Receiver Initiated MAC Design for Ad Hoc Networks Based on Multiuser Detection

Jinfang Zhang, Zbigniew Dziong, Francois Gagnon and Michel Kadoch
Department of Electrical Engineering, Ecole de Technologie Superieure
Montreal, Quebec, Canada, H3C 1K3

***Abstract*—Recent technological developments in code division multiple access (CDMA) with multiuser detection (MUD) make multiple packets reception a more appropriate model for the physical layer of future wireless networks. To take advantage of the new features, a shift of responsibility from transmitters to receivers is suggested. This paper proposes a novel receiver initiated multimedia access control (MAC) protocol and a distributed Generic Additive Increase Multiplicative Decrease (GAIMD) fair scheduling scheme to configure efficient transmissions in Ad Hoc networks. The schemes are very simple to implement. Simulation results demonstrate that the throughput can be significantly improved when compared to a transmitter initiated scheme [1] at the price of increased queueing delay of priority voice packets.**

I. Introduction

Multihop mobile Ad Hoc networks have recently been the subject of extensive research due to its ubiquitous potential applications in military, emergency or conference, etc. environments. Various MAC protocols for Ad Hoc networks based on collision avoidance have been proposed over the past few years. Traditional collision avoidance MAC protocols are usually transmitter-initiated. However, the advent of diversity modulation and signal processing has changed the underlying assumption of the conventional collision model that at most one packet can be received at each node. The use of directional antenna arrays [2] and CDMA make multiuser reception (MUR) a more appropriate model for the physical layer of future wireless networks. MUR suggests a shift of responsibility from transmitters to receivers. Specifically, since the transmitter might not be aware of the receiver MUR capacity and the traffic directed to the receiver, it is better that receiver decide which nodes should transmit. Thus, it appears that receiver controlled protocols have the inherent advantage in exploiting the MUR property. Though some research has been done in receiver-initiated collision avoidance in Ad Hoc networks, the solutions in [2] are based on intelligent directional antenna technologies which are far from mature to be implemented in unpredictable Ad Hoc networks. In addition, potential collisions cannot be completely avoided if two nodes cannot satisfy a certain minimum angular separation. Another promising approach is to use CDMA multi-channel transmissions to increase the spectrum reuse. A MAC scheme based on CDMA with 3 orthogonal codes is proposed and evaluated in a Manhattan network topology in [3]. However, [3] cannot avoid transmission collision when two or more neighboring nodes use the same code channel simultaneously. In fact, transmission collision cannot be effectively avoided if each node in two hops' distance cannot be allocated with a unique code.

The efficiency of CDMA based MUR can be further increased by applying MUD [4]. In this case, mutual interference of received signals is effectively mitigated at the expense of increased complexity. Recent technological advances allow integration of a CDMA MUD based receiver on one chip. This development allows application of MUD for Ad Hoc networks in order to take advantage of both: spectrum reuse improvement due to MUR and capacity gain due to MUD. Although MUD has been known for a long time, most of the studies focus on the physical layer [5]. The studies on MAC and network performance with the application of MUD technology in Ad Hoc networks appear in the literature only recently. In [1], a MUD based MAC scheme is designed for Ad Hoc networks. Nevertheless, it is a transmitter-initiated scheme where significant signaling has to be exchanged before data transmission for fairness provisioning and QoS satisfaction. As a result, the data throughput decreases with the increasing number of neighboring nodes.

As a node has no knowledge of the transmission requests and packet priorities of potential transmitters, another challenging topic for receiver-initiated multiple access is how to dynamically determine whether the node should be in transmission or in reception status which will directly affect the radio resource utilization and QoS performance. The scheme proposed in [2] assumes a random time duration between two successive reception states, while [3] uses a fixed value. Both schemes are not practical without even considering the dynamically varying traffic load in contending nodes.

To solve these problems, this paper proposes a novel receiver initiated MAC protocol and a distributed GAIMD fair scheduling scheme based on MUD. In GAIMD fair scheduling, each node adaptively determines its transmission or reception status based on a) candidate packets waiting at this node; b) contending packets destined to this node. The proposed scheme provides QoS satisfaction by selecting candidate packets for transmission according to their priorities. The effective cooperation among neighboring nodes achieves fairness requirements. Finally, each receiver organizes the transmissions with power allocation to maximize throughput. The remainder of the paper is organized as follows. Section II presents a framework for the proposed MAC design and scheduling. The applied MUD model is also reviewed. A distributed receiver initiated MAC protocol and a GAIMD fair scheduling scheme are proposed in section III. Simulation results are presented to evaluate the

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performance of the proposed schemes, followed by conclusion remarks in section V.

II. System Model

The network works on a single frequency band and the connection between nodes is assumed to be symmetric. Each node is equipped with a half-duplex CDMA multiuser detector so that if a signal arrives at the node when it is transmitting, the signal cannot be correctly received, which is denoted as primary collision. The maximum number of signals that MUD can detect is referred to as capacity of the MUD receiver. Two kinds of neighbors, transmission neighbors and sense neighbors are defined. Transmission neighbors are neighboring nodes that connections can be established to and sense neighbors are neighboring nodes that their codes can be sensed and used in interference mitigation. The maximum number of sense neighbors, D, is limited by both the MUD capacity and the physically existing neighboring nodes. The maximum number of transmission neighbors is denoted as neighborhood capacity M. We assume that MUD capacity is greater than or equal to the neighborhood capacity in order to have each node capable of MUD reception from all its transmission neighbors at the same time. Since the complexity of MUD receiver grows exponentially with its capacity, M constitutes an important parameter for MUD optimization. Because interference from sense neighbors is mitigated by using MUD, unique code assignment should be in two hops' sense distance. On the other hand, signal exchanges and data transmission only take place among transmission neighbors.

A. Radio structure

The proposed frame structure of a synchronous time division CDMA (TD-CDMA) system is illustrated in Fig. 1. Transmission synchronization at frame level can be achieved by tracing a common timing source, such as GPS [6]. Two types of frames

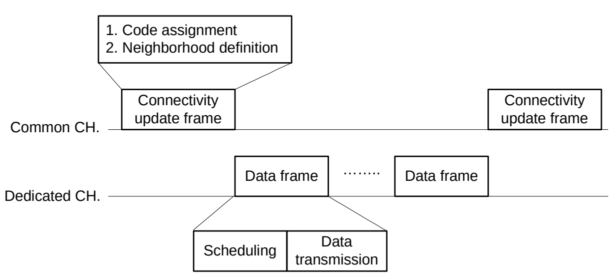

Fig. 1. Synchronous transmission

are defined. The first frame is for a connectivity update protocol which is carried on common code channel [7]. In this frame, each node in two hops' sense range is allocated a unique code channel to facilitate MUD. Besides, transmission neighbors are identified at each node. We will not review the detailed implementation of this frame and interested readers can refer to [7]. The focus of this paper is on data frame which is comprised of a scheduling slot and a data transmission slot. In scheduling slot, each node determines its status to be either in transmission or in reception by taking into account the radio access fairness and throughput objectives. The designated receivers then initiate transmission process and configure transmission in neighborhood with power allocation. It is noted that the scheduling slot and data transmission slot both operate in the MUD mode with dedicated code channels for collision avoidance.

B. Linear multiuser detector

MUD improves the performance of spread spectrum systems by exploiting the structure of the multi-access interference when demodulating the signal of a user. We consider a minimum mean square error (MMSE) detector as it is an optimal linear detector that maximizes the signal-to-interference-plus-noise ratio (SINR). Under random spreading sequences, the SINR at receiver k from transmitter i is [5]

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

where

$$I(p_j, p_i, \gamma(i,k)) = \frac{p_j p_i}{p_i + p_j \gamma(i,k)} \tag{2}$$

is the effective interference from interfering node j, $j \in D$. p_i and p_j are the received power from node i and j at node k; σ^2 is the noise power at node k; L is the processing gain; and D is the number of signals being processed by node k. By analyzing the effective interference in Eq. (2), we note that when $p_j \ll p_i$, the effective interference is simply p_j; but when the interference power increases, for instance when the interfering node j moves closer to the receiver k than the transmitter i, $p_j \gg p_i$, the effective interference seen by node k is bounded by $p_i/\gamma(i,k)$. If the signal and the interference power are comparable in amplitude, the effective interference can also be reduce by a factor of $1 + \gamma(i,k)$. In summary, MUD can efficiently mitigate interference as

$$I(p_j, p_i, \gamma(i,k)) = \begin{cases} p_j & p_j \ll p_i; \\ \frac{p_j}{1+\gamma(i,k)} & p_j \approx p_i; \\ \frac{p_i}{\gamma(i,k)} & p_j \gg p_i. \end{cases} \tag{3}$$

Eq. (1) holds when the number of nodes (D) and the processing gain (L) both go to infinity, with $D/L \to \theta$, a constant. In a real system, the SINR expression is an approximation to the actual SINR, and the approximation becomes more accurate as the system scales up.

III. Receiver initiated MAC protocol and GAIMD fair scheduling

To facilitate presentation, we first review the traffic model and the scheduling principles considered in this paper. We assume two classes of traffic: realtime voice and non realtime data. Each traffic class has different SINR and delay requirements. To accommodate arriving voice and data packets, each node requires two buffers for each of its neighboring node. The packet length is corresponding to the frame length although this is not a constraint as packets of variable length can be fragmented or assembled [8]. A newly generated packet has its delay set to 0 and this delay value is increased by one every frame till the packet is scheduled to transmit or the value equals the delay bound. In the latter case, the packet is discarded. The delay value defines the packet priority with delay equal to 0 representing the lowest priority within each class. We assume that voice packets always have precedence for transmission over data packets.

A. Receiver initiated MAC protocol

In IEEE 802.11 standard [9], transmitter-initiated MAC protocols use RTS/CTS handshakes to reserve a channel before data transmission. Exponential backoff is introduced if RTS collision happens which makes the radio resource utilization inefficient. The MAC protocol proposed in [1] applies multiple CDMA code channels to separate nodes. With a multiuser detector equipped at each node, a receiver can detect multiple signals simultaneously to improve throughput. This scheme is optimal in that it achieves the most efficient transmission configuration with the knowledge of contending nodes to provide best fairness. Therefore, this scheme can be referred to as a benchmark in evaluating the performance of scheduling schemes. Nevertheless, this scheme requires substantial signaling overhead, so, the basic idea of the proposed receiver-initiated MAC protocol in this paper is to reduce the signaling overhead by initiating transmission process from receivers rather than transmitters. A minor modification to the commonly used RTS/CTS handshakes is required by introducing a Ready-To-Receive (RTR) message and the extended RTS/CTS messages.

A node can be in one of the two states defined as transmission and reception in each data frame. Nodes in transmission status are referred to as potential transmitters and nodes in reception status are referred to as potential receivers. Determination of the status will be discussed in the following fair scheduling section. At the beginning of each scheduling slot, each potential receiver uses its dedicated code channel to broadcast an RTR message at the maximum transmission power. The potential transmitters collect the RTR messages in its neighborhood with MUD. As in transmitter-initiated RTS exchanges, primary collision happens at potential receivers as shown in Fig. 2, where nodes N1, N2 and N5 serve as potential receivers and N3 and N4 serve as potential transmitters. The lines represent the connections among nodes and the arrows represent the RTR directions. Here, due to primary collision, N1 cannot receive the RTR message from N2 because N1 is transmitting at the same time, and vice versa. However, primary collision at receivers does no harm to packets scheduling and transmission because the RTR message is intended for potential transmitters only. A receiver list is then formed at each potential transmitter. Based on design objectives, fairness provisioning or throughput maximization for instance, potential transmitters select their associated receivers. Here, transmission priority is given to waiting packet with the highest priority. Therefore, if the waiting packet of the highest priority finds its destination in the receiver list, this packet is selected as a candidate packet for potential data transmission. Otherwise, the second highest priority packet is checked. This process continues in the order of decreasing packet priority till a packet is found or it reaches the end of the waiting buffer. In the latter case when a potential transmitter cannot find a receiver in neighborhood, it will wait for the next data frame if there are waiting packets in buffer. The RTR message is formatted as shown in Fig. 3

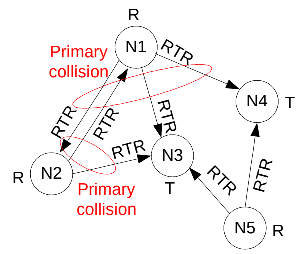

Fig. 2. RTR primary collision

Frame control	Sender ID	FCS
8bits	8bits	16bits

Fig. 3. RTR message

Once a candidate packet is selected, the potential transmitter sends out a RTS message formatted as in Fig. 4 to the destination. The delay value indicates the priority of the candidate packet. It is noted that primary collisions will not occur to RTS messages since the receivers are already known to transmitters. A receiver can receive all the RTS messages with MUD and then executes transmission scheduling and transmission power allocation for each transmission request. Let N_v and N_d denote the number of voice and data transmission requests at a receiver and define $\alpha_v = N_v/L, \alpha_d = N_d/L$. 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

Frame control	Sender ID	Receiver ID	Delay	FCS
8bits	8bits	8bits	8bits	16bits

Fig. 4. RTS message

$$\begin{cases} \gamma_v = \frac{p_v}{\sigma^2+(\alpha_v-\frac{1}{L})I(p_v,p_v,\gamma_v)+\alpha_d I(p_d,p_v,\gamma_v)+\frac{1}{L}\sum_i I(p_i,p_v,\gamma_v)+I_o} \\ \gamma_d = \frac{p_d}{\sigma^2+\alpha_v I(p_v,p_d,\gamma_d)+(\alpha_d-\frac{1}{L})I(p_d,p_d,\gamma_d)+\frac{1}{L}\sum_i I(p_i,p_d,\gamma_d)+I_o} \end{cases}. \tag{4}$$

where $\frac{1}{L}\sum_i I(p_i,p_v,\gamma_v)$ and $\frac{1}{L}\sum_i I(p_i,p_d,\gamma_d), i \in D$ are the effective interference from neighboring nodes whose destinations are other nodes; I_o is the interference from outside of the sense neighbors. For large networks, γ_v/p_v and γ_d/p_d are a constant[5]. Based on Eq. (4), the minimum required received power p_v and p_d to achieve γ_v and γ_d for voice and data traffic can be obtained at each receiver. If either p_v or p_d exceeds power limit, the low priority transmission requests will not be scheduled to transmission in the following data slot. In this case, p_v and p_d need to be recalculated. With channel estimation according to the signal attenuation on RTS message, the minimum required transmission power for each code channel is estimated. Then, the candidate packets allowed for final transmissions are selected. Transmission permissions which include the transmission power for each candidate packet are passed via CTS messages as formatted in Fig. 5 back to the potential transmitters.

Frame control	Sender ID	Receiver ID	Transmission power	……	FCS
8bits	8bits	8bits	8bits	16(M-1)bits	16bits

Fig. 5. CTS message

It is noted that the RTS message is expanded with a delay field and the CTS message with a transmission power field. Delay field assists packet transmission decision making and a more precisely estimated transmission power level not only can provide efficient power consumption, but also can reduce interference for efficient spectrum reuse. The newly added RTR message makes potential transmitters aware of the receivers in neighborhood and assist in efficient transmission configurations. Moreover, the shift of transmission responsibility from transmitter to receiver effectively solves the primary collision problem that is inevitable in transmitter-initiated schemes. The MAC protocol described here gives transmission preference to voice packets over data packets, however, different design objectives can be achieved without any change to this signaling structure. For instance, based on channel estimation, a transmitter can select a candidate packet with the best channel quality for maximum throughput. Also, a transmitter can send out multiple transmission requests for all the neighboring receivers, and a receiver can reject some requests for traffic load balancing in networks. We do not consider these options in this paper.

B. GAIMD Fair scheduling

This section studies how to schedule a node either in transmission or in reception status alternately. Before discussing the proposed fair scheduling for transmission and reception at each node, we first review the self-regulated Transmission Control Protocol (TCP)[10] in internet. TCP does not assume any explicit knowledge about network internals and other sessions. If a congestion signal is captured, TCP sender aggressively reduces its congestion window (cwnd) by a half, or even reinitializes cwnd for severe congestion. Otherwise, TCP probes for unused bandwidth conservatively by enlarging cwnd by one segment per round-trip time. This self regulated congestion control is referred to as AIMD. Instead of the increase by one and decrease by half strategy, a more Generic AIMD (GAIMD)[11][12] algorithm uses a pair (α, β) to additively increases and multiplicatively decreases its cwnd. GAIMD scheme is inspiring in that: a) it is totally distributed, the congestion window is adaptively regulated based only on the local information; b) each node attempts to achieve maximum throughput by increasing congestion window additively if no congestion is sensed; c) yet the scheme can promptly react to congestion to achieve fairness among contending flows. So by modifying and applying these features, we propose a GAIMD based fair scheduling for distributed Ad Hoc networks.

At initial time, each node randomly selects an integer transmission window (twnd) $w(i), w \in [0, W], i = 0$, where W is the maximum transmission window size. The twnd represents the number of continuous data frames that a node is in transmission status, and a node with $w(i) = 0$ indicates that it is in reception status. When $w(i) > 0$, a node is in transmission status and its twnd decreases by 1 every data frame till it reaches 0. Then the node serves as a receiver for potential data packets destined to this node. $w(i)$ number of continuous data transmission frames plus 1 frame for reception are defined as the ith transmission cycle. If data packets are received during reception frame, which means that neighboring nodes are contending radio resource with this node, the node aggressively decreases its twnd in the $(i+1)$th transmission cycle to $\beta, 0 \le \beta \le 1$, of its previous value $w(i+1) = \beta w(i)$. Otherwise, if there is no data packets received in reception frame, the node increases its twnd by $\alpha, \alpha \ge 0$, $w(i+1) = w(i) + \alpha$. If a node has no waiting packet in buffer, it stays in reception status till there is newly generated packet. A scheduling procedure for node N1 is shown in Fig. 6 where the (α, β) pair is set to $(1, 0.5)$. T rep-

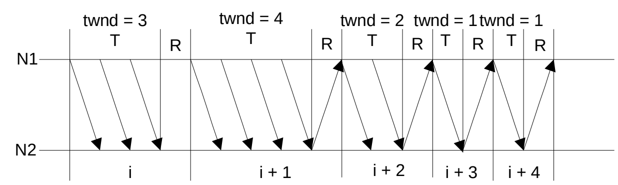

Fig. 6. Transmission and reception

resents the transmission frame and R represents the reception frame. In the ith transmission cycle, N1 has twnd equal to 3. When there is no transmission request received during reception frame, N1 increases its twnd by 1 in the $(i+1)$th cycle. Once a transmission request is detected in the $(i+1)$th cycle, N1 aggressively decreases its twnd by half in the $(i+2)$th transmission cycle as shown in Fig. 6. The twnd is further decreased to 1 in the $(i+3)$th transmission cycle when one more packet request is detected in the $(i+2)$th transmission cycle. Then N1 and N2 fair share the radio resources with average throughput of 0.5 packets/frame.

In general, when a new traffic flow injects into the network contending for radio resources, the existing flow will reduce its transmission rate exponentially with the twnd of $w(i+j) = \beta^j w(i)$ after j number of transmission cycles. Here we assume all nodes have the same weight or priority. The maximum twnd W indicates how efficiently the radio resource is utilized and how quickly the system reacts to traffic variation. A small W can react to newly generated traffic rapidly, but the radio resource is wasted as a heavily loaded node will have to schedule to reception status every $W+1$ data frames. On the contrary, a large W value can support a long period of transmission without interruption, but it will take long for the newly added traffic to achieve its maximum transmission rate. As the long convergence time cannot adapt to the fast varying traffic load promptly, a moderate maximum transmission window W is preferred in an unpredictable Ad Hoc network configuration.

As each receiver schedules its transmission and reception frames in a distributed way, deadlock happens when two or more competing nodes have the same transmission window and are synchronized in transmission and reception frames. Deadlocked nodes cannot establish communication with each other. Nevertheless, this phenomenon is very scarce due to two facts. First, it is noted that neither α nor β needs to be an integer. Therefore, it is straightforward that either $w(i+1) = w(i) + \alpha$ or $w(i+1) = \beta w(i)$ is not necessary an integer as well. A practical $w(i+1)$ with transmission window size increased by α can be $\lceil w(i+1) \rceil$ or $\lfloor w(i+1) \rfloor$ based on the following algorithm written in pseudocodes, where $\alpha \geq 0$ and $Rand()$ is a random variable generator between $[0, 1]$.

1 If $(\alpha > 1)$
1.1 $w(i+1) = w(i) + \lfloor \alpha \rfloor$;
1.2 If $(Rand() < \alpha - \lfloor \alpha \rfloor)$
1.2.1 $w(i+1) + +$; // $w(i+1) = w(i) + \lceil \alpha \rceil$
2 Else
2.1 If $(Rand() < \alpha)$
2.1.1 $w(i+1) + +$; // $w(i+1) = w(i) + \lceil \alpha \rceil$
2.2 Else
2.2.1 $w(i+1) = w(i)$; // $w(i+1) = w(i) + \lfloor \alpha \rfloor$

Based on this algorithm, with $(\alpha - \lfloor \alpha \rfloor)$ probability, $w(i+1) = w(i) + \lfloor \alpha \rfloor + 1$, and with $1 - (\alpha - \lfloor \alpha \rfloor)$ probability, $w(i+1) = w(i) + \lfloor \alpha \rfloor$. So, with probability $(\alpha - \lfloor \alpha \rfloor)(1 - (\alpha - \lfloor \alpha \rfloor))$, two deadlocked nodes are unlocked. Besides in a practical network where there are usually more than 2 contending nodes in neighborhood, if 2 nodes are synchronized and deadlocked, the other contending nodes can also unlock them by randomly injected traffic since the probability of deadlock among more than 2 contending nodes is very small. $w(i+1) = \beta w(i)$ is treated the same way and we will not go into details here due to the space limit.

IV. Simulation Results

A discrete event driven simulator written in C language is developed to evaluate the performance of the proposed receiver initiated MAC and GAIMD scheduling schemes. To compare with the scheme proposed in [1], the simulation environment set up for [1] is used in this simulation as described below. The carrier frequency is 450Mhz and the spreading gain is set to 128. Each channel supports a transmission rate of 1Mbps. We focus on a circular simulation area with radius of 1km accommodating 60 nodes. A mobility model which mimics human and vehicle movement behavior is applied [13]. The speed limit is 50km/h. To avoid boundary effect, the nodes moving out of the circular area will reenter the simulation area. We assume a reliable wireless communication and a free space propagation model. The transmission frame is synchronized with a interval of 10msec. Each node has the maximum transmission power of 33dBm. The neighborhood threshold is set to -46dBm and the one sided noise PSD is 10^{-9}W/Hz.

The number of packets a node generating to each of its transmission neighbors is proportional to the received power. Packets are uniformly generated with average rate of p per node. Voice packets occupy 20% of the total generated packets. With BPSK modulation, the voice and data traffic have SINR requirement of 7dB and 10dB, respectively, for service satisfaction. The delay bound for voice and data traffic are set to 15 and 150 frames, respectively. As a destination node may move out of the neighborhood of a sender, waiting packets with this destination are reassigned a destination selected from the updated transmission neighbor list. Without stated otherwise, each connection supports a fixed transmission of one packet during each frame. The maximum transmission window size is set to $W = 10$ and the duration of each simulation experiment is 20000 frames.

To validate the fairness of the proposed GAIMD scheduling, we use fairness index defined as [14]

$$F_{index} = \frac{(\sum_f T_f/\phi_f)^2}{N \times \sum_f (T_f/\phi_f)^2}, \tag{5}$$

where T_f denotes throughput of flow f, and ϕ_f denotes weight of flow f. Because this work considers only packet level, so a flow is replaced by a node and the fairness among flows is replaced by fairness among competing nodes. Because each node is equally distributed in both functionality and location, so here ϕ_f is the same for each node.

First we select an optimal (α, β) pair which can achieve maximum throughput and at the same time provide QoS satisfaction. Here the QoS parameter we are interested in is the average voice queueing delay in one hop's transmission. The neighborhood capacity is set to $M = 16$ and the traffic load is set to $p = 0.8$. Figs. 7, 8 and 9 show the obtained throughput, the one hop voice packet delay and the fairness index with the varying β values as x-axis. The figures plotted with varying α values are not shown here since the same optimal (α, β) pair is chosen. It is seen that when (α, β) is set to $(1, 0.75)$,

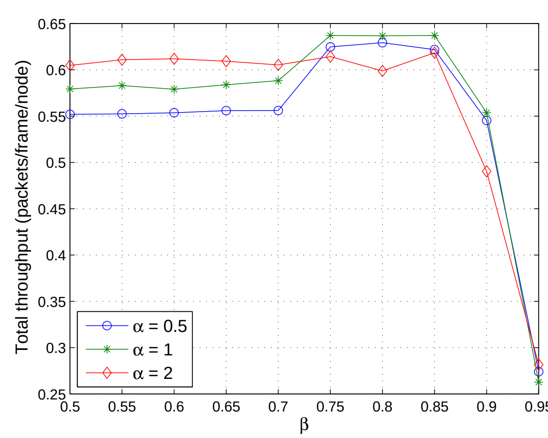

Fig. 7. Throughput versus (α, β) pairs

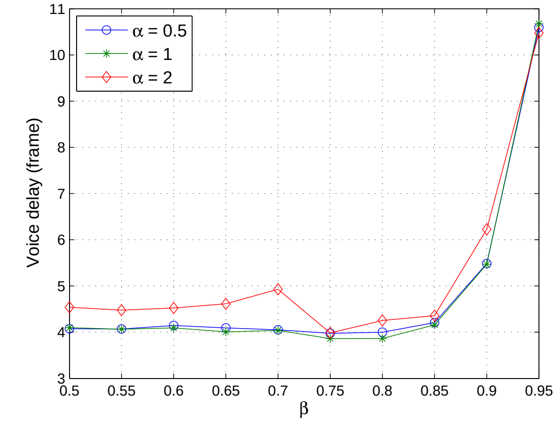

Fig. 8. Voice delay versus (α, β) pairs

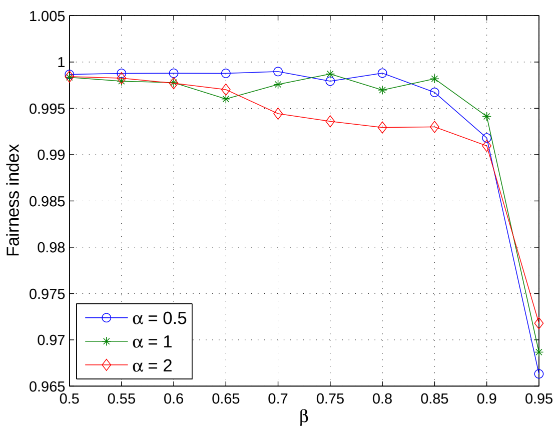

Fig. 9. Fairness index versus (α, β) pairs

the throughput can reach the maximum value of 0.637 packets/frame/node. Meanwhile, the voice delay achieves the minimum value. When β is small, the twnd is reduced significantly on detection of resource contention. Therefore, nodes are generally scheduled to receive frequently and the radio resource is wasteful due to frequent transmission interruption. On the other hand, with large β values, the throughput also decreases significantly because nodes in transmission status dominate the system with very small number of receivers. The effect of α is also straightforward as shown in Fig. 7. A small α value results in more receivers and a large α value causes too many transmitters in the system. Both these two cases cannot achieve good throughput performance. Fig. 9 also verifies that the proposed receiver initiated GAIMD scheduling can achieve fairness among contending nodes.

We now apply the selected $(\alpha, \beta) = (1, 0.75)$ to study the effectiveness of the proposed receiver initiated MAC and GAIMD scheduling schemes (referred to as R-GAIMD). Comparisons are made with the random transmission and reception scheduling scheme (referred to as RAND) and the transmitter initiated MAC and scheduling scheme proposed in [1] (referred to as T-initiated). In RAND case, the transmission window is a random variable which is uniformly distributed between $[0, W_{RAND}]$. We apply $W_{RAND} = 4$ for comparison purpose as it achieves the maximum throughput when $M = 16$ and $p = 0.8$.

Fig. 10 shows the total throughput with varying traffic load p when $M = 16$. It is seen that the proposed R-GAIMD

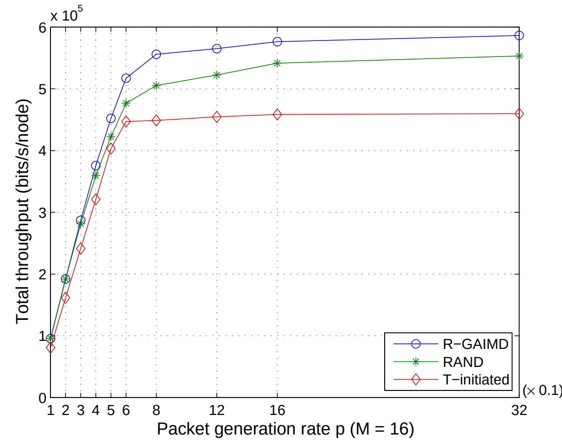

Fig. 10. Throughput comparison with varying traffic load

MAC scheme outperforms the RAND scheme, and both receiver initiated MAC scheme outperform the transmitter initiated T-optimal MAC scheme with higher obtained throughput. This is because with receiver initiated MAC schemes, the transmitters are aware of the receivers in neighborhood and can always find waiting packets for efficient transmission organization. Nevertheless, Fig. 11 shows that the transmitter initiated T-initiated MAC scheme can achieve a better voice delay performance. This is because the T-initiated MAC scheme associates transmitters and receivers based on the knowledge of packet priorities of contending nodes so that the scheduling configuration is for fairness preference. On the other hand in the receiver initiated MAC design, nodes are blind of the packet priorities of contending nodes so the average queueing delay is increased.

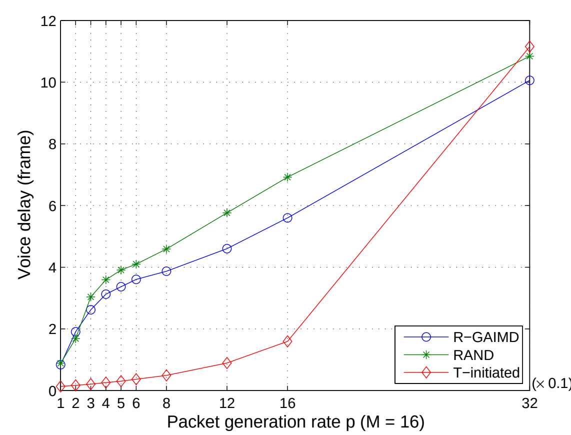

Fig. 11. Voice delay comparison with varying traffic load

Fig. 12 presents the throughput comparisons of the same three schemes with the varying neighborhood capacity M for given traffic load of $p = 0.8$. It is seen that when the number of neighborhood capacity is small, $M = 1, 2, 3$ and 4, the transmitter initiated T-initiated fair scheduling scheme outperforms both the R-GAIMD and RAND schemes. However, as M further increases, the throughput obtained with T-initiated scheme decreases rapidly because of: (1) the linearly increasing signaling overhead with M; (2) the high possibility of blocked transmissions due to power limitation as transmitters do not have enough knowledge of potential receivers around. And the proposed R-GAIMD fair scheduling outperforms the T-initiated fair scheduling by around $17\%, 22\%$ and 49%, respectively when $M = 12, 16$ and 32. Fig. 13 compares the average voice queueing delay. As expected, the transmitter initiated scheme can achieve better delay performance compared to the receiver initiated R-GAIMD and RAND schemes. However, we observe that even under the extreme heavy traffic load of $p = 3.2$ or at single reception capability condition of $M = 1$, the proposed scheme still can guarantee the voice delay within the delay bound of 150msec. Therefore, the proposed scheme can work properly within at least one hop distance for tolerable conversations.

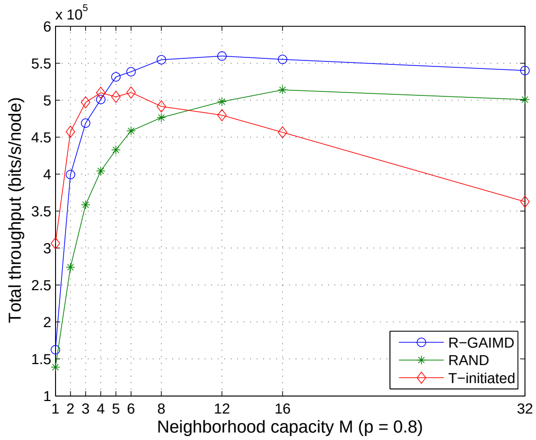

Fig. 12. Throughput comparison with varying neighborhood capacity

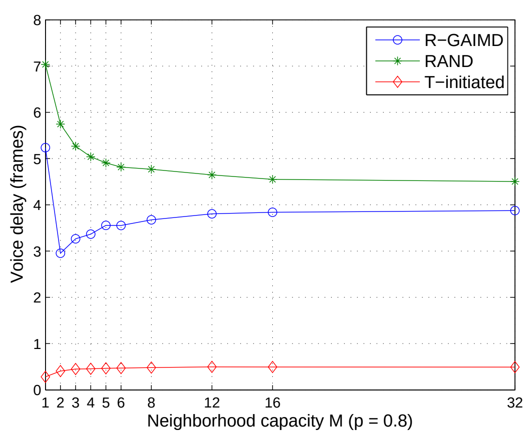

Fig. 13. Voice delay comparison with varying neighboring capacity

V. Conclusions and Future Work

A novel receiver-initiated MAC protocol and a GAIMD fair scheduling scheme based on MUD are proposed and their performance are evaluated. It is observed that the MAC protocol is simple to implement and only little modification is needed to make it compatible with the commercial 802.11 products. By broadcasting one extra signaling message RTR from potential receivers and adaptive transmission and reception frame scheduling at each node, the proposed GAIMD fair scheduling scheme can efficiently organize transmissions to achieve both throughput improvement and fairness provisioning. As this work assumes a perfect wireless transmission environment, research work on a more practical channel model and the associated opportunistic scheduling with crossed physical and MAC design is undergoing.

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