# A New MAC Scheme Supporting Voice/Data Traffic in Wireless Ad Hoc Networks

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**Abstract**—In wireless ad hoc networks, in addition to the well-known hidden terminal and exposed terminal problems, the location-dependent contention may cause serious unfairness and priority reversal problems. These problems can severely degrade network performance. To the best of our knowledge, so far, there is no comprehensive study to fully address all these problems. In this paper, a new busy-tone-based medium access control (MAC) scheme supporting voice/data traffic is proposed to address these problems. Via two separated narrow-band busy-tone channels with different carrier sense ranges, the proposed scheme completely resolves the hidden terminal and exposed terminal problems. Furthermore, with the use of transmitter busy tones in the node backoff procedure, the proposed scheme ensures guaranteed priority access for delay-sensitive voice traffic over data traffic. The priority is also independent of the user locations, thus solving the priority reversal problem. The fairness performance for data traffic in a nonfully connected environment is also greatly improved (as compared with the popular IEEE 802.11e MAC scheme) without the need of extra information exchanges among the nodes.

**Index Terms**—Wireless ad hoc networks, medium access control, fairness, busy tone, priority, hidden and exposed terminals.

## 1 Introduction

In recent years, with an increasing demand for multimedia applications (including voice, video, and data), wireless ad hoc networks are expected to provide heterogeneous services. Different applications have different quality-of-service (QoS) requirements. Transmission delay and jitter are the main QoS parameters of real-time traffic. Throughput and fairness are the QoS indication for data traffic. QoS provisioning at the medium access control (MAC) layer is desired in order to provide priority access to real-time traffic in the presence of data traffic, meanwhile achieving a certain level of QoS for data. Also, due to the scarce radio resources, the MAC scheme must ensure efficient channel utilization.

In contrast to centrally controlled networks (e.g., cellular networks [1], [2] or infrastructure-based WLANs [3]), a distributed MAC scheme should be implemented among all the nodes to coordinate the transmissions in ad hoc networks without a central controller. Developing such a distributed MAC scheme is not a trivial task because each node does not have the complete information of other contenders, and there is no efficient way to let one node control the behaviors of others. Further, because of the limited node transmission ranges, wireless ad hoc networks are usually nonfully connected. The nonfully connected environment presents more challenges to the MAC scheme design. The hidden terminals bring more collisions, and the exposed terminals lead to inefficiency of channel utilization [4]. Moreover, in a nonfully connected environment, the locations of the contending flows may significantly affect the channel access opportunity of each flow. The location-dependent contention results in serious unfairness (starvation of some flows) and priority reversal problems (i.e., a high-priority flow gets less chances to access the channel than its low-priority counterpart) [5].

In the literature, some distributed MAC schemes have been proposed, to address one or more of the aforementioned problems in wireless ad hoc networks:

- For hidden terminal problem—To alleviate the hidden terminal problem, a request-to-send (RTS)/clear-to-send (CTS) dialog is used in many MAC schemes (e.g., in the most popular IEEE 802.11 and IEEE 802.11e MAC [6]). However, the RTS/CTS dialog is less effective to avoid collisions in a relatively crowded region with hidden terminals [7]. Another popular approach is to protect the receiver's DATA frame reception by an additional busy-tone channel (which is separated from the information channel) to indicate whether or not the receiver is receiving a DATA frame [4], [8]. The busy-tone solution avoids DATA frame collisions. However, RTS collisions caused by hidden terminals cannot be avoided. On the other hand, for voice transmissions, due to the small DATA frame size, usually no RTS/CTS dialog is adopted. None of the preceding schemes gives a solution to avoid voice-DATA frame[1] collisions caused by hidden terminals.

1. In this paper, a voice-DATA frame means a DATA frame from a voice traffic source, while a data-DATA frame means a DATA frame from a data traffic source.

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- For exposed terminal problem—The dual busy tone multiple access (DBTMA) [4] has been proposed via dual busy-tone channels. However, acknowledgment (ACK) frames are omitted in DBTMA (which is not reasonable for unreliable wireless channels) because, otherwise, collisions may occur when a sender is receiving an ACK frame while another nearby sender is transmitting a DATA frame.
- For priority access and fairness—Most of the previous work focuses on fully connected networks. Only limited work addresses the characteristics of a nonfully connected environment. A contention-based scheme is proposed to provide priority scheduling in nonfully connected networks using busy tones [5]. A reservation-based scheme is presented in [9], under the assumption that all the nodes are synchronized, which is difficult to implement in practice. In [10], an ideal fairness model is proposed for ad hoc networks, taking spatial reuse into consideration. This model requires complete information of network topology, imposing extreme complexity for implementation. In self-coordinating localized fair queueing [11], the service tag information should be exchanged among neighboring nodes, leading to a certain level of information exchange overhead.

To the best of our knowledge, so far, there is no comprehensive study to fully address all the problems of hidden terminals, exposed terminals, unfairness, and priority reversal associated in a nonfully connected environment. Without solving all these problems, QoS provisioning for multimedia applications in ad hoc networks is difficult to achieve. The contribution of this paper lies in that it is the first work to propose an effective MAC scheme to address all these problems. As a follow-up of our previous work [12] (which focuses on a fully connected network), this work aims at addressing the hidden/exposed terminal, long-term unfairness, and priority reversal problems associated with a nonfully connected network. Our proposed MAC scheme utilizes two narrow-band busy-tone channels and one information channel. Similar to all other busy-tone schemes [4], [13], extra hardware cost is incurred to implement the busy-tone channels. However, as mentioned in [4], the wireless transceiver architecture proposed in [14] can help to set up the busy-tone channels with low hardware cost.

The rest of this paper is organized as follows: Section 2 describes the system model and the problems associated with a nonfully connected environment. The proposed MAC scheme is presented in Section 3, and its performance is analyzed in Section 4. Section 5 is devoted to the numerical performance evaluation of the proposed scheme, followed by concluding remarks in Section 6.

## 2 The System Model and Problem Statement

Consider a wireless ad hoc network where real-time voice and elastic data applications are supported. A voice application generates constant bit rate (CBR) voice packets with a fixed packet size. Data traffic flows are long-lived file transmissions. Voice traffic is assigned a higher priority over data traffic. There is a single information channel in the network, through which all the nodes send their frames. Any overlap of transmissions at a receiver causes a collision, and none of the overlapped frames can be correctly received. The successful simultaneous transmissions that do not interfere with each other are possible due to spatial reuse. Since this research focuses on the MAC layer, one-hop transmissions are considered. The physical layer of the ad hoc network is IEEE 802.11b based since it is the most widely adopted technology.

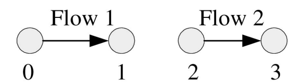


Fig. 1. Priority reversal due to location-dependent contention.

In the following, we take the IEEE 802.11 MAC as an example to describe the problems to be addressed in this research. Note that these problems are not just associated with the IEEE 802.11 MAC. Many contention-based MAC schemes (e.g., [15], [16], and [17]) that use a backoff mechanism similar to that of IEEE 802.11 may suffer from the same problems.

### 2.1 Hidden and Exposed Terminal Problems

These are two notorious problems and have been discussed extensively in the literature (e.g., [4] and references therein). To alleviate the collisions caused by hidden terminals, the RTS/CTS approach is widely adopted. However, RTS and CTS frames themselves are still subject to collisions. Although busy-tone-based schemes (e.g., [4]) have been proposed to solve the CTS collision problem, RTS frame (and voice-DATA frame that does not have a prior RTS/CTS handshaking) collisions caused by hidden terminals cannot be avoided. According to IEEE 802.11b, the RTS frame size is 20 bytes. Considering a physical layer overhead (192 $\mu$s), 272 $\mu$s is needed to transmit an RTS frame at the required basic rate (i.e., 2 Mbps). With one slot time equal to 20 $\mu$s, one RTS transmission time approximately equals 14 slots. Consider two terminals that are hidden to each other and have not suffered from collisions previously. They randomly pick up their backoff timers from the initial contention window (CW) denoted by $CW_{min}$ (31 in IEEE 802.11b) and start to count down their backoff timers simultaneously. The probability that their RTSs collide with each other is as high as 66.6 percent (the probability that, for two integers each randomly chosen from 0 to 31, the difference between them is less than 14). For voice-DATA frames (without a preceding RTS/CTS dialog), this probability can be higher since a voice-DATA frame size is normally larger than an RTS frame size. If more than two hidden terminals exist, the collision probability will be higher, resulting in a reduced system throughput. In order to achieve a large throughput in the network with hidden terminals, protection of only data-DATA frames is not sufficient. RTS and voice-DATA frame collisions caused by hidden terminals should also be completely avoided, if possible.

### 2.2 Priority Reversal Problem

An example of priority reversal problem [5] is shown in Fig. 1, where flow 1 (from node 0 to node 1) has a higher priority than flow 2 (from node 2 to node 3). Flow 1 and flow 2 conflict with each other since node 1 and node 2 are neighbors. It is likely that flow 1 may lose its priority when competing with flow 2. The reason is that node 2 is a hidden terminal of node 0 and cannot be aware of the transmission of node 0. Even though node 0 may start its transmission earlier than node 2, it is possible that node 2 starts its transmission to node 3 before node 0 completes its transmission to node 1 (i.e., their transmission durations overlap), resulting in a collision at node 1. The reception at node 3 is successful in this scenario. As a result, the low-priority flow (i.e., flow 2) rather than the high-priority flow (i.e., flow 1) delivers its frame successfully, which is not desired.

Furthermore, in a nonfully connected environment, it is possible that a node having a high-priority flow and more neighbors experiences a higher contention level than its contenders with low-priority traffic and fewer neighbors. As a result, the priority access cannot be ensured either [5].

### 2.3 Long-Term Unfairness Problem

Although the IEEE 802.11 MAC is characterized by inherent short-term unfairness [18], it does have a good performance of long-term fairness[2] in a fully connected network. However, when applied to a nonfully connected environment, the IEEE 802.11 MAC suffers from serious long-term unfairness, i.e., some flows may be starved. Consider the same topology shown in Fig. 1, where flow 1 and flow 2 have the same priority. When the IEEE 802.11 MAC is deployed, flow 1 is almost starved and flow 2 occupies almost all the channel capacity for the following reason. When node 2 is transmitting, node 1 cannot correctly receive node 0's RTS. Without getting the response from node 1, node 0 will keep retransmitting the RTS without success. Its CW size is doubled each time when the RTS transmission fails, and eventually reaches the maximum value $\mathrm{CW}_{\max}$. On the other hand, if node 0 is transmitting, node 2 knows exactly the ending time of node 0's transmission (by overhearing node 1's CTS), and thus defers its own transmission until node 0 finishes its transmission. Hence, node 2 maintains the minimum CW size $\mathrm{CW}_{\min}$. As a result, node 0 is unlikely to get the channel (due to the much larger CW size) and will get starved.

## 3 The Proposed MAC Scheme

In our proposed MAC scheme, the total channel bandwidth is divided into three parts with sufficient spectral separation (which is similar to DBTMA [4]): information channel, transmitter busy-tone (BTt) channel, and receiver busy-tone (BTr) channel. The difference of our scheme from DBTMA is that, by adjusting the receiver's sensitivity, we set the channels' carrier sense ranges such that the BTt channel's carrier sense range covers the two-hop neighborhood of the sensing node, while the BTr channel's carrier sense range covers the one-hop neighborhood of the sensing node.[3] For presentation clarity, we assume that, when a node is receiving a frame, only its one-hop neighbors' transmissions may corrupt its reception. In reality, a node's interference range may be larger than its transmission range, so the nodes beyond one hop of a receiver may still be able to corrupt the reception. In this case, our scheme can still work if we adjust the BTt channel's carrier sense range to be the interference range plus the transmission range, and the BTr channel's carrier sense range to be the interference range. Further discussion on this is omitted.

Similar to the IEEE 802.11e, voice and data traffic are assigned different arbitration interframe space (AIFS) values, i.e., AIFS[voice] $<$ AIFS[data]. Before its contention, each contending node should wait for the two busy-tone channels idle for a duration of its AIFS. Each node also keeps a backoff timer, the initial value of which is randomly selected from its CW. After the AIFS idle time of the two busy-tone channels, the node starts to send a busy tone in the BTt channel (instead of starting to count down its backoff timer, as in the IEEE 802.11e). The duration of the busy tone equals its backoff timer (in the unit of slot time). Upon the completion of its busy tone, the node senses the BTt channel again. If a busy BTt channel is sensed (i.e., another node is sending a busy tone), the node selects a new backoff timer (from its current CW), and starts its busy tone after AIFS idle time again of both the busy-tone channels. If the BTt channel is idle after the node finishes its busy tone transmission:

- For the case of voice traffic, the voice transmitter sends its voice-DATA frame, and simultaneously sends a busy tone in the BTt channel until the completion of the voice-DATA frame, for the purpose of protecting the voice-DATA frame from being corrupted by hidden terminals (to be discussed in Section 3.1). Upon reception of the voice-DATA frame, the receiver sends a busy tone in the BTr channel, which serves as an ACK.
- For the case of data traffic, the data transmitter sends an RTS frame, and simultaneously sends a busy tone in the BTt channel until the completion of the RTS to prevent interferers. Upon reception of the RTS, the data receiver sends a busy tone in the BTr channel, which serves the same function as CTS. Upon reception of the BTr busy tone, the data transmitter transmits its data-DATA frame. When the data receiver is receiving the data-DATA frame, it keeps sending a BTr busy tone to prevent interferers. If data-DATA frame has been received successfully, the data receiver continues to send a BTr busy tone for a small duration (i.e., the busy-tone detection time), which serves as an ACK.

If the traffic source node (i.e., voice or data transmitter) does not receive the BTr busy tone after its transmission of an RTS or DATA frame, a collision is inferred. The source node will double its CW (until the maximum value $\mathrm{CW}_{\max}$ is reached), select a new backoff timer, and start its next contention after the two busy-tone channels have been sensed to be idle for its AIFS again. The CW is reset to the initial value $\mathrm{CW}_{\min}$ upon a successful transmission. Note that voice and data nodes keep the same $\mathrm{CW}_{\min}$ and $\mathrm{CW}_{\max}$ in our scheme (to be further discussed in Section 3.3), unlike the IEEE 802.11e.

Details of the operation procedure of the proposed MAC scheme are presented in Appendix A. The subsequent four sections will demonstrate how our proposed scheme can

2. For long-term fairness, fair service shares are achieved among all the contending nodes in a relative large time scale (e.g., 10 seconds). On the other hand, short-term fairness should be achieved in a small time scale (e.g., 10 ms).

3. All the nodes that are within the transmission range of a node (say node A) are one-hop neighbors of node A. All the nodes that are beyond the transmission range of node A but within two times the transmission range of node A are two-hop neighbors of node A. In a wireless network, the carrier sense range varies with the receiver's sensitivity. The feasibility of such a setting method can be found in [19].

solve the problems of hidden terminal, exposed terminal, priority reversal, and unfairness, respectively.

### 3.1 Solution to the Hidden Terminal Problem

To achieve good performance, not only data-DATA frame collisions but also RTS and voice-DATA frame collisions caused by hidden terminals should be completely avoided if possible. In our MAC scheme, the use of an increased carrier sense range in the BTt channel can help to achieve this target. To protect an RTS (or a voice-DATA) frame from being corrupted by hidden terminals, when a sender starts to transmit its frame, it also transmits a busy tone in the BTt channel, and stops it when the RTS (or voice-DATA) frame transmission is finished. Because of the increased carrier sense range of the BTt channel, all the potential hidden terminals that may interfere with this ongoing transmission can sense the BTt channel being busy, and thus defer their own transmissions and avoid corrupting the RTS (or voice-DATA) frame transmission. Further, for data traffic, when the sender completes the RTS transmission and the destination node recognizes that it is the intended receiver, the destination will send a busy tone immediately in the BTr channel (i.e., serves the same function as CTS). All the potential hidden terminals of the sender can hear this busy tone, thus deferring their transmissions. The destination continues sending the BTr busy tone during the whole data-DATA frame reception. Therefore, collision is avoided from the beginning of the RTS transmission to the end of the data-DATA frame transmission.

### 3.2 Solution to the Exposed Terminal Problem

With the use of the BTr busy-tone channel, our scheme can resolve the exposed terminal problem. When a desired receiver receives an RTS (or DATA) frame, instead of responding with a CTS (or ACK) frame in the information channel, the receiver sends a busy tone in the BTr channel that serves the same function as an CTS (or ACK) frame. After sending out an RTS (or DATA) frame, the sender senses the BTr channel. The status of a busy BTr channel indicates that the RTS (or DATA) frame has been successfully received by the receiver; otherwise, a collision has occurred. Replacing the CTS and ACK frames with the BTr busy tones allows multiple senders within one-hop neighborhood to send their frames simultaneously (as long as they do not interfere with each other at the receivers) without the problem that the feedback from a receiver may be corrupted by other ongoing DATA transmissions, since the feedback and DATA transmissions are in different channels.

To replace the CTS and ACK frames with the BTr busy tones, it is essential to ensure that, when a sender senses a BTr busy tone after completing its frame transmission, this busy tone must be from its own destination rather than from any other nodes (since the BTr busy tone does not carry any bit information). This is achieved in our scheme as follows: When a sender is sending an RTS (or DATA) frame, all the potential receivers (which are the destinations of other nodes) within this sender's one-hop neighborhood cannot correctly receive their own frames; and therefore, none of them will send the BTr busy tone as a feedback.

### 3.3 Solution to the Priority Reversal Problem

To address the priority reversal problem, it is desired to ensure the channel access priority for voice traffic independent of the node location. Our approach is to let all potential hidden terminals of the voice node be aware that the voice node is contending for the channel, so that they defer their own contentions. This is achieved with the use of the BTt busy tone. In our scheme, after waiting for both the busy-tone channels to be idle for an AIFS[voice], the voice node sends a BTt busy tone. Thus, for data nodes, if there exists one or more voice contenders within its two-hop neighborhood, they will sense the BTt busy tone (from voice nodes) during the AIFS[data] ($>$ AIFS[voice]), and defer their transmissions. Therefore, the voice node avoids the priority reversal problem no matter where it is located, benefiting from the doubled carrier sense range of the BTt channel.

Note that the proposed MAC scheme not only avoids priority reversal in a nonfully connected environment but also ensures guaranteed priority access for voice traffic. Although IEEE 802.11e and our scheme use the same AIFS settings, our scheme can achieve guaranteed priority access for voice over data in each contention, while IEEE 802.11e can only achieve statistical priority access[4] for voice over a long term. The advantage of our scheme comes from the different BTt busy tone starting moments of voice and data nodes. This also explains why the same $\mathrm{CW_{min}}$ and $\mathrm{CW_{max}}$ are adopted by voice and data nodes in our scheme, unlike the IEEE 802.11e.

### 3.4 Solution to the Unfairness Problem

In 802.11e, the nodes with the smallest backoff timer transmit. When a node transmits successfully, its CW is reset to the initial value, and thus, its chance to win the next contention is still large. When a packet transmission is collided, the CW of the source node is doubled (up to the maximum value), and thus, its chance to win the next contention is small.

On the contrary, due to the different BTt busy tone starting moments of voice and date nodes in our scheme, either of the following two scenarios will happen at the beginning of each contention: 1) one or more voice nodes start BTt busy tone transmission after AIFS[voice] idle time of both the busy-tone channels, and all data nodes defer; and 2) there is no voice contender, and one or more data nodes start BTt busy tone transmission after AIFS[data] idle time of both the busy-tone channels. This means that, if some nodes transmit BTt busy tones at the beginning of each contention, they should have the same AIFS value, and thus their BTt busy tone start instants should also be the same. Therefore, when a node senses an idle BTt channel after it completes its BTt busy tone (the condition for transmission of an RTS or a voice-DATA frame), this means that the node has the largest backoff timer among all the contending nodes, since the duration of its busy tone is equal to its backoff timer. In other words, in our

4. Statistical priority access means that the prioritized access for high-priority traffic is only guaranteed in a long term, but not for every contention. Since each node continues to count down its backoff timer once the channel becomes idle for an AIFS, a low-priority node with a probably large initial backoff timer will eventually count down its backoff timer to a small value, most likely smaller than the backoff timer of a newly backlogged high-priority node. Then, the low-priority node gains the channel, resulting in the high-priority node waiting for a long time for the next competition [5].

scheme, the node(s) with the largest backoff timer transmit. Upon a successful transmission, a node's CW is reset to the initial value, so that its chance to have the largest backoff timer among all the nodes and win the next contention is small. Upon a collision of its packet, the node's CW is doubled (up to the maximum value), so it has a large chance to have the largest backoff timer among all the nodes and win the next contention. This means that the channel access time is shared more fairly among the contending nodes in our scheme than in IEEE 802.11e. Since the BTt busy-tone channel has a larger carrier sense range, fairness can be achieved in a larger range. Use Fig. 1 as an example. Suppose that after one contention, node 2 resets its CW to $\mathrm{CW}_{\min}$ (because of a successful transmission) while node 0 doubles its CW (because of the failed RTS). Then, in the next contention, it is very likely that node 0 has a larger backoff timer than node 2, and thus sends a longer BTt busy tone (which can be heard by node 2). Therefore, node 0 will win the access to the channel and node 2 will defer its transmission.

It may seem that the waiting time (before getting the channel) of a node is longer with our scheme than that with IEEE 802.11e, since the node with the largest backoff timer gets the channel. However, as shown in our previous work [12], the CW setting can be set to small values in our scheme. The CW setting {3:15} (i.e., $\mathrm{CW}_{\min} = 3$ and $\mathrm{CW}_{\max} = 15$) works well for a large number (up to 500) of contending nodes, and the throughput and collision probability are quite stable when the number of contending nodes increases (discussed in details in Appendix B). Therefore, the negative effect of the longer backoff time can be neglected. The insensitivity of the system performance to the number of nodes also facilitates network configuration.

The idea to let the node with the longest busy tone win the channel is inspired by the black-burst scheme [20]. However, the original idea of black-burst is proposed to provide QoS guarantee for real-time traffic and cannot be directly applied to solve the unfairness problem. Here, we adopt the "jamming" nature of black-burst, and modify the backoff procedure (as aforementioned). A result of the modification is good fairness performance. Furthermore, in the original black-burst scheme, only two traffic classes are supported. Our scheme can support more traffic classes as long as they have different AIFS values, though only two traffic classes are considered in this paper.

### 3.5 Advantages of Our Scheme Compared with Related Work

Although both our scheme and DBTMA have two busy-tone channels, they are different in several aspects: 1) The double sense range of the BTt channel in our scheme can solve the RTS and voice-DATA frame collision problem caused by hidden terminals (which cannot be solved by DBTMA).[5] 2) DBTMA solves the exposed terminal problem by omitting the ACK messages, which is impractical for unreliable wireless channels. On the contrary, our scheme solves the exposed terminal problem by letting DATA frames and their acknowledgments be transmitted in different channels. 3) In DBTMA, the BTt busy tone is sent only when the DATA frame is sent. In our scheme, the BTt busy tone is sent not only during DATA frame transmissions but also during the backoff procedure. The extra use of BTt busy tones in the backoff procedure brings about more significant benefits, i.e., it provides guaranteed priority access for voice over data by different starting points of the BTt busy tones, improves fairness performance because the backoff procedure enhanced with the BTt busy tones favors previously unsuccessful nodes, and solves the priority reversal problem.

In terms of fairness, there are a number of research efforts [11], [21] in the literature to enhance the fairness performance of carrier sense multiple access (CSMA)-based MAC. However, information of other nodes (such as received service, service tag, etc.) should be exchanged and stored at a node, thus leading to an extra information exchange overhead. On the contrary, in our scheme, the backoff procedure automatically enhances the fairness performance, without extra overhead. On the other hand, fairness and throughput are two contradicting objectives in terms of network performance [22]. There exists a tradeoff between these two objectives. The goal of this research is not to achieve absolute fairness (which is normally obtained at the cost of sacrificing channel utilization), but to avoid serious unfairness problem (i.e., some flows are starved) that is undesired to both customers and network service providers. Meanwhile, throughput is another QoS metric in our work. We increase the channel utilization by avoiding collisions due to hidden terminals and by letting exposed terminals transmit simultaneously.

5. Note that the proposed scheme can completely solve the RTS/voice-DATA collisions caused by hidden terminals in a reasonably good propagation environment. However, if there are obstacles between the sender and the hidden terminals, the collisions may still happen. In the case, the proposed scheme can alleviate but not completely solve the problem.

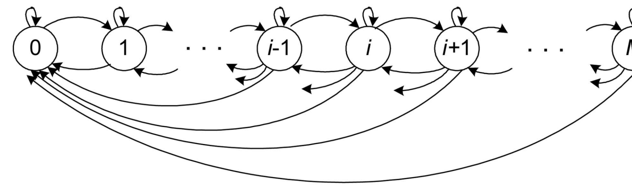

Fig. 2. The state transition diagram of $m(t)$.

## 4 PERFORMANCE ANALYSIS

In this section, we present the analysis of data throughput in our proposed scheme in a fully connected network without voice traffic. As the analytical result for a nonfully connected network with voice traffic is difficult to obtain, we resort to extensive simulations for the performance in a nonfully connected network.

Consider $M$ data source nodes. For simplicity of presentation, the CW of each data node takes values from the set $\{\mathrm{CW}_1, \mathrm{CW}_2\}$ (i.e., $\mathrm{CW}_{\min}[\mathrm{data}] = \mathrm{CW}_1$, $\mathrm{CW}_{\max}[\mathrm{data}] = \mathrm{CW}_2$), where $\mathrm{CW}_2 = 2 \cdot (\mathrm{CW}_1 + 1) - 1$. Our analysis can be easily extended to the cases with more choices of CWs. Let $m(t)$ denote the number of data nodes with CW size $\mathrm{CW}_1$ at time instant $t$, and therefore, $M - m(t)$ data nodes are with CW size $\mathrm{CW}_2$. Define a *transmission event* as a successful transmission or a collision. We sample the value of $m(t)$ at the beginning of each transmission event, and form a discrete-time Markov process, as shown in Fig. 2.

For state $m(t) = i$, let $j_1$ and $j_2$ denote the number of nodes that transmit in the next transmission event, with CWs $\mathrm{CW}_1$ and $\mathrm{CW}_2$, respectively.[6] The probability of the largest backoff timer value $l$ in such a transmission event (i.e., $j_1$ nodes with $\mathrm{CW}_1$ and $j_2$ nodes with $\mathrm{CW}_2$ choose a backoff timer $l$ and all other nodes choose backoff timers less than $l$) is given by (1). In (1), $j_1 + j_2 \geq 1$. The event is a successful transmission if $j_1 + j_2 = 1$, or a collision if $j_1 + j_2 > 1$.

$$P_{j_1,j_2,l|i} = \begin{cases} \binom{i}{j_1}\left(\frac{1}{\mathrm{CW}_1+1}\right)^{j_1}\left(\frac{l}{\mathrm{CW}_1+1}\right)^{i-j_1} \\ \quad \cdot\binom{M-i}{j_2}\left(\frac{1}{\mathrm{CW}_2+1}\right)^{j_2}\left(\frac{l}{\mathrm{CW}_2+1}\right)^{M-i-j_2}, \\ \quad 0 \leq l \leq \mathrm{CW}_1, & \text{if } j_1 \neq 0, \\ \min\left\{1, \left(\frac{l}{\mathrm{CW}_1+1}\right)^{i}\right\} \\ \quad \cdot\binom{M-i}{j_2}\left(\frac{1}{\mathrm{CW}_2+1}\right)^{j_2}\left(\frac{l}{\mathrm{CW}_2+1}\right)^{M-i-j_2}, \\ \quad 0 \leq l \leq \mathrm{CW}_2, & \text{if } j_1 = 0. \end{cases} \tag{1}$$

Fig. 2 illustrates the state transition diagram of $m(t)$. For state $m(t) = i$, after a transmission event, the process will

- remain at state $i$ if one node with $\mathrm{CW}_1$ transmits successfully (i.e., $j_1 = 1$ and $j_2 = 0$) with probability $\sum_{0\leq l\leq \mathrm{CW}_1} P_{1,0,l|i}$, or a collision occurs in which no nodes with $\mathrm{CW}_1$ but at least two nodes with $\mathrm{CW}_2$ are involved (i.e., $j_1 = 0$ and $j_2 \geq 2$) with probability $\sum_{0\leq l\leq \mathrm{CW}_2,\ 2\leq j_2\leq M-i} P_{0,j_2,l|i}$;
- transit to state $i+1$ if one node with $\mathrm{CW}_2$ transmits successfully (i.e., $j_1 = 0$ and $j_2 = 1$), with probability $\sum_{0\leq l\leq \mathrm{CW}_2} P_{0,1,l|i}$; and
- transit to state $i-k(1 \leq k \leq i)$ if a collision occurs in which $k$ nodes with $\mathrm{CW}_1$ are involved, with probability $\sum_{0\leq l\leq \mathrm{CW}_1, 1\leq j_2\leq M-i} P_{1,j_2,l|i}$ when $k = 1$, or $\sum_{0\leq l\leq \mathrm{CW}_1,\ 0\leq j_2\leq M-i} P_{k,j_2,l|i}$ when $k > 1$.

Based on the transition probabilities among the states in Fig. 2, we can obtain the steady-state probabilities of all the states, $[\pi(0), \pi(1), \ldots, \pi(M)]$. Let $t_{\mathrm{slot}}$ denote a slot time, $t_s$ and $t_c$ the time durations of a successful transmission and a collision (not including the backoff time), respectively, given by

$$\begin{cases} t_s = \mathrm{AIFS[data]} + S_{\mathrm{RTS}}/R_{\mathrm{basic}} + t_{\mathrm{det}} + S_{\mathrm{d\_DATA}}/R + t_{\mathrm{det}}, \\ t_c = \mathrm{AIFS[data]} + S_{\mathrm{RTS}}/R_{\mathrm{basic}} + t_{\mathrm{det}}, \end{cases} \tag{2}$$

where $S_{\mathrm{RTS}}$ and $S_{\mathrm{d\_DATA}}$ are the RTS and data-DATA frame sizes in bits, respectively, $R_{\mathrm{basic}}$ and $R$ are the basic rate (for RTS transmission) and information transmission rate (for DATA transmission), and $t_{\mathrm{det}}$ is BTr busy-tone detection time (i.e., the sender detects the BTr busy tone after an RTS and/or a data-DATA frame transmission). Then, the average time in a transmission event of state $i$ is

6. Here, we omit the time index $t$ for $j_1$ and $j_2$.

TABLE 1
Simulation Parameters

| Parameter | Value |
|---|---|
| Slot time $t_{\mathrm{slot}}$ | 20 $\mu$s |
| SIFS | 10 $\mu$s |
| AIFS[voice] | 30 $\mu$s |
| AIFS[data] | 50 $\mu$s |
| $\mathrm{CW}_{\min}$ (for the proposed scheme) | 3 |
| $\mathrm{CW}_{\max}$ (for the proposed scheme) | 15 |
| $\mathrm{CW}_{\min}$[voice] (for IEEE 802.11e) | 15 |
| $\mathrm{CW}_{\max}$[voice] (for IEEE 802.11e) | 127 |
| $\mathrm{CW}_{\min}$[data] (for IEEE 802.11e) | 31 |
| $\mathrm{CW}_{\max}$[data] (for IEEE 802.11e) | 1023 |
| $\mathrm{CW}_{\min}$ (for DBTMA) | 15 |
| $\mathrm{CW}_{\max}$ (for DBTMA) | 255 |
| PHY preamble | 192 $\mu$s |
| MAC header | 36 bytes |
| $S_{\mathrm{RTS}}$ (RTS frame size) | 20 bytes |
| $S_{\mathrm{CTS}}$ (CTS frame size, IEEE 802.11e) | 14 bytes |
| $S_{\mathrm{ACK}}$ (ACK frame size, IEEE 802.11e) | 14 bytes |
| $S_{\mathrm{d\_DATA}}$ (data-DATA frame size) | 1000 bytes |
| Link rate (for IEEE 802.11e) | 11 Mbps |
| Information channel rate (for the proposed scheme and DBTMA) | 10.9 Mbps |
| Basic rate (RTS/CTS transmission) | 2 Mbps |
| $t_{\mathrm{det}}$ (busy-tone detection time) | 10 $\mu$s |

$$\overline{t_e(i)} = \sum_{l;\ j_1+j_2=1} P_{j_1,j_2,l|i} \cdot (l \cdot t_{\mathrm{slot}} + t_s) + \sum_{l; j_1+j_2>1} P_{j_1,j_2,l|i} \cdot (l \cdot t_{\mathrm{slot}} + t_c). \tag{3}$$

Thus, we can calculate the average system throughput as

$$T = \frac{S_{\mathrm{d\_DATA}} \cdot \sum_{0\leq i\leq M} \pi(i) \cdot \sum_{l;\ j_1+j_2=1} P_{j_1,j_2,l|i}}{\sum_{0\leq i\leq M} \pi(i) \cdot \overline{t_e(i)}}. \tag{4}$$

## 5 Performance Evaluation

To evaluate the performance of our proposed scheme, we compare it with IEEE 802.11e and DBTMA. We compare our scheme with IEEE 802.11e in all cases and with DBTMA in the cases with hidden and exposed terminals, since DBTMA focuses on the issues of hidden and exposed terminal problems, but not on priority and fairness issues. Since DBTMA does not explicitly specify its backoff mechanism, for fair comparison, we use the same backoff mechanism as that in our scheme. For DBTMA, only data traffic is considered. We choose the GSM 6.10 codec as the voice source as an example. The voice payload size is 33 bytes and the frame interarrival period is 20 ms. Long-lived data traffic is considered (each data node always has frames to send).

As shown in our previous work [12], both voice and data traffic can choose the CW setting {3:15} in our scheme. For IEEE 802.11e and DBTMA, it is not appropriate to use the same small CW sizes as those in our scheme. For 802.11e, the small CW setting leads to serious collisions and low throughput when the number of contending nodes increases. A detailed discussion is presented in Appendix B on the reason that our scheme but not IEEE 802.11e can choose a small CW setting. For DBTMA, from simulations, we find that a small CW setting leads to a very low throughput in the network with hidden terminals. The simulation parameter values are listed in Table 1. The

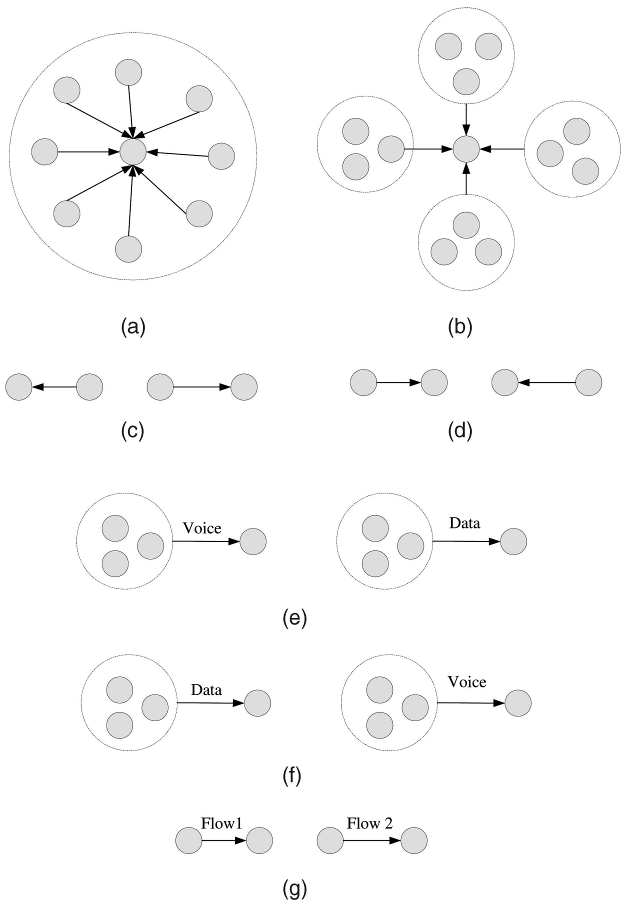


Fig. 3. The network topologies used in simulation.

system performance is first evaluated under some specific network topologies as shown in Fig. 3. Then, random topologies are simulated for more comprehensive evaluation. The simulation is done in Matlab.

### 5.1 Throughput in a Scenario with Hidden Terminals

We compare the performance in scenarios (a) and (b) (see Fig. 3). Scenario (a) is a fully connected network with $N(=4, 12, 20, \ldots)$ senders. Scenario (b) is a network with hidden terminals. The network has four groups,[7] each containing $N/4$ senders. In both scenarios, all the senders send data traffic to a common receiver. Fig. 4 shows the RTS collision probability (which is approximated by the ratio of the collided RTS frame number to the total transmitted RTS frame number). For IEEE 802.11e and DBTMA, the RTS collision probability in scenario (b) is much higher than that in scenario (a). Correspondingly, the aggregate throughput in scenario (b) is much lower than that in scenario (a), shown in Fig. 5. The gap is contributed by the hidden terminals. On the contrary, with our scheme, the RTS collision probability and the aggregate throughput almost remain the same in both scenarios. The hidden terminals in scenario (b) do not introduce more RTS

7. Each group represents a set of nodes, which are in the transmission range of each other and are contending with each other. The nodes in the same group have the same characteristics. The nodes in one group are beyond the transmission range of any node in other groups.

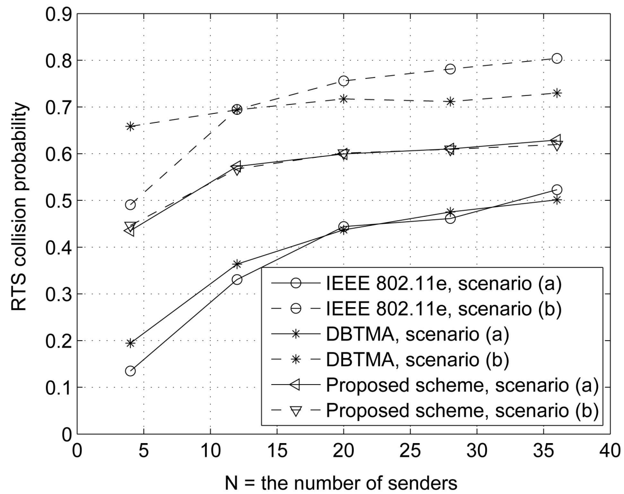


Fig. 4. RTS collision probability for IEEE 802.11e, DBTMA, and our scheme in scenarios (a) and (b).

collisions, indicating that our scheme effectively avoids RTS collisions caused by hidden terminals. Note that completely avoiding RTS collisions caused by hidden terminals does not mean that the RTS collisions do not occur. Actually, in scenario (a) without hidden terminals, RTS collisions still exist. Such collisions occur when more than one contending nodes choose the same backoff timer. We resolve those collisions by doubling the CW of the collided nodes as in IEEE 802.11e.

From Fig. 4, we notice that, in scenario (a), the RTS collision probability of our scheme is higher than that of IEEE 802.11e and DBTMA. It is because our scheme uses a smaller CW size than those for IEEE 802.11e and DBTMA (see Table 1). The smaller the CW, the higher the RTS collision probability. However, the backoff time is also reduced significantly in our scheme. As a result, the aggregate throughput in our scheme is still higher than those of IEEE 802.11e and DBTMA, as shown in Fig. 5.

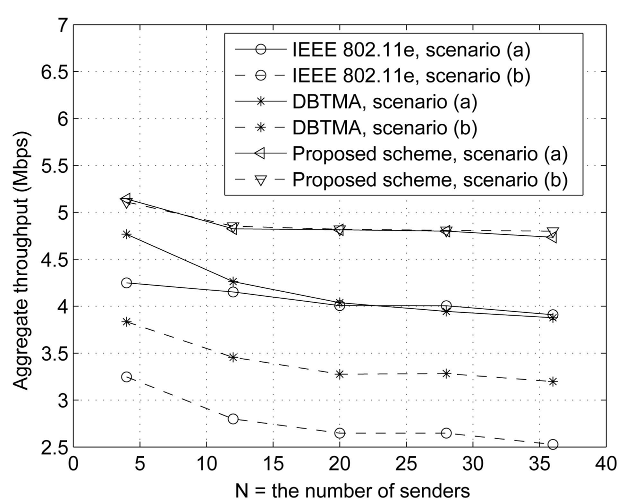


Fig. 5. The aggregate throughput of IEEE 802.11e, DBTMA, and proposed scheme in scenarios (a) and (b).

TABLE 2
The Aggregate Throughput (in Megabits per Second) in Scenarios (c) and (d)

| | Scenario (c) | Scenario (d) | Single-flow |
|---|---|---|---|
| Proposed | 11.95 | 11.96 | 5.98 |
| IEEE 802.11e | 4.22 | 3.84 | 3.87 |
| DBTMA | 10.87 | 10.97 | 5.48 |

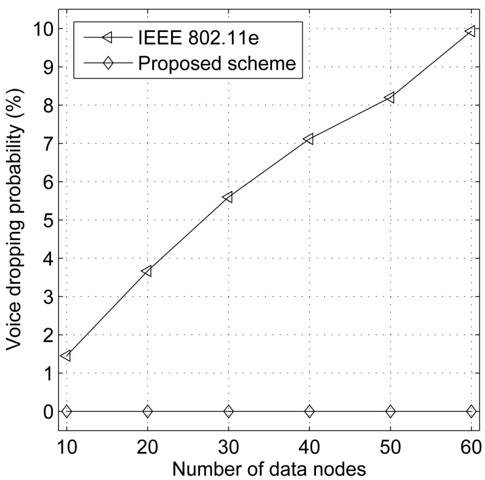


Fig. 6. Voice frame dropping probability versus the number of data nodes in a fully connected network with 20 voice nodes.

### 5.2 Throughput in Scenarios with Exposed Terminals

In scenario (c) in Fig. 3, the senders are exposed terminals, while in scenario (d), the receivers are exposed terminals. In both scenarios, senders send data traffic to the corresponding receivers. Table 2 compares the aggregate throughput of the proposed scheme, IEEE 802.11e and DBTMA in these two scenarios. For comparison, the throughput of a single data flow (i.e., only a data flow exists in the network) is also presented. We can see that the aggregate throughput of IEEE 802.11e in either scenario (c) or scenario (d) is similar to the single-flow throughput, indicating that IEEE 802.11e suffers from the exposed terminal problem. On the contrary, the aggregate throughput of our scheme and DBTMA in scenarios (c) and (d) are almost two times the single-flow throughput, indicating that our scheme and DBTMA allow simultaneous transmissions among exposed terminals.

### 5.3 Priority Access

First, we evaluate the priority performance of the proposed scheme supporting voice/data traffic in a fully connected network. The delay bound of a voice frame is set to be 40 ms. This means a voice frame will be discarded by the sender if it is unable to be delivered to the receiver within the bound. Fig. 6 shows the voice frame dropping probability in different MAC schemes, for 20 voice source nodes when the number of data source nodes changes from 10 to 60. No voice dropping is observed in our scheme, while in IEEE 802.11e, the voice frame dropping probability increases with the data node number. The results indicate that our proposed scheme (which provides guaranteed priority access) has better QoS provisioning capability than IEEE 802.11e (which provides statistical priority access).

Next, we choose two specific scenarios (e) and (f) (see Fig. 3) to study whether or not the priority access is dependent on the locations of the flows. In both scenarios, a group of voice nodes is contending with a group of data nodes. Each group contains $N$ nodes, sending traffic to a common receiver. We use the average voice flow access delay (which is the time duration from the instant that the frame is at the head of the buffer to the instant that the frame has been successfully transmitted) as the performance metric, given in Table 3. For IEEE 802.11e, the voice access delay is quite large in scenario (e), from 9.9 to 686.0 ms with the increase of $N$; while in scenario (f), the delays are around 1 ms for all the $N$ values. These results indicate that the priority access performance of IEEE 802.11e is location dependent. On the contrary, in our scheme, the voice access delay almost remains the same (around 1 ms) in both scenarios and for all the $N$ values, indicating that our scheme provides a stable priority access, independent of the flow locations.

### 5.4 Fairness

First, we compare the short-term fairness performance of IEEE 802.11e and our scheme in a fully connected network with only data traffic, as shown in scenario (a) in Fig. 3. The fairness is measured by Jain's Fairness Index given by $\frac{(\sum_{i=1}^{N_d} T_i)^2}{N_d \cdot \sum_{i=1}^{N_d} T_i^2}$ [23], where $T_i$ is the throughput of the $i\text{th}$ data node over a time window, and $N_d$ is the number of data nodes. The higher the Fairness Index value, the better the fairness performance. We sample the Fairness Index values after each duration over which each data node transmits six frames on average. Fig. 7 compares the average Fairness Index values. As expected, our scheme shows better short-term fairness performance than IEEE 802.11e. The aggregate throughputs of the two schemes are also shown in Fig. 7. We can see that the analytical and simulation results

TABLE 3
The Average Voice Access Delay (in Milliseconds) with Different Node Number $N$ within a Group in Scenarios (e) and (f)

| $N$ | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 10 | 20 |
|---|---|---|---|---|---|---|---|---|---|---|---|
| 802.11e | Scenario (e) | 9.9 | 27.1 | 45.7 | 57.3 | 97.1 | 101.6 | 165.3 | 175.5 | 228.3 | 686.0 |
| | Scenario (f) | 1.38 | 1.17 | 1.27 | 1.19 | 1.31 | 1.21 | 1.26 | 1.24 | 1.25 | 1.32 |
| Proposed | Scenario (e) | 1.27 | 1.15 | 1.11 | 1.11 | 1.12 | 1.10 | 1.08 | 1.11 | 1.11 | 1.11 |
| | Scenario (f) | 1.09 | 1.07 | 1.06 | 1.16 | 1.06 | 1.10 | 1.10 | 1.15 | 1.11 | 1.13 |

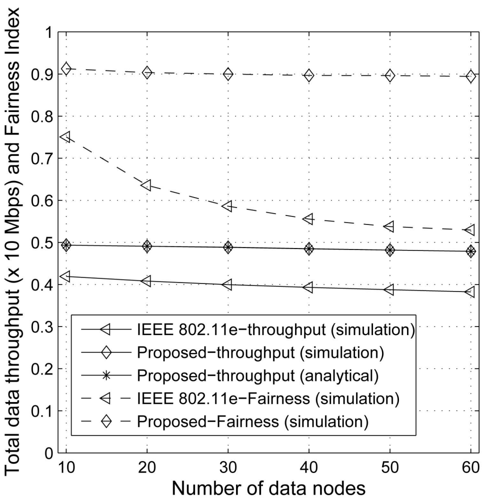


Fig. 7. The total data throughput and Fairness Index versus the number of data nodes in a fully connected network with only data traffic.

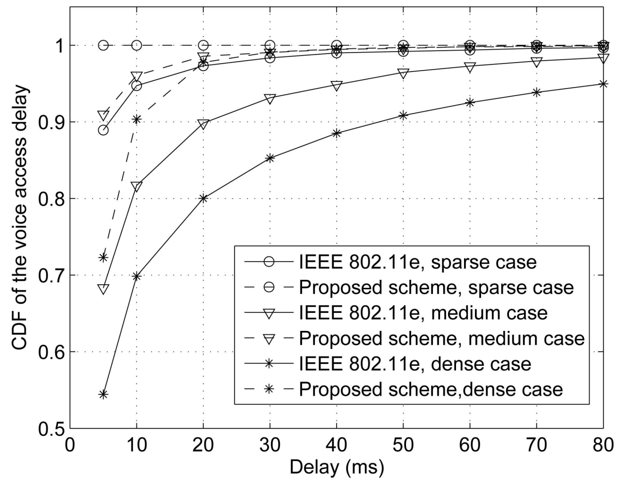


Fig. 8. The CDF of voice access delay in random topologies.

for our scheme match well. The aggregate throughput of our scheme is larger than that of IEEE 802.11e.

Next, we compare the long-term fairness performance of IEEE 802.11e and our scheme in scenario (g) in Fig. 3 (since even long-term fairness is difficult to achieve in this scenario, we do not consider short-term fairness, but compare the achieved throughput of each flow). Both flows are for long-lived data transfers. It is found that flows 1 and 2 achieve a throughput of 3.46 and 2.31 Mbps, respectively, in our scheme, but 0.2 and 3.77 Mbps, respectively, in IEEE 802.11e. In IEEE 802.11e, flow 1 is almost starved while flow 2 occupies the channel almost all the time, while in our scheme, each flow gets a certain share of the channel time. Note that our scheme improves the long-term fairness performance to some degree as compared with IEEE 802.11e, but does not yet achieve absolute fairness. To achieve absolute fairness in a distributed manner is extremely challenging. Extra information needs to be exchanged among the nodes, and a controller is needed to coordinate the transmissions from the nodes, making the scheme not scalable. Therefore, there exists a tradeoff between scalability and absolute fairness.

### 5.5 Performance in Random Topologies

We consider a 1,000 m $\times$ 1,000 m service area, where the transmission range of each node is 200 m. The nodes are evenly distributed in the area. The flows are randomly chosen from the nodes which are one hop away. Half of the flows are voice flows and the remaining are data flows. Here, we simulate three cases: sparse (36 nodes with 10 flows), medium (121 nodes with 50 flows), and dense (441 nodes with 200 flows). The node density is measured as $x$, where the number of nodes is $N = x^2$. Initially, we choose $x = 6$ for sparse case. We increase $x$ by 5 for medium case, and further by 10 for dense case. In our experiment, we use the different node density to reflect the different contention level of the network. When the node density increases within a fixed area (i.e., the number of nodes increases), the number of flows (and traffic load) will increase, so does the contention level. Since flows are randomly generated, and may encounter different levels of contention, we do not consider fairness for this case.

To compare the priority access performance, we show the cumulative distribution function (CDF) of voice access delay in Fig. 8. The vertical axis is the probability that the voice access delay is equal to or smaller than the delay specified in the horizontal axis. It is clear that our scheme has a smaller voice access delay than IEEE 802.11e in all the three cases. In the sparse case, all voice frames' access delays are below 5 ms in our scheme, and approximately 11 percent voice frames have an access delay larger than 5 ms in IEEE 802.11e. With the increase of user density, the voice access delay is increased in both schemes because the voice flows encounter a much higher contention level and are more likely to collide. In the dense case, around 30 percent (10 percent) voice frames have an access delay larger than 10 ms in IEEE 802.11e (our scheme). The aggregate data traffic throughput of IEEE 802.11e is 14.56, 21.77, and 11.54 Mbps in the sparse, medium, and dense cases, respectively, while for our scheme, it is 23.41, 46.16, and 30.21 Mbps, respectively. That is, our scheme has a higher throughput than IEEE 802.11e in all the three cases. Note that the system throughput in the medium case is larger than those in the other two cases. In the sparse case, as a small number of flows contend for the channel, the network capacity is not fully utilized. With an increased flow number in the medium case, the system throughput is increased. When the flow number further increases (in the dense case), more resources are used by voice traffic, resulting in a reduced throughput of data traffic.

### 5.6 Sensitivity of the Proposed Scheme to Carrier Sense Ranges

With an appropriate carrier sense range setting, the hidden/exposed terminal, priority reversal, and unfairness problems are eliminated and, at the same time, the resources are efficiently utilized. In reality, the carrier sense ranges may not be set exactly as required, resulting in reduced efficiency or effectiveness of the proposed scheme. To investigate the sensitivity of the proposed scheme to different carrier sense ranges, random topologies with different node densities are considered. The random topologies and flows are generated in the same way as in the preceding section. The BTt channel's carrier

TABLE 4
The Average Voice Access Delay (in Milliseconds) and Aggregate Data Throughput (in Megabits per Second) with Different Carrier Range Settings

| $\alpha$ and $\beta$ values | | $\alpha$=1.6 $\beta$=1.0 | $\alpha$=1.8 $\beta$=1.0 | $\alpha$=2.0 $\beta$=1.0 | $\alpha$=2.2 $\beta$=1.0 | $\alpha$=2.4 $\beta$=1.0 | $\alpha$=2.0 $\beta$=1.1 | $\alpha$=2.0 $\beta$=1.2 |
|---|---|---|---|---|---|---|---|---|
| Sparse case | Voice access delay | 0.66 | 0.67 | 0.74 | 0.74 | 0.80 | 0.74 | 0.75 |
| | Data throughput | 23.57 | 23.56 | 23.41 | 23.41 | 22.80 | 23.41 | 23.42 |
| Medium case | Voice access delay | 2.75 | 2.64 | 2.50 | 2.50 | 2.57 | 2.68 | 2.29 |
| | Data throughput | 49.38 | 49.47 | 46.16 | 42.86 | 41.42 | 46.15 | 35.7 |
| Dense case | Voice access delay | 5.76 | 5.07 | 5.05 | 4.73 | 4.95 | 4.64 | 3.95 |
| | Data throughput | 35.74 | 33.15 | 30.21 | 26.07 | 21.27 | 28.94 | 27.72 |

sense range is set to $\alpha$ times the transmission range, where $\alpha$ varies from 1.6 to 2.4. The BTr channel's carrier sense range is set to $\beta$ times the transmission range, where $\beta$ varies from 1.0 to 1.2. Typically, the carrier sense range is no less than the transmission range, so we do not consider the case that the BTr channel's carrier sense range is less than the transmission range.

Table 4 compares the average voice packet delay and the aggregate data throughput with different BTt/BTr carrier sense range settings in the network with different node densities. It can be seen that the average voice access delay changes slightly with different carrier sense range settings. For the aggregate data throughput, it remains almost the same in the sparse case, and reduces slightly in the medium and dense cases when the BTt/BTr channel's carrier sense range increases. In the medium and dense cases, when the BTt channel's carrier sense range is larger than the coverage of two-hop neighborhood, some nodes may unnecessarily defer their transmissions, resulting in a reduced resource utilization. When the BTt channel's carrier sense range is less than the coverage of two-hop neighborhood, a slightly higher data throughput is achieved with the cost of unfairness. When the BTr channel's carrier sense range is larger than the coverage of one-hop neighborhood, the receivers may prevent some nodes (which may not corrupt their receptions) from transmitting concurrently. However, in the sparse case, since flows are likely to be far away from each other, the carrier sense ranges have little impact on the resource utilization.

## 6 Conclusion and Further Discussion

In this paper, we have presented a novel busy-tone-based distributed MAC scheme supporting voice and data traffic in wireless ad hoc networks. Although many busy-tone-based MAC schemes have been proposed in the literature, most of them are designed to solve hidden (or exposed) terminal problem. The newly proposed scheme is the first one to utilize the busy tones to address not only the hidden and exposed terminal problems but also the priority reversal and unfairness problems associated with wireless ad hoc networks. The simulation results demonstrate that the system throughput is significantly increased by resolving the hidden and exposed terminal problems. As compared with the IEEE 802.11e, our scheme greatly reduces voice traffic delay (by ensuring guaranteed priority access for voice traffic, independent of the traffic locations), and significantly improves short-term and long-term fairness performance for data traffic.

In this paper, we do not consider the capture effect where a receiver may be able to correctly receive its desired frame even when a collision occurs. With the capture effect, the following may happen: when a sender senses a BTr busy tone after completing its frame transmission, the busy tone may not be from its own destination because another receiver (not the target sender's receiver) with its sender close enough may receive correctly its desired frame despite the target sender's transmission at the same time, and thus sends a BTr busy tone. Further research efforts are needed to deal with the case with the capture effect.

## Appendix A

## The Operation Procedure of the Proposed Distributed MAC Scheme

Fig. 9 illustrates the state transition diagram of the proposed MAC scheme. The ellipses represent the states of one node, and the name of each state transition is labeled along the path. At the initialization of the network, every node is at the *Idle* state. Detailed state transition procedure is as follows:

- *Transition S_IB/S_IC:* When a node is at the *Idle* state and has traffic to send, it sets its CW and AIFS according to the traffic type, and chooses a random backoff timer from [0, CW]. Then, the node senses the BTr and BTt channels for the duration of AIFS[voice] (or AIFS[data]). If no busy-tone signal for AIFS[voice] (or AIFS[data]), the node will send a busy tone to jam the BTt channel, and the length of the busy tone (in the unit of slot time) is equal to its

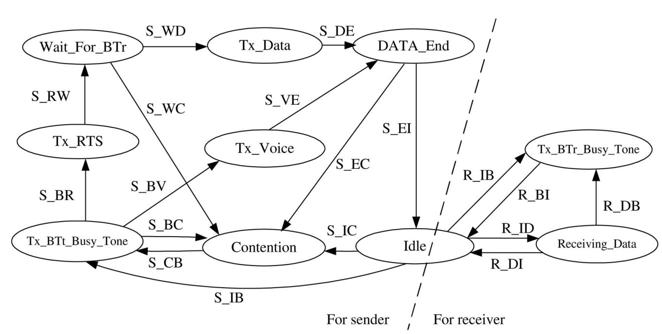


Fig. 9. The state transition diagram of the proposed scheme.

backoff timer. Then, the node goes into the *Tx_BTt_Busy_Tone* state. If the node senses a busy-tone signal in either channel, it goes into the *Contention* state.

- *Transition S_BR/S_BV/S_BC:* When a node is at the *Tx_BTt_Busy_Tone* state, after the completion of its own BTt busy tone, the node monitors both BTt and BTr channels for one slot time. If both channels are idle (which means the node is sending the longest BTt busy tone), the node will transmit. If the node has a data packet to send, it transmits its RTS frame in the information channel, at the same time sends busy tone in the BTt channel, and goes into the *Tx_RTS* state; if the node has a voice packet to send, it transmits voice-DATA frame in the information channel, at the same time sends busy tone in the BTt channel, and goes into the *Tx_Voice* state. If either the BTt channel or the BTr channel is busy, the node goes into the *Contention* state.
- *Transition S_RW:* When a node is at state *Tx_RTS*, it keeps transmitting an RTS frame at the information channel and the BTt busy tone at the BTt channel. At the end of RTS frame transmission, the node stops its BTt busy tone, and sets a timer (equal to the busy-tone detection time), and goes into the *Wait_For_BTr* state.
- *Transition S_VE:* When a node is at state *Tx_Voice*, it keeps transmitting voice-DATA frame in the information channel and the BTt busy tone in the BTt channel. At the end of voice-DATA frame transmission, the node stops its BTt busy tone, and sets a timer (equal to the busy-tone detection time), and goes into the *DATA_End* state.
- *Transition R_ID/R_IB:* When a node is in the *Idle* state and has no backlogged traffic to send, it keeps monitoring the information channel to check whether there is any RTS (or voice-DATA) frame destined to it. If an RTS destined to it is received correctly, the node sends a busy tone immediately in the BTr channel as an indication of successful reception, sets a timer equal to the data-DATA frame transmission duration, and goes into the *Receiving_Data* state. If a voice-DATA frame destined to it is received successfully, the node goes into the *Tx_BTr_Busy_Tone* state.
- *Transition R_BI:* When a node is at state *Tx_BTr_Busy_Tone*, it continues its BTr busy tone for the busy-tone detection time, then stops the BTr busy tone and goes into the *Idle* state.
- *Transition S_WD:* When a node is at the *Wait_For_BTr* state, it senses the BTr channel. If a BTr busy tone is sensed, it sends a data-DATA frame immediately in the information channel, and goes into the *Tx_Data* state.
- *Transition S_WC:* When a node is at the *Wait_For_BTr* state, it senses the BTr channel. If the node does not sense a BTr busy tone (which means a collision may occur), upon timeout, the node doubles its CW (up to $\mathrm{CW}_{\max}$) and goes into the *Contention* state.
- *Transition S_DE:* When a node is at the *Tx_Data* state, it keeps transmitting its data-DATA frame. When the data-DATA transmission is finished, the node sets a timer (equal to busy-tone detection time), and goes into the *DATA_End* state.
- *Transition R_DB:* When a node is at the *Receiving_Data* state, it receives a data-DATA frame. If the data-DATA frame is successfully received, it goes into the *Tx_BTr_Busy_Tone* state.
- *Transition R_DI:* When a node is at state *Receiving_Data*, it receives a data-DATA frame. If it does not successfully receive a data-DATA frame, upon timeout, it stops the BTr busy tone immediately and goes into the *Idle* state.
- *Transition S_EI/S_EC:* When a node is at the *DATA_End* state, it senses the BTr channel. If it senses a BTr busy tone (which means the destination successfully receives the DATA frame), the node resets its CW to the initial value $\mathrm{CW}_{\min}$, and goes into the *Idle* state. If the node does not sense a BTr busy tone in the *DATA_End* state, upon timeout, it doubles its CW (up to $\mathrm{CW}_{\max}$) and goes into the *Contention* state.
- *Transition S_CB:* When a node is at the *Contention* state, it randomly chooses a backoff timer from its current CW. Then, the node senses the BTr and BTt channels for the duration of AIFS[voice] (AIFS[data]). If no busy tone is sensed, the node will send a busy tone to jam the BTt channel, and the length of the busy tone (in the unit of slot time) is equal to its backoff timer. Then, the node goes into the *Tx_BTt_Busy_Tone* state. If the node senses a busy-tone signal in either channel, it remains at the *Contention* state.

## APPENDIX B

## THE CW SETTINGS IN THE PROPOSED SCHEME AND IN IEEE 802.11E

For presentation simplicity, in this appendix, we assume that all the nodes belong to one class (e.g., data traffic).

In IEEE 802.11e, as long as two or more nodes have the same backoff timer value, a collision happens. On the other hand, in our scheme, a node first sends a BTt busy tone (the duration of which is the node's backoff timer value in the unit of time slot), and transmits if its BTt busy tone is the longest (i.e., its backoff timer is the largest) among all the nodes. After each contention, each node selects a new backoff timer from its CW. That is, actually the backoff timer does not count down, unlike the case in IEEE 802.11e. A collision happens only when 1) two or more nodes have the same backoff timer and 2) the backoff timer is the largest among the backoff timer values of all the nodes. Condition 2 determines that the collision probability in our scheme is not high. Let us use the case with CW setting {3:15} and 500 active nodes as an example. Suppose initially all the nodes have $\mathrm{CW} = 3$. In the first contention, approximately $500/(3+1) = 125$ nodes select a backoff timer value 3 (the largest value), and thus transmit. A collision happens. The 125 nodes increase their CW to 7. In the second contention, each node selects a new backoff timer from its CW.

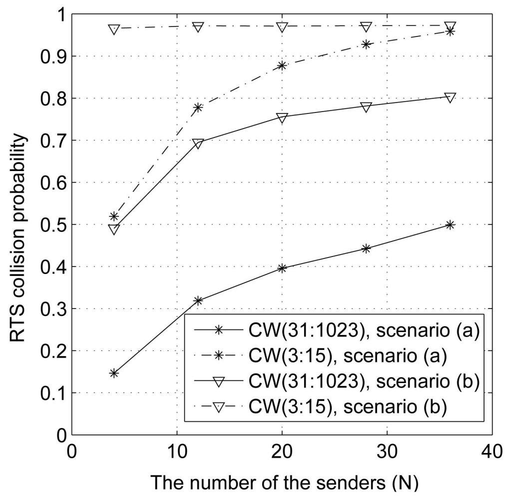


Fig. 10. The RTS collision probability versus the number of senders for IEEE 802.11e with different CW settings.

Approximately $125/(7+1) \approx 16$ nodes (from nodes with $\mathrm{CW} = 7$) select a backoff timer value 7 (the largest value), and thus transmit. Since a collision happens, the 16 nodes further increase their CW to 15. In the third contention, each node selects a new backoff timer, and approximately $16/(15+1) = 1$ node (from nodes with $\mathrm{CW} = 15$) selects the backoff timer value 15 (the largest value), and transmits. So in the third contention, a successful transmission is very likely to happen. And subsequently, it is also very likely that a successful transmission happens in the fourth contention, since very likely one of the remaining nodes with $\mathrm{CW} = 15$ selects the largest backoff timer value and transmits. From this example, we can see our scheme can work well with 500 active nodes. In our previous work [12], we have shown by simulation that when the number of active nodes increases from 60 to 500, the system throughput decreases only by 5 percent. Since the CW setting in our scheme is insensitive to the number of active nodes (this is actually another advantage of our scheme), we suggest that each node fix its CW setting to {3:15}.

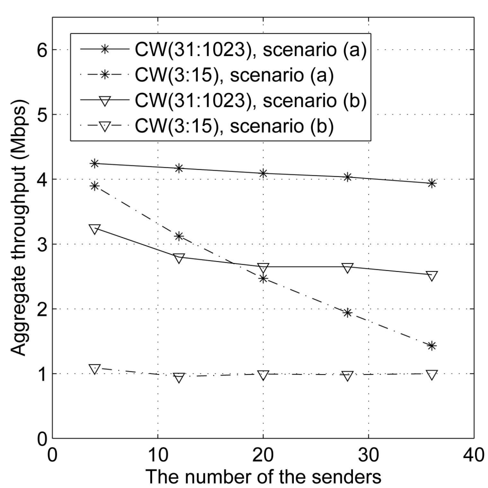


Fig. 11. The aggregate throughput versus the number of senders for IEEE 802.11e with different CW settings.

On the other hand, for IEEE 802.11e, it is not appropriate to use the same small CW setting as in our scheme, since a small CW setting may lead to severe collisions and low throughput when the number of contending nodes increases. Figs. 10 and 11 compare the collision probability and throughput of IEEE 802.11e with CW setting {3:15} and {31:1,023} in scenarios (a) and (b) shown in Fig. 3, respectively. It is clear that IEEE 802.11e has worse performance with a smaller CW setting. For fair comparison, we choose CW setting {31:1,023} (as recommended by the standard) for IEEE 802.11e.

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