An Uplink Medium Access Protocol with SDMA Support for Multiple-Antenna WLANs

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Abstract— In this paper, we propose a contention based uplink Medium Access Control (MAC) protocol design for Wireless Local Area Networks (WLANs) with Spatial Division Multiple Access (SDMA) support. Our protocol does not require sophisticated smart antenna equipments, and it can be implemented in simple omni-directional multiple-antenna WLANs. Different from the super-frame based approaches, the proposed one is a pure contention based MAC protocol and can be easily implemented into standard 802.11 systems with slight modifications. By jointly considering the the physical and the MAC layer situations, dynamic system parameter adjustment is designed to enhance throughput and protocol efficiency. In addition, our protocol provides interface for user scheduling, which makes it more extensible. Simulation results show that our scheme can achieve high network throughput, and discussions regarding different system factors are also included.

I. INTRODUCTION

The growing demand for high speed wireless Internet access has led to extensive research on the development of Wireless Local Area Networks (WLANs), especially on improving throughput and protocol efficiency. The new IEEE 802.11n draft [3] takes Multiple Input Multiple Output Orthogonal Frequency Division Multiplexing (MIMO-OFDM) as the physical layer solution, which greatly increases the transmission rate by using spatial diversity or multiplexing. On the other hand, facilitating multiple transmissions or receptions in WLANs has been proposed recently [5] [6] [7] [8] [9], which is mostly realized by using Spatial Division Multiple Access (SDMA) with multiple antennas equipped on the access point (AP) and the user stations. Adopting SDMA in multiple-antenna WLANs is beneficial due to the following reasons. First, allowing several users to transmit simultaneously, SDMA can avoid collisions and reduce the number of inter-frame spaces to relieve the protocol burden, which can not be achieved by merely increasing data rate [4]; moreover, due to the capability and space limitation of the user stations, the number of antennas on the user stations is less than that of the AP, then there will be more degrees of freedom to be used by SDMA to further increase the sum-data rate.

However, the MAC layer enhancement in 802.11n does not include SDMA support. In order to exploit SDMA in WLANs, two issues have to be considered in MAC protocol design: channel estimation and frequency synchronization. Thanks to the RTS/CTS scheme in 802.11 MAC, stations and the AP can make use of RTS/CTS exchanges to estimate the channel signature and do frequency/time synchronization. Hence the key issue is transformed to another problem: where and how should the RTS/CTS exchange take place in order to allow sufficient channel estimation and synchronization for multiple user stations, while keeping the efficiency high. This also requires the newly deigned MAC protocol exploiting the interaction between the physical and MAC layers.

In Ref. [5], an SDMA solution is proposed regarding both the physical layer and the MAC layer. In this work, the MAC protocol is based on HIPERLAN-II [2], of which the superframe is divided into parts to allow random access and data transmission. In the random access period, channel estimation and frequency synchronization can be accomplished by access requests. Being one of the pioneering works in this field, it is an omni-directional multiple-antenna solution. Recently there appears some approaches relying on smart antenna systems, most of which are multi-beam directional antenna ones [6] [7] [8]. However, multi-beam antenna systems may not work well with unevenly distributed user stations, and collecting the location information of user stations leads to additional protocol overhead. Considering economic and practical issues on the user station side, it is more appropriate to use omni-directional multiple-antenna in WLANs especially for the *uplink*. On the other hand, in order to allow channel estimation and synchronization for multiple users, the MAC layer protocols they have proposed are all super-frame based with fixed length random access period, which is not flexible and not compatible with the widely adopted DCF (Distributed Coordination Function) mode of 802.11 MAC, so pure contention based MAC protocol is preferred. While most of the existing MAC protocols do not consider the channel condition, the one proposed in Ref. [9] is channel state information based, and can be used to enhance performance in WLANs with CDMA or OFDMA as the physical layer. For SDMA, channel characterization is more complex, thus MAC protocols similar to that in [9] can not be directly exploited.

In this paper, we propose a pure contention based uplink MAC protocol supporting SDMA in omni-directional multiple-antenna WLAN systems. While providing significant network throughput gain, the protocol can be easily realized in standard 802.11 MAC with slight modifications. In addition, with alterable random access period accomplishing channel estimation and synchronization, our scheme is more flexible and efficient, and can adapt itself to both the channel condition

and the network load by jointly considering the physical and MAC layer situations. Moreover, under the framework of our proposed protocol, various user scheduling schemes can be implemented, and it gives chances to realize central scheduling in contention based MAC.

The rest of the paper is organized as follows. Section II describes the detail of our proposed protocol. In Section III, we theoretically analyze the throughput performance of the protocol, based on which the system parameter adaptation scheme is described. Simulation results are provided and the performance of our protocol is evaluated in Section IV. Finally, Section V concludes the paper.

II. PROPOSED MAC PROTOCOL

A. System Description and Framework

We consider the uplink access of a single cell with one AP. Each user station has a single antenna for low cost, while the AP has an array of up to 4 antennas. The antennas of the stations and the AP are omni-directional. The physical layer is MIMO-OFDM as described in the new 802.11n draft [3]. No more than 4 user stations can simultaneously transmit OFDM modulated symbols to the AP. Then the AP uses the Minimum Mean Squared Error (MMSE) detector to separate the signals of the user stations, according to the channel knowledge obtained by RTS/CTS exchanges.

Our proposed MAC protocol is based on standard IEEE 802.11 MAC protocol [1] with Distributed Coordination Function (DCF) mode. In order to support SDMA transmission, the channel contention scheme is extended. Nevertheless, the MAC protocol is still contention based, no additional central control message is needed. As schematized in Fig. 1 and Fig. 2, the whole transmission procedure is divided into two periods: random access period, and data transmission period. The data transmission period includes parallel uplink data transmissions and a broadcasting ACK. Generally speaking, the data transmission time is decided by the data frame length, and for simplicity, we set it to a constant length. The random access period allows several RTS/CTS exchanges for stations to compete for the channel. By RTS/CTS exchanges, stations and the AP can also accomplish channel estimation and frequency synchronization.

In our protocol, the AP holds the following two parameters: M_{random} : Maximum possible number of random transmission requests during the random access period. (If all of them are accepted by the AP, then M_{random} equals to the number of simultaneous transmissions in the data transmission period.) T_{timeout} : Maximum possible length of random access period.

In the random access period, if within T_{timeout} there has already been M_{random} RTS/CTS exchanges, then the random access period immediately ends, after which the data transmission period starts (see Fig. 1). Otherwise, if the load is low or the number of stations is less than M_{random} , timeout will occur (see Fig. 2). In this situation, before the data transmission starts, the AP will wait for the whole T_{timeout} . Hence, in our protocol, the length of the random access period varies, depending on the two parameters the AP holds and the traffic

Fig. 1. Protocol example, for $M_{\text{random}} = 3$ and T_{timeout} is relatively long (comparing to the load))

Fig. 2. Protocol example, for $M_{\text{random}} = 3$ and T_{timeout} is relatively short (comparing to the load)

load. As one of the main feature of our proposed scheme, the non-fixed length of the random access period is different from the fixed one of the schemes with super-frames¹. Hence, our scheme is more efficient and flexible.

B. Protocol detail

In this section, we give the details of our proposed MAC protocol. There are two kinds of CTS the AP sends (Refer to Fig. 1 and Fig. 2). The first kind is denoted as PCTS (Pending CTS), and it acts like an acknowledge to the RTS that AP has just received, but data transmission will not follow after the PCTS. The network allocation vector (NAV) of PCTS is set to zero, and the remaining T_{timeout} is included. The second kind is denoted as FCTS (Final CTS), and it acts not only like an acknowledge to the last RTS the AP has received, but also an message to announce the start of data transmission. The NAV of FCTS is set to be $SIFS + H_{data} + T_{data} + SIFS + ACK,$ where T_{data} represents the payload time and H_{data} represents the header (physical and MAC) of the data frame. We have designed so that the sizes of PCTS and FCTS are the same.

¹There are some adaptive schemes like the one in Ref. [6]. However, the transmission still needs to wait for the access time to end even if the intended access requirement has already been satisfied earlier, and the AP has to inform stations about the change of the super-frame parameters every time when the change happens, and that will also lead to protocol burden.

Any station who wants to send data first behaves as conventional 802.11 DCF to send an RTS to the AP (Including carrier sensing, waiting for DIFS idle time, and choosing a random back-off time). However, the detail of the RTS should be slightly modified, the NAV is set equal to $SIFS + CTS$ in order to protect the RTC/CTS exchange and avoid silence of other stations who want to compete for the channel after this time RTS/CTS exchange. The information of the data amount should be included in the RTS, but since the data transmission period is of constant length, the station should control the number of packets merged in the data frame according to the transmission period and its transmission rate.

Upon receiving the RTS, the AP can first get the channel state information (CSI) of the station. Then the AP should decide which kind of CTS to reply. If the number of received RTSs is less than M_{random} and T_{timeout} is not over, the PCTS will be replied. With PCTS received, user station can get the training sequence used for the data transmission, and apply frequency synchronization with the AP, but the station has to wait, and further RTS/CTS exchange may occur.

If the AP has already collected M_{random} RTSs (shown in Fig. 1) or T_{timeout} is over from the first PCTS sending timestamp (shown in Fig. 2, note the start point of T_{timeout}), the FCTS will be sent. Before sending the FCTS, the AP can make certain decisions according to the CSI collected:

- Which station can transmit during the coming transmission period.
- Which transmit rate to use for each station.

The selection algorithm may vary according to what performance goal the system is designed to achieve, which is open to specific considerations. *Actually, it is an interface for realizing cross-layer access control*. Hence, our protocol provides such extensibility here. For example, throughput maximization or fairness consideration can be taken into account in the user selection algorithm. In FCTS, the information that indicates which stations to transmit and the transmit rate to use for each selected station is contained.

When the stations receive FCTS, it will perform data transmission immediately after one SIFS. Since all the stations can hear the CTS, the stations waiting (those who transmit RTS and receive PCTS beforehand) can also start data transmission at the same time. Finally, after the data transmission, the AP broadcasts the ACK immediately after one SIFS following the data transmission.

Comments: For those stations that have not sent RTS, since they can still hear the PCTS, from the remaining T_{timeout} value included in the PCTS, they can behave according to it in order to protect the FCTS sent by the AP in case of timeout.

C. Practical issues

1) Compatibility: In our protocol, the modifications to the standard 802.11 MAC are slight. For the user stations, they only need to distinguish the two kinds of CTS, and then decide to transmit or to wait. The duty of the AP is more complex. However, since the AP has stronger manage ability and the number of APs is much less than that of user stations, the modification to the AP is worthwhile.

2) Frequency Offsets in OFDM: Since the physical layer is MIMO-OFDM, the sensitivity to Carrier Frequency Offsets (CFO) of OFDM should be taken into account. Fortunately, after multi-user detection, the CFO impacts of different users have slight interaction [14], so it is just like the CFO problem in single user OFDM systems, and the solution can be found in conventional 802.11a WLANs with OFDM physical layer.

3) Imperfect CSI: There are two aspects of imperfect CSI: channel estimation error and outdated CSI due to time varying channel. During RTS/CTS exchanges, multiple antennas are used to enhance diversity, slight channel estimation error does not affect performance too much. In the data transmission period, since we consider uplink, with pre-allocated training symbols, the performance is close to that of single user MIMO multiplexing transmission, which has been widely studied and used in the new 802.11n WLAN. Since the indoor wireless channel is stable enough over fairly long time (longer than 100 ms [10]), the problem of outdated CSI has slight effect.

III. THROUGHPUT ANALYSIS AND PARAMETER ADJUSTMENT

In this section, we first analyze the saturated throughput performance of the proposed protocol, and the optimal number of maximal parallel transmissions M^{opt} regarding channel quality is derived. For unsaturated traffic, we propose a dynamic parameter adjustment to enhance system efficiency.

A. Saturated Throughput Analysis

The data rate of each simultaneous transmission is set according to the post-Multiuser-Detection SNR (post-MD SNR) as shown in Table I. When we have M users transmitting simultaneously, and the AP is equipped with A antennas $(A > M)$, the post-MD SNR is calculated as (for MMSE) detector) [15]:

$$
\eta_k = \frac{1}{[(\rho H^H H + I)^{-1}]_{k,k}} - 1,\tag{1}
$$

where η_k represents the post-MD SNR of the kth user (among the M simultaneous users), ρ represents the average SNR, H is the channel matrix, which is an $A \times M$ complex Gaussian distributed random matrix. Then the rate of the kth user can be determined by checking Table I, and we denote this process as $R_k = \Gamma(\eta_k)$. Hence the sum-rate is $R = \sum_{k=1}^{M} R_k$. In order to get the system throughput, we need the average sumrate, denoted as $\overline{R}(M,\rho) := E(R) = M \cdot E(R_k)$, where we emphasize that the average sum-rate is a function of both M and average SNR ρ , and the last equivalence holds because of the assumption that the channel statistics of different users are the same. The dependence of $\overline{R}(M, \rho)$ on ρ and M actually reflects the diversity-multiplexing tradeoff as shown later.

We are now ready to calculate the saturated system throughput, and we assume that there are no hidden terminals and capture effect. The throughput is then approximated as

$$
S \approx \frac{T_{\text{data}}\overline{R}(M,\rho)}{T_s + M \cdot T_m} = \frac{T_{\text{data}}M \cdot E(R_k)}{T_s + M \cdot T_m},\tag{2}
$$

TABLE I

RATE ADAPTATION TABLE

post-MD SNR (dB)	Data Rate	post-MD SNR (dB)	Data Rate
> 24.56	54 Mbps	>10.79	18 Mbps
> 24.05	48 Mbps	> 9.03	12 Mbps
> 18.80	36 Mbps	> 7.78	9 Mbps
>17.04	24 Mbps	other	6 Mbps

where we have

$$
\begin{cases}\nT_s = \text{SIFS} + T_{\text{data}} + H_{\text{data}} + \text{SIFS} + \text{ACK}, \\
T_m = \alpha + \text{DIFS} + \text{RTS} + \text{SIFS} + \text{CTS},\n\end{cases}
$$

where RTS, CTS and ACK are the time length of these packets including physical header. T_{data} and H_{data} are the payload duration and the data frame header respectively. Parameter α is a function of total number of stations n and the contention window size CW_{min} and CW_{max} [13]. The value of α depends on every RTS/CTS exchange, which can be considered independent with the assumption of moderate number of user stations, so α can be calculated using the model in Ref. [13]. Because RTS/CTS access mode leads to small collision cost and CW_{max} is very large, the transmission probability is mainly decided by CW_{min} . We further approximate α as $\alpha \approx \xi \cdot CW_{\text{min}}$, where ξ is a constant given the number of stations, and as a simplest approximation, we can set $\xi = 0.5$. We now have T_{data} , T_s and T_m as constants despite the change in average SNR and M . From (2) we can see that for a given $E(R_k)$, larger M leads to higher throughput. Moreover, given ρ, since the diversity order of each stream is $A - M + 1$ [15], $E(R_k)$ decreases as M increases, but the decreasing trend is weakened with the increasing of SNR, and with adaptive modulation, when SNR is very high, $E(R_k)$ is almost unrelated to M. The dependence of S on ρ and M is a tradeoff due to two factors: one is the diversity-multiplexing tradeoff, and the other is the different protocol overhead with various values of M. As a result, given ρ , there exists an integer M^{opt} ranging from 1 to 4 that can provide the largest throughput. We first define SNR boundaries $\{\rho_i\}_{i=0}^4$, with $\rho_0 = 0$ and $\rho_4 = +\infty$, the M^{opt} is decided as

$$
M^{\text{opt}} = i, \quad \text{if } \rho \in [\rho_{i-1}, \rho_i) \tag{3}
$$

where ρ_1 , ρ_2 and ρ_3 can be decided by numerical search that satisfies

$$
\frac{T_{\text{data}}R(i,\rho_i)}{T_s + i \cdot T_m} = \frac{T_{\text{data}}R(i+1,\rho_i)}{T_s + (i+1) \cdot T_m}. \quad i = 1,2,3 \quad (4)
$$

In order to calculate $\overline{R}(i,\rho)$, we need the distribution of η_k . However, the exact distribution of the MMSE post-MD SNR is not well studied, so we can use ZF detector to approximate MMSE detector². After ZF detector, η_k is Gamma distributed with diversity order $A - M + 1$ [15]. With this distribution, which is identical to nakagami-m distribution with $m = A M+1$ and mean ρA , we can use the model in [12] to calculate the average rate $E(R_k)$ of user k, then the relation $\overline{R}(i,\rho) =$ $i \cdot E(R_k)$ is used to get $\overline{R}(i,\rho)$.

With the above analysis, the maximal number of the parallel transmissions can be set to M^{opt} according to the periodically selected average SNR ρ . When the traffic load is saturated and the number of user stations is more than M^{opt} , M_{random} can be set to M^{opt} if no scheduling scheme is specified (i.e., all the transmission requests are accepted). For the unsaturated traffic case or when the number of stations is smaller than M^{opt} , it is necessary to adopt dynamic parameter adjustment as described in the next sub-section.

B. Dynamic parameter adjustment

Once M^{opt} is decided according to the channel condition, the choices of M_{random} and T_{timeout} further depend on the number of user stations and the traffic load, both of which can be represented by the number of *active stations* defined as the user stations having packets to send at the same time. For example, if there are only 2 active stations having packets to send, while at the AP side M_{random} is set to 3, then for every transmission procedure there will be a timeout, which leads to unnecessary protocol cost. It is obvious to see that at this time, M_{random} should be 2. So the goal of parameter adjustment is to enhance system efficiency, and the duty of parameter adjustment is to dynamically choose appropriate value of M_{random} and T_{timeout} . If M_{random} is suitable for the current system situation, timeout will rarely happen, so the value of T_{timeout} is not very important as long as it can act like a warning message to help adjust M_{random} . We set T_{timeout} related to M_{random} as

$$
T_{\text{timeout}} = M_{\text{random}}(\text{RTS} + 2 \cdot \text{SIFS} + \text{CTS}) + (M_{\text{random}} - 1)(\text{DIFS} + 4 \cdot CW_{\text{min}} \cdot \text{Slottime}).
$$

The AP can record following information of each transmission period in the memory: number of users accessed, the user MAC addresses, and whether there happens a time out. The length of the buffer can be defined according to experiences. For example, in the simulations described later, we set the buffer length to 30, which means the buffer can record information of 30 *recent transmission procedures*. The AP checks the buffer at the end of each transmission, then makes the following adjustment to M_{random} if necessary:

- a) *Decrease* M_{random} . If over 70 percent of the transmission records are timeout, and M_{random} is more than 1, AP will decrease M_{random} to $M_{\text{random}} - 1$, and then clear the buffer.
- b) *Increase* M_{random} . We assume the number of all single transmissions (each one of the parallel simultaneous transmissions) recorded in the buffer is N_s , among which the records number of the *i*th user station is N_i , and the total number of users recorded (the users who have at least one single transmission recorded in the buffer) is U , if the following criterion is satisfied:

$$
\sum_{i} \text{sgn}(\frac{N_i}{N_s} - \frac{0.5}{U}) > M_{\text{random}},
$$

²As long as the SNR is not too low, the approximation is acceptable.

TABLE II

SIMULATION PARAMETERS

Parameter	Value	Parameter	Value
SIFS	$10 \mu s$	Data Rate	Depends on SNR
RTS	160 bits	Basic Rate	6 Mbps
DIFS	$28 \mu s$	Slot Time	$9 \mu s$
CTS	132 bits	PHY Header	$32 \mu s$
ACK	114 bits	$T_{\rm data}$	889 μ s (1000bytes/9Mbps)
\overline{CW}_{\min}	32 slots	CW_{\max}	$\overline{1024}$ slots

where function $sgn(x)$ is defined as:

$$
sgn(x) = \begin{cases} 1 & x > 0 \\ 0 & x \le 0, \end{cases}
$$

and if M_{random} is less than M^{opt} described in the previous sub-section, the AP will increase M_{random} to $M_{\text{random}} + 1$. Then the buffer is cleared.

Comments: We can see that M_{random} is tightly related to the number of *active stations*, which not only reflects the number of the user stations, but also the overall traffic load, because when the traffic load is low, there are few stations having packets to send at the same time.

IV. SIMULATION RESULTS

In this section, simulation results are offered to show the performance gain of our proposed SDMA MAC comparison to the conventional 802.11 MAC and the effectiveness of the parameter adjustment. We use ns-2 as the simulation tool, and the simulation parameters are set as shown in Table II according to the 802.11n drafts [3]. The sizes of CTS and ACK are modified in our protocol, and as mentioned, we have designed so that the sizes of PCTS and FCTS are the same. The rate adaptation parameters are set as shown in Table I.

We will present three sets of simulations. The first two are saturated load cases, but the difference between them is whether the channel variation is taken into account or not. The last one is unsaturated case to evaluate our dynamic parameter adaptation scheme. Since user scheduling is not detailed (which is open to the usage of our protocol), all the M_{random} RTS requests will be accepted for transmission. In all the simulation figures, M is short for parameter M_{random} .

A. Saturated load without channel variation

In this set of simulations, the channel variation is turned off, and SNR is set to 60dB, which is so high that almost every simultaneous transmission can be accomplished with top rate: 54Mbps. Fig. 3(a) shows the saturated throughput performance of our proposed protocol. Since it is a saturated scenario, adaptive parameter adjustment is not implemented, and timeout rarely happens. From Fig. 3(a) we can see the network throughput gain is significant due to parallel simultaneous transmissions with high M_{random} . The throughput is not sensitive to the number of users. However, the performance degradation of large number of users is more apparent with high value of M_{random} , because high M_{random} requires more RTS/CTS exchanges, which is more easily affected by the frequently happened collisions when the number of users is large.

B. Saturated load with channel variation

In this set of simulations, we assume Rayleigh fading between each of the transmit-receive antenna pairs, and the number of user stations is set to be 8. The saturated throughput is shown in Fig. 3(b), from which we can see that when average SNR is low, SDMA scheme is worse than standard 802.11, because with less simultaneous users, we can benefit more from diversity gain under low average SNR scenario and light protocol overhead. When average SNR increases, the advantage of our SDMA scheme becomes more significant, since the fading will not affect the data rate so much with high average SNR, and the increasing protocol overhead is compensated by high multiplexing gain. The result matches the throughput analysis in Section III, and the corresponding boundaries ρ_1 , ρ_2 and ρ_3 are shown in the figure, from which M^{opt} can also be easily inferred.

The network throughput is limited by the highest data rate: 54Mbps, since we do not have the SNR table for higher transmit rates in 802.11n. However, from Fig. 3(b), we can still predict that the throughput will be higher if we allow higher data rate when SNR is high, and even though average SNR is not so high, there will still be possibility that high rate being used because of the randomness of fading. Thus, our protocol can achieve network throughput larger than 100Mbps if implemented in 802.11n systems.

Because the data frame duration is fixed while the transmission rate is changing, the MAC layer packet duration is not a constant value, and it is not easy to directly measure the packet delay under this scenario. We make the following assumption in order to measure the packet transmission delay: the packet length is fixed to be 1000 bytes, and each station aggregates or separates packets according to the transmit rate. For example, if the channel is so bad that it cannot allow a 1000-byte-packet completed within a single frame transmission, the station can divide the packet into parts according to the transmit rate, and the delay is the time between the send time of the first part and the receive time of the last part; if the channel is good enough to allow multi-packets to be transmitted, packet aggregation takes place.

Fig. 3(c) shows the delay performance of our protocol, and we can see that no matter how the channel condition is, with high M_{random} , the delay performance is always better. Moreover, delay performance is not sensitive to the channel condition. Though it sounds strange at first glance, if we note that the delay is actually the delay of *successfully* transmitted packets, the result makes sense. At low SNR, the cost is the increase of packet loss, which is equal to the throughput gap between low SNR and high SNR in the saturated scenario.

C. Unsaturated load with parameter adjustment

In this set of simulations, the dynamic parameter adjustment scheme is tested. Different from the previous simulations, the traffic is not saturated and the average packet load of

(a) Throughput (Without channel variation)

Fig. 3. Simulation results with saturated traffic

Fig. 4. Unsaturated throughput (With parameter adjustment)

each station is changing with time. We assume no channel variation in this simulation set and M^{opt} is 4. Then our dynamic parameter adjustment scheme is only related to *active stations*, which represent those having packets to send. In the simulation, with totally four users, we turn on or turn off the CBR sources (with source rate as 40Mbps and random packet arrival) of each user at different time in order to change the number of active users. The result is shown in Fig. 4. In this figure, the dashed line represents the throughput when parameter adjustment is not implemented and $M_{\rm random}$ is set to be 3. It is obvious that parameter adjustment scheme can track the number of "active stations", and outperforms much more than the non-adaptive scheme does.

V. CONCLUSION

In this paper, we have proposed a contention based uplink medium access control protocol for wireless LANs with SDMA capability. High network throughput has been achieved by allowing multiple parallel uplink data transmissions. The scheme has been designed to be pure contention based in nature, and be suitable for all MIMO configurations in WLANs. The proposed protocol can adapt itself according to the channel condition and the load change of the network by dynamic

parameter adaptation, which makes the protocol more flexible. Extensibility is another feature of our protocol for it provides interface to realize user scheduling. Our work slightly changes the standard 802.11 MAC, especially on the mobile station side, so it can be easily implemented in existing systems.

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