rDCF: A Relay-enabled Medium Access Control Protocol for Wireless Ad Hoc Networks

Hao Zhu
Department of Electrical & Computer Engineering
Florida International University
Miami, FL, 33174
Email: hao.zhu@fiu.edu

Guohong Cao
Department of Computer Science & Engineering
The Pennsylvania State University
University Park, PA 16802
E-mail: gcao@cse.psu.edu

Abstract— **It is well known that IEEE 802.11 provides a physical layer multi-rate capability, and hence MAC layer mechanisms are needed to exploit this capability. Several solutions have been proposed to achieve this goal. However, these solutions only consider how to exploit good channel quality for the direct link between the sender and the receiver. Since IEEE 802.11 supports multiple transmission rates in response to different channel conditions, data packets may be delivered faster through a relay node than through the direct link if the direct link has low quality and low rate. In this paper, we propose a novel MAC layer relay-enabled distributed coordination function (DCF) protocol, called rDCF, to further exploit the physical layer multi-rate capability. We design a protocol to assist the sender, the relay node and the receiver to reach an agreement on which data rate to use and whether to transmit the data through a relay node. Considering various issues such as bandwidth utilization and channel errors, we propose techniques to further improve the performance of rDCF. Simulation results show that rDCF can significantly improve the system performance when the channel quality of the direct link is poor.**

Index Terms: IEEE 802.11, simulations, MAC, wireless networks.

I. Introduction

With the advantage of low cost and high data rate, IEEE 802.11 based wireless networks are becoming extremely popular. In order to improve the network performance, it is fundamentally important to design good media access control (MAC) protocols to efficiently utilize the limited spectrum [2], [6], [20], [22]. Two different MAC mechanisms are supported by the IEEE 802.11 standard [11]: one is called *distributed coordination function* (DCF), which is based on carrier-sense multiple access with collision avoidance. With DCF, the mobile nodes can spontaneously form an ad hoc network without any pre-installed infrastructure. Such networks can be quickly deployed in civilian and military environments such as battlefield, disaster recovery, group conference and wireless office; the other is called *point coordination function* (PCF), which is based on polling and is built on the top of DCF. Currently, the PCF protocol has not been commercialized yet [12].

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IEEE 802.11 has physical-layer multi-rate capability [11], which means that data can be transmitted at a number of rates according to the channel condition. For example, when the signal-to-noise ratio (SNR) is high, i.e., error detection and recovery is not that important [9], an aggressive and efficient modulation scheme can be applied to increase the rate. When the SNR is low, a conservative and redundant modulation scheme should be applied to reduce the bit error rate. In practice, IEEE 802.11b supports transmission rates of 1, 2, 5.5, and 11 Mbps, and IEEE 802.11a supports data rates of 6, 9, 12, 18, ..., 54 Mbps [9], [19].

To exploit the physical layer multi-rate capability, researchers have proposed various protocols. At the network layer, some channel state aware routing schemes [6], [2], [20] have been studied to improve the end-to-end throughput by taking into account the channel condition as one of the route selection metrics. However, due to the long latency of route updates and the high control overhead, these schemes cannot quickly react to dynamic channel condition and can not achieve high bandwidth utilization. At the MAC layer, [9], [14], [19] have been proposed to exploit the multi-rate capability. The basic idea of these schemes is to let the sender select a proper transmission rate according to the history of the successful transmissions; or to let the receiver sense the channel condition before the transmission, and notify the sender via a control packet (e.g. the clear-to-send (CTS) packet). However, these schemes only utilize the data rate of the direct link between the sender and the receiver. In many cases, data may be delivered much faster through multiple links that have high transmission rates than through the direct link with a low transmission rate.

In this paper, we propose a novel DCF-based MAC protocol called *relay-enabled DCF* (rDCF) to further exploit the multi-rate capability of IEEE 802.11. Based on the channel condition among mobile nodes, rDCF can intelligently apply multi-hop (mainly two-hop in this paper) data transmission to achieve higher transmission rate. Specifically, when the direct link between the sender and the receiver can only support a low transmission rate, but there exists a relay node such that both the links from the sender to the relay node and from the relay node to the receiver can support high transmission rates, the impending packet can be delivered from the sender to the receiver faster by two-hop high speed transmission via the relay node. With rDCF, each mobile node senses the channel conditions among its neighbor nodes. Based on the collected channel conditions, if a node can become a relay node of its neighbors, it periodically advertises the relay information. When the sender sends the packet to the receiver, if it can find a relay node, a triangular handshake is formed among the sender, the relay node and the receiver so that they can quickly agree on whether to perform relay and which rate to use according to the real-time channel condition. To deal with issues such as bandwidth utilization and time-varying channel condition, we propose techniques to enhance the rDCF protocol. We evaluate the rDCF protocol in various scenarios, and the simulation results show that rDCF can significantly reduce the packet delay, improve the system throughput, and alleviate the impact of channel errors on fairness.

The remaining of this paper is organized as follows. Section II describes the background and the related work. Section III gives the motivation of the work. The details of rDCF are presented in Section IV. Section V analyzes rDCF. Section VI evaluates the performance of rDCF through simulations. Section VII concludes the paper.

II. Background and Related Work

A. System Model

We consider a wireless network based on IEEE 802.11b that can support transmission rates of 1, 2, 5.5 and 11 Mbps. The wireless medium is shared among multiple contending mobile nodes, i.e., a single physical channel is available for wireless transmission. The DCF with request-to-send (RTS)/clear-to-send (CTS) handshake is used for medium access control since it has been shown that the RTS/CTS mechanism is effective to solve the hidden terminal problem [3] and to improve the system performance when the packet size is large [4]. According to the channel condition, a packet could be transmitted at different transmission rates. We assume that data packets can be transmitted at different transmission rates, but control packets (e.g. RTS, CTS, ACK) are transmitted with the base rate which is 2 Mbps in this paper. For simplicity, we assume that each node transmits its packets using a constant transmission power. The wireless channel between the sender and the receiver is assumed to be almost symmetric. In this paper, we will not consider security issues and the motivation for nodes to relay. Many existing techniques [5], [10], [16] can be used to address security issues and the motivation for relay.

Based on the distance, the sensing power and the modulation scheme, a node can be in different range of the sender: the *transmission range* and the *carrier sensing range*.

- *transmission range:* within this range, the node can receive and correctly decode the packet.
- *carrier sensing range:* within this range, the node can sense the signal but cannot decode the packet.

B. The IEEE 802.11 DCF Protocol

The standard DCF protocol is described in [11]. After a transmitting node senses an idle channel for a time period of a *distributed inter-frame space* (DIFS), it backs off for a time period which is chosen uniformly from the range of 0 to its contention window size (CW). After each successful data transmission, the window size is set to CW_{min}, which denotes the pre-specified minimum contention window. After the backoff timer expires, the node sends a RTS to the receiver. If the receiver successfully receives the RTS, it replies a CTS after a time period of *short inter-frame space* (SIFS). When the sender receives the CTS, it transmits the impending packet. For the purpose of reliability, the receiver needs to reply an ACK after it receives the packet correctly. Any other node overhearing either the RTS or the CTS extracts the information contained in the packet and updates its *network allocation vector* (NAV), which indicates the time period reserved for data transmissions. Then, the node defers its transmission until its NAV expires. For each transmission failure, which may be caused by collisions or channel errors, a binary exponential backoff is applied to double the backoff window, and the window size is bounded by the maximum contention window (denoted by CW_{max}).

C. Related Work

Kamerman and Monteban [14] designed the auto rate fallback (ARF) protocol to utilize the multi-rate feature of IEEE 802.11. In ARF, the sender adapts the rate of each data transmission based on the history of previous successful transmissions. Since ARF is a sender-initiated protocol, it does not work well when the channel condition becomes unstable. Holland *et al.* [9] proposed a receiver-based auto rate (RBAR) protocol. With the rate feedback by the receiver, RBAR can adapt the channel condition more promptly than ARF. Later, the opportunistic auto rate (OAR) scheme was proposed in [19]. OAR utilizes the fragment burst in IEEE 802.11 [11], which allows more than one packets to be transmitted when the sender is granted medium access. OAR outperforms RBAR only when the channel condition between the sender and the receiver can support a high transmission rate (say 11 Mbps). ARF, RBAR and OAR only consider the channel quality between the sender and the receiver. When the channel quality between the sender and the receiver is poor, the performance of these schemes would be significantly degraded.

The channel quality has been used as a metric for route selection in some routing protocols [2], [6], [7], [20]. A path with overall best channel condition is selected to improve the end-to-end throughput [2], [6], [20] or power efficiency [7]. However, compared to MAC layer relay, network layer relay has higher control overhead and may incur a long queuing delay. When the channel condition changes frequently, due to the slow response of the routing protocols, network layer relay cannot react quickly to exploit the opportunities to deliver data at a high transmission rate.

In [22], a relay enabled PCF protocol, called rPCF has been proposed to utilize the multi-rate capability via two-hop MAC layer relay. In rPCF, each mobile node reports the sensed channel condition to the access point. Based on the collected information, the access point decides and notifies the node at which rates to apply relay through the polling packet. Compared to rPCF, the design of rDCF is much more challenging: First, rDCF needs to operate in a distributed way, and then it requires different techniques to coordinate the sender, the relay node and the receiver in rDCF. Second, we need to consider the exposed terminal problem and the hidden terminal problem in rDCF, which does not exist in rPCF.

III. Motivations

A. *Advantage of two-hop relay*

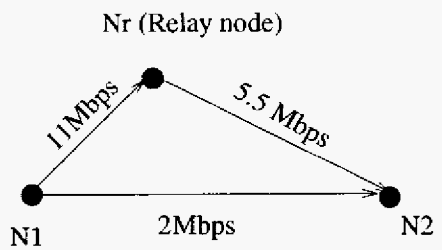


Fig. 1. The advantage of using the relay node

Since the channel condition varies with time and is location dependent [18], the multi-rate capability can be further exploited by enabling MAC layer multi-hop transmission. For example, as shown in Figure 1, suppose N_1 needs to send data to N_2, and the channel of $N_1 \rightarrow N_2$ only supports a transmission rate of 2 Mbps. At the same time, the channel conditions of $N_1 \rightarrow N_r$ and $N_r \rightarrow N_2$ are much better, and they can support data rates of 11 Mbps and 5.5 Mbps respectively. With a packet length of L, if the data can be transmitted along $N_1 \rightarrow N_r \rightarrow N_2$ at the MAC layer, the transmission delay is approximately $(\frac{1}{11} + \frac{1}{5.5})L$. Thus, the actual transmission rate is approximately equal to $\frac{5.5\times 11}{5.5+11} = 3.7Mbps$, which is much larger than 2 Mbps, when the packet is transmitted along $N_1 \rightarrow N_2$. Even after considering of the control overhead, when the packet size is not very small, the overall time to deliver the data packet can still be significantly reduced (see Section V for details). Although it is possible to have more than one relay nodes, considering the control overhead of the coordination among related nodes, we focus on two-hop MAC layer relay in this paper, which is sufficient in most cases.

There may be doubts on whether the relay mechanism will work since the channel conditions of $N_1 \rightarrow N_2$ and $N_2 \rightarrow N_r$ may be unstable, and then the actual transmission rate that can be achieved with relay could be lower than that with direct transmission. Fortunately, as stated in [19], when the node does not move very fast, i.e., less than 20 m/s, the coherence intervals [18], [19][1], are on the order of multiple packet transmission times. In most cases, since mobile nodes move fairly slow (say less than 5 m/s) in ad hoc networks, it is feasible to exploit relay opportunities for each packet transmission (if there exists a suitable relay node) so that the performance of the system can be significantly improved.

[1] The coherence interval is the average time interval during which the channel conditions are correlated.

B. *MAC layer relay vs. Network layer forwarding*

As we mentioned in Section II-C, the function of exploiting multi-rate capability can be performed through MAC layer relay or network forwarding. MAC layer relay is better than network layer forwarding in three aspects:

1) Packets relayed at the MAC layer do not have queuing delays, whereas packets forwarded at the network layer would experience a long queuing delay if the relay node has many packets in the queue.
2) Because each network forwarding evolves a RTS/CTS handshake plus an ACK, the control overhead of network forwarding is higher than that of MAC layer relay.
3) Network layer forwarding may affect the bandwidth allocation of the relay node, and then forwarding the packets of other nodes may affect the delivery of its own packets. In contrary, with MAC layer relay, because each relayed packet does not enter the queue of the relay node, MAC layer relay does not interfere the node's transmission opportunity. This property is helpful to apply some rewarding schemes [5] to motivate the relay.

IV. The Relay-Enabled DCF

In this section, we first present the basic protocol of rDCF, and then propose techniques to enhance it. Finally, we discuss the various impacts of the relay and some implementation issues.

A. *The Basic Protocol*

1) The Service Advertisement: Similar to most existing work [9], [19], we apply receiver-initiated channel condition measurement and let the receiver notify the sender of the transmission rate via CTS. With rDCF, each node promiscuously listens to all ongoing RTS and CTS packets. By extracting the piggybacked transmission rate in the CTS, a node knows the channel condition between the sender and the receiver. Meanwhile, it can measure the channel quality between the sender or the receiver and itself by sensing the signal strength of RTS or CTS respectively. Since CTS packets do not have the MAC address of the packet sender, a node needs to infer the sender of the CTS according to the semantic of CTS. In particular, suppose N_r overhears a RTS from N_i to N_j. If it overhears a CTS addressed to N_i after a SIFS, N_r can infer that the sender of the CTS is N_j.

For a given flow between a pair of sender and receiver, with the measured channel quality, if a node finds that the packets can be transmitted faster with the MAC layer relay, it adds the identity (e.g. MAC address) of the sender and the receiver into its willing list. In order to reduce the control overhead, we can limit the length of the willing list (i.e. 10 entries). Periodically, each node advertises its willing list to its one-hop neighbors. Some schemes such as [21] can be used to improve the reliability of the broadcast. Once a node, say N_i, receives a willing list from N_r, and finds that $N_i \rightarrow N_j$ is in the list, it adds N_r into its relay table (Note that it is possible that there are more than one relay nodes available for $N_i \rightarrow N_j$). As an optimization, the number of redundant service advertisements for a given flow can be reduced as follows: Before sending the advertisement, if N_r has overheard more than m advertisements containing $N_i \rightarrow N_j$ from other nodes, it knows that at least m other nodes have claimed to be the relay node for $N_i \rightarrow N_j$, and then deletes $N_i \rightarrow N_j$ from the willing list. In this paper, we set the value of m to be 3.

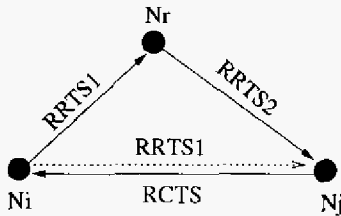


Fig. 2. An illustration of the triangular handshake

2) The Triangular Handshake: In the standard DCF protocol, the RTS/CTS handshake is required for each unicast packet transmission in order to prevent collisions. In [9], [19], this handshake is further utilized to probe the channel condition on a per-packet basis. Following these principles and considering backward compatible to the standard DCF, we modify DCF and refer this new protocol as the basic protocol of rDCF. As shown in Figure 2, where the dashed line pointed to N_j means that N_j can overhear the packet. When a node N_i has a packet for N_j, it first searches the relay table using N_j as index. If N_i cannot find a relay node, the standard DCF with receiving based rate feedback [9] is applied. Otherwise, N_i picks a relay node N_r and starts to coordinate the communication with N_r and N_j. Specifically, N_i sends a new packet, called *relay RTS* (RRTS1), to N_r. When N_r receives the RRTS1, it generates another relay RTS (RRTS2) and sends it to N_j. By sensing the signal strength of RRTS1 , N_r and N_j individually determines the achievable transmission rate of $N_i \rightarrow N_r$, $N_i \rightarrow N_j$, denoted by R_1, R_{dir} respectively. According to the signal strength of RRTS2, N_j determines the transmission rate of $N_r \rightarrow N_j$, denoted by R_2. Based on R_1 (piggybacked by RRTS2), R_{dir} and R_2, N_j replies CTS which piggybacks R_{dir} if the packet cannot be transmitted faster with relay. Otherwise, N_j replies a *relay CTS* (RCTS), which piggybacks R_1 and R_2, to N_i.

If N_i receives a CTS, it sends the data packet directly to N_j with the transmission rate of R_{dir}. If N_i receives a RCTS, as shown in Figure 3, it sends the data packet to N_r with the transmission rate of R_1. After N_r receives the packet, it relays the packet to N_j with the transmission rate of R_2 after a SIFS. If the packet is correctly received by N_j, N_j replies an ACK to N_i. If the transmission fails, the sender can detect it with

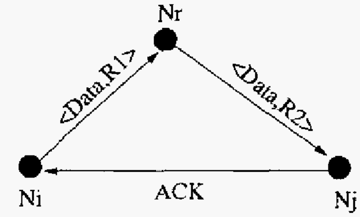


Fig. 3. An illustration of the MAC layer relay

a timeout mechanism similar to the standard DCF [11].

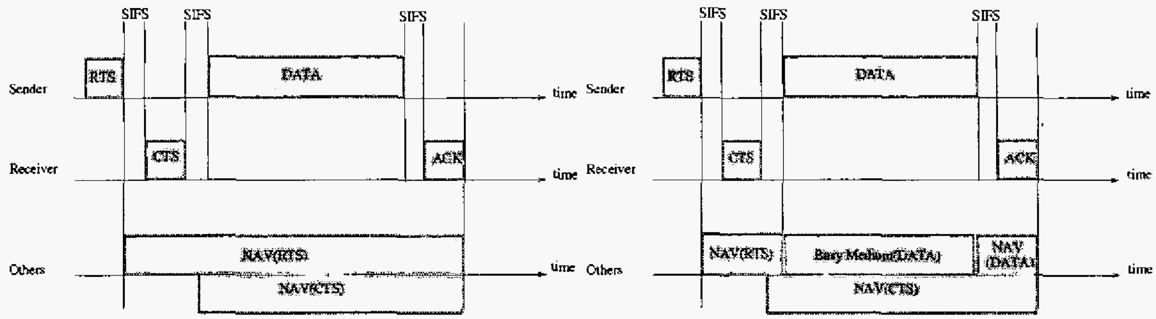
B. Enhancements of rDCF

The basic protocol of rDCF describes the basic mechanism to achieve relay-enabled DCF. However, considering the bandwidth utilization, the dynamical nature of wireless channels and the impact of multi-rate transmissions, we propose techniques to further improve the performance of rDCF.

1) Dealing with Multi-rate Transmission: With IEEE 802.11 DCF, carrier sensing is performed using physical carrier sensing as well as virtual carrier sensing. As shown in Figure 4 (a), when the data is transmitted with a fixed rate, the sender can easily calculate the duration of the packet transmission based on the packet length and the transmission rate. However, when the transmission rate can be adaptively changed, the sender cannot precisely calculate the length of the duration before sending the RTS, since it does not know the transmission rate of the impending packet in advance. In the solution of [9], the sender chooses a data rate based on some heuristic; i.e., the most recent rate that was successfully used for transmission. This solution is not good enough for rDCF since the sender needs to estimate the transmission rates for both hops of the relay, and it may be difficult to get a precise estimate.

Our approach: We designed a new carrier sensing scheme for rDCF, which is shown in Figure 4 (b). Instead of estimating the possible transmission rates and calculating the duration of the data transmission, the sender first calculates the duration of the RTS and CTS transmissions only[2]. The duration can be precisely calculated since all control packets (e.g. RTS, CTS, ACK, ...) are transmitted at the base rate, say 2 Mbps. After the sender receives CTS or RCTS, it calculates the durations of the data packet (if necessary) and the ACK based on the piggybacked transmission rate(s). In this way, our scheme can guarantee that other nodes within the transmission range of the sender and the receiver would defer medium access exactly for the data packet transmission time. Compared to the standard approach, our approach can achieve better bandwidth utilization in some situations. For example, suppose a CTS is lost at the sender due to collision or channel error, since the standard approach has longer duration piggybacked in the RTS than our approach, the neighbor nodes of the sender would defer for a longer time period in the standard DCF. Table I lists the duration for each packet used in rDCF. In the table, σ is the

[2]In case of relay, it needs to calculate the duration of RRTS1, RRTS2 and RCTS transmissions.



(a) The standard scheme (b) The new scheme

Fig. 4. The comparison of two different carrier sensing schemes

Packet Type	The Duration
RTS	$CTS + \sigma + 2SIFS$
CTS	$DATA(L, R_{dir}) + \sigma + 2SIFS + ACK$
RRTS1	$RRTS2 + RCTS + 2\sigma + 3SIFS$
RRTS2	$RCTS + DATA(L, R_1) + 2\sigma + 3SIFS$
RCTS	$DATA(L, R_1) + DATA(L, R_2) + 2\sigma + 3SIFS$ $+ACK$
Data_{dir}	$ACK + \sigma + SIFS$
Data_1	$DATA(L, R_2) + ACK + 2\sigma + 2SIFS$

TABLE I

THE CALCULATIONS OF THE DURATION IN rDCF

maximum propagation delay, $DATA(L, r)$ is the time needed to transmit the packet with length of L at rate r. Note that the calculation of each duration includes the transmission time of both PHY layer header and MAC layer header. $Data_{dir}$ refers to the data packet with direct transmission, and $Data_1$ is the data packet sent from the sender to the relay node. Other unlisted packets have a duration of 0.

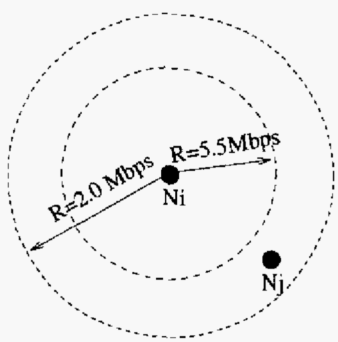


Fig. 5. An illustration of different transmission ranges

Besides the impact on virtual carrier sensing, different transmission rates also result in different transmission ranges. For a given receiving power level, the packet transmitted with higher rate may have higher bit error rate. As shown in Figure 5, suppose N_i and N_j are far away from each other and the channel quality can only support 2 Mbps. N_j may not be able to decode a packet if N_i sends the packet at the rate of 5.5 Mbps. In this case, N_j is out of the transmission range of N_i. Based on this fact, when the sender sends data at high rate, some one-hop neighbors may stay within its carrier sensing range but cannot extract the information of the duration piggybacked in the packet. To deal with such problems, we adopt the *reservation-sub-header* (RSH) in [9]. Specifically, a RSH is inserted preceding the data frame and is sent at the same or lower rate compared to RTS. Different

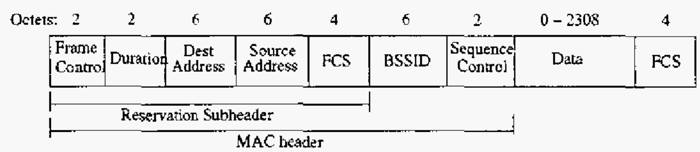


Fig. 6. Data packet frame format in [9]

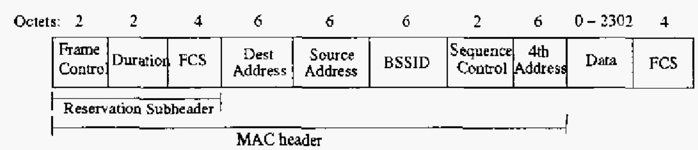


Fig. 7. Data packet frame format in our scheme

from [9], as shown in Figure 6 and Figure 7, our RSH does not need to include the MAC addresses of the sender and the receiver because the revised carrier sensing scheme would not incur any incorrect medium reservation of RTS. As a result, the overhead of our RSH is smaller than that in [9]. Since RSH is transmitted at a low rate (2 Mbps in this paper), all one-hop neighbor nodes can extract the duration in the RSH and update their NAV values accordingly.

2) Dealing with Dynamic Channel Condition: The channel condition may change frequently in wireless networks [18], which may have significant impacts on the performance of rDCF. In order to alleviate the impacts of dynamic channel conditions, it is desirable to adaptively decide when to perform relay according to the channel conditions.

We design a simple randomized algorithm as follows: Each relay node in the relay table of N_i is associated with a *credit* ranging in $[0.0, 1.0]$. To exploit successful relays, each time when N_i finds a relay node for the receiver N_j, N_i chooses the one with the largest credit. After selecting the relay node, N_i generates a random number in $[0.0, 1.0]$ and sends RRTS1 to the chosen relay node if the credit is greater than or equal to the random number. Otherwise, N_i applies DCF and sends RTS to N_j. When a node N_r successfully relays a packet for N_i, which is indicated by receiving the ACK, the credit of N_r is increased by 0.1. When a relay via N_r fails, the credit is decreased by 0.1. When N_i receives that willing list from N_r and finds itself in the list, the credit of N_r is enhanced by 0.5.

Some types of transmission failures can be detected and recovered quickly in rDCF to reduce the cost of failures. As shown in Figure 2, suppose N_i has a packet for N_j and finds the relay node N_r. We add two optimizations to the basic protocol as follows:

- If RRTS1 is lost, N_j can detect it if no packet is overheard after $SIFS + \sigma$ when the transmission of RRTS1 is finished. Then, it replies a CTS to N_i;
- If the data packet sent from N_i to N_r is lost, N_i can detect it if no packet is overheard after $SIFS + \sigma$. Then, N_i backoffs based on the binary exponential backoff protocol for re-transmission.

C. Impacts of Relay

In multi-hop ad hoc networks, the relay node may have some impacts on the system performance. In this section, we discuss some issues caused by relaying packets, and show that these impacts are very small in most cases through analysis.

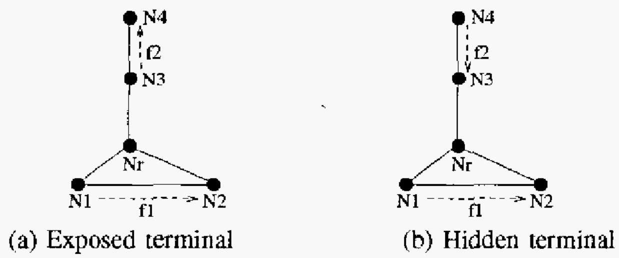


Fig. 8. An illustration of the impact of rDCF on spatial reuse

1) The Impact on Spatial Reuse: As packets being relayed, rDCF may have impacts on the spatial reuse of the network. As shown in Figure 8 (a) and (b), any pair of nodes connected by a solid line can hear each other. With the standard DCF, f_1 and f_2 can simultaneously transmit data since they don't contend with each other for the medium. When N_r relays packets for flow f_1, N_3 has to defer its transmissions in order to avoid collisions, which may cause exposed or hidden terminal problems [3], [4]. At a first glance, if N_r always relays packets for f_1, the performance of f_2 may be significantly affected. After looking into the carrier sensing mechanism of IEEE 802.11, we can see that the impact is quite small in most cases.

Suppose N_r relays a packet for f_1 at time t. For exposed terminal problem, there are two cases:

- **Case 1:** N_3 is in the transmission range of N_r at t, which means that it can extract the packet duration. N_3 can defer medium access for the exact time period of the ongoing data transmission, and then start to contend for the medium again. As a result, in the long run, N_3 and N_1 have similar opportunities to access the channel.
- **Case 2:** N_3 is within the carrier sensing range of N_r so that it cannot extract the packet duration. In this case, N_3 resumes contending the medium only when the medium is idle for an extended inter-frame space (EIFS), which is equal to 364 μs [11]. As a result, N_3 may defer the medium access to sometime later after N_1 receives the ACK. Since the expected time of (ACK+the post backoff[3]+$DATA(L, R_{1\to r})$) is greater than EIFS, where $R_{1\to r}$ is the transmission rate of $N_1 \to N_r$, we can see that N_3 would not be starved and can eventually obtain the medium access.

When N_3 transmits a packet to N_4, N_r sets its NAV to be either the data transmission time from N_3 to N_4 or EIFS (when a collision happens). When N_1 sends packets to N_r at this time, N_r will not send RRTS2 to N_2 since its NAV has not expired. In this case, the receiver applies the optimization technique in Section IV-B.2 and the impending packet of N_1 is served with DCF.

For the hidden terminal problem, the impact of relay could be greater since the sender of f_2 will double its current contention window size and backoff again. However, similar to the exposed terminal problem, since N_3 does not always sense busy medium, this impact would not significantly affect the performance of f_2.

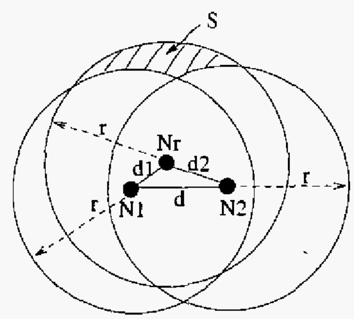


Fig. 9. An illustration of the extended sensing area

We also analyze the extended sensing area caused by N_r. As shown in Figure 9, the extended sensing area S is N_r's sensing area which does not overlap with the sensing areas of N_1 and N_2. It is not difficult to see that, for a given distance (d) between N_1 and N_2, the size of S increases as $d_1 + d_2$ increases. To meet the criteria of relay, $d_1 + d_2 \leq D_{5.5} + D_{11}$ should hold, where $D_{5.5}$ and D_{11} are the maximum transmission range of 5.5 Mbps and 11 Mbps respectively. By setting d_1 and d_2 to be $D_{5.5}$ and D_{11} respectively, we can calculate the upper bound of S.

We give some numerical results on the upper bound of increased sensing area as a function of d. Following ns-2 [8], we set r, $D_{5.5}$ and D_{11} to be 550 m, 200 m and 100 m respectively. d changes from 210 m to 250 m. The numerical

[3]After receiving the ACK, the sender is required to backoff for a random period between 0 and CW_{min}

d (meters)	210	220	230	240	250
Upper bound of increased sensing area (%)	11.5	10.5	9.2	8.2	7.2

TABLE II
THE IMPACT OF RELAY ON THE SENSING AREA

results are shown in Table II. As can be seen, compared to the total sensing area of the sender and the receiver, the increased sensing area is small.

2) The Impact of Hidden Relay: Based on the location of the relay node, some node may be able to hear from the sender, but unable to hear from the relay node. For example, as shown

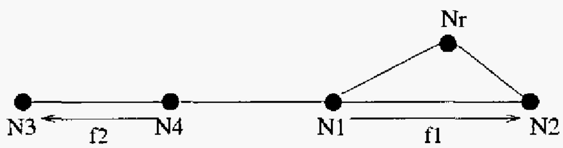


Fig. 10. An illustration of the impact of hidden relay node

in Figure 10, N_4 can hear from N_1 but cannot hear from N_r. This may cause collisions at N_1 since N_4 may not defer medium access for the period of one data transmission when N_r relays a packet for f_1. In the following, we analyze this impact, and show that it is very small. Suppose N_1 sends a packet to N_r at time t, there are two cases:

- **Case 1:** N_4 can extract the duration from the packet, and defer medium access accordingly. Since the duration is equal to the time needed for relaying the data packet, N_4 would not contend for the medium before N_1 gets the ACK.
- **Case 2:** N_4 cannot extract the duration from the packet, and set its NAV to be EIFS. With DCF, EIFS can be used to guarantee that the sender can receive the ACK. However, it may not always hold in rDCF. Since EIFS could be smaller than $DATA(L, R_{r\to 2}) + ACK + DIFS$, N_4 may send a packet to N_3 before N_1 receives the ACK, since it does not sense the signal of the packets sent by N_r and N_2. As a result, it is possible that the packet sent by N_4 collides with the ACK at N_1.

When Case 2 happens, N_1 needs to re-transmit the data packet. As stated in Section IV-B.2, N_1 also reduces the credit of N_r by 0.1 since the previous relay operation failed. Even if the flow rate of f_2 is high, the occurrence of Case 2 is bounded since the credit of N_r will eventually be small enough so that N_r would not be chosen for relay.

D. Implementation Issues

In this section, we describe how rDCF can be incorporated into IEEE 802.11. The MAC layer header and the format of the MAC frame used for unicast is shown in Figure 7. Similar to the standard [11], each MAC frame has four address fields to indicate the BSS identifier (BSSID), source address (SA), destination address (DA), and the fourth address. These addresses may appear in different order and in different type of frames. In order to support rDCF, some minor modifications to the standard 802.11 frames are required: Each relay related data or control frame (e.g., RRTS1) uses all four address fields in the order of SA, DA, BSSID, and the fourth address. The first and second hop relay can be differentiated by the subtype value[4] in the frame control field. With SA, DA and the fourth address fields, the addresses of the sender, the relay node and the receiver can be stored in each frame. In order to identify the piggybacked transmission rates, we append an 8-bit *rate tag* to the frame if necessary. The tag is divided into two 4-bit fields, which can be used to represent two transmission rates. Since many functions of DCF (e.g. RTS/CTS, rate adaptation) are implemented in firmware [12], these modifications can be easily done.

V. ANALYSIS OF rDCF

In this section, we analyze the saturation throughput gain of rDCF over the single rate DCF (operating at 2 Mbps). For simplicity, we assume the channel condition is ideal (i.e. no hidden terminals and capture [4]) and all flows are always backlogged. The cases with dynamic channel condition are studied through simulations (see Section VI).

The analytic model developed by Bianchi [4] is used for throughput analysis because it can be used to model various CSMA/CA based MAC protocols provided that collision avoidance follows binary exponential backoff. Let CW_{min} denote the minimum contention window size (in the number time slots), and assume that each node applies the binary exponential backoff scheme with the maximum backoff stage m (i.e. $CW_{max} = 2^m * CW_{min}$). For a fully connected topology with n flows, the probability τ that a flow transmits in a slot time is obtained from the following function:

$$2(2(1-\tau)^{n-1}-1)-(2(1-\tau)^{n-1}-1)(CW_{min}+1)\tau+ \\ (1-(1-\tau)^{n-1})CW_{min}(1-(2^m(1-(1-\tau)^{n-1})^m)\tau = 0 \quad (1)$$

Since we do not consider capture, as shown in Figure 4, the carrier sensing scheme of rDCF is exactly equivalent to that of DCF. In rDCF, for each node other than the sender and the receiver sending the packet, the node defers its own transmission in the same way as in DCF, no matter it relays the packet or not. With the fact that rDCF and DCF have the same backoff scheme, we can see that the process of the channel contention at each node in rDCF is the same as that in DCF. Consequently, the time spent in contention for each node in rDCF is the same as that in DCF. The following shows the average time for the channel being sensed busy under DCF and rDCF which are denoted as T_s^{DCF} and T_s^{rDCF} respectively, and average time spent in contention, denoted as T_c:

$$\begin{aligned} T_s^{DCF} &= RTS + CTS + ACK + DATA(L, R_b) \\ &\quad +4SIFS + 4\delta + DIFS &(2)\\ T_s^{rDCF} &= RRTS1 + RRTS2 + RCTS + ACK \\ &\quad +DATA(L, R_1) + DATA(L, R_2) \\ &\quad +5SIFS + 5\delta + DIFS &(3)\\ T_c &= RTS + DIFS + \delta &(4) \end{aligned}$$

[4]The subtype value can be selected from the reserved ones between 1000 and 1111 (binary).

where L is the packet length, R_b is the base rate (i.e. 2 Mbps), and R_1 and R_2 are the transmission rate of the first hop and the second hop relay respectively. Note that the time spent by each packet includes the overhead of PHY and MAC header, which is obtained according to each frame format in DCF and rDCF respectively. With the results of [4], the ratio between the saturation throughput of rDCF and that of DCF, denoted by γ, follows:

$$\gamma = \frac{(1-P_{tr})\sigma + P_{tr}P_sT_s^{DCF} + P_{tr}(1-P_s)T_c}{(1-P_{tr})\sigma + P_{tr}P_sT_s^{rDCF} + P_{tr}(1-P_s)T_c} \quad (5)$$

and

$$P_{tr} = 1-(1-\tau)^n \quad (6)$$

$$P_s = \frac{n\tau(1-\tau)^{(n-1)}}{P_{tr}} \quad (7)$$

where σ is one time slot.

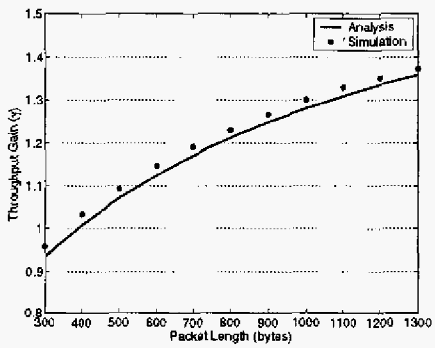


Fig. 11. Throughput gain: analysis versus simulation

With Eq 5, we show the numerical results of the throughput gain as the function of packet length L. We also validate our analysis through simulations. We assume that $n = 5$, $CW_{min} = 32$, $m = 5$, and each flow has a relay node which provides $R_1 = 5.5Mbps$ and $R_2 = 11.0Mbps$. As shown in Figure 11, the results between analysis and simulation are quite close. We can see that the throughput gain increases as L increases. In particular, when L is too small (say less than 400 bytes), rDCF performs worse than DCF. The reason is that when L is too small, the reduced transmission time by relaying data packet cannot combat the extra control overhead of rDCF (e.g. RRTS2 packets).

VI. Performance Evaluation

A. The Propagation Model

When the wireless channel is assumed to be stable, we use the propagation model in ns-2 [8], which combines the Friis free space propagation model and the two-ray ground propagation model [18]. Basically, when the sender and the receiver are close, the Friis free space model is applied so that the path loss exponent is 2. Otherwise, the two-ray ground propagation model and the path loss exponent becomes 4.

When there is multi-path fading or relative movement between the sender and receiver, the channel condition between them may change frequently. The frequency of this change depends on the relative speed of the mobile node with respect to its surroundings. We use the Ricean fading model [18] to simulate the fading channel conditions. The Ricean distribution is given by:

$$p(r) = \frac{r}{\alpha^2}e^{-(\frac{r}{2\alpha^2}+K)}I_0(2Kr) \quad (8)$$

where K is the distribution parameter representing the line-of-sight component of the received signal, α^2 is the variance of the background noise, r is the received power, and $I_0(.)$ is the modified Bessel function of the first kind and zero order [18].

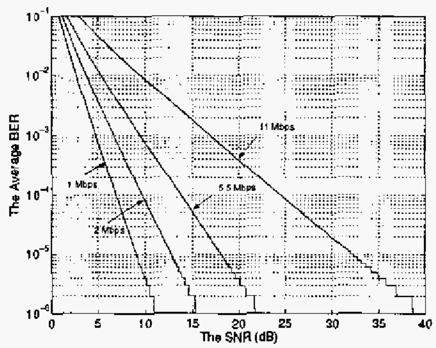


Fig. 12. The BERs under different transmission rates

When a node receives or overhears a packet, it determines whether the packet is corrupted according to the packet length, the SNR and the corresponding bit error rate (BER). With the BER of BPSK given by [15] and the approximate BER performance using different modulation techniques in [1], we have the BERs at different transmission rates shown in Figure 12. The probability that p can be successfully received, denoted by P_{succ}, is calculated by:

$$P_{succ} = (1 - BER(\gamma))^L \quad (9)$$

where $BER(\gamma)$ is the BER with the SNR of γ, and L is the packet length.

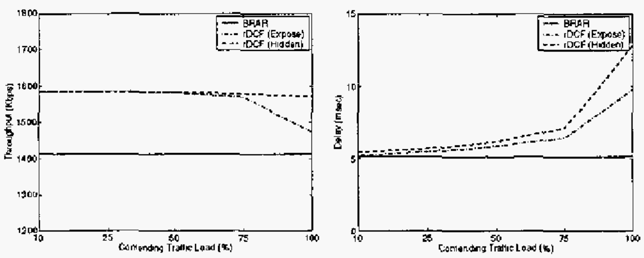
B. The Simulation Setup

Our simulation is based on ns-2 and its extensions [17], [8]. Similar to [19], the distance thresholds for 11Mbps, 5.5Mbps, and 2Mbps are 100m, 200m, and 250m respectively. The thresholds for different data rates are chosen based on the distance range. The mean period for service advertisements is 1.0 second. The data packet length is set to be 1000 bytes and the simulation time is set to be 100 seconds. Based on the analytical results in Section V, we set the packet size threshold for relay to be 400 bytes. We run each case 5 times and use the average as the simulation result.

We compare rDCF with the state-of-the-art protocol called *receiving based auto rate* (RBAR) protocol [9]. It has been shown that RBAR outperforms the standard DCF and the sender-based rate adaption protocol called auto rate fallback (ARF). We do not compare rDCF with the opportunistic auto rate (OAR) protocol since OAR degrades to RBAR when the link quality between the sender and the receiver is poor. The RBAR protocol works as follows. The receiver measures the channel quality based on the signal-to-noise ratio of the arriving RTS packet. Then, it sets the transmission rate according to the highest feasible value allowed by the channel condition, and piggybacks the rate with the CTS packet. After receiving the CTS, the sender sends out the data packet with the piggybacked transmission rate.

We use throughput and delay to measure the performance. The throughput is the total amount of data (in bits) delivered divided by the simulation time. The packet delay is the time interval from the packet entering the sender's queue to the time being delivered to the receiver. Note that the control overhead is also counted in the measurement.

C. Simulation Results



(a) Throughput of Flow 1 (b) Delay of Flow 2

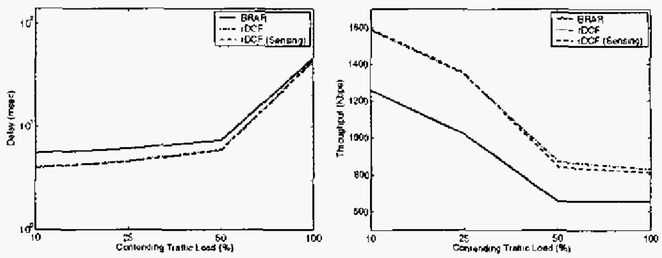
Fig. 13. The impact of rDCF on spatial reuse

1) Impacts on Spatial Reuse: In this experiment, we evaluate the impacts of rDCF on the spatial reuse, and assume the channel condition is stable. The topologies used are shown in Figure 8 (a) and (b), under which the performance results are denoted as rDCF (Exposed) and rDCF (Hidden) respectively. The channel quality between the sender and the receiver of each flow can only support 2 Mbps. N_r and N_3 are within the carrier sensing range of each other. The *contending traffic load* (CTL) (in percentage of the saturation throughput) of flow 1 (flow 2) increases as the aggregated traffic of the flows that are spatially close to flow 2 (flow 1) increases, and vice versa.

We first evaluate the impacts of CTL on the throughput of flow 1. Suppose flow 1 is backlogged. As shown in Figure 13 (a), when the CTL of flow 1 is not high (e.g. 50%), the throughput of flow 1 under rDCF is not affected and is much higher than that under RBAR. In case of rDCF (Expose), when the CTL of flow 1 is high (i.e. over 75%), the throughput of flow 1 decreases. As discussed in Section IV-C.1, since N_r frequently defers the medium access of flow 2, many data packets are transmitted with direct transmissions. In case of rDCF (Hidden), the impact of flow 1's CTL is very small since N_3 only sends short packets (i.e. CTSs and ACKs). Note that N_3 and N_r are within the carrier sensing range of each other.

We then evaluate the impacts of CTL on the delay of flow 2. The rate of flow 2 is fixed to be 160 Kbps (or 20 pkt/sec). As shown in Figure 13 (b), in case of both rDCF (Expose) and rDCF (Hidden), when the CTL of flow 2 is not high (i.e. less than 50%), its impact on flow 2's delay is quite small. When the CTL of flow 2 is very high (i.e. near 100%), the delay of flow 2 increases. The reason of the prolonged delay has been discussed in Section IV-C.1, and the result conforms our claim that flow 2 would not be starved.

D. The Impact of Hidden Relay



(a) Delay of flow 1 (b) Throughput of flow 1

Fig. 14. The impact of hidden relay on rDCF

We study the impact of hidden relay in this section. The topology has been shown in Figure 10. We assume that the channel is stable. By default, in rDCF we assume N_4 can extract the duration of each data packet sent by N_1. rDCF (Sensing) denotes the situation that N_4 cannot extract the duration field. As stated in Section IV-C.2, the impact of hidden relay does not exist in the default rDCF, but it exists in rDCF (Sensing).

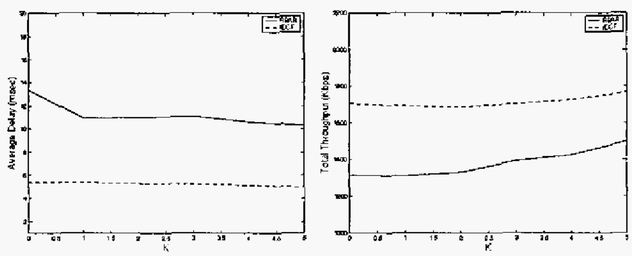
We evaluate the impact of CTL on the delay of flow 1. The rate of flow 1 is fixed to be 160 Kbps. As shown in Figure 14 (a), when the CTL of flow 1 is low, because of relay, the delay of flow 1 in rDCF and rDCF (Sensing) is much smaller than that under RBAR. As the CTL of flow 1 increases, the delay of flow 1 under rDCF and rDCF (Sensing) increases and becomes close to that under RBAR. Since N_4 and N_1 can hear each other, they compete the medium access. As a result, as the CTL of flow 1 increases, N_1 takes more time to contend the medium. From the figure, we can also see that the delay of flow 1 under rDCF (Sensing) is almost the same as that under rDCF, which shows that the impact of hidden relay on the delay of flow 1 is almost negligible.

We then examine the impact of CTL on the throughput of flow 1, which is always backlogged. As shown in Figure 14 (b), the throughput of flow 1 under rDCF and rDCF (Sensing) is always greater than that under RBAR. Only when the CTL of flow 1 is high (say more than 50%), we can see the difference between rDCF and rDCF (Sensing). As expected in Section IV-C.2, due to collisions caused by N_4, the throughput of flow 1 under rDCF-S is less than that under rDCF. However, the throughput difference is small, which shows that the impact of hidden relay on the throughput of flow 1 is not a big issue.

E. Fully Connected Topology

In this subsection, we study the performance of rDCF in a fully connected topology where nodes can hear each other. We put 20 nodes in the area (220m $\times$ 220m). Among them, 10 nodes act as either the sender or the receiver of the five flows. To examine the effectiveness of relay, we assume the average channel condition between the sender and the receiver of each flow can only support 2 Mbps. The remaining 10 nodes are randomly distributed in the area. We use the Ricean propagation model to emulate the dynamic channel condition and evaluate the impacts of the line-of-sight parameter K and the mobility.

1) Impact of K: The channel condition could be quite dynamic due to various factors. One important factor is the line-of-sight parameter K. A large K means a good channel quality while a small K means a poor channel quality. We first set the rate of each flow to be 160 Kbps and evaluate the packet delay under rDCF and RBAR. As shown in Figure 15 (a), the delay under rDCF is much smaller than that under RBAR and the impact of K on rDCF is smaller than that on RBAR. We then evaluate the system throughput under rDCF and RBAR by letting all the flows always backlogged. As shown in Figure 15 (b), under rDCF and RBAR, the system throughput increases as K increases, since the system-wide channel condition becomes better when K is larger. Compared to RBAR, rDCF can have much higher system throughput (at least 25% more). The performance gain is mainly due to the high transmission rate achieved by the MAC layer relay.



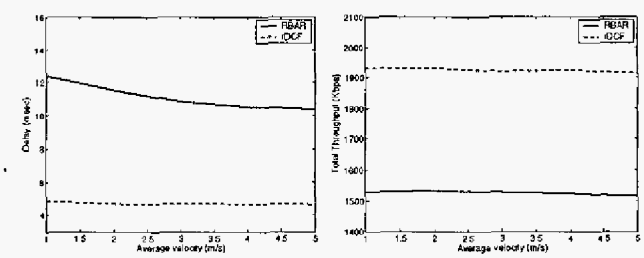
(a) Delay (b) Throughput

Fig. 15. The performance comparison between RBAR and rDCF under different K

Fig. 16. The fairness comparison between RBAR and rDCF

After looking at the throughput of each flow, we found that the impact of channel errors on fairness can be significantly reduced by rDCF. Figure 16 shows the throughput of each flow when K=0. As can be seen, under RBAR, the throughput of flow 3 and flow 5 is much less than that of flow 1, flow 2 and flow 4. The reason is that the distance between the sender and the receiver of flow 3 and flow 5 is longer than that of other flows. As a result, the accumulated time period when the channel condition is poor becomes larger, which causes more packets of flow 3 and flow 5 being lost due to channel errors. Consequently, due to the binary exponentially backoff, the accumulated backoff time of flow 3 and flow 5 becomes more than other flows. However, as shown in the figure, this unfairness does not exist under rDCF. The reason is that most packets from flow 3 and flow 5 can be delivered via relay, where both the channel conditions between the sender and the relay node and between the relay node and the receiver are more stable than the direct link. As a result, the number of transmission failures due to channel errors can be significantly reduced by using relay.

2) Impact of Mobility: Mobility affects the channel condition in two ways. First, it changes the node's location which may affect the value of K and the strength of the received signal strength. Second, due to Doppler shift in frequency of the received signal, it may reduce the channel coherence time period. We evaluate the impact of mobility on the performance of rDCF. Similar to [9], each receiver of a flow keeps moving back and forth. More specifically, it moves toward the sender until the distance between them is equal to 200m, and then moves back until the distance between them is 250m. Similar to [19], K is fixed to be 5. As shown in Figure 17 (a), the delay under rDCF slightly decreases as the mean moving speed increases. This can be explained as follows: as the moving speed increases, the receiver may have more chances to move closer to the sender, which makes the average channel quality between the sender and the receiver better. With relay, rDCF outperforms RBAR because it can have higher transmission rate when the sender and the receiver are far away from each other. For the same reason, as shown in Figure 17 (b), the total throughput under rDCF is much better than that under RBAR.



(a) Delay (b) Throughput

Fig. 17. The performance comparison between RBAR and rDCF under different velocities

	Delay (msec)	Throughput (Kbps)
RBAR	92.1	270.3
rDCF	17.5	387.5

TABLE III

THE PERFORMANCE COMPARISON UNDER THE MULTI-HOP TOPOLOGY

F. Multi-hop Topology

We evaluate the performance of rDCF under multi-hop topology in which 30 nodes are randomly distributed in a rectangular area of 1000m $\times$ 400m. All nodes are assumed to move around the area randomly and the mean moving speed is 3 m/s. The line-of-sight parameter K is set to be 5. Similar to [9], we simulate a single flow in the system and the routing protocol is the dynamic source routing (DSR) [13]. The end-to-end delay (when the flow rate is 160 Kbps) and the end-to-end throughput (when the flow is always backlogged) are shown in Table III. As can be seen, compared to RBAR, rDCF has significantly shorter delay since the impact of channel error has been largely relieved via relay. With the same reason, rDCF also achieves much better throughput than RBAR.

VII. CONCLUSIONS

In this paper, we presented a novel relay-enabled DCF protocol, called rDCF, to exploit the physical layer multi-rate capability. According to the channel condition, data can be transmitted with different rates, and some data packets may be delivered faster through a relay node than through the direct link if the direct link has low quality and low rate. The basic protocol of rDCF is proposed to help the sender, the relay node and the receiver coordinate to decide what data rate to use and whether to use a relay node. Considering the bandwidth utilization and the dynamic nature of wireless channels, we also propose several techniques to enhance the performance of rDCF. Simulation results showed that rDCF outperforms the receiver-based auto rate (RBAR) protocol in terms of throughput and delay in various scenarios. In the future, we can enhance the analysis considering various distributions of node positions and data rates. We can further improve the performance of rDCF by considering power efficiency in ad hoc networks. For example, when there are more than one relay nodes available, the remaining power level of these relay nodes can be used as a factor for the sender to choose the relay node.

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