# A Rate-adaptive MAC Protocol Based on TCP Throughput for Ad Hoc Networks in fading channels

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***Abstract*— Wireless technology is becoming a leading option for future Internet access. Transmission Control Protocol (TCP) is one of the protocols designed on the basis of the transmission characteristics in wired networks. It is known that the TCP performance deteriorates drastically under a wireless communication environment. On the other hand, many wireless networking standards such as IEEE 802.11a, 802.11b, and 802.11g have multirate capability. Therefore, adaptive rate control methods have been proposed for ad hoc networks. However, almost methods require the modification of the request to send (RTS) and clear to send (CTS) packets. Therefore, the conventional methods are not compatible with the standardized system.**

**In this paper, we propose adaptive rate control mechanisms for ad hoc networks. Our mechanisms are based on the RTS/CTS mechanisms. However, no modifications to the RTS and CTS packets are required in the proposed method. Therefore, our proposed method can attempt to satisfy the conventional IEEE 802.11 standards. Moreover, an adequate transmission rate is selected based on an estimated TCP throughput performance. From simulation results, it is observed that the proposed method can improve the throughput performance without any modification of packet structures.**

***Keywords*— Ad hoc networks, Cross-layer design, Rate Control, Media Access Control, Fading channel**

## I. Introduction

Wireless technology is becoming a leading option for future Internet access. Transmission Control Protocol (TCP) is one of the protocols designed on the basis of the characteristics of wired networks. In TCP, senders detect a segment loss caused by network congestion which is a characteristic feature of the wired network. The senders also determine an adequate transmission rate on detecting the segment loss. However, under wireless communication environments, considerable segment losses occur due to transmission errors in the wireless link. Therefore, the TCP sender erroneously judges the segment loss caused by the transmission error to be congestion. The sender suppresses the transmission rate itself, even if the obtained bandwidth is adequate. Hence, since most transmission of data is no longer transmitted, the TCP performance deteriorates drastically[1], [2].

On the other hand, many wireless networking standards such as IEEE 802.11a, 802.11b, and 802.11g have multirate capability. With the multi-rate mechanisms, transmission can take place at various rates according to channel conditions. Therefore, an adaptive rate control method has been proposed for ad hoc networks[3], [4]. Auto Rate Fallback (ARF)[6], Receiver Based Auto Rate (RBAR)[7] and Opportunistic Auto Rate (OAR)[5] have been investigated for as Media Access Control (MAC) layer mechanisms. The ARF is a sender-based MAC protocol that supports multirate capability. In the ARF, if ten consecutive acknowledgment (ACK) packets are received successfully or the timer expires, the transmission rate is increased. On the contrary, if two consecutive ACK packets are not received, the subsequent transmissions are made at the next lower transmission rate. Therefore, the ARF scheme cannot handle fast channel variations. The RBAR is a more effective scheme that uses multirate capability. In the RBAR, after receiving a request to send (RTS) packet, the receiver calculates the transmission rate to be used by the upcoming data packet on the basis of the signal-to-noise ratio (SNR) of the received RTS packet; the selected transmission rate is sent to the sender through a clear to send (CTS) packet. Therefore, the RBAR can handle fast channel variations. However, the RBAR requires the modification of the RTS and CTS packets to include the data packet size and the selected transmission rate. Hence, the RBAR is incompatible with IEEE 802.11 standards. The OAR is a similar scheme to the RBAR. The key idea of the OAR is to opportunistically utilize high quality channels. Therefore, the OAR allows multiple packet transmissions under the good channel condition. However, it is also incompatible with IEEE 802.11 standards. Finally, although almost methods can improve the communication performance, more discussions about a condition for the data transmission are required. Therefore, it is difficult to set adequate values to the control parameters to improve performance.

In this paper, we propose an adaptive rate control mechanism for ad hoc networks. Our mechanism is based on the RTS/CTS mechanisms such as the RBAR and OAR. The key idea of the proposed mechanism is to utilize the network allocation vector (NAV), which indicates the transmission duration, in the RTS and CTS packets to convey the required information. However, no modifications to the RTS and CTS packets are required. Therefore, our proposed method can attempt to satisfy the conventional IEEE 802.11 standards.

Moreover, the transmission rate is selected based on the TCP throughput performance, because it is known that the TCP throughput deteriorates rapidly under wireless communication environments. From simulation results, it is observed that the proposed method can improve the throughput performance without any modifications of packet structures.

## II. Adaptive rate control methods

A variety of adaptive rate control methods has been proposed. In order to control the transmission rate, some information should be exchanged between a sender and a receiver. Therefore, almost every method requires some modifications of the packet structures in the MAC layer. However, MAC layer protocols such as IEEE 802.11 are standardized. Therefore, compatibility with the standardized MAC protocols is lost if the packet structures are modified for the new scheme. Hence, it is not practical to modify the packet structure in order to implement conventional methods.

In this paper, we propose an adaptive rate control method without packet structure modifications. Therefore, our method is compatible with the IEEE 802.11 standards. In the proposed method, the receiver estimates the channel condition and packet error rate for the upcoming data packet and also estimates the TCP throughput performance for the transmission rate. The receiver then decides the adequate transmission rate for each data packet and conveys it to the sender through a CTS packet. As a result, the proposed method can handle various fading channels.

### A. Estimation of packet error rate

The packet error rate is generally calculated by the bit error rate (BER) and the packet length. Assuming the IEEE 802.11 to be the ad hoc network protocol, the receiver estimates the SNR for the RTS packet and obtains the BER $P_b(r)$ with transmission rate $r$ by the equations derived from an analytical model.

The RTS packet of the IEEE 802.11 includes the network allocation vector (NAV), which indicates the transmission duration; thus, the receiver calculates the packet length of the upcoming data packet by using the transmission duration and transmission rate in the wireless channel. Since the initial NAV, $NAV_0$, is set into the RTS packet, it includes the transmission duration of a CTS packet, the data packet and an ACK packet. The initial NAV value is obtained as follows.

$$NAV_0 = 3D_{SIFS} + D_{DATA(R_0)} + D_{CTS} + D_{ACK} \quad (1)$$

In this expression, $R_0$ is the base transmission rate that the sender uses at initial transmission; $D_{SIFS}$, the duration of a short interframe space (SIFS); $D_{DATA(R_0)}$, the transmission duration of the data packet with the transmission rate $R_0$; $D_{CTS}$, the transmission duration of the CTS packet; and $D_{ACK}$, the transmission duration of the ACK packet. From NAV value, the receiver can obtain the packet length without the modification of the packet structures. The packet length is as follows.

$$L = R_0(NAV_0 - 3D_{SIFS} - D_{CTS} - D_{ACK}) \quad (2)$$

We can estimate the bit error rate from the received SNR of the RTS packet and the packet length. Specifically, the packet error rate of the upcoming data packet is obtained as follows.

$$P_f(r) = 1 - (1 - P_b(r))^L \quad (3)$$

### B. Selection of transmission rate

In order to select an adequate transmission rate, some criteria can be considered. However, we employ the estimated TCP performance as the criteria for the transmission rate selection in this paper. Because TCP is designed on the basis of the transmission characteristics of wired networks, and it is known that the TCP performance deteriorates sharply under wireless communication environments. The TCP performance generally depends on the segment error rate. Therefore, we estimate the TCP throughput in advance by using an analytical model[9], [10] or simulation results. Figure 1 shows the TCP performance which is obtained by computer simulations. From this figure, we can find that the TCP performance deteriorates rapidly as the segment error rate increases. The estimated TCP throughput $T_{TCP}(r)$ is as follows.

$$T_{TCP}(r) = TCP_{throughput}(P_f(r)) \quad (4)$$

If the transmission rate is different, the transmission duration is also different. Therefore, the consumed wireless resource is also different. Therefore, we take into consideration the consumed wireless resource for each transmission rate. The estimated TCP throughput considering the consumed wireless resource is

$$T_{TCP-RATE}(r) = T_{TCP}(r)r/R_{MAX} \quad (5)$$

where $R_{MAX}$ is the maximum transmission rate. Finally, we can select an adequate transmission rate $R_s$ by using the TCP performance and consumed wireless resource.

$$R_s = \max_r(T_{TCP-RATE}(r)) \quad (6)$$

### C. Estimation of selected transmission rate

In order to convey the selected transmission rate to the sender, the receiver calculates a new NAV value based on the selected transmission rate. If the transmission rate is changed to $R_s$, the transmission duration of the data packet is as follows.

$$D_{DATA(R_S)} = L/R_s \quad (7)$$

Since the new NAV value is included in the CTS packet, it should include the transmission duration of the data packet and the ACK packet. Therefore, the new NAV $NAV_{new}$ is obtained as follows.

$$NAV_{new} = 2D_{SIFS} + D_{DATA(R_S)} + D_{ACK} \quad (8)$$

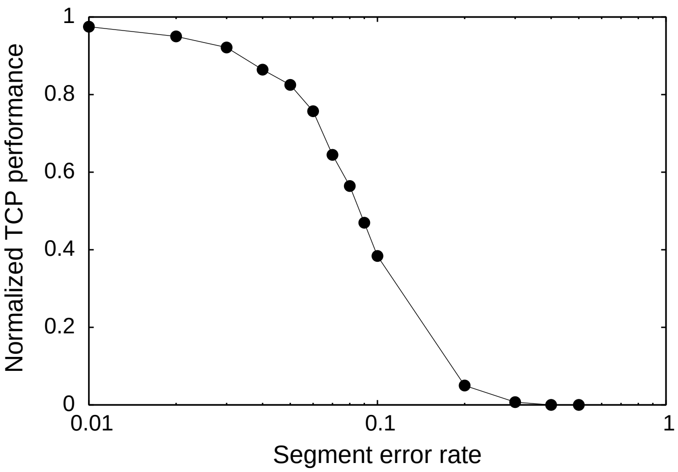


Fig. 1. TCP Performance.

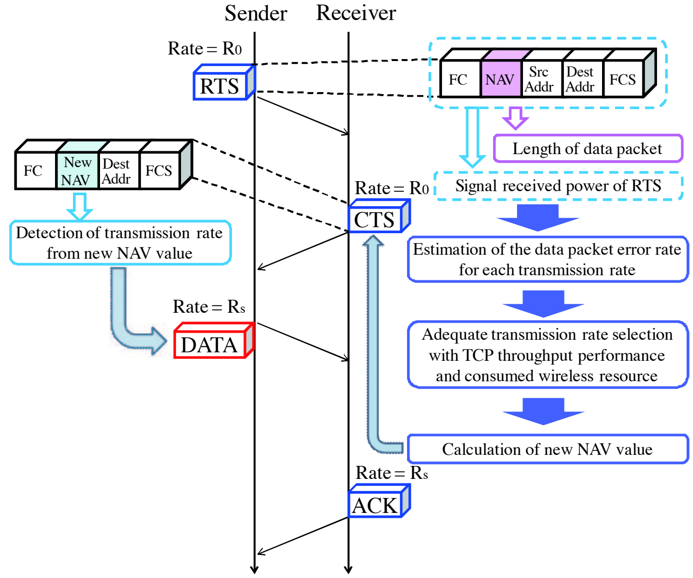


Fig. 2. Operations in the transmission rate selection method.

The receiver sets the new NAV value to a duration field in the CTS packet. After receiving the CTS packet from the receiver, the sender estimates the selected transmission rate from the new NAV. The estimated transmission rate $\hat{R}_s(r)$ is as follows.

$$\hat{R}_s(r) = L/(NAV_{new} - 2D_{SIFS} - D_{ACK}) \qquad (9)$$

As a result, the sender and receiver can select the adequate transmission rate with a conventional RTS and CTS mechanism, and therefore, the proposed method is compatible with the IEEE 802.11 standards.

## III. Operations in the Proposed Method

### A. Transmission rate selection method

Figure 2 shows the operations in the proposed method. In this case, a sender transmits a RTS packet to a receiver with an initial NAV value based on the base transmission rate $R_0$, and the receiver selects the rate $R_s$ as an adequate transmission rate. The procedures of the proposed method are follows.

1) When the sender has the data to be transmitted, it calculates a NAV value with a base transmission rate $R_0$ and the packet length $L$. Then it transmits an RTS packet including the calculated NAV. This process is same as the IEEE 802.11 standards.
2) When the receiver receives the RTS packet from the sender, it calculates the length of the upcoming data packet and estimates the SNR using the received signal power of the RTS packet.
3) The receiver estimates the packet error rate of the incoming data packet by using the estimated packet length and calculated bit error rate. Then, it estimates the TCP throughput for each transmission rate with the calculated TCP throughput in advance.
4) The receiver selects the transmission rate that maximizes the TCP throughput by considering the consumed wireless resource.
5) The receiver recalculates the new NAV using the selected transmission rate and the packet length, and replies with a CTS packet that includes the new NAV.
6) The sender calculates the selected transmission rate using the new NAV in the CTS packet and the packet length, and transmits the data packet with the selected transmission rate.

### B. Example communication.

In the proposed method, the NAV value in a RTS packet may be different from that in a CTS packet, because our method uses the NAV value in order to convey a selected transmission rate to the sender from the receiver. Therefore, neighbor nodes also perform a little different operation from the IEEE 802.11 standards.

Figure 3 shows an example node placement. In this example, each node can communicate with next neighbor nodes. Therefore, the node A can communicate with the sender. The sender can communicate with the node A and the receiver. The receiver can communicate with the sender and a node B. The node B can communicate with the receiver.

Figure 4 shows an example communication in the propose method. In this example, we focus on the neighbor nodes of the sender and the receiver, because the neighbor nodes need to change the waiting time based on the NAV value.

1) The sender transmits a RTS packet with a duration based on base transmission rate $R_0$.
2) The neighbor node A receives the RTS packet, and set the NAV value to a waiting time. In Fig. 4, this NAV value is the upper side duration which includes $D_{CTS}$, $D_{DATA(R_0)}$ and $D_{ACK}$.
3) The receiver receives the RTS packet, and selects an

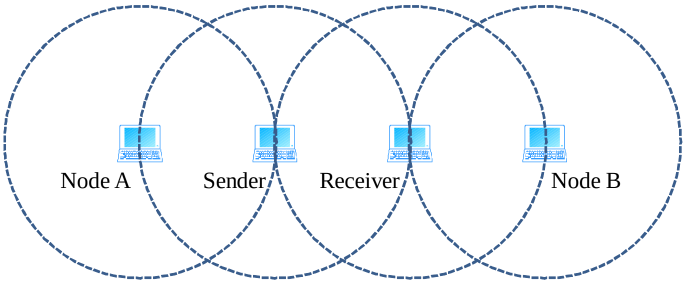


Fig. 3. Example of node placement.

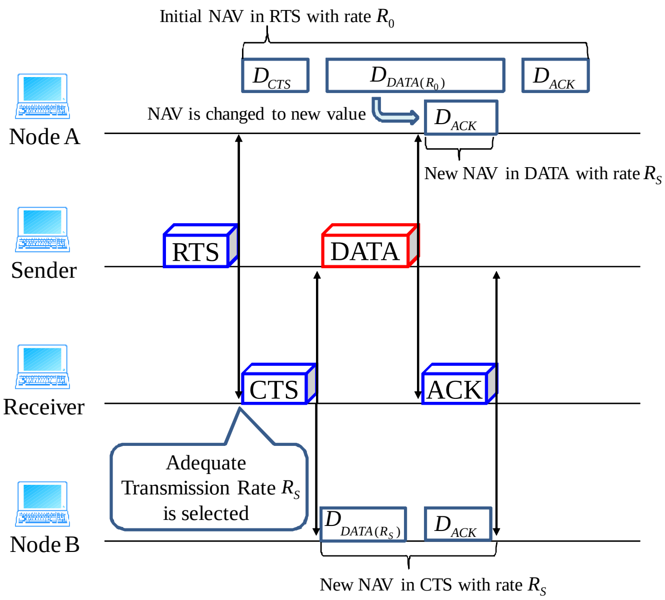


Fig. 4. Example of communication in the proposed method.

adequate transmission rate $R_s$. The new NAV value is calculated based on the selected transmission rate $R_s$. Then the new NAV value is set to a CTS packet. The receiver transmits the CTS packet.

4) The neighbor node B receives the CTS packet, and set new NAV value to a waiting time. In Fig. 4, this new NAV value includes $D_{DATA(R_S)}$ and $D_{ACK}$.
5) The sender receives the CTS packet, estimates the selected transmission rate $R_s$ and transmits a data packet with the transmission rate $R_s$.
6) The neighbor node A receives the data packet, and changes the waiting time from the initial NAV value $D_{DATA(R_0)}$ to the new NAV value $D_{DATA(R_S)}$. In Fig. 4, this NAV value is the lower side duration which includes $D_{ACK}$.

If the neighbor nodes do not support the proposed method, they wait for the initial NAV value $D_{DATA(R_0)}$ based on the base transmission rate $R_0$. Therefore, the performance is deteriorated a little. However, our method can coexistent with the IEEE 802.11 methods.

TABLE I
SIMULATION PARAMETERS

| | |
|---|---|
| Simulator | QualNet |
| Simulation time | 5000 [s] |
| Simulation trial | 100 times |
| Number of nodes | 10 |
| Node placement | 100 [m] apart |
| Relative displacement of nodes | None |
| Nodes velocity | 0.1 - 10 [m/s] |
| Data packet size | 1 [KB] |
| Communication system | IEEE 802.11g |
| Transmission rates | 6, 9, 12, 18, 24, 36, 48 and 54 [Mbps] |
| IP Queue size | 50 [KB] |
| Propagation pathloss model | free space |
| Wireless environment | Rayleigh fading |
| Routing protocol | AODV |
| Application | FTP |

## IV. NUMERICAL RESULTS

In this section, we compare the performance for the proposed method with those for the conventional method with the basic RTS/CTS mechanisms for the fixed transmission rates 6 M, 9 M, 12 M, 18 M, 24 M, 36 M, 48 M, and 54 M [bps] in IEEE 802.11g by using computer simulation. The simulations are performed by the network simulator QualNet[11]. The proposed method can be applied to several routing protocols. In this study, we assume an ad hoc on-demand distance vector (AODV) [8] protocol to be the ad hoc routing protocol. In the simulations, ten wireless terminals are placed 100 [m] apart, and the relative displacement of nodes is not changed. The wireless channels are assumed to be Rayleigh fading channels with a velocity between 0.1 and 10 [m/s]. The application is File Transfer Protocol (FTP) and data packets with the length of 1 [KB] are transferred for 5000 [s]. Detail simulation parameters are shown in Table I.

Figure 5 shows the utilized percentage of transmission rate. From this result, we can find that our proposed method mainly selects three transmission rates 12 M, 18 M and 24 M [bps] in this condition, and an average transmission rate is about 20 M [bps]. Moreover, our method can handle the variety of channel condition when the node velocity is changed.

Figure 6 shows the number of retransmissions due to ACK timeout in the MAC layer. From this result, it is shown that our method can decrease the retransmissions due to the packet losses. This is because our method can select the adequate transmission rate according to the channel condition. Additionally, we can find that the proposed method has better performance than the conventional method of the 18 M [bps], which is a similar transmission rate to the proposed method. This is because our method can transmit the packets with higher transmission rates under the good channel conditions.

Figure 7 shows the TCP throughput performance per 1 [MHz] versus the node velocity. From this result, it is evident that our proposed method can maintain the highest throughput. This is because our method estimates not only the packet error rate of the upcoming data packet according to the RTS packet but also the TCP throughput for each transmission rate. As a result, our method can select an adequate transmission rate for the current channel condition. Moreover, we can find that our method can handle fast channel variations and keep higher throughput when the node velocity is fast. This is because the adequate transmission rate is selected by using the RTS/CTS mechanisms.

On the contrary, the throughputs for the conventional methods with 6 M, 9 M, and 12 M [bps] are fairly good. Because a low transmission rate can help to achieve a low packet error rate, the TCP can maintain a large congestion window. However, the throughputs for 18 M, 24 M, 36 M, 48 M, and 54 M [bps] do not yield good performances. This is because the TCP itself halts the transmission due to many packet errors. Moreover, the adequate transmission rate depends on the wireless channel condition. Therefore, another transmission rate may be adequate when the node placement is different.

Figure 8 shows the number of fast retransmits per segment. From this result, it is found that low transmission rates of 6M and 9M can keep the lowest number of fast retransmits. Because the low transmission rate can achieve a low packet error rate, almost all TCP segments are not lost by transmission errors. On the contrary, high transmission rates of 18M, 24M and 36M suffer from many fast retransmits. Because high transmission rate can not achieve a low packet error rate in this case, many TCP segments are lost due to transmission errors. Moreover, transmission rates of 48M and 54M have curious performance. In this case, few fast retransmits occur for low node velocity and some occur for high velocity. This is because almost segments are lost by transmission errors and thus TCP suspends the transmission of segments for low node velocity, whereas the high transmission rate can transmit a segment successfully in short time even if the channel condition varies fast.

Finally, our method can also keep the low number of fast retransmits even if the higher transmission rates are used. Moreover, we can find that our method is better suited for the moderately changing channel, because our method selects the adequate transmission rate.

Figure 9 shows the number of timeouts per segment. From the result, we can find that low transmission rates of 6M and 9M also keep the lowest number of timeouts. Additionally, our method can keep almost the same performance as these two cases. This is due to the similar reason to fast retransmits. On the contrary, high transmission rates suffer from many timeouts because almost segments are lost by transmission errors with high transmission rates.

Figure 10 shows the number of route constructions per segment. From the result, it is evident that our proposed method can maintain the lowest number of route constructions with the higher transmission rate than 6 M, 9 M and 12 M [bps]. This means that our method can keep the continuous and high rate transmission with few route reconstructions. This is because the sender transmits a packet with a lower transmission rate for the poor channel conditions and higher rate for better conditions.

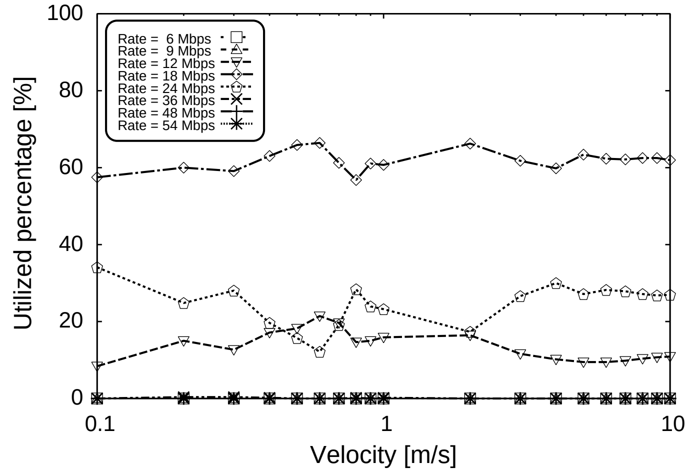


Fig. 5. Utilized percentage of transmission rate.

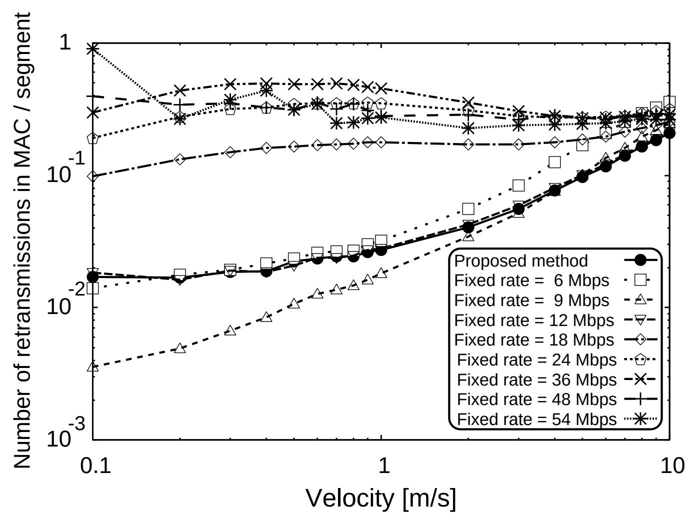


Fig. 6. Retransmissions due to ACK timeout in MAC.

## V. CONCLUSIONS

We have presented a new MAC protocol that improves the TCP performance of ad hoc networks in fading channels. By estimating the packet error rate for the upcoming data packet and the TCP throughput performance, the receiver selects an adequate transmission rate for the data packet. Our proposed method does not modify the packet structures and is compatible with the IEEE 802.11 standards. In addition, our proposed method does not require any additional wireless resource to estimate the channel conditions. Our proposed

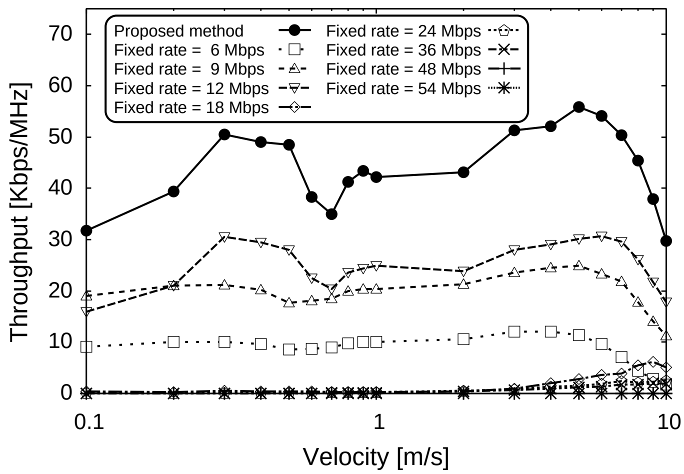


Fig. 7. Throughput performance.

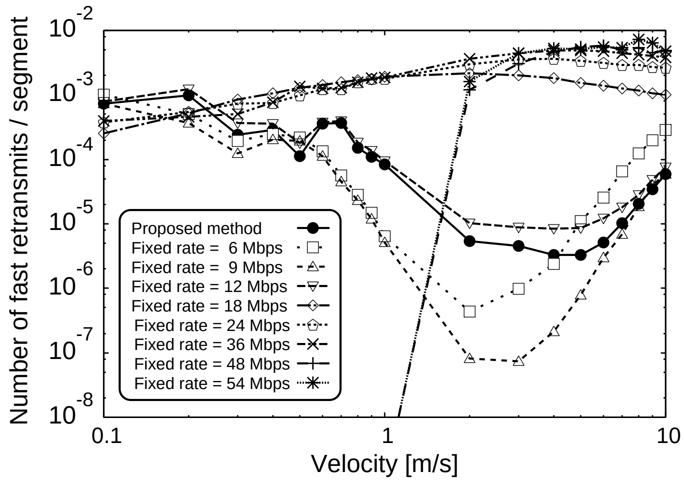


Fig. 8. Fast Retransmit performance.

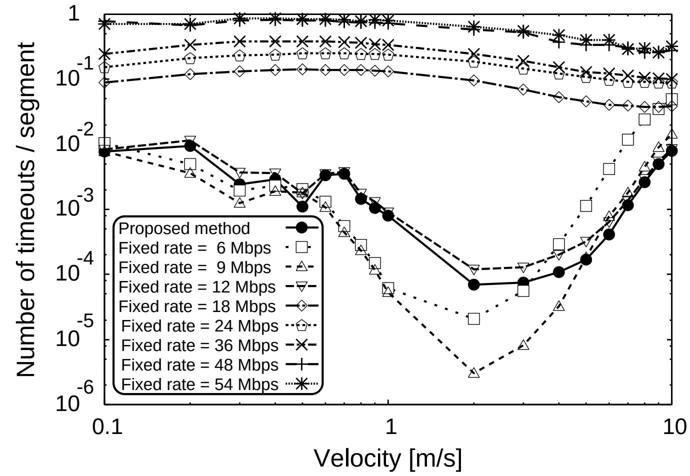


Fig. 9. Timeout performance.

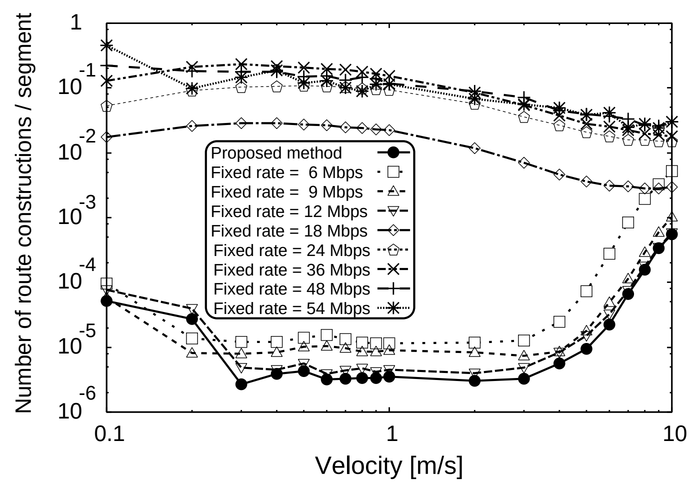


Fig. 10. Route construction.

method can handle the dynamic channel condition and achieve the high throughput performance. Finally, if the neighbor nodes do not support the proposed method, they wait for a little longer than the proposed method. However, the performance is not so deteriorated. Hence, our method can coexistent with the IEEE 802.11 methods.

## ACKNOWLEDGMENT

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