A Novel Channel-Adaptive Uplink Access Control Protocol for Nomadic Computing

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Abstract—We consider the uplink access control problem in a mobile nomadic computing system, which is based on a cellular phone network in that a user can use the mobile device to transmit voice or file data. This resource management problem is important because an efficient solution to uplink access control is critical for supporting a large user population with a reasonable level of quality of service (QoS). While there are a number of recently proposed protocols for uplink access control, these protocols possess a common drawback in that they do not adapt well to the burst error properties, which are inevitable in using wireless communication channels. In this paper, we propose a novel TDMA-based uplink access protocol, which employs a channel state dependent allocation strategy. Our protocol is motivated by two observations: 1) when channel state is bad, the throughput is low due to the large amount of FEC (forward error correction) or excessive ARQ (automatic repeated request) that is needed and 2) because of item 1, much of the mobile device's energy is wasted. The proposed protocol works closely with the underlying physical layer in that, through observing the channel state information (CSI) of each mobile device, the MAC protocol first segregates a set of users with good CSI from requests gathered in the request contention phase of an uplink frame. The protocol then judiciously allocates channel bandwidth to contending users based on their channel conditions. Simulation results indicate that the proposed protocol considerably outperforms five state-of-the-art protocols in terms of packet loss, delay, and throughput.

Index Terms—Mobile computing, distributed data access, wireless systems, adaptive protocols, error control.

1 Introduction

Tetherless nomadic computing environments are envisioned to proliferate in the near future due to the unprecedented advancements in hardware and wireless communication technologies. The use of wireless communication links allows users to continue computing while on the move. However, to realize a truly user-friendly mobile computing system entails tackling a number of intricate resource management and coordination chores such as efficient and reliable data dissemination, maintenance of data consistency, queries management, etc. [3]. The difficulties encountered in handling these tasks are aggravated by the fact that they have to be dealt with under a whole new set of system constraints like low data rates, high error rates, minimal power supply in the mobile devices, etc. [6], [20]. Indeed, as it is widely believed that future mobile computing networks will be based on the ubiquitous cellular phone networks such that users are expected to use a cell phone like device for data transmission as well, resource management schemes must take into account the burst error characteristics and low power in the user's handheld devices. While the problems involved in downlink data broadcasting are extensively studied by numerous researchers [5], [16], [17], [18], the uplink access control problem has received much less attention. Efficient solutions to uplink access control are critical for supporting a large user population with a reasonable level of quality of service (QoS). The abundant bandwidth available in the forthcoming third generation (3G) wireless systems does not help to alleviate the problem because the user population size will for sure increase proportionately. In this paper, we propose a novel TDMA-based uplink access protocol which judiciously allocates channel bandwidth to contending users based on their channel conditions. Our channel state dependent allocation is motivated by two observations: 1) when channel state is bad, the throughput is low due to a large amount of FEC (forward error correction) or excessive ARQ (automatic repeated request) is needed and 2) because of item 1, much of the mobile device's energy will be wasted.

The classical design of an uplink access protocol is to arbitrate and statistically multiplex the transmission requests of multiple uncoordinated users and allocate transmission bandwidth to the users in a fair manner. Well-known examples include the ALOHA protocol for a packet radio network and the CSMA/CD protocol for a wired local area network. The key feature of the classic design is that all users are homogeneous—they have the same traffic characteristics. However, in our study, we consider a cellular mobile information system for integrated voice (i.e., isochronous traffic) and file data services[1] (i.e., bursty traffic), for which an effective and intelligent protocol is particularly desired due to the sharing of the precious bandwidth by a dynamically changing population of users with various traffic demands. There are four aspects in characterizing an uplink access control protocol:

- *Request Mechanism*. The mechanism of receiving user requests critically affects the performance of a

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1. We refer to such services as data services hereafter for simplicity.

protocol. For example, in some contention based protocols, too much contention (e.g., a large number of active users) will result in system instability such that users keep on contending without success due to excessive collisions. Under such a *thrashing* situation, most of the information slots are not used. Different protocols employ various techniques to combat this problem.

- *Slots Allocation*. In most protocols, information slots are assigned on a first-come-first-basis and can be reserved in subsequent frames if the user submits a voice request. However, some recently proposed protocols employ more intelligent approaches to further enhance the channel utilization.
- *Frame Structure*. Traditionally, the frame is of a static structure. That is, for example, there is a fixed portion of the frame dedicated for receiving transmission requests, while the remaining portion is for information slots. A major merit of a static frame structure is the ease of implementation and is energy efficient for the mobile devices, which do not need to listen to the channel all the time. Some other protocols, however, employ a dynamic frame structure, with the objective to utilize the bandwidth more efficiently.
- *Performance*. The capacity of the network and QoS depend critically on the performance of the protocol in terms of packet dropping rate, delay, throughput, and utilization.

A scrutiny of the above four aspects reveals a large design space for uplink access control protocols, despite that not many such protocols are proposed for nomadic computing environments. Nevertheless, five recently suggested protocols, based on radically different philosophies, are selected for an extensive performance comparison with our proposed protocol. The protocols chosen are:

- RAMA [2] (resource auction multiple access), a protocol that employs a collision avoidance approach.
- RMAV [12] (reservation-based multiple access with variable frame), a protocol with a dynamic frame with variable length, designed to achieve a short delay at light load and high throughput at high load.
- DRMA [19] (dynamic reservation multiple access), a protocol with a dynamic frame structure in which the portion of bandwidth designated for user requests is dynamically adjusted, designed to maintain system stability at high load.
- D-TDMA/FR [19] a traditional dynamic TDMA protocol with a static frame structure.
- D-TDMA/VR [14] a dynamic TDMA protocol based on a channel-adaptive variable-throughput physical layer.

In general, these previous protocols attempt to accommodate more file data transmission requests, which do not impose constraints on data delay, by exploiting the silence gaps of the voice requests, which require bounded-delay packet transmission and, hence, enjoy a higher transmission priority than data users in that reservation is allowed for the former but not the latter. However, while sophisticated slot assignment strategies with articulated frame structures are proposed in these methods, none of them considers the effect of burst channel errors on protocol performance, let alone the investigation of exploiting or adapting the protocol mechanism to the error characteristics to enhance performance. Essentially, these previous protocols are designed and analyzed based on the assumption that packet transmission through the wireless channel is error-free. However, because the geographically scattered mobile devices inevitably suffer from different degrees of fading and shadowing effects, indeed a common drawback of previous protocols is that they assume the underlying physical layer always delivers a constant throughput and, as such, they may not be able to effectively utilize the precious bandwidth when the channel condition is swiftly varying among different users.

Our proposed uplink access control protocol, called CHARISMA (CHannel Adaptive Reservation-based ISochronous Multiple Access), is based on a channel adaptive physical layer. As will be detailed in Section 4, CHARISMA is a dynamic TDMA (D-TDMA)-based protocol with one major distinctive feature: The user contention requests are gathered by the base station in the first phase of the time frame without immediately announcing the information slots assignment right after each contention minislot. After all requests are received, the information slots are assigned to the users based on their respective CSI ranking. Performance gain is derived by supplying an additional input to the protocol, namely, the channel state information (CSI) from the physical layer.

Link adaptation is not a new concept and many pioneering researchers have explored the idea of channel adaptive transmission. Indeed, the most notable example is the work by Bhagwat et al. [4], in which a downlink channel adaptive approach is proposed for enhancing wireless TCP performance. However, previous work usually just discusses the adaptive scheduling on the downlink direction (which have a totally different design constraint as the uplink direction which involves the contention resolution issue). Furthermore, the channel estimation model in previous work is usually at a more coarse grain time resolution (e.g., observing the ARQ performance in the system suggested by Bhagwat et al. [4]). Gathering CSI at the physical layer and making adaptive resource allocation decisions based on it are relatively less explored. On the other hand, the concepts of link adaptive resource allocation have also appeared in GPRS, EDGE, and HDR wireless standards [13], [8]. However, they are designed for separate voice and packet data paths. Having integrated voice and packet data in a wireless network with link adaptation (GERAN Release 2000 [7]) it is still a host research topic, as well as a major goal in standardization efforts.

We have implemented all six protocols on a common simulation platform, from which extensive performance results are obtained. The protocols are evaluated for test cases with and without request queues, which store transmission requests that survive the contention but fail to get assigned information slots. Three performance metrics, namely, voice packet dropping rate, data delay, and data throughput, are considered. The paper is

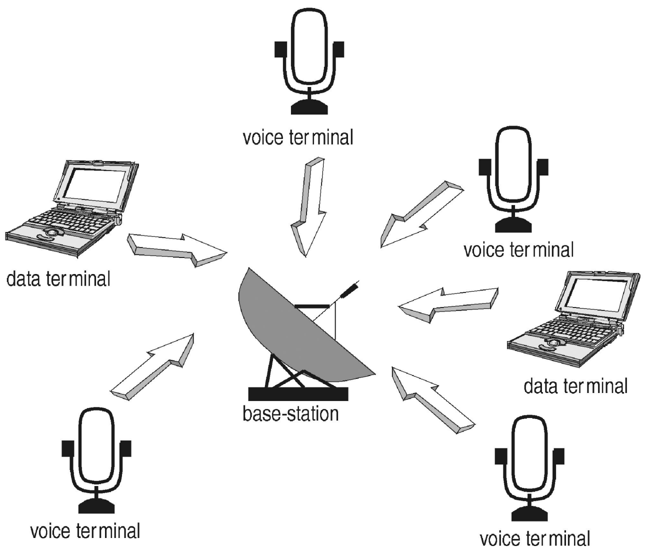


Fig. 1. System model for the uplink access of a wireless system with voice and file data requests.

organized as follows: In the next section, we describe the models we used. Some state-of-the-art protocols are then described in detail in Section 3. In Section 4, we describe in detail the functionality and characteristics of the proposed CHARISMA protocol. Section 5 contains the performance results and our interpretations. The final section provides some concluding remarks.

2 Models

The cellular mobile information system considered in this paper, with a star topology as shown in Fig. 1 (i.e., a cell phone network with file data transmission support), is aimed to support integrated voice and data services. As such, we assume that there are only two types of requests from the mobile devices, namely, the voice requests and the data requests in the system. Voice packets are assumed to be delay sensitive while data packets are assumed to be delay insensitive. Thus, voice packets are labeled with *deadlines*. A voice packet will be dropped by the mobile device if the deadline expires before being transmitted. Such packet dropping has to be controlled to within a certain limit (e.g., below 1 percent as indicated in [10]) in order that the quality of service to the voice users is still acceptable. The source and contention models are summarized below.

- **Voice Source Model.** The voice source is assumed to be continuously toggling between the talkspurt and silence states. The duration of a talkspurt and a silence period are assumed to be exponentially distributed with means t_t and t_s seconds, respectively (as indicated by the empirical study in [10], $t_t = 1$, and $t_s = 1.35$). We assume a talkspurt and a silence period start only at a frame boundary.
- **Data Source Model.** The arrival time of file data generated by a mobile device is assumed to be exponentially distributed with a mean equal to one second. The data size, in terms of number of packets, is also assumed to be exponentially distributed with mean equal to 100 packets. Again, we assume that the packets arrive at a frame boundary.
- **Request Contention Model.** As in most previous studies, to avoid excessive collisions, even if a voice or data request has some packets awaiting to be sent, the mobile device will attempt to send a request at a request minislot only with a certain *permission probability*. The permission probability for submitting voice and data requests are denoted by p_v and p_d, respectively.

A mobile device entering a new voice talkspurt or generating a new stream of data packets transmits an appropriate request packet in one of the request slots of the next frame. If there are more than one packet transmitted in the same request slot, collision occurs and none of the requests will be correctly received (we ignore the capture effect [1] in this paper). At the end of each request slot, the successful or unsuccessful request will be identified and broadcast by the base station. An unsuccessful mobile device (does not receive the acknowledgment announcement in the downlink frame) can retry in the next request slot. On the other hand, a successful mobile device then transmits his/her information packet in the corresponding information slot in the current frame.

3 Descriptions of State-of-the-Art Protocols

Five recent protocols are selected for our quantitative comparison, with the objective to highlight the merits and demerits of different protocol designs. The protocols selected are: RAMA [2], RMAV [12], DRMA [19], D-TDMA/FR [19], and D-TDMA/VR [14]. For completeness, these protocols are outlined below. The reader is referred to the respective references for more detailed information.

3.1 RAMA

The frame structure of the RAMA protocol [2] is composed of two parts: the auction subframe with N_a auction slots and the information subframe with N_i information slots, as shown in Fig. 2a. In each slot of the auction subframe, a bidding process is implemented with the objective of avoiding collisions. Specifically, the auction process is based on the users' IDs, which are randomly generated at the beginning of each slot. The number of digits in each ID is critical and depends on the number of users in the system. Essentially, the number of digits should be large enough such that the probability of two different users generate the same ID be very small. However, the length of the ID cannot be too large because each digit is transmitted using a distinct frequency within the channel bandwidth (as shown in [2], a set of orthogonal frequencies is used). Thus, there is a tradeoff in selecting the ID length. At the beginning of an auction slot, each contending user send, the most significant digit (MSD) of its ID to the base station. After collecting all the digits in the first auction slot, the base station broadcasts the largest digit in a small announcement minislot in the downlink frame. Users with a smaller digit will then drop out of the contending (auction) process. This process is repeated until all the digits are sent to the base

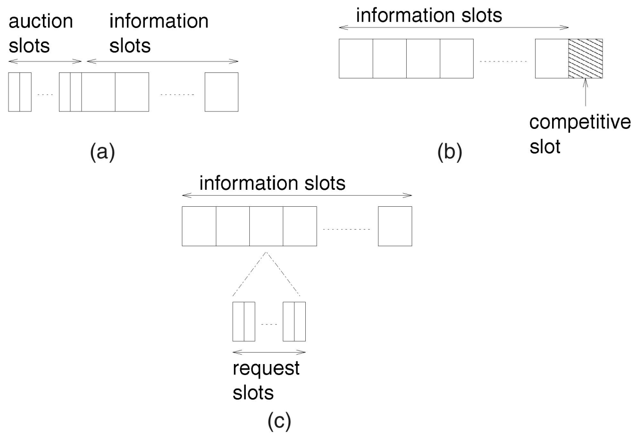


Fig. 2. Frame structures of the (a) RAMA protocol, (b) RMAV protocol, and (c) DRMA protocol.

station and eventually there will be one winner who then gets assigned an information slot in the current frame. If the winner is a voice user, the assigned slot can be used in subsequent frames also (i.e., reservation) without further contention. On the other hand, data users have to contend again. Furthermore, the user ID of a data user is always smaller than that of a voice user so as to give a higher priority to the latter. It can happen that two users send the same digit to the base station in the auction process. But, such a "collision" will very likely (depending on the size of the ID space) be resolved in latter digits. Note that the transmission request process is not deterministic because the users' IDs are randomly generated in each auction slot.

A major merit of the RAMA protocol is that its collision avoidance mechanism can prevent system instability when the traffic load is very high. That is, progress is still maintained and no thrashing will occur. However, the overheads, in terms of channel bandwidth (an auction slot is larger than a normal request slot) and hardware, can be quite significant if the number of users is large.

3.2 RMAV

The RMAV protocol [12] employs a variable frame structure in that the frame length is dynamically adjusted depending on the traffic load. Specifically, as shown in Fig. 2b, there is only one request slot (the last slot), called the competitive slot, in each frame. All other slots are information slots, which are assigned to some users. Thus, a user without any assigned information slot sends a request packet at the competitive slot to contend for one or more information slots (a data user can get more than one information slots bounded above by P_{max}, which is 10 according to [12]) in the next frame. In this manner, the frame length becomes variable because the number of assigned information slots varies from frame to frame. But, the frame length is bounded by nP_{max} if there are n users in the system.

A major merit of the RMAV protocol is that it can achieve very short delay when system load is low and high throughput when system load is high. However, because there is only one slot in each frame for requests contention, system thrashing can occur even when there is only a moderate number of contending users.

3.3 DRMA

The DRMA protocol [19] also employs a variable frame structure in that there is no dedicated frame slots for users to make transmission requests. As shown in Fig. 2c, a frame is composed of N_k information slots only. However, at the beginning of each information slot, the base station broadcasts a small message indicating whether the information slot is assigned or not. If not, the information slot will be "converted" into N_x request slots such that the active users (with information packets to send) will transmit request packets to contend for information slots. Each successful request will get assigned an information slot (if available) in the current frame. As in all other protocols considered in this paper, voice users enjoy a higher priority than data users in that they can reserve information slots in subsequent frames while data users cannot. In the DRMA protocol, users can get a chance to request for information slots only when there are empty information slots, which can be converted into request slots. Thus, when the traffic load is very high, users are given only very little opportunity to contend for information slots and thrashing can therefore be avoided.

A major merit of the DRMA protocol is that similar to the RAMA protocol, it can prevent system instability when the traffic load is high. However, the overheads of implementing the variable frame structure, in terms of channel bandwidth (the base station needs to announce the status of every information slot) and mobile device's power (each user needs to listen to the announcement at each information slot), can also be quite significant.

3.4 D-TDMA/FR

The D-TDMA/FR protocol [19] is among the first improved PRMA type of protocol. The time frame is divided into two parts: N_r request slots and N_i information slots. Specifically, a traditional slots assignment strategy is used in that whenever a request is successfully received in the request phase, information slots, if any, are immediately assigned to the requests. After a voice user has successfully reserved a information time slot in the current frame, the user can use a time slot in each frame every 20 msec until the current talkspurt terminates. A data user will be allocated information slots only if there are remaining information slots in the current frame. The data requests are considered also on a first-come-first-served basis.

3.5 D-TDMA/VR

D-TDMA/VR [14] employs the same access control method as in D-TDMA/FR. However, it differs from D-TDMA/FR in that a *variable-throughput* channel-adaptive physical layer is used. However, although a variable throughput is offered to the access control layer, there is no interaction between the access control layer and the physical layer. In other words, the access control layer is not aware of the current situation in the physical layer in the process of bandwidth allocation. However, with the help of the channel-adaptive physical layer, D-TDMA/VR has twice the average offered throughput compared to D-TDMA/FR.

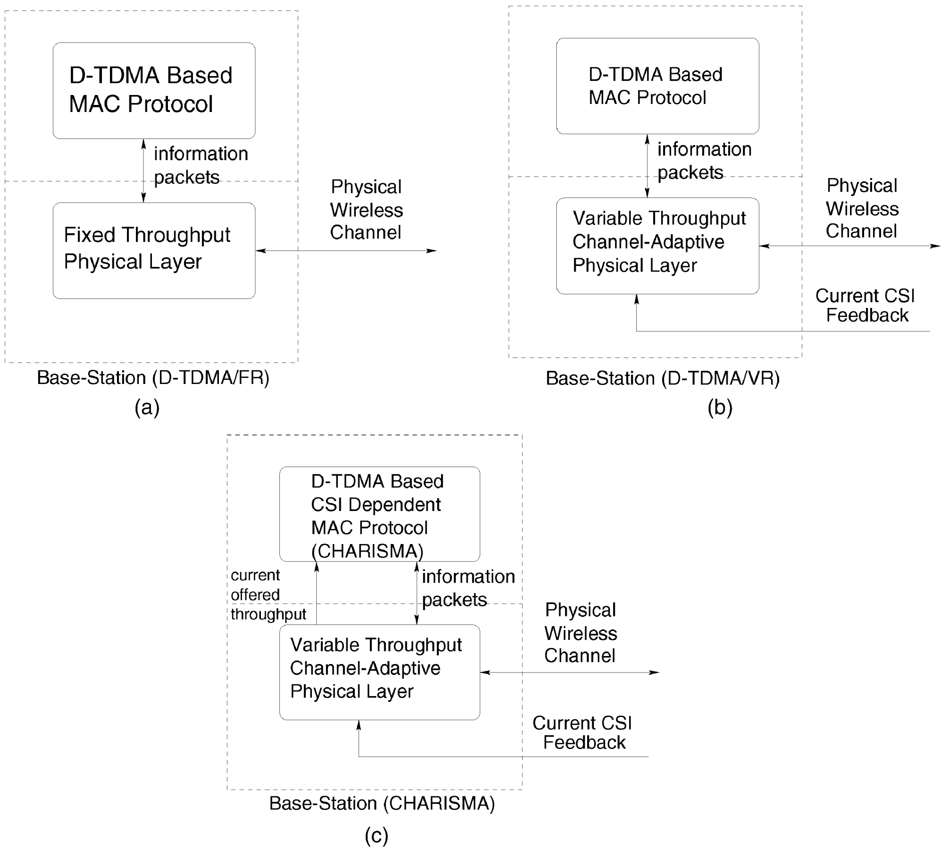


Fig. 3. Conceptual models of the physical and access control layers. (a) D-TDMA/FR (fixed-thoughput channel encoder). (b) D-TDMA/VR (adaptive channel encoder). (c) CHARIMSA (adaptive channel encoder with access control interaction).

4 The Proposed CHARISMA Protocol

The CHARISMA protocol (CHannel Adaptive Reservation-based ISochronous Multiple Access) is based on a novel concept exploiting the synergy between two protocol layers instead of strictly following the traditional information hiding protocol design paradigm. Fig. 3 highlights the differences in the designs among D-TDMA/FR, D-TDMA/VR, and CHARISMA.

4.1 Structures of TDMA Frames

Fig. 4 shows the frame structures for the uplink and the downlink in the CHARISMA protocol. In the uplink, a frame is divided into three subframes as illustrated in Fig. 4a. They are the *request subframe*, *information subframe*, and the *pilot symbol subframe* (note that pilot symbols are known reference symbols [9]). Specifically, there are N_r minislots in the request subframe for voice requests reservation and data requests contention. Again, a data request is not allowed to make reservation in the sense that, even if a data request successfully seizes an opportunity to transmit in the current frame, he/she has to contend again in the next frame if he/she has some more data to send. There are N_i information slots in the information subframe for the transmission of voice or data packets. Finally, there are N_b slots in the pilot symbol subframe. On the other hand, a downlink frame is similarly partitioned into four subframes, namely, the *acknowledgment subframe*, *poll-for-CSI subframe*, *information subframe*, and *announcement subframe*. The number of slots in the subframes are also given by N_r, N_b, N_i, and N_b, respectively. The detailed functions of each subframe will be elaborated later in Section 4.3. The frame duration is 2.5 msec. Such a short frame duration has the advantage of shorter delay and is practicable in wideband systems [21]. Furthermore, the CHARISMA protocol, like other TDMA protocols, operates in a synchronous manner in that the mobile terminals and the base station are aligned by the time frame boundaries (in fact, all TDMA based systems must have the frame boundary synchronized because the time slot boundaries would be common to all mobile users).

Due to the short propagation delay in a cellular network, we assume that the mobile devices can immediately know the request result. This assumption can be justified by the following arguments. First of all, the delay involved from the time when the mobile devices send the request to the base station to the time that the mobile devices realize the result of the contention (by checking the announcement on the downlink) is composed of two parts, namely, the processing delay at the base station and the round trip propagation delay between the mobile devices and the base station. In our study, we focus on high speed wireless access systems, and as such, the propagation delay is on the order of $0.1\mu sec$ (for a 30m separation), which corresponds to less than a minislot. The processing time required at the base station is also rather short because, at the time of sending the acknowledgment to the mobile device, the base station does not need to make any decision on resource

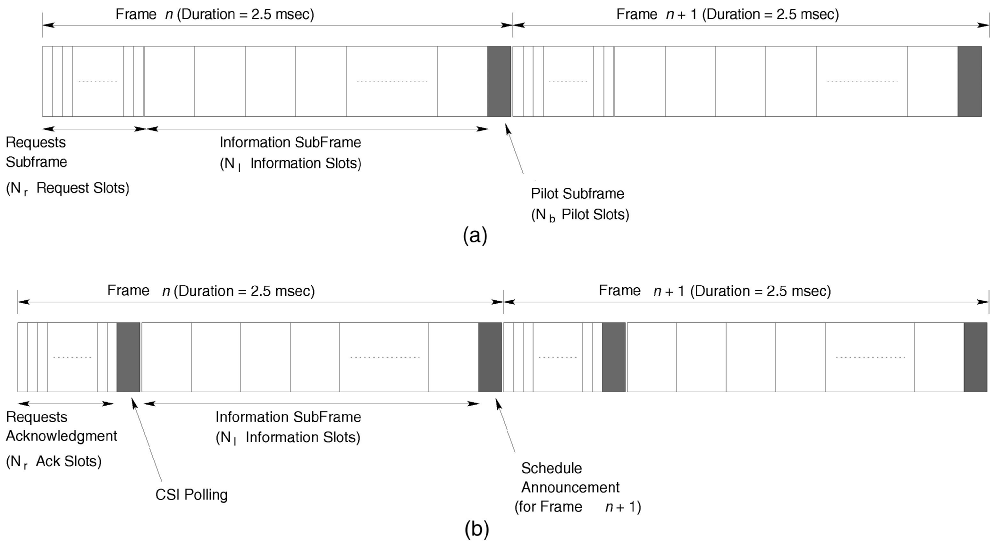


Fig. 4. Frame structures of the proposed CSI-dependent access control protocol. (a) Uplink frame structure. (b) Downlink frame structure.

allocation yet. Thus, the request packet processing delay will be negligible (much shorter than a minislot). Considering the worst case, the delay of the acknowledgment is less than two minislots. Thus, after transmitting the request to the base station, the mobile device simply listens for subsequent downlink acknowledgment minislots (not necessarily the immediately following acknowledgment slot) with a time-out value (which is set to be five minislots in our simulation platform). The mobile device will retry sending another request when the timer expires.

4.2 Variable Throughput Channel-Adaptive Physical Layer

The CHARISMA protocol is designed based on a novel *channel-state conscious* concept such that the burst-error property of the radio channel is exploited to further enhance the multiple access system performance. Specifically, the wireless link between a mobile device and the base station is characterized by two components, namely, the *fast fading* component and the *long-term shadowing* component. Fast fading is caused by the superposition of multipath components and is therefore fluctuating in a very fast manner (on the order of a few msec). Long-term shadowing is caused by terrain configuration or obstacles and is fluctuating only in a relatively much slower manner (on the order of one to two seconds). To illustrate, a sample of measured fading signal is shown in Fig. 5.

Let $c(t)$ be the combined channel fading which is given by:

$$c(t) = c_l(t)c_s(t),$$

where $c_l(t)$ and $c_s(t)$ are the long-term and short-term fading components, respectively. Both $c_s(t)$ and $c_l(t)$ are random processes with a *coherence time* (time separation between two uncorrelated fading samples) on the order of a few milliseconds and seconds, respectively.

Short-Term Fading. Without loss of generality, we assume $\mathcal{E}[c_s^2(t)] = 1$, where $\mathcal{E}[]$ denotes the expected value of a random variable. The probability distribution of $c_s(t)$ follows the Rayleigh distribution which is given by:

$$f_{c_s}(c_s) = c_s \exp\left(\frac{-c_s^2}{2}\right).$$

In this paper, we assume the mean and maximum speeds of the mobile device are 50km/hr and 80km/hr, respectively. Thus, the Doppler spread [23], $f_d \approx 100$Hz. It follows that the coherence time, denoted by T_c, is approximately given by:

$$T_c \approx \frac{1}{f_d}, \tag{1}$$

which is about ten msec.

Long-Term Fading. The long-term fading component, $c_l(t)$, is also referred to as the *local mean* [23], which, as shown by field test measurement, obeys the *log-normal* distribution, $f_{c_l}(c_l)$. That is,

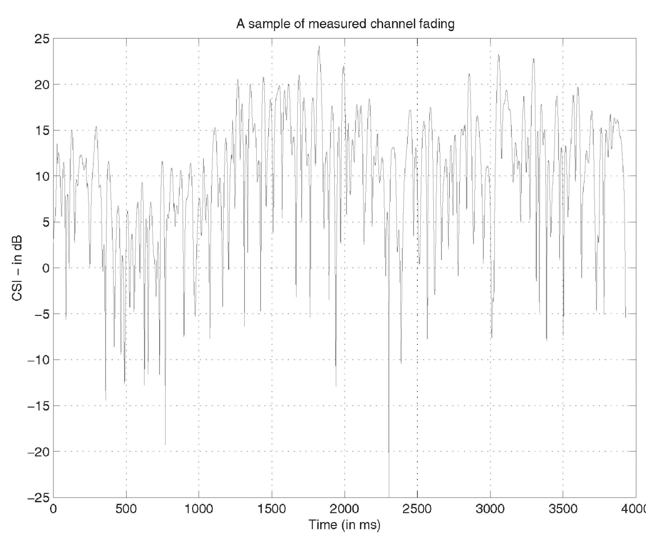


Fig. 5. A sample of channel fading with fast fading superimposed on long-term shadowing.

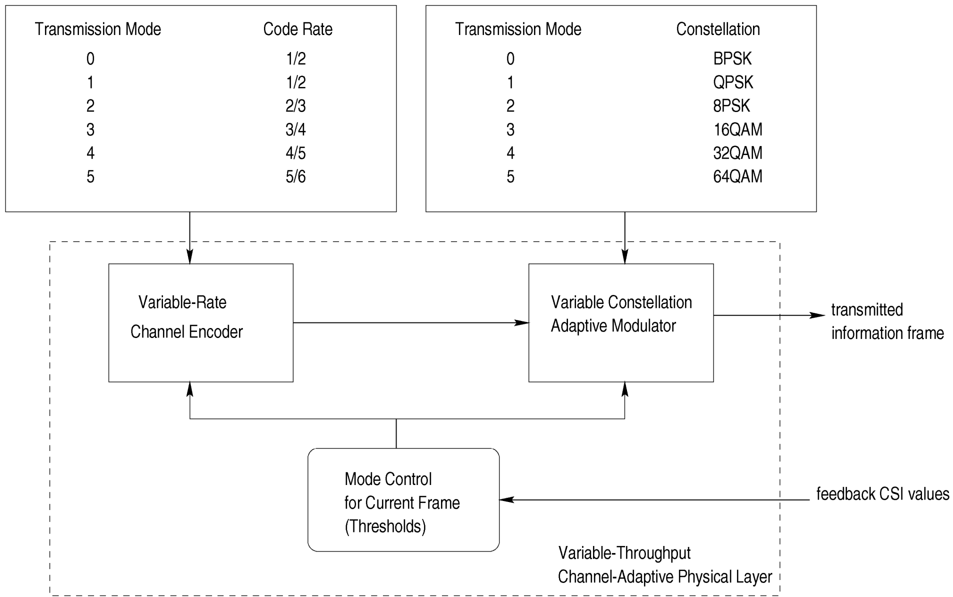


Fig. 6. A conceptual block diagram of the variable-throughput channel-adaptive physical layer.

$$f_{c_l}(c_l) = \frac{4.34}{\sqrt{2\pi}\sigma_l c_l} \exp\left(\frac{-(c_l(dB) - m_l)^2}{2\sigma_l^2}\right),$$

where m_l, σ_l are the mean (in dB) and the variance of the log-normal distribution, i.e., $c_l(dB) = 20\log(c_l)$. Since $c_l(t)$ is caused by terrain configuration and obstacles, the fluctuation is over a much longer time scale. Again, from field test results, the order of time span for $c_l(t)$ is about one second. Since mobile devices are scattered geographically across the cell and are moving independently of each other, we assume the channel fading experienced by each mobile device is independent of each other.

As usual, redundancy is incorporated to the information packet for error protection. To exploit the time-varying nature of the wireless channel, a variable-throughput channel-adaptive physical layer is employed as illustrated in Fig. 6. Channel state information (CSI), $c(t)$, which is estimated at the receiver, is fed back to the transmitter via a low-capacity *feedback channel*. Based on the CSI, the level of redundancy and the modulation constellation applied to the information packets are adjusted accordingly by choosing a suitable transmission mode. Thus, the instantaneous throughput is varied according to the instantaneous channel state. In our study, a 6-mode variable-throughput adaptive bit-interleaved trellis coded modulation scheme (ABICM) is employed [15]. Transmission modes with *normalized throughput*[2] varying from $1/2$ to five are available depending on the channel condition.

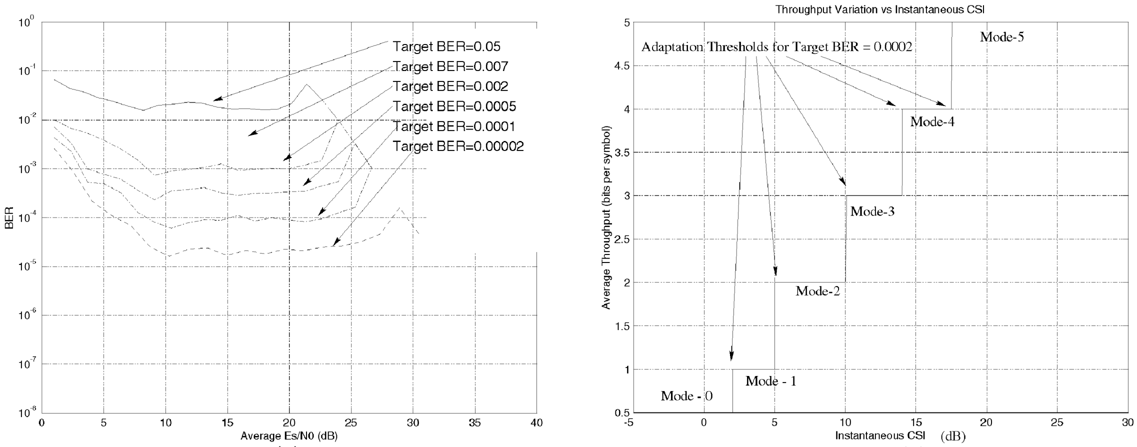
We assume the coherence time of the short-term fading is around ten msec which is much longer than an information slot duration. Thus, the CSI remains approximately constant within a frame and it follows that the transmission mode for the whole frame is determined only by the current CSI level. Specifically, transmission mode q is chosen if the feedback CSI, $\hat{c}$, falls within the *adaptation thresholds*, (ζ_{q-1}, ζ_q). Here, the operation and the performance of the ABICM scheme is determined by the set of adaptation thresholds $\{\zeta_0, \zeta_1, \ldots\}$. In this paper, we assume that the ABICM scheme is operated in the *constant BER* mode [15]. That is, the adaptation thresholds are set optimally to maintain a target transmission error level over a range of CSI values. When the channel condition is good, a higher mode could be used and the system enjoys a higher throughput. On the other hand, when the channel condition is bad, a lower mode is used to maintain the target error level at the expense of a lower transmission throughput. Note that, when the channel state is very bad, the adaptation range of the ABICM scheme can be exceeded such that the throughput (mode-0) becomes so low, making it impossible to maintain the targeted BER level. This adverse situation is illustrated in Fig. 7a.

Given the above considerations about the channel state, the instantaneous throughput offered to the access control layer, denoted by ρ, is also variable and is therefore a function of the CSI, $c(t)$, and the target BER, P_b, denoted by $\rho = f_\rho(c(t), P_b)$. Fig. 7b illustrates the variation of ρ with respect to the CSI.

4.3 Multiple Access Control

The operation of the CHARISMA protocol is divided into two phases, namely, the *request phase* and *transmission phase*. In the request phase, mobile devices which have voice or data packets to transmit will send a request packet in one of the request slots, governed by the respective permission probability. The request packet is very short, occupying only a minislot, as illustrated in Fig. 9a. It contains the mobile device ID, request type (voice or data), packet deadline, number of information packets desired to transmit, as well as some pilot symbols. If more than one mobile device sends request packets in the same request slot, collision occurs and none of the request packets can be successfully received (i.e., capture effect is not considered in this study). After each request slot, an acknowledgment

2. Normalized throughput refers to the number of information bits carried per modulation symbol.



(a) (b)

Fig. 7. BER and throughput of ABICM scheme. (a) Instantaneous BER and the adaption range. (b) Instantaneous throughput versus CSI.

packet will be broadcast from the base station through the acknowledgment slot in the downlink frame as illustrated in Fig. 8a. The acknowledgment packet contains only the successful request packet ID. Mobile devices that fail to receive an acknowledgment will retransmit the request packet in the next request slot, again governed by the permission probability. On the other hand, successfully acknowledged devices will wait for announcement on the allocation schedule from the base station.

Similar to traditional access control protocols, the number of request slots in the CHARISMA protocol, N_r, is slightly larger than the number of information slots, N_i, in order to provide more opportunities for users to contend for information slots. In the CHARISMA protocol, the N_r minislots are used for gathering more mobile device requests as candidates for information time slot allocation. Specifically, the base station first collects all requests in the current request phase as well as the *backlog requests* from the previous frames. All the requests will be assigned priorities which are computed according to the deadline, CSI, service type (voice or data), as well as the waiting time of the request (i.e., the number of elapsed frames since the request is acknowledged). The time slot allocation algorithm is conceptually depicted in Fig. 8b. Since the physical layer offers a variable throughput which is dependent on the CSI, the rationale behind the CHARISMA protocol is to give higher priority to the mobile devices that are in better channel condition in the bandwidth allocation process. The motivation of this strategy is that, a user with better channel condition, with the support of the variable-throughput channel encoder, can enjoy a larger throughput and, therefore, can use the system bandwidth more effectively. Nevertheless, for fairness's sake, information slots should also be allocated to mobile devices that are approaching their deadlines, despite their possibly worse channel states; otherwise, the queued information packets will be dropped. Specifically, the *priority metric* of the *i*th request (which may be a new request or a backlog request), μ_i, is given by the following equation:

$$\mu_i = \begin{cases} g_v(CSI^{(i)},T_d^{(i)})=f_\rho(CSI^{(i)})+\lambda_v(T_d^{(i)})^{-\beta_v}+\Delta V & \text{for voice request} \\ g_d(CSI^{(i)},T_w^{(i)})=f_\rho(CSI^{(i)})+\lambda_d(T_w^{(i)})^{\beta_d} & \text{for data request,} \end{cases} \quad (2)$$

where $T_d^{(i)}$, $T_w^{(i)}$, λ_v, λ_d, β_v, β_d, and ΔV are the deadline, the waiting time, the *forgetting factors* of the voice and data requests, as well as the *priority offset* assigned to the voice requests, respectively. As can be seen from (2), a higher priority will be assigned to requests with a higher throughput (indicated in the first term) as well as requests with a urgent deadline or long waiting time (indicated in the second term). It should be noted that the parameters λ, β, and Δ can be selected to reflect the relative importance of the traffic factors: urgency, channel condition, and traffic type.

With the priority metric defined above, in the allocation phase, information time slots can be allocated to all service requests according to the priority values. If there are not sufficient time slots to service all requests, remaining requests are queued and reconsidered in the next frame. Here, it should be noted that if the deadline for a remaining request has expired, this request will not be queued anymore and the information packet at the mobile device will be dropped. After the request phase, the results of information time slot allocation will be broadcast in the announcement subframe of the downlink frame. The announcement packet contains the time slot allocation schedule as well as the transmission mode as illustrated in Fig. 9b. Mobile devices will then start to transmit information packets on the allocated time slots.

The CHARISMA protocol is reservation-based for voice requests only. As mentioned earlier, for a data request, even if information time slots have been assigned for its successfully acknowledged request, the allocation is meant only for the current frame and the user has to initiate another request if there are still some more data packets to be sent. By contrast, for a voice user, when information time slots have been assigned for its successfully acknowledged request, additional requests will be *automatically generated* by the base station (hence, reservation) periodically at 20 msec time intervals (the voice packet period). Thus, the voice user does not need to contend for request slots anymore for the whole remaining time in the current

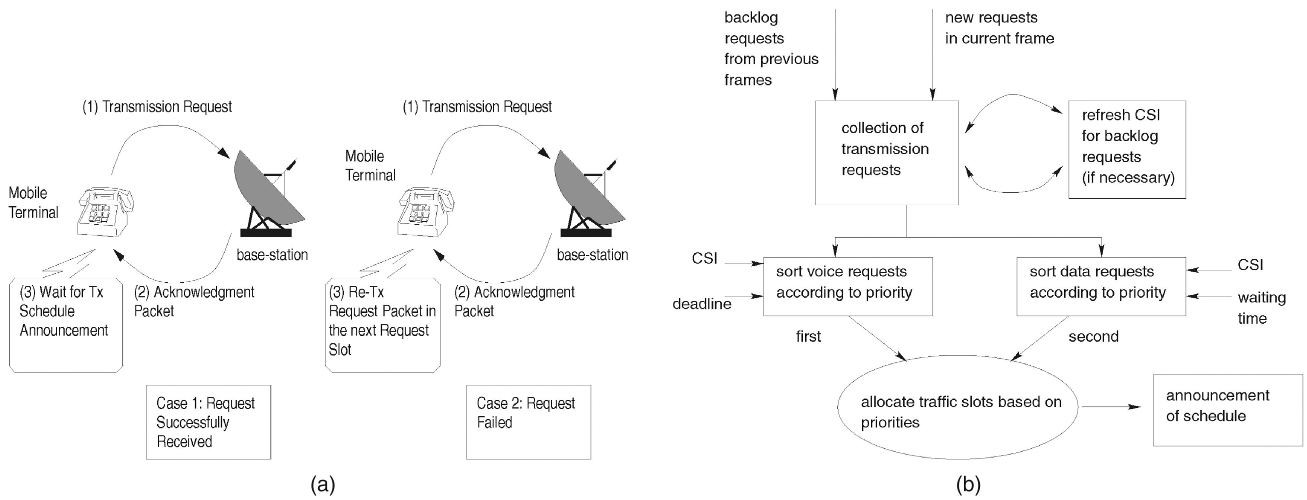


Fig. 8. Operations of the proposed CSI-dependent access control protocol. (a) Operation of request and acknowledgment. (b) Conceptual time-slot allocation.

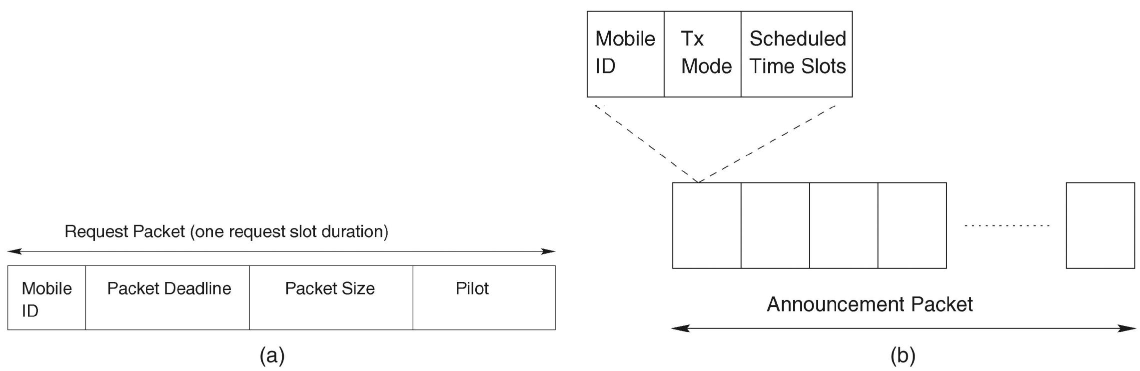


Fig. 9. Formats of the request and announcement packets. (a) Request packet format. (b) Announcement packet format.

talkspurt. By avoiding unnecessary requests, the advantage of this reservation strategy is to reduce the contention collisions by the request slots. A mobile device (voice or data) will recognize a contention failure when it does not receive an acknowledgment announcement in the downlink frame generated by the base station.

4.4 Gathering Channel State Information

A critical component in the proposed CHARISMA protocol is the determination of the current CSI for each user. As mentioned earlier, we assume that the coherence time for short-term fading is around 10 msec as illustrated in Section 4.2 while the frame duration is only 2.5 msec. Thus, the CSI remains approximately constant within at least two frames. For a new request, known pilot symbols are embedded in the request packets so that the CSI can be estimated at the base station (by observing the error pattern in the known symbols) and, most importantly, this estimated CSI is valid even for the next frame duration. However, for a backlog request, the estimated CSI value obtained previously during a past request phase may be obsolete and, thus, a mechanism is needed to obtain update the CSI. The CSI updating procedure used in the CHARISMA protocol is illustrated in Fig. 10a.

At the beginning of each frame, the base station short-lists N_b backlog requests (those with their CSI values expired) according to their priorities. A *CSI polling packet* is then broadcast to the mobile devices in the *CSI-polling subframe*. The CSI polling packet contains the mobile device IDs that are short-listed by the base station. The structure of the polling packet is shown in Fig. 10b. Mobile devices listed in the polling packet transmit known pilot symbols at the appropriate pilot symbol slot according to the order specified in the polling packet. Thus, the base station can obtain the higher priority backlog requests' CSI values, which are valid for two consecutive frames.

4.5 Request Queue

All the above mentioned protocols, except RMAV,[3] can incorporate a *request queue*, which stores at the base station the previous requests that survive the contention but are not allocated information slots. Such a request queue can further alleviate the capacity loss due to requests contention, especially when the traffic load is high. However, including a request queue inevitably increases the implementation complexity of the protocols. Thus, there is a tradeoff. In our simulation study, we examine both the with and without queue versions of each protocol.

3. The RMAV protocol [12] inherently does not require a request queue because there is only one request contention opportunity in each frame so that there is only one winner.

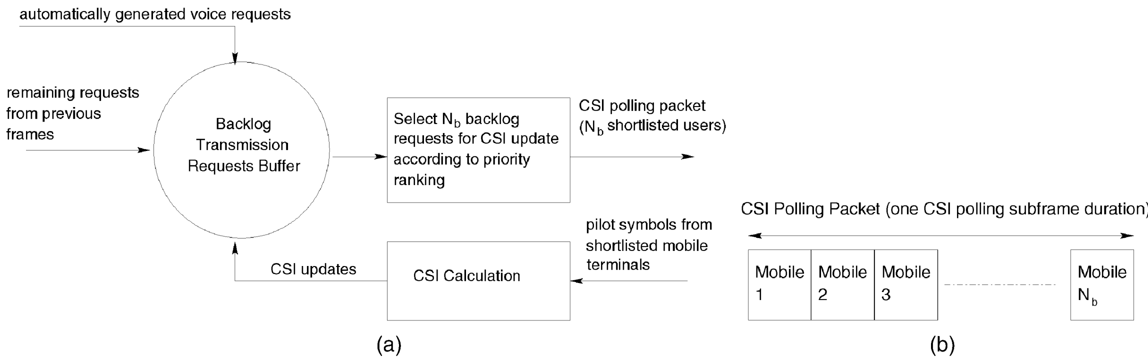


Fig. 10. Illustration of CSI refresh mechanism for backlog requests. (a) CSI updating mechanism for backlog requests. (b) Structure of CSI-polling packet.

5 Performance Results

We have implemented the six protocols on a common simulation platform. In our experiments, we assume a transmission bandwidth of 320 KHz for the TDMA frames. Bit rate of the speech source is 8 Kbps which conforms to the values in GSM and CDMA systems. Table 1 summarizes the parameters we used.

5.1 Performance for Voice Requests

Because the quality of voice communication is determined by the average packet loss rate, we quantify the performance of voice requests by *packet loss rate*, P_{loss}. Note that the packet loss probability is composed of two factors, namely, *packet dropping* and *packet transmission error*. On one hand, voice packet is delay sensitive and, hence, voice packets are labeled with deadlines. A voice packet has to be discarded if its delay exceeds the deadline.[4] Such discarding constitutes the packet dropping at the mobile device. On the other hand, transmitted packet could be corrupted due to channel error and, thus, packet transmission error results. The packet loss rate, P_{loss}, is given by:

$$P_{loss} = \frac{N_{tx} - N_{rv}}{N_{tx}}, \tag{3}$$

where N_{tx} and N_{rv} are the number of transmitted voice packets and the number of voice packets received without error, respectively. The performance of the six protocols in terms of packet loss rate versus the number of active voice users is shown in Fig. 11 for cases without request queue and with request queue, respectively. In the following, we denote the number of data users by N_d.

A glance at Fig. 11 clearly reveals that the CHARISMA protocol outperforms the other five protocols by a considerable margin in terms of voice packet dropping rate, while the relative rankings among the other five are not very consistent in the six test scenarios. In addition, at low traffic, the CHARISMA protocol almost incurs no packet loss but the other five protocols still have a certain level of packet loss. A close scrutiny reveals that such low load losses are due to transmission errors. Another general observation is that the RMAV protocol quickly becomes unstable even with a moderate number of voice users (e.g., 10). This demonstrates clearly that providing just one request contention opportunity can easily lead to instability. Specifically, consider the case without request queue and $N_d = 0$ (see Fig. 11a). At the 1 percent voice packet dropping rate threshold, we can see that CHARISMA can accommodate approximately 100 voice users, while both DRMA and D-TDMA/VR can support only about 80 voice users (the former is slightly better than the latter in this case). Furthermore, the number of voice users supported by both RAMA and D-TDMA/FR is about 60. Thus, for this test scenario, the ranking of the six protocols is: CHARISMA, DRMA, D-TDMA/VR, RAMA, D-TDMA/FR, and RMAV. Next, let us consider the case without request queue and $N_d = 10$ (see Fig. 11c). As can be seen, all the protocols can only accommodate about 80 percent of the number of voice users compared with the case in which $N_d = 0$. However, the protocol ranking is the same as in the previous case. Finally, consider the case without request queue and $N_d = 20$. Again, each protocol supports approximately 20 percent less voice users compared with the case in which

4. In this paper, the deadline of a voice packet is assumed to be 20 msec after it is generated by the source.

TABLE 1
Simulation Parameters

Parameter	Value
N_r	13
N_i	8
N_b	12
N_k	10
N_x	8
N_a (number of auction slots in RAMA)	4
t_t	1000 msec
t_s	1350 msec
p_v	0.3
p_d	0.2
λ_v	4.0
λ_d	0.5
β_v	2.0
β_d	0.2
ΔV	40.0
transmission bandwidth	320 Kbps
speech bit rate	8 Kbps
file data rate	19.2 Kbps
number of frames	2×10^6

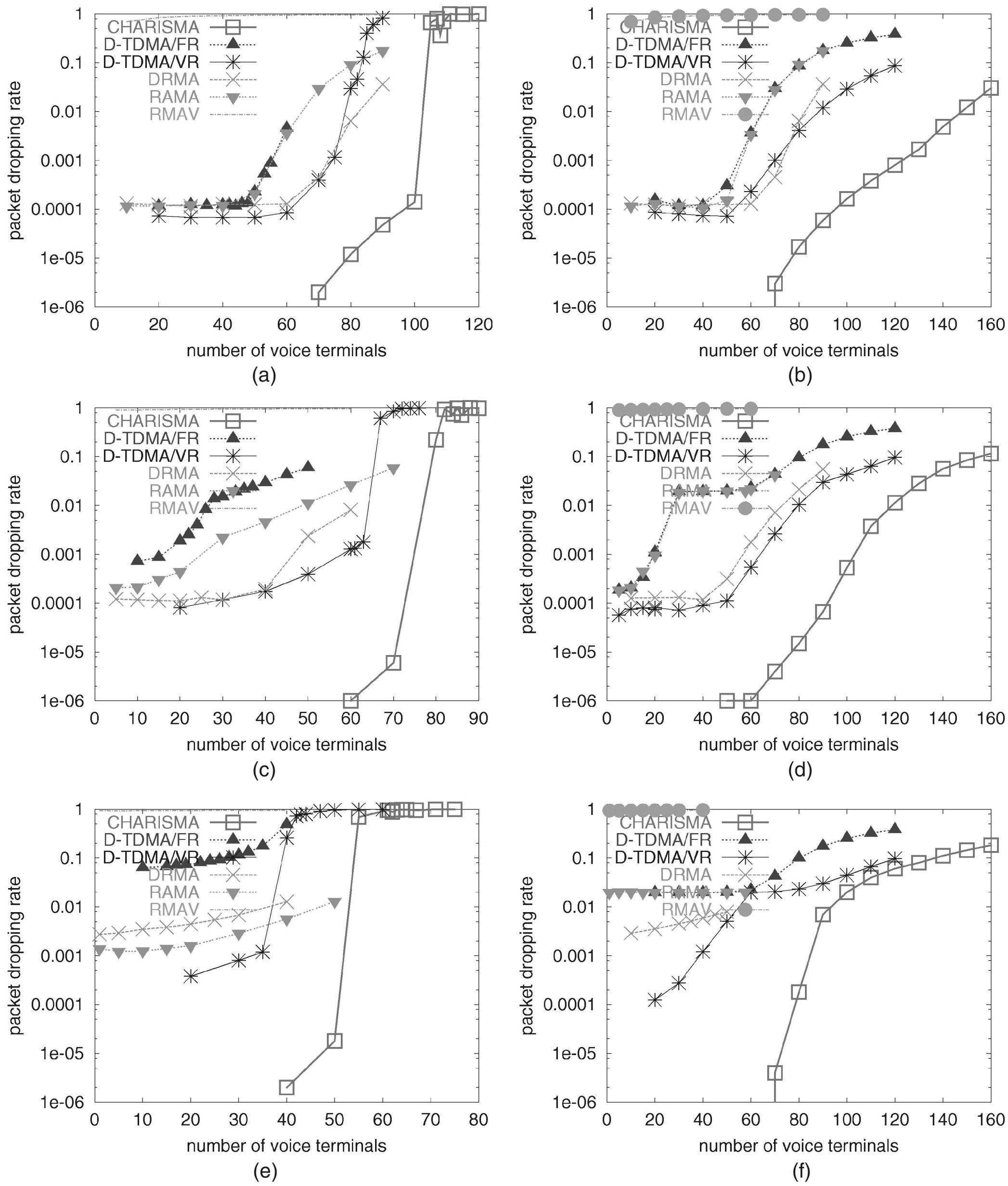


Fig. 11. Voice performance comparison (packet dropping rate versus traffic load). (a) Without request queue and $N_d = 0$. (b) With request queue and $N_d = 0$. (c) Without request queue and $N_d = 10$. (d) with request queue and $N_d = 10$. (e) Without request queue and $N_d = 20$. (f) With request queue and $N_d = 20$.

$N_d = 10$. Moreover, the ranking is different also: CHARISMA, RAMA, DRMA, D-TDMA/VR, D-TDMA/FR, and RMAV. To summarize, the results without request queue indicate the following findings:

- With the intelligent CSI dependent scheduling, the CHARISMA protocol is able to utilize the bandwidth much more effectively. Indeed, because the scheduling process avoids allowing requests with poor channel states to get information slots, packet loss due to transmission errors is greatly reduced. Furthermore, at high load, the CHARISMA protocol can keep the packet dropping due to timeout at a low level.
- Despite that D-TDMA/VR also employs a variable-throughput physical layer, it does not outperform DRMA and RAMA by a great margin (i.e., those with fixed-throughput physical layer). Indeed, by examining the simulation traces, we find that the major benefit of using a variable-throughput physical layer in D-TDMA/VR is that packet loss due to transmission errors is reduced due to the added protection. Thus, it appears that exploiting the synergistic effect between the access control and physical layer (i.e., the CSI dependent scheduling in CHARISMA protocol) is much more important than using the variable-throughput physical layer alone.
- The RAMA protocol, with its collision avoidance property, exhibits a much more graceful performance degradation compared with CHARISMA, D-TDMA/VR, D-TDMA/FR, and RMAV, when

the traffic load becomes very high. Similarly, the DRMA protocol also does not become unstable when the system load is high due to its dynamic frame structure, which simply does not allow users to make requests at high load. Furthermore, the auction mechanism is particularly effective when there are more data users which generate more contention requests.

- When there are data users in the system, the request contention load becomes much higher because a data user repeatedly transmits requests until information slots are granted, as governed by the data permission probability p_d. Furthermore, as a data burst may consist of a few tens of packets, a data user that successfully gets some information slots may also repeat transmitting requests because reservation is not allowed for data users. Thus, every protocol accommodates less voice users when there are data users in the system.

Next, we consider the performance of protocols with request queues. Comparing with those without request queues, the capacity of the CHARISMA and D-TDMA/VR protocols increase significantly. For example, consider the case with $N_d = 0$, the CHARISMA protocol can accommodate about 160 voice users at the 1 percent packet dropping rate threshold, while for the case without request queue, only 100 voice users can be supported. The capacity of D-TDMA/VR also increases by about 25 percent. For the CHARISMA protocol, incorporating a request queue can further increase the *selection diversity* in the CSI dependent scheduling process and, thus, the channel bandwidth can be utilized even more effectively. On the other hand, the request queue in the D-TDMA/VR protocol helps in reducing the packet losses due to severe requests contention.

A very interesting observation is that adding a request queue improves the performance of DRMA and RAMA only slightly. For the DRMA protocol, such an unexpected phenomenon can be explained as follows: The request queue in DRMA usually only contains a few pending requests because an inherent property of DRMA is that users are allowed to participate in requests contention when there are unused information slots. However, the existence of unused information slots implies that successful requests will very likely be assigned information slots in the current frame, and thus, will not be queued. Thus, using a request queue does not help much. Put it in another way, the DRMA protocol inherently exhibits a "distributed requests queueing" property—user requests are "queued" at the user side (i.e., distributed) when there are no empty information slots. On the other hand, for the RAMA protocol, due to its collision avoidance property, all the auction slots will result in successful requests and, the queued requests, if any, are very unlikely to have chances of getting services, especially when the traffic load is high. In summary, the fundamental function of a request queue is to alleviate the request contention and, as such, it is not useful for protocols with an inherent stablizing property such as DRMA and RAMA. Finally, the ranking of protocols with request queue is: CHARISMA, D-TDMA/VR, DRMA, RAMA, D-TDMA/FR, and RMAV.

5.2 Performance for Data Requests

Data packets are delay insensitive and, as such, they will not be discarded at the transmitter. However, they may experience transmission errors when the channel state is too bad. Lost data packets are retransmitted (through the datalink layer). This introduces additional retransmission delay. That is, transmission error is translated into retransmission delay for data requests. To quantify the data request performance, we use the average throughput and the delay as the performance measures. The average data throughput, $\bar{\rho}$, is defined as the average number of data packets successfully received at the base station per frame. The average delay, $\bar{D}_d$, is defined as the average time that a data packet spends waiting in the buffer until the beginning of the successful transmission. Figs. 12 and 13 illustrate the performance of data requests in terms of $\bar{\rho}$ and $\bar{D}_d$, respectively, for cases with and without request queue.

Again, the CHARISMA protocol outperforms the other five protocols by a great margin and the RMAV protocol very quickly becomes unstable. In addition, the rankings in terms of throughput and delay are quite consistent: CHARISMA, D-TDMA/VR, DRMA, RAMA, D-TDMA/FR, and RMAV. When the traffic load is high, the system is in a highly congested state so that the average per-user throughput drops and the average per-user delay also increases dramatically. These adverse phenomena are detrimental to the data users' quality of service (QoS), which depends critically on the parameters pair `(delay, throughput)`. For example, at a QoS level of `(1 sec, 0.25)`, the capacity of the CHARISMA protocol is approximately 1.5 times that of D-TDMA/VR and three times that of RAMA and DRMA. Both results with and without request queue concur with these observations.

5.3 Discussion

It is demonstrated that, by the experiment results described above, the CHARISMA protocol outperforms the other five approaches by a considerable margin. We further explicate and discuss the performance improvements in the following sections.

5.3.1 Slots Utilization

During a voice talkspurt, the mobile device may experience a deep fading for a long time when it is affected by shadowing. In the other five protocols, bandwidth allocation in the access control layer is carried out regardless of the current channel condition as detected in the physical layer. Thus, information slots could also be allocated to such a user and the transmitted packets will be very likely lost due to the poor channel condition. In other words, assigned slots are simply wasted. This kind of wasteful allocation is avoided in the CHARISMA protocol.

In the simulations, all the protocols were governed by the same channel model with a certain bit error rate. Thus, as far as transmission bit error is concerned, all protocols were simulated under the same conditions. However, in the results, we measured the "loss" of data at the receiver and the loss rate depends on both the bit error rate and the packet dropping rate at the sender (due to time-out). Consequently, despite that the D-TDMA/VR protocol also employs link adaptation (transmission bit error rate is

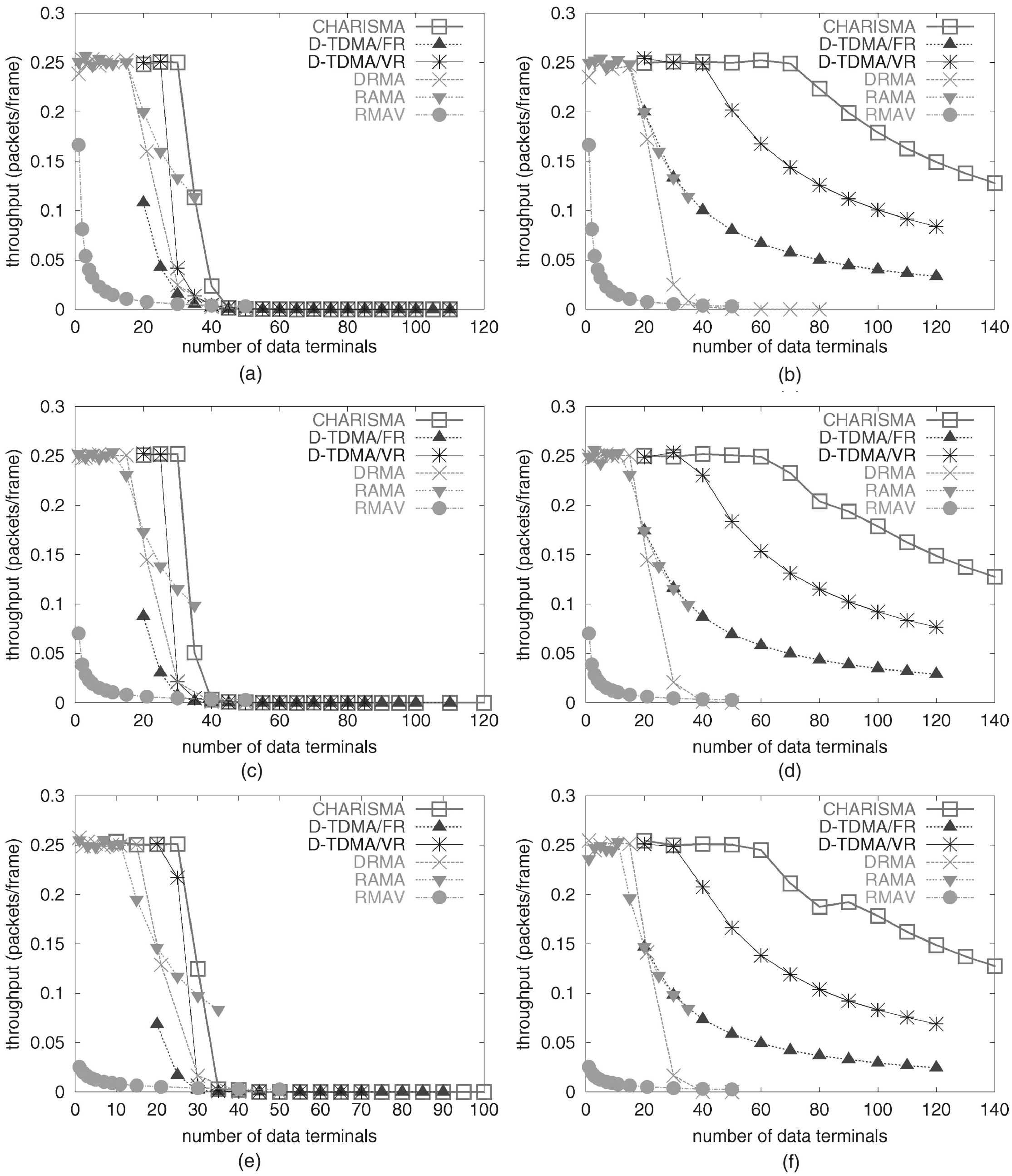


Fig. 12. Data throughput against traffic load. (a) Without request queue and $N_v = 0$. (b) With request queue and $N_v = 0$. (c) Without request queue and $N_v = 10$. (d) With request queue and $N_v = 10$. (e) Without request queue and $N_v = 20$. (f) With request queue and $N_v = 20$.

lowered by adding more redundancy), the loss rate is still higher than that of CHARISMA because the former does not exploit the link adaptation mechanism in the scheduling process (first-come-first-served is used instead).

5.3.2 Selection Diversity

Selection diversity is implicitly incorporated in the CHARISMA protocol. Through the priority-based assignment process, every frame is packed with a selected group of information packets with good channel states. Thus, the effective delivered throughput per frame achieved in CHARISMA can be much higher than that in D-TDMA/VR. In CHARISMA, a large number of transmission requests are collected first before allocation of information slots. From the collection of requests, there is a high likelihood that a sufficient number of requests with good channel states can be selected to fully utilize the information slots in an effective manner (i.e., high throughput). For those requests with poor instantaneous channel states, their transmissions are deferred until when the CSI improves or the deadlines are approaching. By contrast, in the other five protocols, requests are served in a first-come-first-served manner due to the traditional strategy of immediately assigning slots upon successful receipt of requests. Thus, the channel states of such requests are highly diverse and, most importantly, some requests with bad channel states (hence, very low throughput) are also served, whereby causing inefficient bandwidth utilization. For example, a voice request may experience a very good CSI for a long time (out of shadowing). In other protocols, this user, however, may fail to successfully

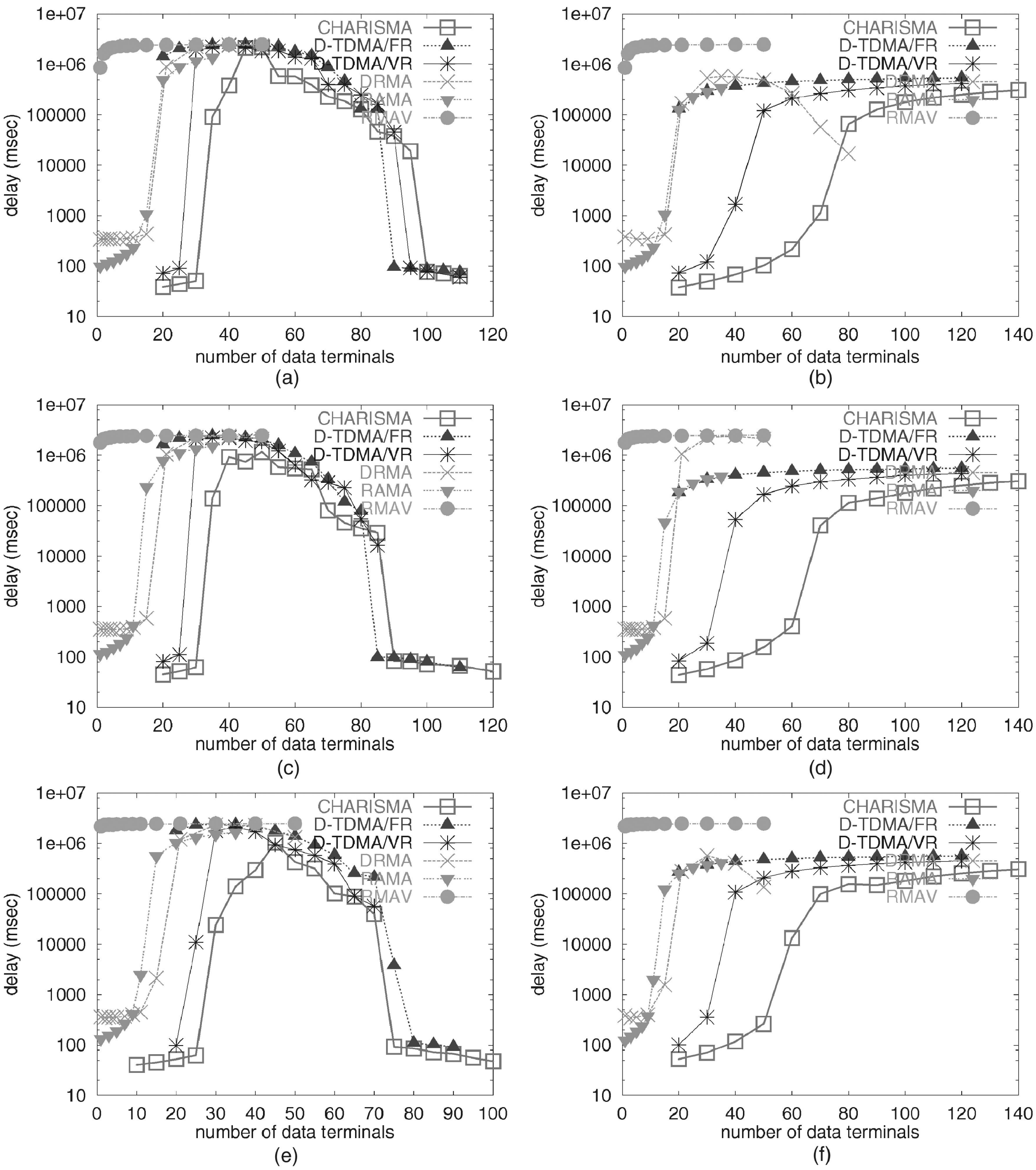


Fig. 13. Data delay against traffic load. (a) Without request queue and $N_v = 0$. (b) With request queue and $N_v = 0$. (c) Without request queue and $N_v = 10$. (d) With request queue and $N_v = 10$. (e) Without request queue and $N_v = 20$. (f) With request queue and $N_v = 20$.

transmitted a request to the base station, probably because of excessive collisions in the request phase. In comparison, our proposed scheme gathers a large number of requests through successive frames and allocates time slots to the users that can use the system bandwidth more effectively. Thus, the likelihood of "missing" a user with a good channel state is much lower and the utilization of bandwidth is therefore higher.

In general, the performance improvements with request queue is dramatically improved compared to the situation without request queue because the queue helps improving system stability and, for the CHARISMA protocol, enhancing the selection diversity.

5.3.3 Mobile Speed and CSI Usage

It is also interesting to note the performance of the CHARISMA protocol under different mobile device speeds, which will affect the CSI gathering and estimation mechanisms. In this regard, we have also obtained simulation results for a range of speed from 10km/hr to 80km/hr. The performance of the CHARISMA protocol remains unchanged even for slow speed region (from 10km/hr to 50km/hr). Due to the space limitations, the results on effect of speed is not included. Let us summarize our observations qualitatively as follows.

As mentioned above, the performance gains of the CHARISMA rely on the exploitation of spatial diversity as a result of independent distributed user geographical positions (and, hence, independent fading between users) and the CSI estimation. When transmission requests of different users are collected during the request subframe, the MAC protocol estimates the CSIs of each user request and allocate resources (informtion slots) based on the estimated CSI. Obviously, this CSI-based resource allocation is meaningful only if the CSI of the users change slowly over the whole frame. Thus, we incorporate a novel CSI refresh mechanism in the CHARISMA protocol to update the estimated CSI values when the requests are queued up. However, when the mobile speed is high, the CSI is changing rapidly over a frame. There could be cases where a user request with good estimated CSI is allocated information slots (those situated at the end of the physical frame) because the MAC protocol believes that the user (with good CSI at the moment) could enjoy a higher throughput and, hence, use the allocated resource more effectively. If the user is moving at high speed, the actual CSI may have changed when the user actually transmit information packets at the allocated slots and the realized throughput may not be as high as that perceived by the MAC protocol at the base station. Fortunately, from our simulation results, even at high speed such as 80km/hr, the performance of the protocol drops slightly only (less than 5 percent). A scruntiny of the requests' behavior reveals that most of the "inaccurate" CSI cases can be resolved by the CSI refresh mechanism. Only a few high speed requests that happen to be at the very end of the frame will waste a small fraction of the allocated bandwidth (a further reason is that the CSI is not changed by too much also). Finally, it is worth mentioning that, in popular standards such as those specified in IMT2000 [11], the targeted mobile speed is at most 15-20 km/hr and the resulting coherence time is about 40-50 msec and thus, our assumption that the CSI remains constant within a 10 msec time frame is justified.

6 Conclusions and Future Work

We have presented a new channel-adaptive uplink access control protocol for a mobile computing system. The proposed protocol, called CHARISMA, employs a variable-throughput adaptive, channel encoder and modulator in the physical layer. An extensive performance comparison of CHARISMA and five recently proposed related protocols, namely, D-TDMA/VR, D-TDMA/FR, DRMA, RAMA, and RMAV are also described. These protocols are carefully selected such that they are devised based on rather orthogonal designed philosophies. The protocols are evaluated for test cases with and without request queues, which store transmission requests that survive the contention but fail to get assigned information slots. Our simulation results reveals that using a variable-throughput physical layer (in the CHARISMA and D-TDMA/VR protocols) can help reducing voice packet dropping due to transmission errors. In addition, protocols with contention collision avoidance (i.e., the RAMA protocol) and protocols with a controlled requesting process (i.e., the DRMA protocol) can provide a high system stability such that performance degrades gracefully even when the traffic load is very high. Furthermore, request queues generally can help improving system performance because the request contention is greatly alleviated. Finally, in view of the fact that the CHARISMA protocol outperforms the other five protocols by a considerable margin, the knowledge of the channel condition reported to the access control layer by the physical layer (in the CHARISMA protocol) is indeed a very useful component in achieving even higher performance in a wireless communication system where burst errors are the norm rather than exception.

There are two possible avenues for further research. The first one is about the priority assignment in scheduling the pending requests. One possible enhancement to our current method is to incorporate a formal notion of "fairness" in assigning priorities by taking into account the different channel conditions, as we have recently proposed in [22]. Another challenging problem is to extend the proposed protocol, probably in a hierarchical manner, to handle the coordination between different base stations, in particular, in the aspect of handoff—that is, when a nomadic user travels into the range of some other base stations, to which new base station should the user attach, from a channel quality point of view? We are currently working on these two problems.

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