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A distributed MAC scheme supporting voice services in mobile *ad hoc* networks‡

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Summary

Future mobile *ad hoc* networks are expected to support voice traffic. The requirement for small delay and jitter of voice traffic poses a significant challenge for medium access control (MAC) in such networks. User mobility presents unique difficulties in this context due to the associated dynamic path attenuation. In this paper, a MAC scheme for mobile *ad hoc* networks supporting voice traffic is proposed. With the aid of a low-power probe prior to DATA transmissions, resource reservation is achieved in a distributed manner, thus leading to small packet transmission delay and jitter. The proposed scheme can automatically adapt to dynamic path attenuation in a mobile environment. Statistical multiplexing of `on`/`off` voice traffic can also be achieved by partial resource reservation for `off` voice flows. Simulation results demonstrate the effectiveness of the proposed scheme. Copyright © 2009 John Wiley & Sons, Ltd.

KEY WORDS: medium access control; code-division multiple access; mobile *ad hoc* networks; statistical multiplexing

1. Introduction

Mobile *ad hoc* networks are expected to support multimedia services such as voice and video with quality-of-service (QoS) requirements. This task is challenging because there is no central controller in such networks. User mobility makes the situation even worse because of the time-varying network topology and propagation loss between any two nodes. QoS support in mobile *ad hoc* networks includes two approaches: routing at the network layer and medium access control (MAC) at the link layer. The function of routing is to search for a network path (i.e., a route) from a traffic source node to its destination node, and to react (e.g., by selecting a new route) to possible route failures and/or network congestion. On the other hand, the function of MAC is to coordinate the channel access in an orderly manner among the mobile nodes such that efficiency can be achieved. Here, we focus primarily on the MAC in a mobile *ad hoc* network supporting voice traffic.

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In such a network, a MAC scheme should be: (1) distributed (because of the scalability requirement); (2) tolerant to the hidden terminal problem; (3) ensuring small delay/jitter (due to the real-time nature of voice traffic); and (4) adaptive to user mobility.

In this paper, we propose an effective MAC scheme with the above desired features. The scheme is fully distributed, thus scaling well to a large network. The hidden terminal problem does not exist in our scheme. Radio resource requirements are met by resource reservation, thus achieving small delay and jitter. Our scheme can also adapt to user mobility to a certain extent. The rest of this paper is organized as follows. Related work is discussed in Section 2. The proposed MAC scheme is presented in Section 3. Statistical multiplexing of voice traffic in our scheme is investigated in Section 4. Performance evaluation is given in Section 5, followed by concluding remarks in Section 6.

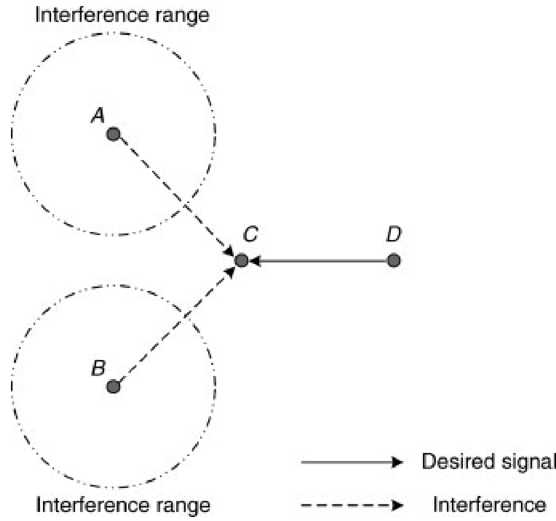

Fig. 1. An example in which the ideal assumption is not valid.

2. Related Work and Discussion

The most prevalent MAC schemes for *ad hoc* networks are carrier-sense multiple access (CSMA) and its variants, and busy-tone based schemes. The CSMA schemes are inherently distributed. However, it is well accepted that CSMA schemes suffer from the hidden terminal problem. In addition, these schemes are based on an ideal assumption, i.e., each transmitter has an interference range so that a receiver of another link outside the interference range does not receive any interference from that transmitter. However, in practical networks, the interference from a transmitter to any receiver depends on the transmission power, the distance between the two nodes, and the propagation environment between them. This means that the interference level is *continuous* in general. The principle behind the preceding ideal assumption is to quantize the continuous interference level to two discrete levels: full interference or no interference, thus making network analysis much easier. However, it may not reflect the reality in an actual network. For instance, consider the situation shown in Figure 1. We have a transmission link from node D to node C. The receiver, i.e., node C is beyond the interference range of two interferers, nodes A and B. According to the ideal assumption, node C receives no interference from either A or B. However, in reality, although the interference from either node A or node B is not significant enough to affect node C's reception, the summation of interference received from nodes A and B may make node C unable to receive its desired signal successfully. On the other hand, one motivation of busy-tone based schemes [1] is to better solve the hidden terminal problem. Each target receiver sends a *receive busy tone* to notify the nearby hidden terminals not to transmit. However, the hidden terminal problem is not eliminated. For instance, collisions of request-to-send (RTS) frames can still occur due to the hidden terminal problem [2]. It is also challenging to determine the busy-tone coverage. Traditionally the busy-tone coverage is set to be the same as the interference range. However, similar to the case in CSMA, if there are multiple interferers beyond a target receiver's busy-tone coverage, the aggregate interference from them can be strong enough to corrupt the target receiver's desired reception, since the interferers cannot hear the busy tone and thus, start their transmissions. In a mobile environment, the multipath fading can cause the busy-tone coverage to vary with time. This can severely degrade the system performance because a strong interferer in the DATA frequency band may not be notified by the busy tone due to instantaneous deep fading in the busy tone frequency band. This is the problem of different channel gains in two frequency bands.

The contention-based nature of CSMA schemes and busy-tone based schemes makes it very difficult to bound delay and jitter to small values for voice traffic. This observation implies that we should instead resort to reservation of channel resources. Resource reservation can avoid contentions, and voice traffic can be transmitted continuously on the reserved channel, thus bounding the transmission delay and jitter.

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Reservation based MAC schemes have been extensively studied in the literature [3–8]. In these schemes, each node first contends in a contention period to send a reservation request. Once the reservation is granted, the node can transmit in the reserved time slot without contention. However, these schemes still suffer from the aforementioned 'ideal assumption' problem. Thus in this research we consider a code-division multiple access (CDMA)-based mobile *ad hoc* network, in which the air interface supports multiple simultaneous transmissions in a neighborhood. In our previous work [9], we have proposed a MAC scheme for a CDMA-based wireless mesh backbone, referred to as the *MESH* scheme here. Before DATA[‡] transmissions, each potential sender first transmits a low-power probe to the network. Upon reception of the probe, each active receiver estimates a potential interference increase due to the potential sender's DATA transmission. The active receiver transmits a busy-tone signal in a separate busy-tone channel to notify the potential sender if the potential interference increase is intolerable. If the potential sender does not detect a busy tone, it successfully reserves a code channel. However, targeted at a wireless mesh backbone with fixed topology and large traffic intensity, the MESH scheme is not effective or efficient in a mobile *ad hoc* network with time-varying network topology and time-varying channel gain between any two nodes. Accordingly, in the following we propose a MAC scheme for a mobile *ad hoc* network supporting voice traffic, based on a similar 'probing' mechanism.

3. The Proposed MAC Scheme

3.1. System Model

Consider a mobile *ad hoc* network having N mobile nodes. At the MAC layer, a source node can communicate with one or more of its neighbors, with a separate queue for each destination node. Each node is assigned a unique transmitting code and a unique receiving code [10]. The network topology changes as the mobile nodes move.

We use a time frame structure as shown in Figure 2, in which time is partitioned into fixed-length frames. In each frame, there are M information slots (ISs), each followed by a short blocking slot (BS). The ISs are

[‡]A 'DATA' packet means an information packet from the traffic source.

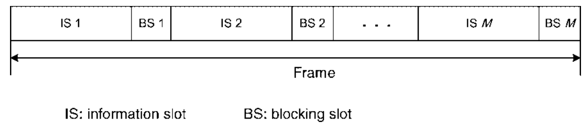

Fig. 2. Time frame structure.

used for information (i.e., DATA and ACK) and probe transmissions, while the BSs are used for transmitting blocking messages (to be discussed). Only constant rate voice traffic is considered.[§]

Each voice link requires a minimal transmission rate (e.g., voice packet generation rate from the codec) so that its delay and jitter can be kept at a very low level. To keep the transmission accuracy of the voice link, a signal to interference plus noise ratio (SINR) threshold is required for each link, i.e., for the link from sender i to receiver j, the following inequality should hold:

$$\frac{G_i P_{ij}^t g_{ij}}{I_j + \eta_j} \geq \Gamma_d \tag{1}$$

where G_i is the spreading gain, P_{ij}^t the transmit power, g_{ij} the path gain from node i to node j, I_j the interference (from other links) received at node j, η_j the background noise power at node j, and Γ_d is the required SINR threshold for accurate information transmission.

Unlike the situation for the MESH scheme, all the transmissions are within a frequency band in our proposed scheme, using CDMA technology. Since Rake reception can collect signal energy from different paths, we assume that there is no fast (multipath) fading,[‖] and the transmit power is attenuated only due to path loss with exponent β and shadowing. A node cannot transmit and receive at the same time, since each node's transmission and reception are within the same frequency band.

[§]Note that our scheme can still work when there are other traffic types such as video and data. Each time frame can be (dynamically) divided into three portions, for voice, data, and video traffic, respectively. Then our scheme can be applied in the voice portion of each frame.

[‖]For a non-ideal case in which fast fading is not fully addressed by a Rake receiver, the impact of the fluctuations due to fast fading can be represented by an increase in the packet loss rate. The additional packet loss rate depends on the propagation environment, the Rake receiver structure, the error-correction technology, user mobility pattern, and so on.

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3.2. Power Allocation for DATA and ACKs

Random access wireless networks usually use one of two power allocation strategies: common power allocation and linear power allocation [11]. A common transmit power is used by all the nodes in the common power allocation strategy [12]. In the linear power allocation strategy [13,14], the transmit power is proportional to the path attenuation level from the sender to the receiver. In a mobile *ad hoc* network, it is difficult to estimate the path attenuation in advance from a sender to a receiver due to user mobility. Here we consider a common power allocation strategy, in which each traffic source node uses a constant power P to transmit its DATA frames to its destination, and the traffic destination uses the same power level to feed back ACKs.

3.3. Multiple Access Procedure

The multiple access control of the network is based on resource reservation. Consider the case of a new call arrival at source node i in time frame 0 to destination node j. Note that our MAC scheme deals only with unidirectional traffic (from node i to j) rather than bidirectional traffic (from node i to j and *vice versa*) for the following reason. In a mobile *ad hoc* network, it is likely that there is no direct link between a pair of users (say user A and user B). In this case, it is not necessary that traffic from user A to user B follows the same path as that in the reverse direction. Therefore, our scheme reserves resources only for unidirectional communications, and the routing protocol in the network layer is expected to ensure that reservation is performed in both directions.

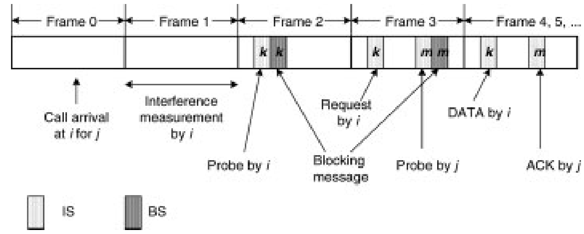

Fig. 3. The multiple access procedure.

The multiple access procedure is illustrated in Figure 3, while the associated finite state machine is depicted in Figure 4.

3.3.1. Selection of DATA IS

Before the call arrival, node i is in state `Idle`. Upon the new call arrival, node i first selects an IS for DATA transmission, referred to as its *DATA IS*, i.e., node i transits to state `DATA_IS_Sel`. The ISs at which the destination node j is sending traffic should be skipped, because node j cannot send and receive at the same time at the same frequency band. In frame 1, node i measures its experienced interference, and monitors the transmitting code of node j. Among the ISs over which node j's transmitting code is not detected, node i selects one with the minimum interference level, say IS k, as its DATA IS. If no IS is available, the call is dropped, i.e., node i proceeds to state `Call dropping`.

3.3.2. Transmissions of probe and blocking messages

Prior to DATA transmission in the selected DATA IS, node i should ensure its DATA transmission does not

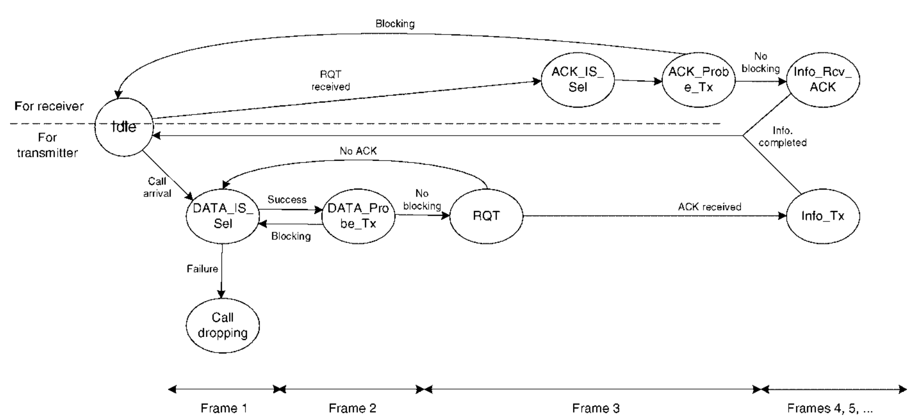

Fig. 4. The finite state machine of the medium access protocol.

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corrupt any existing reception at any active receiver[¶] in the IS. To achieve this goal, node i first sends a low-power test signal into the network, i.e., in state `DATA_Probe_Tx`. Specifically, at IS k of frame 2, node i sends a probe spread by a common probe code with a large spreading gain, and with transmit power level $\alpha \cdot P$ ($\alpha \ll 1$). No data is carried in the probe. With its low power, the probe is very unlikely to corrupt any existing reception at the IS. Although the probe is sent with low power, we assume that it can be received by active receivers at the IS because of the large spreading gain.

At IS k, an active receiver, say node l, first measures the received probe power level denoted by $P_l^{r,p}$ and its own interference level due to other existing transmissions. It estimates the potential interference increase due to DATA transmission of the probe sender as $P_l^{r,p}/\alpha$, and determines whether the potential interference increase can corrupt its own reception. If so, it sends a blocking message at BS k. The blocking message is spread by a common blocking code.[#] No data is carried by the blocking message. The power allocation for the blocking message will be discussed in Section 3.4.

At BS k, node i monitors the blocking code. If the detected blocking message power is above a threshold $P_{th}^{r,b}$, it gives up on IS k for DATA transmission and tries another IS until no blocking message is detected.

3.3.3. Selection of ACK IS

If no blocking message is detected, node i sends a request message to destination node j, at IS k in frame 3 spread by node j's receiving code, and sends its first DATA frame at IS k in frame 4 spread by node i's transmitting code, both with power P. That is, the node proceeds to state `RQT`.

Upon receiving the request message, node j selects an IS for its ACK, referred to as *ACK IS*, i.e., node j proceeds to `ACK_IS_Sel`. Generally it is desired that, at the ACK IS, the interference experienced by node i is small so that the ACK can be received correctly. So in the request message, node i indicates (among the ISs except IS k) the IS with minimal experienced interference, say IS m. After node j has selected IS m as the ACK IS, it proceeds to state `ACK_Probe_Tx`. That is, it sends a probe at IS m in frame 3 in the same manner[**] as node i sends a probe for DATA IS. If no blocking message is detected at BS m, node j sends an ACK spread by its transmitting code at IS m in frame 4 for the DATA frame received in frame 4, i.e., node j proceeds to state `Info_Rcv_ACK`.

If node i does not receive an ACK at IS m of frame 4, it goes back to state `DATA_IS_Sel`, i.e., selects another DATA IS, and repeats the preceding procedure with a maximal number of attempts. If an ACK is received, it proceeds to the next state `Info_Tx`.

3.3.4. Continuous transmissions at reserved DATA and ACK ISs

At subsequent time frames from frame 5, node i in state `Info_Tx` transmits its DATA frames spread by its transmitting code at IS k, and node j in state `Info_Rcv_ACK` transmits the ACKs spread by its transmitting code at IS m, until the completion of the call. It can be seen that call-level resource reservation is performed. Upon the call completion, no link termination procedure is necessary, because our scheme is based on real-time interference measurement, and other active receivers can measure the interference level reduction when the target link stops using the reserved resource.

3.4. Power Allocation for the Blocking Message

If an active receiver at an IS is to send a blocking message in response to a probe from a potential sender,[††] the transmit power of the blocking message should be designed carefully so that the potential sender can be notified. The objective of blocking message power allocation is to ensure that the potential sender can detect a blocking message with power above threshold $P_{th}^{r,b}$.

At an IS, it is possible that one or more potential senders may send probes for resource reservation. Since the resource reservation is performed at the call

[¶]Here an active receiver can be an existing traffic destination node for DATA reception, or an existing traffic source node for ACK reception.

[#]Unlike the scheme considered in Reference [9] in which a busy tone is used over a separate control channel, the blocking message in our scheme is sent in the data channel, thus avoiding additional hardware cost to implement the busy-tone channel.

[**]The probe is sent at IS m in frame 3 when $m > k$, or in frame 4 when $m < k$. For simplicity of presentation, we use the former case as an example in this paper.

[††]Here a potential sender can be a traffic source node for DATA transmission or a traffic destination node for ACK transmission.

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level, it is reasonable to assume that at an IS there are at most two potential probe senders in a neighborhood. This assumption is reasonable due to the facts that the duration of a time frame (usually in the order of 10 ms) is much shorter than a call duration (usually in the order of minutes) and that each voice user may only need one request (and thus one probe) within its call duration.

We assume channel reciprocity in terms of shadowing effects. Consider the case in which an active receiver, say node l, detects the probe power level $P_l^{r,p}$, and wants to send a blocking message. Node l sends a blocking message with transmit power

$$P_l^{t,b} = \max\left\{\frac{2 \cdot P_{th}^{r,b} \cdot \alpha P}{P_l^{r,p}}, \frac{P_{th}^{r,b} \cdot P}{\Lambda_l}\right\} \tag{2}$$

where Λ_l is the *interference margin* [15,16] of node l, defined as the maximum extra interference that can be tolerated by node l. From the interference margin definition, we have

$$\frac{G_l P g_l}{I_l + \eta_l + \Lambda_l} = \Gamma_d \tag{3}$$

where G_l and g_l are the spreading gain and path gain for the transmission to node l, respectively. Then Λ_l is given by

$$\Lambda_l = \frac{G_l P g_l}{\Gamma_d} - I_l - \eta_l \tag{4}$$

We first consider the scenario in which only one potential sender, node i, sends a probe at an IS. The path gain from node i to node l is

$$g_{il} = \frac{P_l^{r,p}}{\alpha P} \tag{5}$$

From Equation (2) we have

$$P_l^{t,b} \geq \frac{2 \cdot P_{th}^{r,b} \cdot \alpha P}{P_l^{r,p}} = \frac{2 \cdot P_{th}^{r,b}}{g_{il}} \tag{6}$$

Due to the reciprocity of the channel, $g_{li} = g_{il}$, and so we have

$$P_l^{t,b} \geq \frac{2 \cdot P_{th}^{r,b}}{g_{li}} \tag{7}$$

Then the received blocking message power at the potential sender i is

$$P_l^{t,b} \cdot g_{li} \geq 2 \cdot P_{th}^{r,b} \tag{8}$$

which is greater than the detection threshold $P_{th}^{r,b}$. Hence the potential sender is notified to give up the IS.

Next, we consider the scenario in which two potential senders, node i_1 and node i_2, send probes at an IS. As a common probe code is used, and no data is carried by a probe, node l can collect energy from both probes via a Rake receiver. The total received probe power at node l is also denoted by $P_l^{r,p}$. Suppose that node l wants to send a blocking message. Without loss of generality, assume the path gain from node i_1 to l is not less than that from i_2 to l, i.e., $g_{i_1 l} \geq g_{i_2 l}$. We have

$$P_l^{r,p} = \alpha P \cdot g_{i_1 l} + \alpha P \cdot g_{i_2 l} \leq 2\alpha P \cdot g_{i_1 l} \tag{9}$$

When node l sends a blocking message with power given by Equation (2), the received blocking message power at node i_1 is

$$\begin{aligned} P_l^{t,b} \cdot g_{li_1} &= P_l^{t,b} \cdot g_{i_1 l} \\ &\geq \frac{2 \cdot P_{th}^{r,b} \cdot \alpha P}{P_l^{r,p}} \cdot g_{i_1 l} \\ &\geq \frac{2 \cdot P_{th}^{r,b} \cdot \alpha P}{2\alpha P \cdot g_{i_1 l}} \cdot g_{i_1 l} \\ &= P_{th}^{r,b} \end{aligned} \tag{10}$$

As a result, node i_1 will be notified to give up the IS.

In the following, we show that, if node i_2's potential information transmission (i.e., DATA or ACK) generates extra interference that is intolerable at node l, node l's blocking message with power given by Equation (2) can reach node i_2 with a power level above the threshold $P_{th}^{r,b}$.

If node i_2's information transmission can corrupt node l's reception, this means

$$P \cdot g_{i_2 l} \geq \Lambda_l \tag{11}$$

We have

$$g_{i_2 l} \geq \frac{\Lambda_l}{P} \tag{12}$$

When node l sends a blocking message with power given by Equation (2), the received blocking message power at node i_2 is

$$P_l^{t,b} \cdot g_{li_2} = P_l^{t,b} \cdot g_{i_2 l}$$

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$$\geq \frac{P_{th}^{r,b} \cdot P}{\Lambda_l} \cdot g_{i_2 l}$$
$$\geq \frac{P_{th}^{r,b} \cdot P}{\Lambda_l} \cdot \frac{\Lambda_l}{P}$$
$$= P_{th}^{r,b} \tag{13}$$

As node i_2 detects blocking message power above the threshold, it gives up the IS.

In summary, when there are two probe senders in an IS, and the aggregate potential interference from their information transmissions is intolerable by an active receiver, the potential sender with stronger interference will be blocked. The potential sender with weaker interference will be blocked if its potential interference to the target active receiver is larger than the receiver's interference margin.

3.5. Impact of User Mobility and Solutions

The impact of user mobility on each active link at the MAC layer is two fold. One is due to the time-varying path attenuation of the desired signal, and the other is due to the varying interference level. The path attenuation consists of three components: path loss, shadowing, and fast (multipath) fading. Fast fading can be addressed by the Rake receiver that collects signal energy from different paths. The time-varying path loss, shadowing, and interference level make the SINR of each link fluctuate with time. Generally our scheme can automatically adapt to the varying SINR of each link, because each receiver determines whether or not to admit new links into the network based on real-time measurement and estimation of its desired signal power and interference levels. However, it is possible that the SINR of some ongoing links at an IS cannot remain above the threshold Γ_d because of the fluctuations. Thus, a *link failure* may happen. Although a new route searched by the routing scheme can help to solve the problem, it is desirable to deal with the problem at the MAC layer first before resorting to the routing scheme, considering the overhead and time needed for a new route search and the duration needed to set up a new route.

To address the link failure problem at the MAC layer, the traffic destination node of each active link indicates in its ACK two candidate DATA ISs at which the destination node experiences the minimum interference and does not transmit. The destination node also indicates a detected link failure (if any) in the ACK. If the traffic source node (say node i) does not receive correctly the ACK from its destination (say node j), this means the ACK is corrupted.[‡‡] As node i does not have the information of whether the DATA transmission is corrupted or not, we let the link be re-established in a new DATA IS and a new ACK IS. The main difference from the establishment procedure given in Section 3.3 is as follows. Node i sends two probes in the two candidate DATA ISs indicated in the previous ACK. At each candidate IS, a request is sent if no blocking message is detected. The request also indicates two candidate ACK ISs in which node i experiences the minimum interference and does not transmit. If at least one request is received at node j (which means a new DATA IS is established), node j sends two probes at the candidate ACK ISs. If at least one probe at a candidate ACK IS is not blocked by active receivers at that IS, a new ACK IS is established. If the new DATA IS and ACK IS cannot be established simultaneously, the call is dropped.

On the other hand, if node i successfully receives an ACK which indicates that a failure of DATA transmission happens, a similar procedure is executed, except that only the establishment of a new DATA IS is needed.

Generally, if the link failure is due to large interference, it is likely that the call can be established successfully in other DATA and/or ACK ISs at which nodes generating large interference are not active. However, if the link failure is due to a large path loss of the desired signal (e.g., the traffic source and destination are separated by a large distance) or due to shadowing of the desired signal (e.g., the path between the traffic source and destination is blocked by a high building), it is very likely that the link quality of the target call at other ISs is not good enough either, and thus, the call will be dropped. In this case, a new route should be selected by the routing scheme.

3.6. Summary

The proposed MAC scheme has the desired features discussed in Section 1. The scheme is performed in a distributed manner at the mobile nodes, and is thus scalable to large networks. The hidden terminal problem does not exist, since each active receiver makes an admission decision on a new call based on real-time measurement of its own desired signal power and interference levels. The reservation nature of our scheme

[‡‡]If an expected DATA frame is not received, the traffic destination node responds with a NACK. For simplicity of presentation , we use 'ACK' to represent both ACK and NACK in this paper.

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can guarantee small delay and jitter for voice traffic as long as the minimum required rate can be met. Our scheme can adapt automatically to user mobility to a certain extent, also benefiting from the fact that each active receiver makes real-time measurements of desired signal power and interference levels. In addition, the 'ideal assumption' required by CSMA and busy tone based schemes (as discussed in Section 2) is unnecessary for our scheme. Furthermore, although the blocking message has similar functionality to the busy-tone in busy tone based schemes, our scheme does not have the problem of different channel gains in the separate busy tone and DATA frequency bands, because the blocking message and DATA are sent in the same band.

4. Statistical Multiplexing of Voice Traffic

The resource reservation procedure in the preceding section is suitable for constant rate voice traffic, i.e., the sender generates packets at a constant rate during the call lifetime. However, voice traffic is normally characterized by an `on`/`off` nature. No information packets are generated in an `off` state, thus posing some challenges on the resource reservation procedure. When an `on` voice flow turns `off`, other active receivers measure less interference. If subsequently one or more new calls are admitted at the serving ISs of the `off` voice flow, and later the `off` voice flow switches to the `on` state, it is likely that some active receivers at the voice flow's serving ISs will experience more interference than they can tolerate, thus leading to link corruption. This problem is due to the inaccurate interference estimation when a voice flow turns `off`. Furthermore, it is more desirable to achieve multiplexing gain of `on`/`off` voice traffic. To address these problems, the following solutions are proposed.

4.1. Solution to Inaccurate Interference Estimation

When a voice flow is in an `off` state, it uses a probe (referred to as `off` *probe*) with a different code from that in Section 3.3 to inform other links of the interference that will be generated when the `off` voice flow turns `on` again. Specifically, when a voice flow from node i to node j turns `off`, its source and destination nodes send an `off` probe on the DATA and ACK IS, respectively, with power αP and a large spreading gain. Each active receiver keeps monitoring the `off` probe. At an active receiver l at the DATA or ACK IS, let $P_l^{r,p,\mathtt{off}}$ denote the received `off` probe power. Then, when a new call sender sends a probe (for admission) that arrives at node l with power level $P_l^{r,p}$, node l will send a blocking message if

$$\frac{P_l^{r,p}}{\alpha} + \frac{P_l^{r,p,\mathtt{off}}}{\alpha} > \Lambda_l \qquad (14)$$

This means the potential interference of `off` voice flow from node i to node j is counted in node l's current interference level, and thus, the inaccurate interference estimation problem can be addressed. When the voice flow from node i to node j turns `on` later, nodes i and j can send the DATA and ACK frame at the DATA IS and ACK IS, respectively, without any re-admission procedure.

4.2. Solution for Statistical Multiplexing

In Equation (14), each active receiver at an IS counts 100% of the potential interference from `off` voice flows, and thus, there is no statistical multiplexing gain. If an active receiver counts only a portion ξ (<1) of the potential interference from `off` voice flows, only part of the reserved capacity for `off` voice flows is maintained. So we propose that, when a new call sender sends a probe (for admission) that arrives at an active receiver node l with power level $P_l^{r,p}$, node l will send a blocking message if

$$\frac{P_l^{r,p}}{\alpha} + \xi \cdot \frac{P_l^{r,p,\mathtt{off}}}{\alpha} > \Lambda_l \qquad (15)$$

where ξ is referred to as *reservation factor*. By this means, only a portion, ξ, of the potential interference from `off` voice flows is counted by the receiver node l, and thus, only a portion, ξ, of the capacity for the `off` voice flows is reserved. In other words, new calls can use a portion, $1 - \xi$, of the capacity for the `off` voice flows.

On the other hand, when an `off` voice flow turns `on` and its source and destination nodes start to transmit in its previous DATA and ACK ISs, respectively, it is possible that the transmission accuracy of other links at the ISs may be corrupted. So re-admission of `off`-to-`on` voice flows is necessary as follows. When an `off` voice flow turns `on`, the voice source node sends a probe in its previous DATA IS and in another IS at which the voice destination node experiences the least interference (as indicated in the last ACK). At either IS, if no blocking

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message is detected, the voice source node sends a request which also indicates a candidate ACK IS (different from its previous ACK IS) at which it experiences the least interference and does not transmit. If the voice destination node receives at least one request, it sends a probe in its previous ACK IS and the candidate ACK IS. A new ACK IS is established if at least one probe is not blocked. The call will be dropped if a DATA IS and an ACK IS cannot be established simultaneously.

Note that the admission of an off-to-on voice request is different from that of a new call request. For a new call request, a certain portion of the capacity should be reserved for off voice flows, as shown in Equation (15). On the other hand, for an off-to-on voice request, such reservation is not necessary, and thus, we have the following criterion for blocking an off-to-on voice request

$$\frac{P_l^{r,p}}{\alpha} > \Lambda_l \tag{16}$$

Therefore, when an off voice flow turns on, the source or destination node should use a different probe code from that used by the source or destination of a new voice call (i.e., different from that used in Section 3.3). By this means, other active receivers can distinguish off-to-on voice requests from new voice call requests.

5. Performance Evaluation

We use computer simulation to evaluate the performance of the proposed scheme. Traditional QoS metrics for voice traffic include delay, jitter, and packet loss rate. From Section 3, it can be seen that, as long as a call is admitted into the network, the transmitter can send voice packets continuously in the reserved slot. So the voice delay is fixed with the maximum value being a time frame duration, and there is no jitter. Therefore, in the following, packet loss rate is used as the major QoS metric for voice in the simulation. The simulation is done using Matlab.

Consider an *ad hoc* network consisting of N nodes, with $N/2$ voice source nodes and $N/2$ voice destination nodes. The source nodes are randomly distributed in a 1000 m × 1000 m square, each associated with a destination node. All the source nodes are fixed, and all the destination nodes move with a speed v in a randomly selected direction. A destination node is assumed to move within a circle with the radius being 150 m and the center being of the associated source node.[§§] Each time frame has a fixed duration of 20 ms, and is further divided into 8 ISs and 8 BSs. The spreading gain is 32 for DATA and ACKs, and 3200 for probes. The path loss attenuation exponent β is 2.4, and the shadowing is modeled by the first-order autoregressive process given in Reference [17]. The voice call inter-arrival time at each source node is exponentially distributed with mean value 48 s, and the duration of a call is exponentially distributed with mean value 30 s. When a call is active, one DATA frame is generated over each time frame. The SINR requirement Γ_d is 10 dB. In the following, the simulation for each set of parameters ends when on average each source node finishes 100 calls.

Table I. The percentage of the ISs with different numbers of probes.

Number (N) of nodes	20	40	60	80	100
$v = 10$ kph					
1 probe	1.000	1.000	0.998	0.996	0.997
2 probes	0	0	0.002	0.004	0.003
≥ 3 probes	0	0	0	0	0
$v = 20$ kph					
1 probe	1.000	0.996	1.000	0.995	0.996
2 probes	0	0.004	0	0.005	0.004
≥ 3 probes	0	0	0	0	0

In the simulation, the number N of nodes varies from 20 to 100, and the destination node velocity v varies from 0 to 20 km/h (kph). We first verify our assumption in Section 3.4 that the probability of more than two probe senders at an IS is negligible. Among the ISs with one or more probe senders, Table I lists the percentage of ISs that have one probe sender, two probe senders, and more than two probe senders, respectively. No more than two probes are observed at an IS, thus validating our assumption.

We also compare our scheme with the IEEE 802.11 MAC scheme. In this comparison, $N = 100$, no node is moving, and an active source node generates constant rate voice flows. It is observed that the 95% confidence interval of the packet delay and jitter in the IEEE 802.11 MAC scheme is (1.27, 1.33) s and (0.038, 0.042) s, respectively. On the other hand, our scheme is reservation based. Thus, the packet delay bound is a time frame duration, i.e., 20 ms, and the jitter is zero. These results indicate that our scheme performs better in terms of timely delivery of voice packets.

[§§] Routing may be involved when a destination node moves beyond this distance, a situation which is beyond the scope of this analysis.

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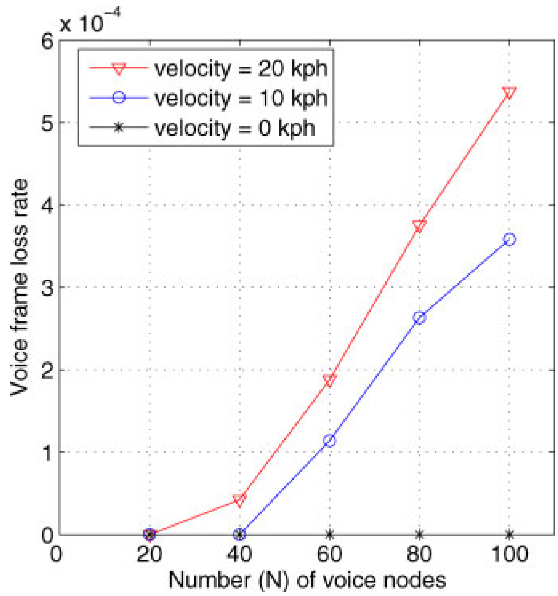

Fig. 5. The voice frame loss rate of a voice call.

Next we evaluate the voice frame loss rate and call dropping rate in our scheme with constant rate voice flows. For an ongoing voice call, if its SINR requirement is not satisfied, it tries to switch from one IS to another. Some voice frames will be lost during the switching due to the switching processing time, and if the switching is not successful, call dropping occurs. Figure 5 shows the voice frame loss rate with different N and v values. A larger value of N or v tends to increase the frequency of switching among the ISs, thus increasing the voice frame loss rate. It can also be seen that all the voice frame loss rates are bounded by 0.1%, which should be within an acceptable range for voice service. It is observed that the ongoing call dropping probability is bounded by 1.2%, which happens when $N = 100$ and $v = 20$ kph. So our scheme can maintain the voice frame loss rate and call dropping rate at low levels.

Table II. Ongoing call dropping probability and new call blocking probability for the cases with `on`/`off` voice flows.

ξ	0	0.5	1
$v = 0$ kph			
Dropping prob. (%)	0.6	0.3	0
Blocking prob. (%)	0	0	0.42
$v = 10$ kph			
Dropping prob. (%)	4.7	4.3	0.9
Blocking prob. (%)	0.2	0.3	3.6
$v = 20$ kph			
Dropping prob. (%)	5.1	4.9	1.2
Blocking prob. (%)	0.2	0.1	4.1

For `on`/`off` voice flows with statistical multiplexing in our scheme, the durations of `on` and `off` states are simulated as being independent and exponentially distributed with mean values 352 and 650 ms, respectively. For $N = 100$, Table II shows the ongoing call dropping probability and new call blocking probability. When the reservation factor increases, more capacity is reserved for `off` voice flows, and thus, the ongoing call dropping probability decreases. However, the new call blocking probability increases correspondingly due to the reduced available capacity for new calls.

6. Conclusions

It is challenging to support voice traffic over a mobile *ad hoc* network due to the delay-sensitive nature of such traffic and due to user mobility. In this paper we have proposed a MAC scheme to support voice traffic by the resource reservation mechanism. In the proposed scheme, user mobility is addressed by real-time signal and interference measurements and estimation at the receivers. Statistical multiplexing is achieved by partial resource reservation for `off` voice flows. The results of this study provide useful insight into the development and deployment of mobile *ad hoc* networks supporting multimedia services.

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