MACAW: A Media Access Protocol for Wireless LAN's

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Abstract

In recent years, a wide variety of mobile computing devices has emerged, including portables, palmtops, and personal digital assistants. Providing adequate network connectivity for these devices will require a new generation of wireless LAN technology. In this paper we study media access protocols for a single channel wireless LAN being developed at Xerox Corporation's Palo Alto Research Center. We start with the MACA media access protocol first proposed by Karn [9] and later refined by Biba [3] which uses an RTS-CTS-DATA packet exchange and binary exponential backoff. Using packet-level simulations, we examine various performance and design issues in such protocols. Our analysis leads to a new protocol, MACAW, which uses an RTS-CTS-DS-DATA-ACK message exchange and includes a significantly different backoff algorithm.

1 Introduction

In recent years, a wide variety of mobile computing devices have emerged, including palmtops, personal digital assistants, and portable computers. While the first portables were designed as stand-alone machines, many of these new devices are intended to function as full network citizens. Consequently, a new generation of wireless network technology is needed to provide adequate network connectivity for these mobile devices. In particular, wireless local area networks (LAN's) are expected to be a crucial enabling technology in traditional office settings where such mobile devices will be initially, and most heavily, utilized.

The media in a wireless network is a shared, and scarce, resource; thus one of the key questions is how access to this shared media is controlled. In this paper, we focus on media access protocols in wireless LAN's. Our research has a dual purpose. One goal is to develop a media access protocol for use in the wireless network infrastructure being developed in the Computer Science Laboratory at Xerox Corporation's Palo Alto Research Center [7, 8]. The other goal is to explore some of the basic performance and design issues inherent in wireless media access protocols. While our specific simulation results may only apply to PARC's particular radio technology, we expect that some of the basic insight gained will be more generally applicable.

Wireless media access protocols for a single channel can typically be categorized as either token-based or multiple access. For reasons we explain in the next section, we choose the multiple access approach.[1] Our work is based on MACA, a Multiple Access, Collision Avoidance protocol first proposed by Karn [9] and later refined by Biba [3]. Using packet-level simulations of the wireless network to guide our design, we suggest several modifications to MACA. We call the resulting algorithm MACAW, in recognition of its genealogical roots in Karn's original proposal. Our design is based on four key observations. First we observe, following Karn [9] and others [4, 12], that the relevant contention is at the receiver, not the sender. This renders the carrier sense approach inappropriate. Second, we note that, in contrast to Ethernets, congestion is location dependent; in fact, the first observation is irrelevant without the second. Third, we conclude that, to allocate media access fairly, learning about congestion levels must be a collective enterprise. That is, the media access protocol should propagate congestion information explicitly rather than having each device learn about congestion independently. Fourth, the media access protocol should propagate synchronization information about contention periods, so that all devices can contend effectively. In particular, this means that contention for bandwidth should not just be initiated by the sending device. While our proposed protocol provides enhanced performance (as compared to MACA), we hasten to note that it is merely an initial attempt to deal with these challenges; there are many remaining unresolved design issues.

This paper has 5 sections. In Section 2 we first provide some background on PARC's radio network and on the MACA media access protocol. We then, in Section 3, discuss our modifications to MACA; we motivate these changes by presenting simulation data for several different network configurations. We discuss remaining design issues in Section 4 and summarize our findings in Section 5.

[1] We expect in future work to revisit the token-based approach and make a more in-depth comparison.

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2 Background

2.1 PARC's Nano-Cellular Radio Network

The Computer Science Laboratory at Xerox Corporation's Palo Alto Research Center has developed 5 MHz "near-field" radio technology [7]; its low operating frequency eliminates multipath effects, and thus it is suitable for use in an indoor wireless LAN. The LAN infrastructure consists of "base stations", which are installed in the ceiling, and "pads", which are custom built portable computing devices (see [8] for a more complete description). There is a single 256kbps channel, and all wireless communication is between a pad and a base station (the base stations are connected together by an Ethernet). All base stations and pads transmit at the same signal strength. The range of transmission is 3 to 4 meters, and the near-field signal strength decays very rapidly ($\approx r^{-3}$, as opposed to $\approx r^{-1}$ in the far-field region). We thus obtain, around each base station, a very small cell (roughly 6 meters in diameter) with very sharply defined boundaries: a "nanocell". Given that the cells are very small and inter-cell interference is negligible, the aggregate bandwidth in a multi-cell environment is quite high.

A "collision" occurs when a receiver is in the reception range of two transmitting stations[2], and is unable to cleanly receive signal from either station. "Capture" occurs when a receiver is in the reception range of two transmitting stations, but is able to cleanly receive signal from the closer station; this can only occur if the signal power ratio is large ($\approx 10db$ or more). This requires a distance ratio of ≈ 1.5. Perhaps surprisingly, this ratio is rather hard to achieve, given that the base stations are in the ceiling and the pads are typically no higher than a meter or so above the floor. Roughly, this gives a minimum pad-to-base distance of just under 2 meters in a cell whose radius is just over 3 meters. Thus, in our environment, capture will be relatively rare, and is not a primary design consideration.

"Interference" occurs when a receiver is in range of one transmitting station and slightly out-of-range of another transmitting station, but is unable to cleanly receive the closer station's signal because of the interfering presence of the other signal. The rather sharp decay in signal strength makes interference rather rare in our environment, and we do not make it a major factor in our design.

Ignoring both capture and interference leads to a very simple model in which any two stations are either in-range or out-of-range of one another, and a station successfully receives a packet if and only if there is exactly one active transmitter within range of it. In designing our protocol, we often use this model accompanied by the additional assumptions that no two base stations are within range of each other, and that no pad is within range of two different base stations. This is an extremely poor model for far-field radios. It is not quite so poor for our near-field radios, but it is still far from realistic. We have not used this naive model in any of our simulations, but we do use it for intuitive justification of some of the algorithms given below.

Controlling access to a shared media is much easier if the locations of the various devices are known. However, in our setting there is no independent source of location information for the pads. There is no way for a pad to know that it is leaving a cell except through the loss of signal from the base station. Furthermore, there is no way for a pad, or base station, to know about the presence of other devices besides explicit communication.

It is important to note that, in the absence of noise, our technology is symmetric; if a station A can hear a station B, then station B can hear the station A. The presence of noise sources (e.g., displays) may interfere with this symmetry, and in our simulations we will consider the effect of noise. However, noise is not so prevalent that we make it the overriding factor in our design; rather, we design our protocol to tolerate noise well but we have done most of our testing in a noise-free setting.

There are many different ways to control access to a single channel. Typically, these approaches are either multiple access or token-based. We chose the multiple access approach over the token approach for two reasons.[3] First, multiple access schemes are typically more robust. This is especially important in a wireless environment where the mobile devices span the gamut of reliability. Second, we expect the pads to be highly mobile and, given the small cell size, these pads will enter and leave cells frequently. This would necessitate frequent token hand-offs or recovery in a token-based scheme.

One common wireless multiple access algorithm, currently used in packet radio, is carrier sense (CSMA). In the next section we discuss its properties and argue, following Karn [9] and others [4, 12], that the CSMA approach is inappropriate for our setting.

2.2 CSMA

In CSMA, every station senses the carrier before transmitting; if the station detects carrier then the station defers transmission (CSMA schemes differ as to when the transmission is tried again). Carrier sense attempts to avoid collisions by testing the signal strength in the vicinity of the transmitter. However, collisions occur at the receiver, not the transmitter; that is, it is the presence of two or more interfering signals at the receiver that constitutes a collision. Since the receiver and the sender are typically not co-located, carrier sense does not provide the appropriate information for collision avoidance. Two examples illustrate this point in more detail. Consider the configuration depicted in Figure 1. Station A can hear B but not C, and station C can hear station B but not A (and, by symmetry, we know that station B can hear both A and C).

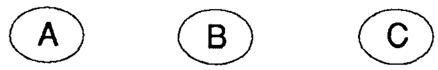

Figure 1: Station B can hear both A and C, but A and C cannot hear each other. A "hidden terminal" scenario results when C attempts to transmit while A is transmitting to B. An "exposed terminal" scenario results if B is transmitting to A when C attempts to transmit.

First, assume A is sending to B. When C is ready to transmit (perhaps to B or perhaps to some other station), it does not detect carrier and thus commences transmission; this produces a collision at B. Station C's carrier sense did not provide the necessary information since station A was

[2] We will use the term station to refer to both pads and base stations.

[3] These reasons are merely intuitive guides for design. We hope, in future work, to explore the token-based approach more fully. Only then can we make a valid comparison between the two approaches.

"hidden" from it. This is the classic "hidden terminal" scenario.

An "exposed" terminal scenario results if now we assume that B is sending to A rather than A sending to B. Then, when C is ready to transmit, it does detect carrier and therefore defers transmission. However, there is no reason to defer transmission to a station other than B since station A is out of C's range (and, as we stated earlier, in our environment there are no "interference effects" from out-of-range stations). Station C's carrier sense did not provide the necessary information since it was exposed to station B even though it would not collide or interfere with B's transmission.

Carrier sense provides information about potential collisions at the sender, but not at the receiver. This information can be misleading when the configuration is distributed so that not all stations are within range of each other. Because carrier sense does not provide the relevant collision avoidance information, we chose to seek another approach based on MACA, which we describe below.

2.3 MACA

Karn proposed MACA for use in packet radio as an alternative to the traditional CSMA media access scheme [9]. MACA is somewhat similar to the protocol proposed in [3] and also to that used in WaveLAN, and both resemble the basic Apple LocalTalk Link Access Protocol [2]. Here we present a very brief and general description of the algorithm ([9] is itself a brief description and does not specify many details).

MACA uses two types of short, fixed-size signaling packets. When station A wishes to transmit to station B, it sends a Request-to-Send (RTS) packet to B; this RTS packet contains the length of the proposed data transmission. If station B hears the RTS, and it is not currently deferring (which we explain below), it immediately replies with a Clear-to-Send (CTS) packet; this CTS also contains the length of the proposed data transmission. Upon receiving the CTS, station A immediately sends its data. Any station overhearing an RTS defers all transmissions until some time after the associated CTS packet would have finished (this includes the time for transmission of the CTS packet as well as the "turnaround" time at the receiving station[4]). Any station overhearing a CTS packet defers for the length of the expected data transmission (which is contained in both the RTS and CTS packets).

With this algorithm, any station hearing an RTS will defer long enough so that the transmitting station can receive the returning CTS. Any station hearing the CTS will avoid colliding with the returning data transmission. Since the CTS is sent from the receiver, symmetry assures us that every station capable of colliding with the data transmission is in range of the CTS (it is possible, though, that the CTS may not be received by all in-range stations due to other transmissions in the area). Notice that stations that hear an RTS but not a CTS because they are in range of the sender but out of range of the receiver can commence transmission, without harm, after the CTS has been sent; since they are not in range of the receiver they cannot collide with the data transmission.

[4] This turnaround time is the time from the reception of the RTS at the receiving antenna to the transmission of the CTS; this includes operating system delays as well as radio transients.

In the hidden terminal scenario in Figure 1, station C would not hear the RTS from station A, but would hear the CTS from station B and therefore would defer from transmitting during A's data transmission. In the exposed terminal scenario, station C would hear the RTS from station B, but not the CTS from station A, and thus would be free to transmit during B's data transmission. This is exactly the desired behavior.

Thus, in contrast to carrier-sense, this RTS-CTS exchange enables nearby stations to avoid collisions at the receiver, not the sender. The role of the RTS is to elicit from the receiver the CTS, whose reception can be used by other stations as an indication that they are in range and thus could collide with the impending transmission. This depends crucially on symmetry; if a station cannot hear station B's CTS then we assume that that station cannot collide with an incoming transmission to B.

If station A does not hear a CTS in response from station B, it will eventually time out (i.e., stop waiting), assume a collision occurred, and then schedule the packet for retransmission. MACA uses the binary exponential backoff (BEB) algorithm to select this retransmission time.

3 Designing MACAW

Our purpose here is to re-evaluate some of the basic design choices in MACA and then produce a revised version suitable for use in PARC's wireless LAN. The MACA algorithm, as presented in [9], is a preliminary design and leaves many details unspecified. We start our investigation by defining these details for an initial version. Appendix A gives the pseudo-code we used to implement MACA.

We mention several aspects of this algorithm here. First, the control packets (RTS, CTS) are 30 bytes long. The transmission time of these packets defines the "slot" time for retransmissions. Retransmissions occur if and only if a station does not receive a CTS in response to its RTS. Retransmissions are scheduled an integer number of slot times after the end of the last deferral period. A station randomly chooses, with uniform distribution, this integer between 1 and BO, where BO represents the backoff counter. The backoff algorithm adjusts the value of BO through two functions, F_{dec} and F_{inc}. Whenever a CTS is received after an RTS, the backoff counter is adjusted via the function F_{dec}: $BO := F_{dec}(BO)$. Whenever a CTS is not received after an RTS, the backoff counter is adjusted via the function F_{inc}: $BO := F_{inc}(BO)$. For BEB, $F_{dec}(x) = BO_{min}$ and $F_{inc}(x) = MIN[2x, BO_{max}]$, where BO_{min} and BO_{max} represent the lower and upper bounds for the backoff counter, respectively. For our simulations we have chosen $BO_{min} = 2$ and $BO_{max} = 64$.

We use packet-level simulations of the protocol to evaluate our design decisions. The simulator we use is a modification of the network simulator we have used in a number of other studies (for example, [5]) of wired networks. The simulator is event-driven and contains the following components: a traffic generator (which can generate data streams according to various statistical models), TCP, UDP, IP, pads, and base stations. The simulator approximates the media by dividing the space into small cubes and then computing the strength of a signal at each cube according to the distance from the signal source to the center of the cube. Errors due to the cube approach can be made arbitrarily small by reducing the cube size. For the simulations mentioned in this

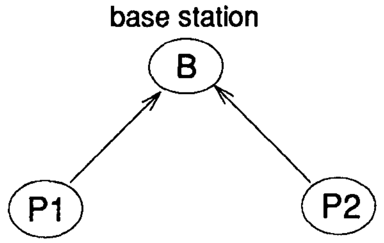

Figure 2: A single cell configuration where all stations are in range of each other and both pads are sending data to the base station (the arrows indicate the direction of the data transmission). The pads are each generating data at a rate of 64 packets per second and are using UDP for transport.

paper the cubes are 1 cubic foot in size.

In our simulations, all pads are 6 feet below the base station height. A station (which can be either a pad or a base station) resides at the center of a cube. Whenever a station starts sending, the strength of the signal is added to the current signal at all nearby cubes. At the end of the transmission, the designated receiving station can correctly receive the packet if the signal strength is greater than some threshold (the signal strength at 10 feet) and is greater than the sum of the other signals by at least 10 dB during the entire packet transmission time[5].

We use the term "stream" to refer to the set of packets going from a particular sender to a particular receiver station. For most of the simulations reported on here, the devices generate data at a constant rate of either 32 or 64 packets per second. All data packets are 512 bytes, and the control packets (RTS, CTS, etc.) are 30 bytes. Simulations are typically run between 500 and 2000 seconds, with a warmup period of 50 seconds. The simulations use a null turnaround time.

We investigate two areas of the media access protocol: the backoff algorithm and the basic RTS-CTS message exchange. We should clearly state our evaluation criterion: the media access protocol should deliver high network utilization and also provide fair access to the media. These goals are not always compatible, and when they are not we choose to deliver fairness[6] over optimal total throughput (note that often the easiest way to optimize total throughput in shared media is to eliminate sharing and turn the media over to one user exclusively).

3.1 Backoff Algorithm

Recall that MACA uses binary exponential backoff (BEB), in which the backoff is doubled after every collision and reduced to the minimal backoff after every successful RTS-CTS exchange. We now show that this does not provide an adequate level of fairness in some simple one-cell configurations. For example, consider the case where there are two pads in a cell, as depicted in Figure 2; the pads are within range of each other (and the base station) and each pad is

[5]These parameters are based on the properties of our hardware, however the specific value should have little effect on the generality of the simulation results.

[6]We do not give a precise definition of fairness. In this section any intuitive notion of fairness is sufficient; as we remark in Section 4, there are deeper allocation questions which do require a more precise definition of fairness.

	BEB	BEB copy
P1-B	48.5	23.82
P2-B	0	23.32

Table 1: The throughput, in packets per second, achieved by the streams in Figure 2.

generating enough UDP traffic to fully consume the channel. As shown in Table 1, when using the BEB algorithm eventually a single pad transmits at channel capacity and the other pad is completely backed off (i.e., its backoff counter is at BO_{max}). This is very similar to the Ethernet behavior noted in [11]. In such a multi-pad single cell environment, if all pads but one have relatively high backoff counters then after every collision it is very likely that the less-backed-off pad will retransmit first and "win" the collision and thereby reset its backoff counter to BO_{min}. This phenomenon keeps recurring after every collision, with the backed-off pads becoming decreasingly likely to win a collision. If there is no maximum backoff, one can show that eventually one pad will permanently capture the channel. This dynamic is driven by having one pad with a significantly lower backoff counter than the other pads, even though intuitively one would expect that the backoff counter should reflect the ambient level of congestion in the cell, which is the same for all pads. Each pad is doing its own congestion calculation based on its own experience and there is no "sharing" of this congestion information; this leads to the different stations having widely varying views of the level of congestion in the cell.

To rectify this, we have modified the backoff algorithm by including in the packet header a field which contains the current value of the backoff counter. Whenever a station hears a packet, it copies that value into its own backoff counter. Thus, in our single cell scenario where all stations are in range of each other, after each successful transmission all pads have the same backoff counter. The results from this algorithm are shown in the right column of Table 1. The throughput allocation is now completely fair. Thus, having the congestion information disseminated explicitly by the media access protocol produced a fairer allocation of resources.

Above we modified the basic structure of the backoff algorithm to allocate bandwidth fairly. An additional minor adjustment to the backoff computation can slightly improve the efficiency of the protocol. The BEB backoff calculation adjusts extremely rapidly; it both backs off quickly when a collision is detected and it also reduces the backoff counter to BO_{min} immediately upon a successful transmission. This produces rather large variations in the backoff counter; in our simple one-cell configuration, after every successful transmission we return to the case where all stations have a minimal backoff counter and then we must repeat a period of contention to increase the backoffs. This is mainly relevant when there are several pads in a cell and demand for the media is high.

To prevent such wild oscillations, we have instead adopted a gentler adjustment algorithm; upon a collision, the backoff interval is increased by a multiplicative factor (1.5) and upon success it is decreased by 1: $F_{inc}(x) = MIN[1.5x, BO_{max}]$ and $F_{dec}(x) = MAX[x-1, BO_{min}]$. This multiplicative increase and linear decrease (MILD) still provides reasonably

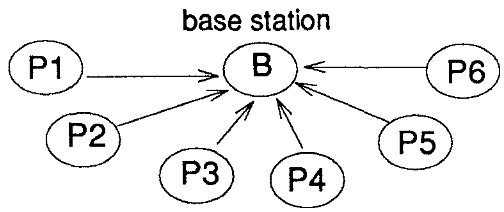

Figure 3: A single cell configuration where all stations are in range of each other. All six pads are sending data to the base station. Each stream is generating data at a rate of 32 packets per second and using UDP for transport.

quick escalation in the backoffs when contention is high but by not resetting the backoff counter to BO_{min} it avoids having to repeat the escalation in backoff counters after every successful tranmission. We tested the relative performance of these algorithms in the configuration depicted in Figure 3, which has six pads all sending data to the base station. Table 2 shows the data from these two backoff algorithms, and illustrates a clear advantage for the MILD algorithm. The performance of MILD and BEB on the two-pad configuration above was essentially identical because the level of contention is sufficiently low that resetting the counters to $BO_{min} = 2$ does not interfere with performance.

	BEB copy	MILD copy
P1-B	2.96	6.10
P2-B	3.01	6.18
P3-B	2.84	6.05
P4-B	2.93	6.12
P5-B	3.00	6.14
P6-B	3.05	6.09

Table 2: The throughput, in packets per second, achieved by the streams in Figure 3.

3.2 Multiple Stream Model

Let us return to the notion of fair allocation of bandwidth. Our initial design had a single FIFO packet queue at each station, with a single backoff parameter BO which controls the transmission (and retransmissions) of the packet at the head of the queue. This design allocates the bandwidth equally to all transmitting stations. Consider the configuration in Figure 4 where there are three pads; the base station is sending data packets to two of the pads, and the third pad is sending packets to the base station. Allocating bandwidth equally to each station gives half of the bandwidth to the pad-to-base-station stream and a quarter to each of the two base-station-to-pad streams. The data in Table 3 shows that when using a single queue in this scenario, each transmitting station (pad or base station) receives an equal share and thus half of the bandwidth goes to the pad-to-base-station stream and a quarter to each of the two base-station-to-pad streams.

Is this the allocation we want to achieve; is this "fair"? Certainly defining a general definition of fairness for this wireless setting is beyond our ken at the moment. However,

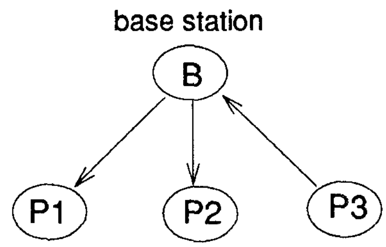

Figure 4: A single cell configuration where all stations are in range of each other. The base station is sending data to two of the pads, and the third pad is sending data to the base station. Each stream is generating data at a rate of 32 packets per second and using UDP for transport.

we can at least state that in a simple single cell configuration we want to treat all streams equally (as opposed to all stations equally); recall that a stream is the flow of data packets between a source-destination pair. That is, to the extent possible, we want to allocate bandwidth equally to streams and not to the stations themselves. This distinction is especially relevant since in our setting all wireless communications must go through a base station, thus base stations are likely to be the source of many streams.

This notion of per stream fairness can be implemented by keeping, in each station, separate queues for each stream and then running the backoff algorithm independently for each queue. When a station is allowed to send (e.g., after a deferral wait) and finds that it has data pending for N destinations, we treat the station as N co-located streams when resolving contention for the media. This can be done as follows. For each destination with data pending, the station flips a coin to determine, based on the backoff counter, how long to wait before sending an RTS to this destination. It then picks the one with the shortest wait time. If more than one of these streams have the same shortest wait time, the station randomly picks one of them as the winner, rather than simulating a collision; because of this, streams originating from a multi-stream station may have a slight advantage over streams in a single stream station.

As can be seen in Table 3, this multiple stream algorithm produces a fair allocation of bandwidth among the competing streams.

	Single Stream	Multiple Stream
B-P1	11.42	15.07
B-P2	12.34	15.82
P3-B	22.74	15.64

Table 3: The throughput, in packets per second, achieved by the streams in Figure 4.

3.3 Basic Message Exchange

In this section we examine the basic RTS-CTS-DATA message exchange and propose four changes. The material on the role of link layer acknowledgment and the need for a carrier-sense like functionality (as in CSMA/CA [2]) is commonly accepted wisdom in the 802.11 community; we repeat the material here for completeness.

3.3.1 ACK

Many of the applications used on mobile devices, such as electronic mail, require reliable delivery of data. At the transport layer these applications use TCP (as opposed to UDP which was used in the previous simulations) to provide that reliability. In MACA, when data packets suffer a collision, or are corrupted by noise, the error has to be recovered by the transport layer. This necessitates a significant wait, as many current TCP implementations have a minimum timeout period of 0.5sec, which was chosen to accommodate both local and long haul data transmissions.

In contrast, recovery at the link-layer can be much faster because the timeout periods can be tailored to fit the short time scales of the media. Thus we have amended the basic RTS-CTS-DATA exchange to include an acknowledgement packet, ACK, that is returned from the receiver to the sender immediately upon completion of data reception. If the ACK is not received by the sender, then the data packet is scheduled for retransmission. If the data packet had indeed been correctly received but the ACK packet was not, then when the RTS for the retransmission is sent, the receiver returns the associated ACK instead of a CTS. The sender increases its backoff if, after sending an RTS, no CTS or ACK arrives before it times out; the sender decreases its backoff when the ACK is received. The backoff counter is not changed if there is a successful RTS-CTS exchange but the ACK does not arrive.

We have simulated the effects of intermittent noise on a single pad-to-base-station stream. Intermittent noise is modeled as a given probability that each packet (regardless of size) is not received cleanly at its intended destination. Table 4 shows the resulting throughput. For the original RTS-CTS-DATA exchange, the dramatic decrease in throughput as the noise level increases is due to the slow recovery at the TCP layer. The decrease in throughput when the ACK is included is much less severe. The overhead due to the inclusion of the ACK packet is only about 9% (36.76 PPS vs. 40.41 in the no noise case), and when the loss rate is only 1 packet in 1000 the two algorithms give essentially identical results. Given that intermittent noise is likely to be present, and that it can have such a deleterious effect on the network throughput, we have decided that the augmented RTS-CTS-DATA-ACK exchange should be used for all reliable data transmissions.

Error Rate	RTS-CTS-DATA	RTS-CTS-DATA-ACK
0	40.41	36.76
0.001	36.58	36.67
0.01	16.65	35.52
0.1	2.48	9.93

Table 4: The throughput, in packets per second, achieved by a single TCP data stream between a pad and a base station in the presence of noise.

3.3.2 DS

In the original analysis of the exposed terminal configuration (Figure 1), we argued that the exposed terminal C should be free to transmit because even though it is in range of the sender B, it is out of range of the receiver A, and the receiver

Figure 5: A two cell configuration where both pads are in range of their respective base stations and also in range of each other. The pads are sending data to their base stations, and each stream is generating data at a rate of 64 packets per second and using UDP for transport.

is the only station that matters. However, C's transmission can benefit only if C can hear a returning CTS. When B is transmitting, C is unable to hear any replies and thus initiating a transfer is useless.[7] Moreover, when C does initiate and does not get any response, its backoff counter increases rapidly. With simple uni-directional transmissions the only relevant congestion is at the receiver; however, with our bi-directional RST-CTS-DATA message exchange, congestion at both ends of the transmission is relevant.

We conclude from this line of reasoning that C should defer transmission while B is transmitting data. Note that because C has only heard the RTS and not the CTS, station C cannot tell if the RTS-CTS exchange was a success and so does not know if B is indeed transmitting data.

There are two approaches to this problem. One can use carrier-sense to avoid sending useless RTS's. A station must defer transmission until one slot time after it detects no carrier (the inclusion of a single slot time of clear air is to ensure that exposed terminals do not clobber the returning ACK). This is essentially the CSMA/CA protocol [2]. We chose a slightly different approach, which does not require carrier sensing hardware. Before sending a DATA packet, a station sends a short 30-byte Data-Sending packet (DS). Every station which overhears this packet will know that the RTS-CTS exchange was successful and that a data transmission is about to occur; these overhearing stations defer all transmissions until after the ACK packet slot has passed.

	RTS-CTS-DATA-ACK	RTS-CTS-DS-DATA-ACK
P1-B1	46.72	23.35
P2-B2	0	22.63

Table 5: The throughput, in packets per second, achieved by the streams in Figure 5.

We have examined the performance of this protocol in the simple two-cell configuration of Figure 5. Here each pad is an exposed terminal to the other pad-to-base-station stream. Table 5 shows the throughput with and without the DS packet. Without the DS packet, one of the pads loses the first contention period and then proceeds to futilely retransmit. The key is that without the DS packet the "losing" pad cannot identify when the next contention period (i.e., the slots after the ACK packet and before the next RTS) starts, therefore it is unable to compete effec-

[7] We should also note that C's transmission does not harm the sender B only if the sender does not need to receive a packet after the CTS. Now that we have included the ACK packet after the DATA packet, this assumption no longer holds. If the exposed terminal begins transmitting after not hearing the CTS, then it is possible that its transmission will collide with the ACK returned from A to B.

Figure 6: A two cell configuration where both pads are in range of their respective base stations and also in range of each other. The base stations are sending data to their respective pads, and each stream is generating data at a rate of 64 packets per second and using UDP for transport.

tively for access. Because DATA packets are large compared to the control packets, an ongoing stream is sending data most of the time. Thus, if the "losing" station is essentially picking random times to retry it will usually end up transmitting its RTS during the middle of an ongoing data transmission which, as we argued above, invariably results in a collision. Thus, to compete effectively, the pad must send its RTS packets during the contention periods, and this requires knowing when the data transmissions start and finish. This need for "synchronizing" information is crucial; in this configuration it is supplied by the DS packet (which informs the other stations about the existence and length of the following DATA packet). In the next section we will add another control packet that provides such synchronizing information for other configurations.

3.3.3 RRTS

Consider the two-cell configuration depicted in Figure 6, where each of the two data streams alone can fully load the media. The first column in Table 6 shows the throughput resulting from the version of the media access protocol incorporating all the amendments we have discussed so far. The B1-P1 stream is almost completely denied access, while the B2-P2 stream is receiving all of its requested throughput. As in the previous section, this is a symmetric configuration (in fact, it is the same configuration as in Figure 5 except that the data flows are reversed) and one of the streams (in this case the B2-P2 stream) wins the initial contention period. After this, due to the relatively large data packet size as compared to that of control packets, most of the time when B1 initiates a data transfer by sending an RTS, the receiving pad P1 cannot respond with a CTS because it is deferring to the data transmission to P2. The only way B1 can successfully initiate a transfer is when its RTS happens to arrive during those very short gaps in between a completed data transmission and the completion of P2's next CTS.

	no RRTS	RRTS
B1-P1	0	20.39
P2-B2	42.87	20.53

Table 6: The throughput, in packets per second, achieved by the streams in Figure 6.

The key problem is again the lack of synchronizing information; B1 is trying to contend with B2 during very short contention periods, but B1 has no way of knowing when those periods start or finish. Notice that the DS packet does not solve this problem because neither of the base stations can hear any part of the other streams message exchange.

Figure 7: A two cell configuration where both pads are in range of their respective base stations and also in range of each other. Base station B1 is sending data to pad P1, and pad P2 is sending data to base station B2. Each stream is generating data at a rate of 64 packets per second and using UDP for transport.

A secondary problem is that B1's backoff counter keeps increasing because it never receives a response from P1.

We can solve both of these problems by having P1 do the contending on behalf of B1. Whenever a station receives an RTS to which it cannot respond (due to deferral), it then contends during the next contention period and sends a Request-for-Request-to-Send packet (RRTS) to the sender of the RTS (if it has received several RTS's during the deferral period, it only responds to the first received RTS). The recipient of an RRTS immediately responds with an RTS. If B1 sends an RTS in response to the RRTS, the normal message exchange is commenced. Stations overhearing an RRTS defer for two slot times, long enough to hear if a successful RTS-CTS exchange occurs. The second column in Table 6 shows the throughput that results from this protocol. Now, both streams have a fair access to the media.

The RRTS packet, however, does not solve all such contention problems. Consider the two-cell configuration depicted in Figure 7. Table 7 shows the throughput resulting from the version of the media access protocol incorporating the amendments we have discussed so far. The B1-P1 stream is completely denied access, while the P2-B2 stream is getting complete channel utilization. This is because most of the time when B1 initiates a data transfer by sending an RTS, P1 cannot hear it due to P2's transmission. The only time B1 can successfully initiate a transfer is when its RTS happens to arrive during those very short gaps in between a completed data transmission and the transmission of P2's next RTS. Again the key is the lack of synchronization information; B1 has no way of knowing when the contention periods are. The RRTS packet is irrelevant here since P1 cannot hear the incoming RTS. We have yet to solve this problem.

B1-P1	0
P2-B2	43.93

Table 7: The throughput, in packets per second, achieved by the streams in Figure 7.

3.3.4 Multicast

So far we have only discussed unicast transmissions, where there is a unique receiver for each packet. For multicast data transmission, there can be multiple receivers for a packet. The RTS-CTS exchange is no longer viable since the multiple receivers cannot coordinate and are likely to collide with each other's CTS. For the time being we have avoided such CTS collisions by having a multicast transmission use an

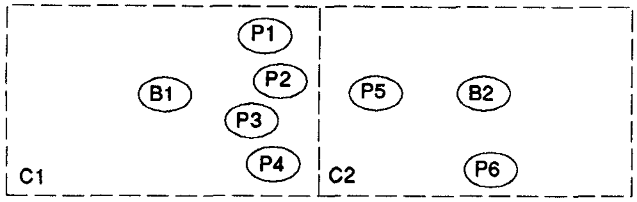

Figure 8: A two cell configuration where all the pads in cell C1 are in range of pad P5. The copying of backoff counters leads to "leakage" of backoff values between the two differently loaded cells.

RTS followed immediately by the DATA packet. The overhearing stations can identify that the RTS is for a multicast address, and therefore all stations defer for the length of the following DATA transmission.

This design, however, has the same flaws as CSMA. Only those stations within range of the sender will defer, and those that are within range of a receiver but not the sender will not be given any signal to defer; in the unicast case the signal is delivered by the CTS packet. We have yet to figure out how to make receiver generated control messages like CTS work in the case of multicast without involving several rounds of contention.

3.4 Backoff Algorithm Revisited

Due to the random nature of multiple access, collisions are unavoidable. MACAW uses a backoff algorithm, in which the transmission of RTS's are delayed for a random number of slots, to reduce the probability of collision and to resolve collisions once they occur. The calculation of this random delay is based on the backoff counter; the value of the backoff value should therefore reflect the level of contention for the media.

As stated earlier, the goal of MACAW is to achieve both a high overall throughput and a "fair" allocation of throughput to streams. The backoff algorithm in MACAW plays a crucial role in achieving both of these goals. For a backoff algorithm to be efficient, the backoff counters must accurately reflect the level of contention. In Section 3.1 we described how we modified the backoff computation from BEB to MILD to achieve a more stable estimate of the contention level. For a backoff algorithm to be fair, all contending stations should use the same backoff counter. In Section 3.1, we also introduced a backoff copying scheme that ensures that contending stations will have the same backoffs.

Notice that this design models the contention level by a single number. This is only appropriate if congestion is always homogeneous. However, in our multicell wireless LAN, congestion is typically not uniform. There is heavy contention for the media in some cells and light contention in others. The single backoff counter algorithm can perform poorly in such cases. Consider, for example, the configuration depicted in Figure 8. There are two adjoining cells, C1 and C2. C1 contains four pads (P1-P4), all near the border with C2. C2 contains only two pads, one of which (P5) is near the border with C1. All of the pads are attempting to transmit to their respective base stations and are individually generating enough data to consume the entire channel. The pads near the border (P1-P5) are within range of each other, and so they overhear each other's packets. If there

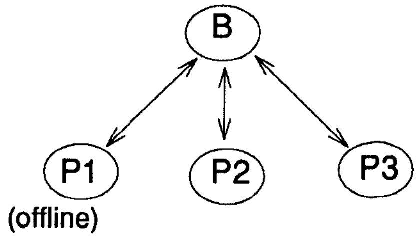

Figure 9: A single cell configuration where all pads are in range of the base stations and also in range of each other. The base station is sending data to each pad, and each pad is sending data to the base station. Each stream is generating data at a rate of 64 packets per second and using UDP for transport. Pad P1 is turned off after 300 seconds.

were no overhearing, then the contention level, and the average backoff counter, would be rather high in C1 and much lower in C2. However, the fact that the border pads can overhear one another leads to "leakage" of the backoff values between the two cells. A high backoff counter from P1 can be copied by P5 and then by B2 and finally by P6. All transmissions in C2 would now have an artificially high backoff value leading to wasteful idle time. Similarly, a low backoff counter from P6 can also traverse the reverse route to make all the transmissions in C1 have an artificially low backoff value leading to wasteful collisions.

Thus, when we use a single number to model congestion, the copying algorithm creates problems. The backoff value can be copied from one region to another, even though the regions have different levels of congestion; we are no longer assured that the backoff counter accurately reflects the ambient contention level in a cell.

There is a second problem with our backoff algorithm. So far we have implicitly assumed that if an RTS fails to evoke a CTS then there has been a collision, and that this collision reflects congestion. However, there are other reasons why an RTS might fail to return a CTS. If there is a noise source close to either the sender (so the returning CTS is corrupted) or the receiver (so the RTS is corrupted), then the RTS-CTS exchange will never succeed. After each unsuccessful attempt, the sender will increase its backoff even though there isn't necessarily any contention in the cell. Similarly, if a base station is trying to reach a pad which has been turned off (or which has left the cell), there will be no response to the base station's RTS but this is not related at all to any contention. In both cases, the sending station will have a high backoff value even though there is little contention for the media. These phenomena are the result of using a single number to reflect the ambient congestion level in the region; they are made worse by the copying algorithm.

For example, consider the configuration depicted in Figure 9. There are three pads in a single cell. The base station is sending to each pad, and each pad is sending to the base station. After some time, pad P1 is turned off. However, the base station continues trying to communicate with pad P1. Every RTS sent to the pad results in a timeout; after a certain number of these the base station gives up (in MACAW we allow a certain number of retries on each packet before discarding the packet; see Appendix B for details). As we discussed in Section 3.2, a random delay interval is chosen for each of the base station's streams and the stream with the earliest retry slot is chosen for transmission. Since there is a single base station backoff counter, all streams have an equal chance of being chosen. Every time stream B-P1 is selected, the backoff will be increased by a factor of 1.5. When either of the other two streams emanating from the base station is chosen and carries out a successful transmission, then the backoff counter is reduced by one. Because the backoff algorithm increases faster than it decreases[8], the backoff counter is eventually driven to very high values.

Because P1 is unreachable, a high backoff is reasonable for transmission to P1. The problem is that this high value of the backoff counter is also used when the base station communicates with the other pads. The problem is exacerbated by the copying algorithm, i.e. this high backoff value is also copied and used by the pads transmitting to the base station. Successful transmissions bring the backoff down and the lowered value is copied back to the base station, but the fact that the multiplicative backoff increases will always dominate the additive backoff decreases means that eventually all the backoffs will be high. As a result, the overall thoughput is low, as shown in the first column of Table 8.

	Single backoff	Per-destination backoff
B1-P2	3.79	8.98
P2-B1	3.78	8.84
B1-P3	3.62	8.68
P3-B1	3.43	8.41

Table 8: The throughput, in packets per second, achieved by the streams in Figure 9.

All three examples (Figure 8, noise next to the sender or receiver, and Figure 9) discussed in this section demonstrate that we must differentiate between the backoffs used to send to different pads. Each station should maintain a separate backoff counter for each stream. As we discussed earlier, the bi-directional nature of the message exchange requires us to account for congestion at both ends of the stream. The backoff value used in a transmission should reflect the congestion at both the destination and at the sender. These congestion levels should be estimated separately, so that this information can be shared by copying, but then should be combined when computing the backoff to be used in transmission.[9] Estimating the congestion at each end requires, in the backoff adjustment algorithm when an RTS-CTS exchange fails, determining whether it was due to the RTS not being received or to the CTS not being received. If an RTS is received but the returning CTS is not, we know that there is congestion at the sender and not at the receiver. If the RTS is not received, we know that there must be congestion at the receiver, but we do not know if there is congestion at the sender as well. In our algorithms, we will not make any change to the congestion estimate at the sender if an RTS fails; similarly, we will not make any change to the congestion at the receiver if a CTS fails. In Appendix B, we describe in detail an algorithm that can determine if the RTS failed or the CTS failed (this determination can only be definitive after the RTS-CTS exchange is finally successful). Given that information, we then adjust the appropriate end's backoff value according to the usual adjustment algorithms.

To achieve fairness, all stations attempting to communicate with the same receiving station should use the same backoff value. This is achieved by our backoff copying scheme, except now there is a separate backoff value for each station, and we now insert the backoff values of both ends into each packet header. The right column of Table 8 shows the simulation results with this per-destination backoff copying algorithm; the overall throughput is no longer affected by the unresponsive pad. This per station backoff copying algorithm enables the congestion information to be specific for a given station yet also copied to all stations that are sending to it.

[8] In some sense, this is a problem of our own making; the BEB algorithm decreases faster than it increases, and so would not suffer the problem we describe here. Of course, the BEB algorithm has, as we have observed, problems of its own

[9] We combine the congestion information by summing the two backoff values.

3.5 Preliminary Evaluation of MACAW

Appendix B contains a detailed description of the MACAW protocol. We take this opportunity to quantify the overhead introduced by the MACAW protocol in a cell with a single UDP stream from a pad to a base station. In Table 9, we show the packets-per-second transmitted under the original MACA RTS-CTS-DATA exchange, and under MACAW's RTS-CTS-DS-DATA-ACK exchange. Note that the addition of the DS and ACK packets decrease the throughput by roughly 8%. MACA achieves a data rate of roughly 217kbps, which is 84% channel capacity. MACAW achieves a data rate of roughly 201kbps, which is 78% channel capacity. However, as we shall see below, this overhead is often more than compensated by superior performance in the presence of congestion and noise.

MACA	RTS-CTS-DATA	53.07
MACAW	RTS-CTS-DS-DATA-ACK	49.07

Table 9: The throughput, in packets per second, achieved by a uncontested single stream.

The scenarios we used to motivate our various design decisions were extremely simple. In this section we present results from two somewhat more complicated network configurations. The first scenario has three cells as shown in Figure 10, which is somewhat similar to Figure 8. C1 contains four pads (P1-P4), all near the border with C2. C2 contains only one pad (P5), which is near the border with C1. There is one pad (P6) which straddles the border between C2 and C3 and is in range of both B2 and B3. The pads near the C1-C2 border (P1-P5) are within range of each other, and so they overhear each other's packets; however, they can only hear their own base station. Each of the pads (P1-P5) has UDP data streams to and from the base station of its cell. Pad P6 is sending a UDP data stream to B3. The data generation rate in each stream is 32 packets per second. Table 10 shows the throughput for each stream; we compare the performance of the MACA algorithm with the MACAW algorithm.

We remark on a few aspects of the simulation results. First, using MACAW over MACA has yielded an improvement of over 37% in throughput. Thus, a superior ability to handle congestion has more than compensated for MACAW's increased overhead. Second, MACAW has yielded a "fairer" division of throughput among the streams in the

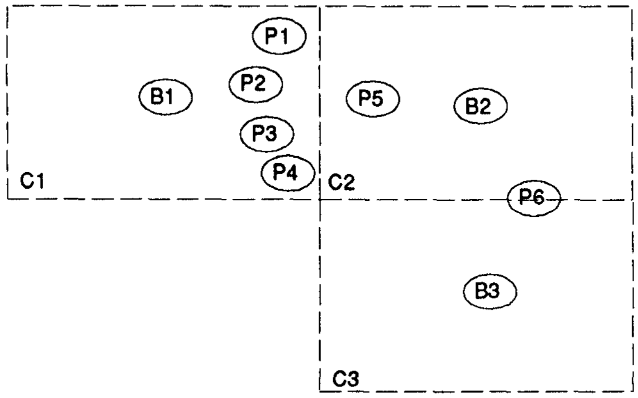

Figure 10: A configuration with three cells with varying levels of congestion.

	MACA	MACAW
P1-B1	9.61	3.45
P2-B1	2.45	3.84
P3-B1	3.70	3.27
P4-B1	0.46	3.80
B1-P1	0.12	3.83
B1-P2	0.01	3.72
B1-P3	0.20	3.72
B1-P4	0.66	3.59
P5-B2	2.24	7.82
B2-P5	3.21	7.80
P6-B3	28.40	25.16

Table 10: The throughput, in packets per second, achieved by the streams in Figure 10.

same cell. In MACAW, the maximum difference between throughput for any two streams in the same cell is only 0.59 packets per second, while in MACA, the maximum difference is 9.60 packets per second. Third, MACAW is able to cope with highly nonhomogeneous congestion, and can shield uncongested neighbours from losing too much throughput due to the presence of a congested neighbour. In both MACA and MACAW, the propagation effect of the congestion across cells is small. Moreover, even though C1 has a much higher contention level than C3 where there is only one data stream running, the two cells achieve about the same media utilization.

The second scenario, as depicted in Figure 11, simulates a small portion of the Computer Science Laboratory at Xerox PARC. There are four cells, which represent an open area (cell C1) flanked by the offices of two researchers (C2 and C3) and a coffee room (C4). Each of the "office" cells C2 and C3 has one pad (P6 in C2, P5 in C3). There are four pads in cell C1 (P1 - P4). There is also a noise source in the cell C1, which is due to the presence of a large electronic whiteboard in the open area. We simulate the effect of the noise by a packet error rate of 0.01 (see Section 3.3.1). Pad P7 is brought into the coffee room (cell C4) from an uncongested cell 300 seconds into the simulated period (which is 2000 seconds long). Each pad sends a TCP data stream to the base station of its cell, with a data generation rate of 32 packets per second. In addition to each pad hearing the base station in its cell, and hearing all other pads in their cell, the configuration is such that P7 can hear P1 and B1 in cell C1, and the pads P4, P5, and P6 can hear each other.

Table 11 shows the throughput for each TCP data stream. MACAW achieves an improvement in total throughput of about 13% over MACA. More importantly, MACAW achieves a fairer distribution of throughput. The P7-B4 and P5-B3 streams respectively capture 46% and 35% of the throughput using MACA, while with MACAW they receive only 32% and 24% respectively. In this simulation, the impact of mobility was not prominent in either case.

	MACA	MACAW
P1-B1	0.78	2.39
P2-B1	1.10	2.72
P3-B1	0.22	2.54
P4-B1	0.06	2.87
P5-B3	18.17	14.45
P6-B2	6.94	14.00
P7-B4	23.82	19.18

Table 11: The throughput, in packets per second, achieved by the streams in Figure 11.

4 Future Design Issues

In Section 3.3, we presented simulation results which supported adding an ACK to the basic RTS-CTS exchange. This data shows the importance of including the functionality of the ACK, but requiring an ACK packet in every basic message exchange is not the only way to achieve this. For instance, unless specifically requested by the sender (through setting a bit in the packet header), ACK's could be piggybacked onto the subsequent CTS packets (by including a field which indicated the sequence number of the most recently arrived packet). Whenever the queue for a stream at a station had more than one packet in it, the sending station would not request the ACK but would merely wait for the piggy-backed acknowledgement on the next CTS. When the queue for a stream at a station had only a single packet, then the sending station would request an ACK.

Alternatively, one could also use NACK's instead of ACK's. Whenever a receiving station did not receive data as expected after sending a CTS, it would send a NACK back to the originator of the RTS. We have not tested either of these alternative ACK'ing schemes. We merely note here that while the case for including the functionality of link-level acknowledgements is strong, there are many ways to implement that functionality and we have not yet fully explored the various options.

Similarly, the data in Section 3.3.2 suggested that stations should not transmit during nearby ongoing data exchanges. We achieved this by the inclusion of DS packets in the message exchange pattern. However, one could equivalently use full carrier-sense, which also inhibits RTS-RTS collisions. There are many intermediate options, such as sensing only clean signals. We have yet to explore the space of such carrier sense mechanisms.

The configuration in Figure 7 poses a problem to which we have, as yet, no answer. This requires some distribution of synchronizing information, but it seems difficult to supply the information to B1 since none of the stations in the

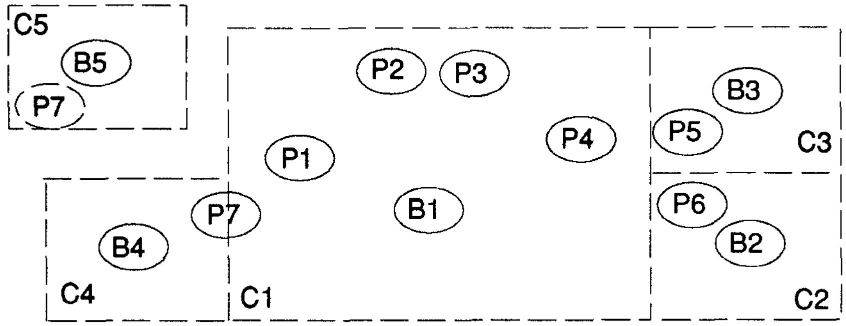

Figure 11: A configuration based on part of the Computer Science Laboratory at PARC.

congested area are aware that B1 is attempting to transmit. Another unsolved problem is multicast; we are not satisfied with our simple RTS-DATA scheme for multicast traffic.

The backoff algorithm we presented is certainly an improvement over the initial single backoff, BEB proposal. However, the performance of such algorithms is exceedingly complex, and the design space is enormous. We have merely scratched the surface both in understanding the behavior of our algorithm and in understanding the other algorithmic options available.

Unlike Karn [9], we did not consider any media access protocols which involved variation in power. Such variations violate the symmetry principle which was so central to our understanding, and thus on our initial pass we did not want to venture so far from familiar territory. Nonetheless, in future work we hope to consider power variations more carefully.

Just as wired networks are moving from offering only a single class of "best-effort" service, wireless networks are also moving towards broadening their service model. For instance, the protocol in [3] supports synchronous service as well as asynchronous service. We have not considered such service yet. We have also not considered approaches other than the multiple access approach. Various token-based schemes, or those involving polling or reservations, are possibilities we hope to explore in future work.

Furthermore, we stated that our criterion for evaluating the performance was utilization of bandwidth and fair allocation of that bandwidth. In homogeneous and localized settings such as an Ethernet, fairness is a well-defined concept. However, in the geographically distributed and non-homogeneous setting of a wireless LAN, fairness is not well defined. For instance, a pad that is on the border of two cells, and can therefore hear both base stations, essentially ties up both base stations when transmitting (in that neither of them can receive other transmissions). Should such a pad receive the same allocation of throughput as pads who are only in range of one of the base stations? Should such pads receive less, because they cause more congestion? Before settling on a final design choice, we must decide what allocation policy we want to implement.

5 Summary

The emergence in recent years of a new generation of mobile computing devices suggests that indoor wireless LAN's will play an increasingly important role in our telecommunication infrastructure, particularly in traditional office settings where the demand for such mobile communication will be highest. The media in such indoor wireless LAN's is a shared, and scarce, resource; thus, controlling access to this media is one of the central design issues in wireless LAN's. In this paper we have discussed the design of a new media access protocol for wireless LAN's: MACAW. It is derived from Karn's earlier proposal, MACA [9]. Our design process relied on four pieces of insight.

First, the relevant congestion is at the receiver, not the sender. This realization, due to Karn [9] and others [4, 12], argues against the traditional carrier-sense approaches (CSMA), and suggests the Appletalk-like use of an RTS-CTS-DATA message exchange. For a variety of performance reasons, we have generalized this to first an RTS-CTS-DATA-ACK exchange and then an RTS-CTS-DS-DATA-ACK exchange.

Second, congestion is not a homogeneous phenomenon. Rather, the level of congestion varies according to the location of the intended receiver. It is inadequate to characterize congestion by a single backoff parameter. We instead introduced separate backoff parameters for each stream, and then for each end of the stream. Care was taken to identify which end of the stream was experiencing collisions.

Third, learning about congestion levels should be a collective enterprise. When each station must rely on its own direct experience in estimating congestion, often chance leads to highly asymmmetric views of a homogeneous environment. To rectify this, we introduced the notion of "copying" the backoff parameters from overheard packets. This idea could be relevant to not only wireless networks but also to Ethernets and other shared media.

Fourth, the media access protocol should propagate synchronization information about contention periods, so that all devices can contend effectively. The DS packet is one example of providing the synchronizing information. Note that this observation implies that contention for access should not just be initiated by the sender of data. In cases where the congestion is mainly at the receiver's location, the sender cannot contend effectively since it cannot know when the data transmissions are over. We introduced the RRTS packet so that the receiving end can contend for bandwidth when it is in the presence of congestion. This packet also allows congestion information to be sent even when data was not being communicated.

These various changes have significantly improved the performance of the media access protocol. However, our design is still preliminary. As we discuss in Section 4, there are many issues which remain unresolved.

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A MACA

The Control rules and the Backoff rules govern the carrier capture and data transmission. The Deferral rules govern collision avoidance. Let us assume station A wants to transmit a data packet to station B. Let stations C and D be two other stations such that C hears only A, and D hears only B.

A pad running MACA can be in one of five states: IDLE, CONTEND, Wait For CTS (WFCTS), Wait For Data (WFData) and QUIET. The Control rules that goven the state transition are the following.

1. When A is in IDLE state and wants to transmit a data packet to B, it sets a random timer and goes to the CONTEND state.
2. When B is in IDLE state and receives a RTS packet from A, it transmits a Clear-To-Send (CTS) packet which contains the machine IDs of B and A, and the permitted number of bytes to send. B sets a timer value and goes to Wait For Data (WFData) state.
3. When A is in WFCTS state and receives a CTS packet from B, it clears the timer, transmits the Data packet to B, and resets the state to IDLE.
4. When B is in WFData state and receives a Data packet from A, it clears the time and resets the state to IDLE.
5. If A receives a RTS packet when it is in CONTEND state, it clears the timer, transmits a CTS packet to the sender and goes to the WFData state.

The Deferral rules are the following.

1. When C hears an RTS packet from A to B, it goes from its current state to the QUIET state, and sets a timer value sufficient for A to hear B's CTS.
2. When D hears a CTS packet from B to A, it goes from its current state to the QUIET state, and sets a timer value sufficient for B to hear A's Data.

The Timeout rules are the following.

1. When A is in CONTEND state and the timer expires, it transmits a Request to Send (RTS) packet which contains the station ID's of A and B, and the requested number of bytes to send. A then sets a timer and goes to WFCTS state.
2. From any other state, when a timer expires, a station goes to the IDLE state.

The descending order of precedence is Deferral rules, Control rules, and Timer rules.

B MACAW

B.1 Message Exchange and Control Rules

As in MACA, MACAW can also be described in terms of Control rules, Deferral rules, and Timeout rules.

A pad running MACAW can be in one of eight states: IDLE, CONTEND, Wait For CTS (WFCTS), Wait for Contend (WFCntend), Wait For DataSend (WFDS), Wait For Data (WFData), Wait For ACK (WFACK) and QUIET.

The Control rules are the following.

1. When A is in IDLE state and wants to transmit a data packet to B, it sets a random timer and goes to the CONTEND state.
2. When station B is in IDLE state and receives a RTS packet from A, it transmits a Clear-To-Send (CTS) packet. B then sets a timer and goes to Wait for DataSend (WFDS) state.
3. When A is in WFCTS state and receives a CTS packet from B, it clears the timer, transmits back-to-back a DS followed by the data packets to B. A then enters WFACK state and sets a timer.

4. When B is in WFDS state and receives a DS packet from A, it goes to WFData state and sets a timer.

5. When B is in WFData state and receives a data packet from A, it clears the timer, transmits an ACK packet, then goes to IDLE state.

6. When A is in WFACK state and receives an ACK packet from B, it resets the state to IDLE, and clears the timer.

7. When B is in IDLE state and receives a RTS for a data packet it acknowledged last time, it sends the ACK again instead of CTS.

8. If A receives a RTS packet when it is in CONTEND state, it transmits CTS packet to the sender, goes to the WFDS state and sets a timer value.

9. If C is in QUIET state and receives an RTS, it goes to the WFContend state.

10. If a station is in IDLE state and receives a Request to Request To Send (RRTS) packet, it transmits a RTS packet to the sender, goes to the WFCTS state and sets a timer value.

The Deferral rules are the following.

1. When C hears a RTS packet from A to B, it goes from its current state to the QUIET state, and sets a timer value sufficient for A to hear B's CTS.

2. When C hears a DS packet from A to B, it goes from its current state to the QUIET state, and sets a timer sufficient for A to transmit the Data packet and then hear B's ACK.

3. When D hears a CTS packet from B to A, it goes from its current state to the QUIET state, and sets a timer value sufficient for B to hear A's data.

4. When B hears a RRTS packet from D, it goes from its current state to the QUIET state and sets a timer value sufficient for an RTS-CTS exchange.

The Timeout rules are the following.

1. When a station is in WFContend state and the timer expires, it sets a random timer and goes to the CONTEND state.

2. When a station is in CONTEND state and the timer expires, it may either transmit a RTS packet to perform a sender-initiated data transmission (A) or a RRTS packet to perform a receiver-initiated data transmission (D), depending on whether it entered CONTEND state from IDLE state, or from WFContend state.

   For sender-initiated transmission, A transmits a RTS packet, containing the station ID's of A and B, and the requested number of bytes to send. A goes to WFCTS state, and sets a timer value. For receiver-initiated transmission, D transmits a RRTS packet and then goes to IDLE state.

3. From any other state, when a timer expires, a station goes to the IDLE state and resets the timer value.

B.2 Backoff and Copying Rules

Each station keeps the following variables:

1. **my_backoff**: the backoff value at this station.

2. For each remote pad:

   local_backoff : the backoff value at this station as estimated by the remote station.

   remote_backoff : estimated backoff value for the remote station.

   exchange_seq_number (ESN) : a sequence number used in packet exchanges with the remote station.

   retry_count : the number of retransmissions.

When a pad P hears a packet, other than an RTS, from Q to R, P updates its estimate about Q and R's contention levels by copying the `local_backoff` and `remote_backoff` values carried in the packet, respectively. In addition, P also copies Q's backoff value as its own backoff, assuming that Q is a nearby station therefore Q's backoff supposedly reflects the congestion level around the neighborhood. RTS packets are ignored because they may not carry the correct backoff values. More precisely,

```
packet(local_backoff, remote_backoff, ESN)
   If packet == RTS, ignore
   else
      Q's backoff = local_backoff;
   if (remote_backoff != I_DONT_KNOW)
      R's backoff = remote_backoff;
   my_backoff = local_backoff;
```

When a Pad P receives a packet from Pad Q to P, if the `exchange_seq_number` has increased, either the packet is initiating a new transmission or a successful handshake has completed. In either case, the backoff values carried in the packet should be the correct ones. Thus P updates the backoff values of its own and that of Q's, increases the ESN and resets the retry_count. Here P's `local_backoff` is a variable used temporarily when attempting an exchange with Q; its value is synchronized with P's `my_backoff` once a successful handshake is done.

On the other hand, if the packet is a retransmission of an old packet, P assumes a collision occurred at Q's end, and increases the backoff value for Q accordingly. Because the sum of the backoff values of the two ends should be the same independently from at which end the collision has occurred, P upates its own backoff value as the difference between the sum and Q's backoff value (as P estimated).

```
packet(local_backoff, remote_backoff, ESN)
   If (ESN > ESN for Q)
      Q's backoff = local_backoff;
      if (remote_backoff != I_DONT_KNOW)
         P's local_backoff = remote_backoff;
         my_backoff = remote_backoff;
      else
         P's local_backoff = my_backoff;
      P's ESN for Q = ESN + 1;
      P's retry_count with Q = 1;
   else  /* the packet is a retransmission */
      Q's backoff = local_backoff + retry_count * ALPHA;
      if (remote_backoff != I_DONT_KNOW)
```

```
      P's local_backoff = (local_backoff + remote_backoff) - Q's backoff;
   else
      P's local_backoff = my_backoff;
   retry_count ++;
```

When pad P sends a packet to Q, it assigns the parameter values in the packet, local_backoff, remote_backoff, and ESN in the following way:

```
If (packet = RTS)  /*or should it be at the beginning of a new packet*/
   local_backoff (used in communicating with Q) = my_backoff;

remote_backoff = Q's backoff (or I_DONT_KNOW);
local_backoff = local_backoff used with Q;
Send packet with local_backoff, Remote_backoff, ESN;
```

When a Pad P times out on a packet to Q:

```
Q's backoff += retry_count * ALPHA;
If reached max_retry_count,
   P's local_backoff used with Q = MAX_BACKOFF;
   Q's backoff = I_DONT_KNOW;
```