Performance Evaluation of A Multiuser Detection Based MAC Design for Ad Hoc Networks

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Abstract— In general, the performance and radio resource utilization of Ad Hoc networks are limited by half-duplex operation and possible collisions. In this paper, we propose a novel approach for MAC design that practically eliminates collisions and significantly increases the bandwidth utilization. The key element of this approach is the CDMA multiuser detection technology that allows receiving several signals in parallel without inflicting self-interference. These features give a promise of significant performance improvements. The main goal of this paper is to assess the range of this gain when compared to other existing alternatives. In particular, we compare the performance of the proposed multiuser detection based MAC design with MAC design based on IEEE 802.11 concept and with MAC design based on multi-code CDMA with one signal reception by one user at a time.

Key words: CDMA Multiuser detection, MAC design, distributed scheduling, Ad Hoc networks.

I. Introduction

Multi-hop mobile Ad Hoc networks have recently been the subject of extensive research. Nevertheless, the performance and bandwidth utilization of the proposed solutions are significantly inferior when compared with the fixed wireless or wired networks. Apart mobility, the main reasons for these limitations are half-duplex operation and possible collisions caused by hidden/exposed terminal problems. While sophisticated adaptive directional antennas give some promise of improvement in the long term, at the moment, this technology seems to be not mature enough to be considered in a difficult Ad Hoc network environment. In this paper, we consider another technology that is more advanced and can significantly improve both the performance and the bandwidth utilization. Namely, recent technological advances allow integration of a CDMA multiuser detection based receiver on one chip and therefore, we consider application of this technology for new MAC design in multi-hop Ad Hoc networks.

There are two main advantages of using multiuser detection in Ad Hoc networks. First, multiple signals from different mobiles can be received at the same time, which practically eliminates the collision problem and can significantly reduce the end-to-end packet transfer delay. Second, the multi-user detection avoids interference between the received signals and this feature can increase the bandwidth utilization by a large factor [1][2][3]. Although multiuser detection is known for a long time, most of the studies focus on the physical layer[3]. According to our knowledge, there are no published studies on application of this technology to mobile Ad Hoc networks. In this context, the main goal of this paper is to evaluate potential gain from application of CDMA multiuser detection compared to other solutions proposed for Ad Hoc networks. In particular, we use as references MAC design based on IEEE 802.11 concept and MAC design based on multi-code CDMA, but with one signal reception by one user at a time.

In order to achieve a gain from multiuser detection, a new MAC layer design is required. This design should realize three main functions. First, distributed dynamic code assignment that avoids code collisions is needed. Then, an implementation of distributed scheduling of transmissions and receptions that maximizes throughput within the required fairness criteria is required. Finally, the data transmission based on multiuser detection has to be organized. Each of these functions constitutes a research topic sufficient for a separate publication of its own. Therefore, in this paper, we give a high level presentation that is focused mainly on an conservative assessment of the potential gain from the application of multiuser detection in Ad Hoc networks. More detailed study of required protocols and algorithms and their optimizations will be given in subsequent publications.

The remainder of the paper is organized as follows. In Section II, we review a multiuser detection model. In the section that follows, we present a framework for the proposed MAC design. It is based on a synchronous time division CDMA (TD-CDMA) structure where each frame is divided into two control slots and a continuous data transmission slot. The control slots are used by a code assignment protocol and a data transmission scheduling protocol. Section IV describes a distributed scheduling mechanism that provides local fairness and priorities for real time traffic based on time deadlines. The numerical results, presented in Section V, indicate that the proposed MAC design increases the throughput by more than an order of magnitude when compared with MAC designs based on IEEE 802.11 concept and by around 100% when compared with MAC designs based on multi-code CDMA. At the same time, the performance characteristics, such as packet delay and loss, are significantly improved. Section VI gives concluding remarks and issues for future work.

II. Linear multiuser detector

Multiuser detection improves the performance of spread-spectrum systems by exploiting the structure of the multi-access interference when demodulating the signal of a user. Fig. 1 demonstrates the throughput improvement multiuser detector over one user detector when two users are transmitting simultaneously. When there is only one communication, the maximum reliable transmission rate is R_1. But when there are two simultaneous communications, with multiuser detection, the reliable communication is possible at rate pair (R_1, R_2) which outperforms the one user communication with $R_1 + R_2 > max(R_1, R_2)$.

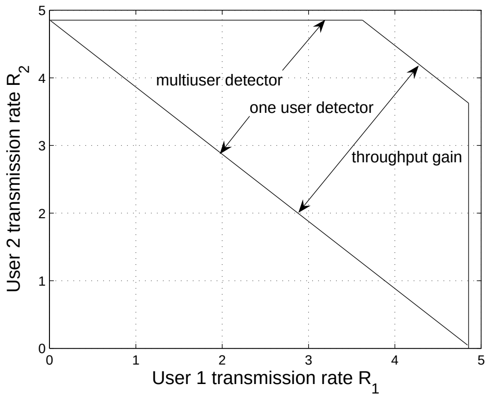

Fig. 1. Throughput improvement of multiuser detection

We consider a minimum mean square error (MMSE) detector as it is an optimal linear detector that maximizes the signal-to-interference-plus-noise (SINR) ratio. Under random spreading sequences, the SINR at node k from node i is [1]

$$\gamma(i,k) = \frac{p_i}{\sigma^2 + \frac{1}{L}\sum_{j=1,j\neq i}^{M} I(p_j, p_i, \gamma(i,k))} \quad (1)$$

where

$$I(p_j, p_i, \gamma(i,k)) = \frac{p_j p_i}{p_i + p_j \gamma(i,k)}$$

is the effective interference from user j. p_i, p_j are the received powers from node i, j at node k; σ^2 is the noise power at node k; L is the processing gain; and M is the number of signals being processed by node k. This equation holds when the number of nodes (M) and the processing gain (L) both go to infinity, with $M/L \to \theta$, a constant. In a real system, the SINR expression is an approximation to the actual SINR. It is noted that the approximation becomes more accurate as the system scales up. This model also gives explanation why the received signals do not inflict normal self-interference and this feature provides significant increase of data transmission throughput.

III. FRAMEWORK FOR MULTIUSER DETECTION BASED MAC PROTOCOLS

We assume that each node is equipped with a half-duplex CDMA multiuser detector and that each node is allocated a dedicated code for transmitter-based data transmission. When a node enters into the system, it is allocated a dedicated code from a set of predefined dedicated code channels that are not used in the node's neighborhood. This allocation can be changed when two nodes with the same code are entering the same neighborhood in order to avoid a code collision.

Due to the nature of multiuser detection, we assume that the transmission times are synchronized at packet level with the help of either in-band signal exchanges or out-of-band solutions, such as GPS. The proposed frame structure of such a synchronous time division CDMA (TD-CDMA) system is illustrated in Fig. 2 where three slots are defined. The first two slots

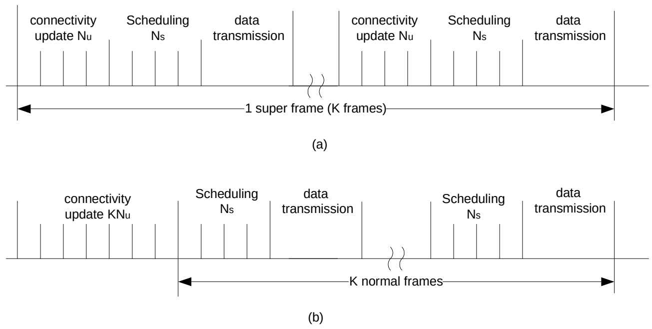

Fig. 2. Synchronous transmission

are associated with control functions and are called connectivity update and scheduling slots, respectively. The third slot serves for a continuous data transmission in the CDMA channels that are selected during the scheduling slot.

The connectivity update slot is used by the connectivity update protocol. In this phase, each node broadcasts its identity information on the common signaling channel so that the neighboring nodes can be detected and code assignment and reassignment can be realized. Since the connectivity update can be executed in a significantly longer time period than the frame cycle, there are two ways to do connectivity update. One is to distribute connectivity update function into K frames and each frame contains N_u minislots for connectivity update that give in total KN_u minislots as shown in Fig. 2 (a). Alternatively, one can introduce a connectivity update slot with KN_u minislots for every K normal frames as illustrated in Fig. 2 (b). In this case, the normal frame only have scheduling and data transmission slots. Connectivity update slot achieves exclusive code assignment in neighborhood to avoid hidden/exposed terminal problems. The broadcasting and code assignment scheme is out of the scope of this paper. The interested readers can refer to [4] for details.

The scheduling slot is associated with the scheduling protocol and its objective is to select transmissions that can be realized in the subsequent data transmission slot. In this phase, the transmitting and receiving nodes have to be chosen together with the data packets to be transmitted and the channel parameters to be used (like power and transmission rate). The transmission selection should take into account the packet priority, the access fairness and throughput objectives. A simple scheduling mechanism is presented in Section IV. Note that the scheduling protocol can use dedicated code channels that are established in the connectivity update phase. In this way, the problem of collisions at receivers which is typical for existing solutions can be effectively eliminated.

As mentioned before, during the data transmission slot, several transmissions are executed in the same neighborhood by taking advantage of CDMA multiuser detection. Here we assume that multicode CDMA is deployed to support variable rate transmission. When a node needs to transmit at m-rate, it first converts its date stream, serial-to-parallel, into m basic-rate streams. Each basic-rate stream is spread with a different code generated by the so called "sub-code concatenation" scheme[5] and superimposed before spread spectrum modulation. The sub-code concatenation generated codes are unique to each mobile.

IV. DISTRIBUTED NEIGHBORHOOD SCHEDULING

To facilitate the presentation of the scheduling algorithm, we first review the traffic model considered in this paper. We consider two classes of traffic: realtime voice and non realtime data. Each node requires two buffers for each of its neighboring nodes to accommodate bursty voice and data packets arrivals. Each of the traffic class has different packet generation rate, SINR, packet loss rate (PLR) and delay requirements. The data transmission slot length corresponds to the transmission time of one voice packet with basic transmission rate on a single code channel. We assume that the data packets can be fragmented or assembled to form transmission packets with the same transmission duration as the voice packets. It is also assumed that each arrival packet that enters the scheduling system is allocated a delay bound (time-out) value expressed in number of frames. This value is decreased by one for every frame interval till the packet is scheduled to transmit or till the value is zero. In the latter case, the packet is discarded. The rate at which voice packets are discarded is referred to as the voice packet loss rate. In the following, we refer to the packets with time-out value equal to one as the most urgent packets (MUPs), the packets with time-out value equal to two as the second MUPs, and so on.

We assume that the voice packets have higher priority than data packets and that, among the same class packets, the smallest time-out value gives precedence for transmission. Based on these rules, each node selects a candidate packet or a set of candidate packets from the same class and destined to one of its neighboring nodes for a possible transmission in the next transmission slot. Then the node priority for transmission contention with other neighboring nodes is determined. Here again, the precedence is given for voice packets. Then, for nodes with candidate voice packets, the precedence is given for nodes with higher overall voice packet loss rate. Analogously, for nodes with candidate data packets, the precedence is given for nodes with higher overall data packet delay.

To resolve the contention for transmission and select the receiving nodes, we apply exchange of RTS/CTS control messages conducted on dedicated channels. Namely, at the beginning of each scheduling slot, each node, with a set of candidate packets for transmission, sends out an RTS message. A RTS message carries the intended receiver ID, the sender ID, a bunch of subcodes which represent the variable transmission rate, and the node priority which contains two fields. The first field is traffic type, either voice or data; and the second field is the measured packet loss rate for voice traffic and the largest delay for data traffic. The distribution of RTS messages in the neighborhood is not straightforward since the targeted RTS receiver may also have an RTS message to be sent. To solve this problem, we propose a protocol that requires only three minislots. In this approach, the nodes are divided into black and white nodes in such a way that in the neighborhood there are at least one black and one white node. Then, in the first minislot, the white nodes transmit the RTS messages on the dedicated channels. These messages are received by all black nodes due to multiuser detection. In the second minislot, the black nodes transmit their own RTS messages together with the RTS messages received in the first minislot. Finally in the third minislot, the white nodes transmit the RTS messages received from the black nodes in the second minislot. Due to the space limitation, we will provide full description of this protocol in a forthcoming publication.

After three minislots of RTS exchanges, each node has the transmission status, either transmission or reception, of its neighboring nodes. A node that receives at least one RTS message determines whether it will receive or transmit in the subsequent transmission slot by comparing its own node priority with the other contending nodes' priorities. In case at least one of the received RTS messages has higher priority, the node becomes a receiver to all contending neighboring nodes; otherwise, it will transmit its own candidate packets. Then, the nodes selected as receivers send out the CTS messages to the senders of received RTS messages in one minislot by means of dedicated channels.

After the first cycle of RTS/CTS exchange, a second round of RTS/CTS exchange is executed. Since this time the receiving nodes are determined, the exchange can be executed in two minislots. This additional exchange has several goals. The first goal is associated with the fact that in the second round the RTS messages are received directly from its source nodes. It gives the selected receivers possibility to estimate the propagation loss related to each connection. Also, it allows estimating the interference induced by connections destined to other nodes. In this way, local power control can be implemented at receivers. Let M_v and M_d denote the number of neighboring voice and data connections which have the same destination. Define $\alpha_v = M_v/L, \alpha_d = M_d/L$. Let γ_v and γ_d denote the SINR requirements and p_v and p_d denote the minimum required received power to achieve γ_v and γ_d, respectively. If we replace Eq. (1) with these parameters and after some manipulation, the SINR expressions for voice and data traffic can be specified as

$$\begin{cases}\gamma_v = \frac{p_v}{\sigma^2+(\alpha_v-\frac{1}{L})I(p_v,p_v,\gamma_v)+\alpha_d I(p_d,p_v,\gamma_v)+\frac{1}{L}\sum_i I(p_i,p_v,\gamma_v)} \\ \gamma_d = \frac{p_d}{\sigma^2+\alpha_v I(p_v,p_d,\gamma_d)+(\alpha_d-\frac{1}{L})I(p_d,p_d,\gamma_d)+\frac{1}{L}\sum_i I(p_i,p_d,\gamma_d)}\end{cases}. \tag{2}$$

The last item in the divisor is the effective interference from neighboring nodes whose destinations are other nodes. For large networks, γ_v/p_v and γ_d/p_d are a constant[1].

Based on Eq. (2), the minimum required received power to achieve γ_v and γ_d for voice and data traffic can be estimated at each receiver. With the estimated propagation attenuation, the minimum required transmission power for each subcode channel is estimated. Under maximum transmission power constraint, the maximum number of packets that can be supported in each connection can also be obtained. The transmission permissions which include the confirmed number of subcodes and the transmission power for each packet transmission are passed via the second CTS message to the senders.

The second goal of the second RTS/CTS round is to allocate a function to the stranded nodes. The stranded nodes are the nodes that do not receive CTS messages addressed to them in the first round since their RTS messages are sent to the nodes that win the contention for transmission. Note that after the first round of RTS/CTS messages, each stranded node knows which function was allocated to each node in the neighborhood (receiver or transmitter or stranded). Based on this information, two options are considered. In the first one, each stranded node selects new candidate packets from the packets destined to one of the receiving nodes. Based on this new selection, the stranded node sends a new RTS message directly to the selected receiving node. In the second option, the stranded nodes organize communication between them. Namely, based on the known neighboring stranded nodes priorities, each node determines whether it becomes a transmitter or a receiver and the transmitters send RTS message directly to the associated receiver.

The scheduling approach proposed in this section is not necessarily optimal but is sufficient for a conservative assessment of the gain from application of the multiuser detection. More sophisticated schemes will be presented in subsequent publications.

V. Numerical results

We consider three major performance measures of the proposed neighborhood scheduling scheme. i) voice packet loss rate; ii) average data packets transmission delay; iii) throughput. Average data packets transmission delay is formulated as $(N_{loss} \times D_{bound} + \sum_m n_{trans.} \times D_{delay})/N$, where N_{loss} is the total number of data packet loss, D_{bound} is the delay bound for data packets, $n_{trans.}$ is the number of successfully transmitted data packets with delay of D_{delay} in frames. Throughput is defined as the accumulated number of packets that are successfully received in one hop's transmission.

The numerical results are obtained by means of a discrete event simulation that models an Ad Hoc network working at 450MHz frequency band with the following parameters. The spreading gain is set to 512. We focus on a circle simulation area with radius of 350m. A mobility model which mimics human and vehicle movement behavior is applied [6]. The speed limit is 120km/h. To avoid boundary effect, the nodes moving out of the circle will reenter the simulation area again. We assume a reliable wireless communication and a free space propagation mode so that the signal attenuation is caused exclusively by transmission distance. To promptly update its connectivity information for neighborhood identification, we apply the radio structure as shown in Fig.2 (b) for connectivity update with an interval of $10 \times$ normal frame length. Each node has the maximum transmission power of 7w. The neighborhood threshold is set to 10^{-6}w and the noise power is set to 10^{-7}w. Each node accommodates two types of traffic, voice and data with SINR requirements of 7dB and 10dB, respectively. Both the voice and the data packet flows are simulated by an ON-OFF model. The delay bound for a packet entering a node is set to 2 and 100 frames for voice and data packets, respectively. The simulation runs 10000 frames to obtain the results.

Since the goal of the presented study is to assess the efficiency of the proposed scheduling scheme, the objective of the applied packet generation model is to achieve large and uniform loading of each node. Therefore, we do not consider the routing algorithm and the packets are generated in each node with indication of the neighboring node to which they should be sent. The ON and OFF durations of packet flows follow an exponential distribution with mean of 1 and 49 frames for voice and 5 and 25 frames for data traffic, respectively. The voice and data packets generation rate at each node is proportional to the number of neighboring nodes and is assumed to be Poisson distributed with mean of 0.1 and 1 times the number of identified neighboring nodes. Every bunch of packets generated in a frame is randomly assigned to a destination chosen from its neighboring node list. As a destination node may move out of the transmission range of a sender, waiting packets with this destination are reassigned to a destination from the updated neighboring node list.

We compare the performance of the proposed MAC design based on multiuser detection with two existing concepts: multiple CDMA connections in the neighborhood [7], and one connection in the neighborhood as in IEEE 802.11. The three communication scenarios are demonstrated in Fig. 3. Note that in

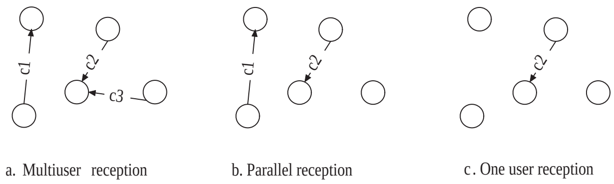

Fig. 3. Simulation scenarios

the first two scenarios, there are several parallel connections in the neighborhood, but only in the multiuser detection case a node can receive several connections simultaneously. In order to facilitate the comparison, we assume that all three communication scenarios use the same frame structure as proposed in section III. This corresponds to an assumption that on average, the ratio between the data communication time and the time associated with exchange of control packets is the same in all scenarios.

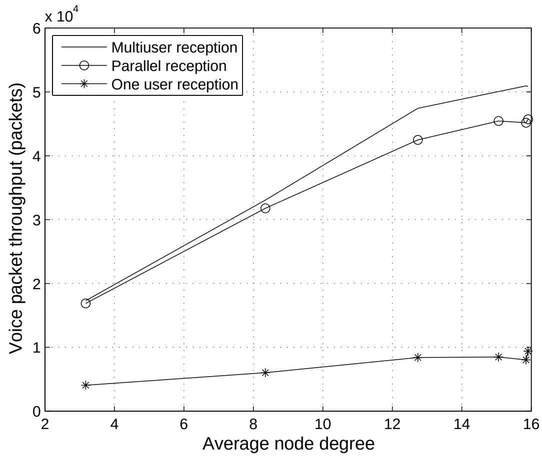

Fig. 4. Voice packet throughput

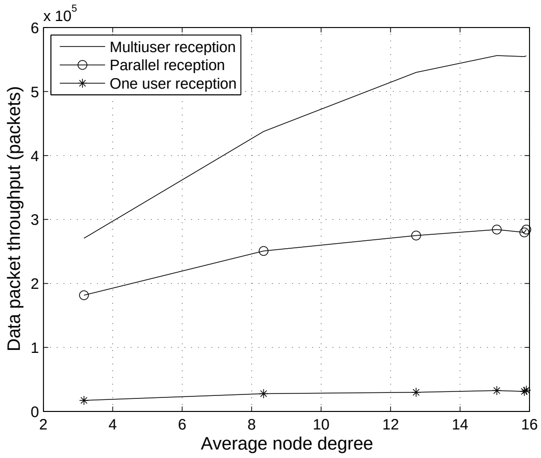

Fig. 5. Data packet throughput

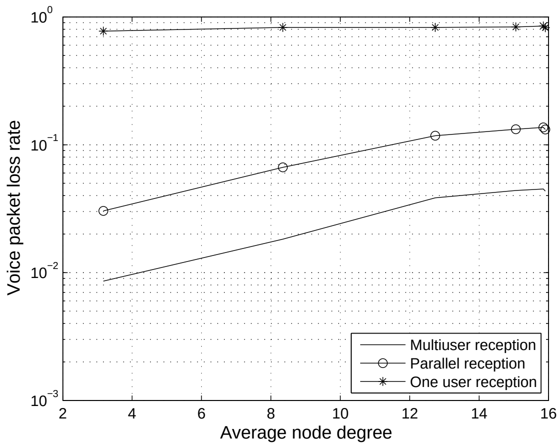

Fig. 6. Voice packet loss rate

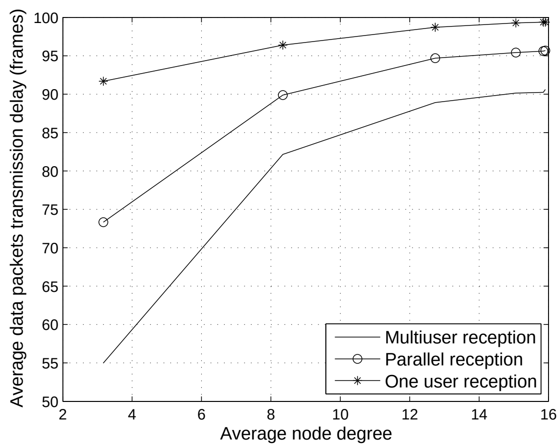

Fig. 7. Average data packets transmission delay

Figs. 4 and 5 show the voice and data packet throughput as a function of average node degree. Node degree is defined as the number of first hop neighboring nodes. As average node degree increases, the proposed MAC design with multiuser detection can provide a total throughput that greatly outperforms the other two solutions. In particular, this gain is in the range of 100% when compared to multiple CDMA channels case and around 15 times when compared to one channel case when the average node degree reaches 16. Obviously, this gain is realized largely by the data traffic since the voice traffic constitutes only around 9% of the total traffic. It is important to underline that at the same time, the multiuser detection scheme provides a lower voice packet loss rate and a smaller data packets transmission delay as illustrated in Figs. 6 and 7. Another observation of Figs. 6 and 7 is that as the average node degree increase, the voice packet loss rate and the average data packet transmission delay increase as well because the increase of network density results in more contentions among neighboring nodes.

VI. CONCLUSIONS

A new MAC design based on multiuser detection is presented in this paper. By using a linear MMSE multiuser detector at each node, the proposed solution can significantly outperform the solutions based on multiple CDMA channels and IEEE 802.11 concepts. Moreover, with described receiver-based power control, each node can save power consumption and reduce the interference induced in other nodes, which can accordingly allow greater spatial reuse.

The presented work is part of a larger project that is supported by both the government and a private company funds and the work is being continued in several directions in both physical and networking layers. Concerning work related to multiuser detection, currently we are working on application of distributed fair queuing algorithm for combined fairness and throughput optimization. Also, a more sophisticated propagation model with consideration of shadowing and fading will be taken into account in opportunistic scheduling for better radio resource utilization.

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