to keep the model simple.

The model implements MAC level acknowledgments and retransmissions, and RTS/CTS (for packets larger than 250 B). This model doesn't include fragmentation (but the compatibility of Packet Frame Grouping and fragmentation has been verified in another set of simulations).

The maximum packet size is 1500 B. All other parameters conform to 802.11 [1] (CWmin = 16 ; SIFS = 28 µs ; Slot = 50 µs ; Headers = 50 B).

5.2 Channel model

The channel model is a very classical and simple radio channel model, including node to node attenuation (80 dB here) and Rayleigh fading (calculated on a per packet basis). The bit rate is 2 Mb/s, and there is no interference.

5.3 Traffic models

There are usually two ways to model the traffic over a network : by using a statistical model or by using network traces (recorded on a real network). As there are no traces over 802.11 or at 2 Mb/s, statistical models have been used for this set of simulations.

5.3.1 Random traffic

The first traffic model used is the well known *random* traffic generator : all packet sizes are uniformly distributed in *]0 ; max packet size]* and packets arrive at the MAC following a Poisson process (random interarrival time with negative exponential distribution).

The main advantage of this traffic model is that it is already widely used and it allows to explore the full range of network loads and packet sizes.

5.3.2 Voice traffic

The main characteristic of voice traffic is a constant load and a fixed arrival rate of packets to the MAC. These traffic characteristics don't apply only to voice, but also to common multimedia applications like video and gaming. For those applications, it is quite common to have a load constant (or bounded) and packets sent at regular intervals.

To simulate those traffics, the *random* traffic generator is modified to use a fixed packet size (no longer random). The *voice* traffic still use a random interarrival time : the fact that these packets may come from another network or a TCP stack adds a variable latency, so packets won't arrive exactly at fixed intervals.

In the simulations, the load of the *voice* traffic is fixed at 32 kb/s, and in each packet an overhead for IP and UDP headers (32 B) is included. Because the packets are so small, the main characteristic of this type of traffic, with regard to the MAC protocol, is the mean interval between packets (the arrival rate at the MAC), the actual traffic load has a much smaller impact.

5.3.3 TCP traffic

The major problem of the *random* model is that it neglects one fundamental property of the network stack : the transport layer adapts to the link bandwidth and try to push as many packets as possible onto the network. The stack will also try to use as many packets of the maximum size (MTU) as possible. This means that when a node has some data to transfer, it will send large back to back packets and saturate the radio medium, so the random interarrival time of the previous model doesn't make sense with a slow network.

The applications on top of the network stack will emphasize this behaviour. Modern computers use some clever cache techniques to minimize the amount of network requests. The result is that most requests will be some long bursts of data over the network (that TCP will translate into long bursts of large packets). This characteristic applies to ftp, http (especially http 1.1), smb, nfs, disk sharing and print jobs.

The first TCP traffic model (*TCP1*)simulates a node sending a large amount of data over a protocol such as TCP. The sender sends packets of the maximum size as fast as possible. The receiving node acknowledges each incoming packet with a short packet (40 B).

The second TCP traffic model (*TCP2*) is a simple bimodal distribution. Each packet is either big (maximum size) or small (40 B), the probability of being small is 1/3 (this simulates medium TCP optimisation). Packets are sent as fast as the link can manage.

These traffic models don't implement any TCP flow control and congestion avoidance mechanisms, but only the packet distribution. With MAC level retransmissions, TCP should always use the full network capacity.

6 Simulation results

All the simulation models have been implemented under the Bones® Designer™ environment. The goal of these simulations was to explore the behaviour of the Packet Frame Grouping scheme in many various scenarios.

6.1 Throughput and latency versus load

The first set of simulations is the very classical Throughput & Latency vs. Load test (using the *random* traffic with variable load). The simulations compare a standard 802.11 MAC with a MAC implementing Packet Frame Grouping (2000 Bytes frame size). The network is composed of 5 nodes, and the load is distributed equally between the nodes. The results show the aggregate throughput and the average latency of the system in these two cases for different loads of the network.

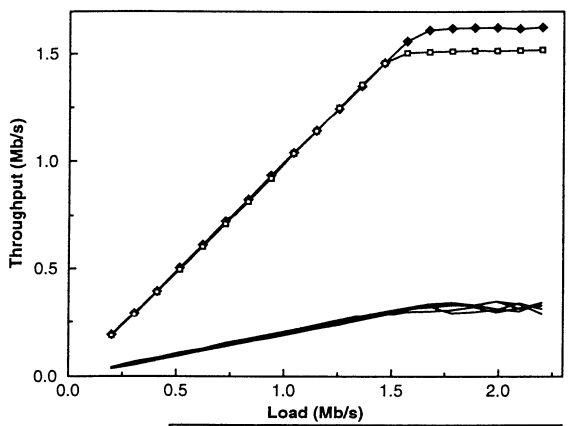


fig. 6.1a

With this type of traffic, Packet Frame Grouping gives a 7 % increase of the maximum throughput (*fig 6.1a*) and a reduction of the average latency (*fig 6.1b*). This shows clearly that Packet Frame Grouping offers a significant improvement of the network efficiency.

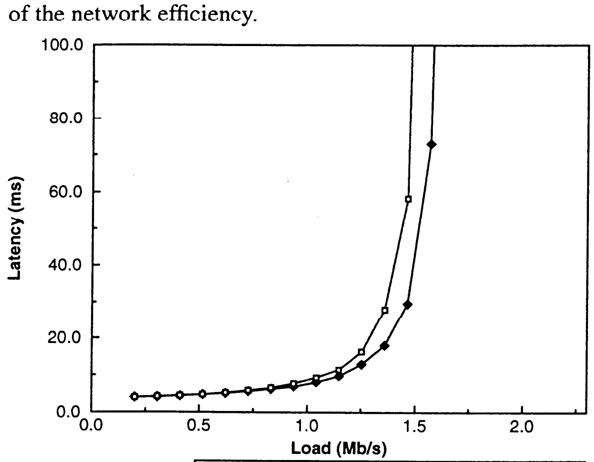


fig. 6.1b

The Throughput vs. Load graph (*fig 6.1a*) includes the individual throughput of each of the 5 nodes using Packet Frame Grouping to show the scheme is fair (each node can send approximately the same amount of data).

6.2 Frame size variations

As explained before, the *random* type of traffic is not very satisfactory. Furthermore, we now need to explore the impact of the different frame sizes on the network performance.

For this, four different configurations have been chosen : the previous one (5 nodes using the *random* traffic), an ftp transfer (a sender and receiver using the *TCP1* traffic), a small TCP network (2 nodes using the *TCP2* traffic) and a medium network (10 nodes using the *TCP2* traffic). As explained in the model description, the last three configurations adapt to the network capacity, but for the 5 nodes using the random traffic, the load has to be adjusted : for the maximum throughput simulation, a load of 1.75 Mb/s has been chosen, and for the average latency a load of 1.45 Mb/s.

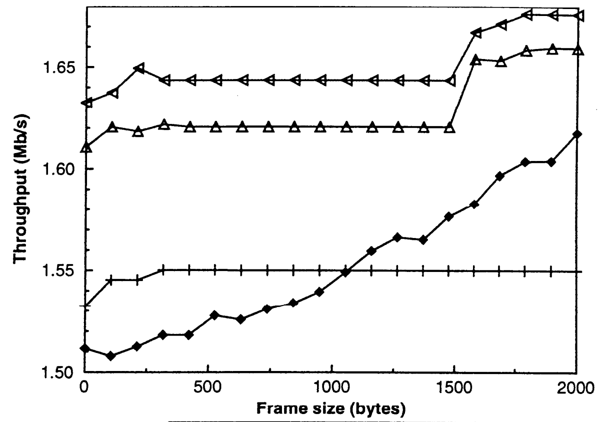


fig. 6.2a

A frame size of 0 is equivalent to no Packet Frame Grouping (scheme disabled) and a larger frame size allows to group more and more packets. The first obvious result is that the improvement given by Packet Frame Grouping depends very much on the traffic considered.

The *TCP1* traffic shows almost no improvement of throughput (*fig 6.2a*) and a huge improvement in latency (*fig 6.2b*). The dramatic improvement of the latency must be seen as an artifact of the *TCP1* model. The huge delay without

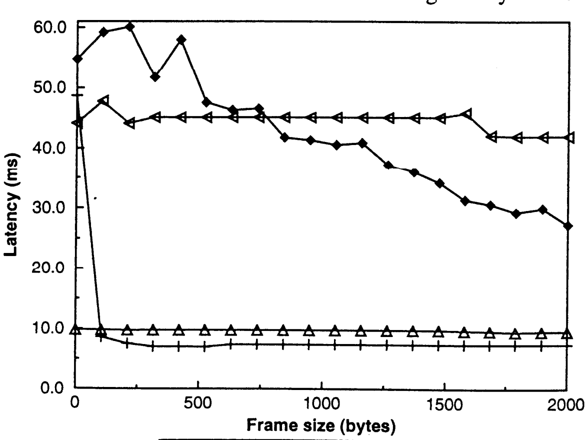


fig. 6.2b

Packet Frame Grouping (frame size = 0) is due to TCP acks being queued. In the real world, there is no way we could queue that many TCP acks in the networking stack and the card.

The lack of increase in throughput (around 1 %) could be explained by the fact that the receiver is the only node sending small packets. As those acks are generated only upon reception of data packets from the sender, the receiver has no real advantage in sending them faster, it still has to wait for the sender.

The *TCP2* traffic shows no improvement in latency (*fig 6.2b*). In this model the packets are generated upon consumption of the previous one by the MAC, there is no queuing in the system and this latency is only the channel access delay, which could explain the lack of improvement due to the scheme. The increases in throughput (around 3 % in total - *fig 6.2a*) correspond to the point where the scheme is able to group two small packets and then a small packet with a large one.

6.3 Multimedia traffic

As explained in the first part, small packets are mainly used by multimedia applications (apart from TCP acks). These are the applications that should benefit the most from a scheme such as Packet Frame Grouping. To verify this assumption, a specific set of simulations has been designed to study a network with a mix of data traffic and multimedia traffic.

To simulate the data traffic, the TCP models of the previous set of simulations have been re-used. The multimedia traffic uses the *voice* model (random interarrival time, fixed packet size). The load of the *voice* model is fixed (32 kb/s + UDP overhead), and different average packet arrival times are used (the main parameter of this model).

The network is composed of two nodes performing a data transfer (using *TCP1*, and then *TCP2*) and 4 nodes exchanging *voice* traffic (giving a total load of 128 kb/s). In each case, the throughput of the data traffic only and the latency of the multimedia traffic only are measured.

6.3.1 TCP1 + Voice traffics

The first graph shows a large throughput improvement (around 9 % - *fig 6.3.1a*) of the data traffic when using

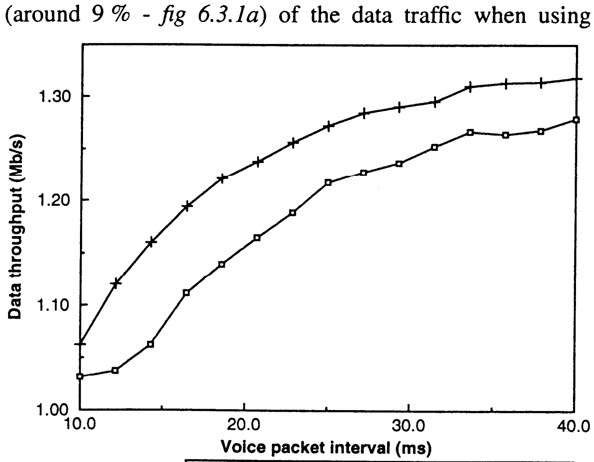


fig. 6.3.1a Packet Frame Grouping / Normal

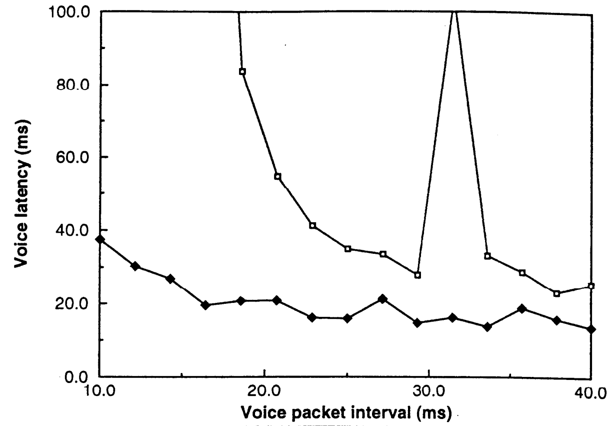


fig. 6.3.1b Packet Frame Grouping / Normal

Packet Frame Grouping. This is quite spectacular because, in the previous set of simulations, the same traffic (*TCP1* - without the addition of the *voice* traffic) was showing no improvement. The fact that the small packets of the *voice* traffic are sent more efficiently leaves more bandwidth for the data packets.

The voice traffic benefits largely from Packet Frame Grouping (*fig 6.3.1b*). For average packet intervals larger than 20 ms, the improvement in latency is around 50 %. Packet Frame Grouping also helps when sending packets with very short interval time (under 20 ms) with tolerable delay, where a standard MAC would not cope.

6.3.2 TCP2 + Voice traffics

The simulation shows the same type of results when using *TCP2* as the data traffic instead of *TCP1*. The data throughput improvement is slightly larger (around 10 % - *fig 6.3.2a*), and the latency improvement smaller (around 25 % - *fig 6.3.2b*). This smaller latency improvement could be explained by the fact that the *TCP2* traffic is more aggressive that *TCP1* (uses more large packets), so tends to use more of the available bandwidth (latencies in this simulation are much higher than with *TCP1*).

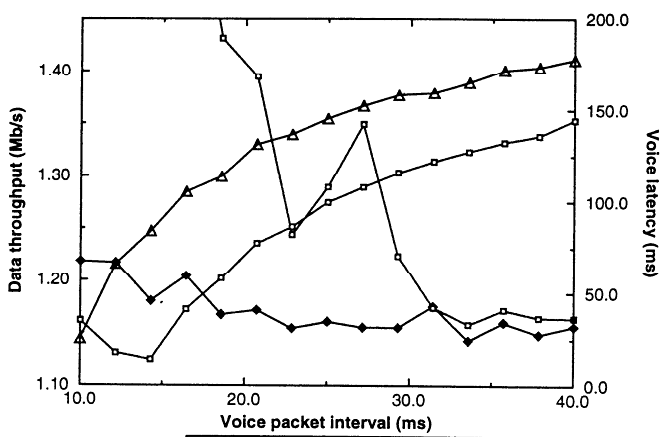


fig. 6.3.2a Packet Frame Grouping / Normal
fig. 6.3.2b Packet Frame Grouping / Normal

7 Conclusions

Because of the radio network architecture constraints, radio MAC protocols are not very efficient in sending small packets. As the bandwidth is scarce, this limits the usage of radio LANs for multimedia applications.

By grouping small packets into a contention free frame, the protocol gets rid of some of the contention overhead and increases the share of the medium available to an application using small packets. This scheme is easy to implement and fits naturally in 802.11 as it just reuses the principle of 802.11 fragmentation.

Simulations of the scheme have showed that it offers an improvement of the MAC in both throughput and latency. The improvement is quite limited in the case of large data transfer, but the network performance increases dramatically for a mix of data and multimedia traffics.

The next step would be to implement the Packet Frame Grouping scheme in a 802.11 wireless LAN to study its effect in real conditions and validate the results of the simulations.

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