Evaluation of adaptive rate control scheme for ad-hoc networks in fading environments

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***Abstract*— In this research, we investigate the adaptive rate control scheme in the IEEE 802.11 ad-hoc network, and evaluate the performance with ad-hoc on-demand distance vector (AODV) and optimized link state routing (OLSR) in fading environments. The proactive routing protocol such as OLSR constructs all routes between nodes, and changes the route in response to link status. In the fading environment, the link status will change frequently. As a result, poor end-to-end connectivity deteriorates in end-to-end throughput performance. Especially, transmission control protocol (TCP) is a special protocol that is sensitive to segment losses. Therefore, TCP suffers from this poor end-to-end connectivity. The adaptive rate control is one of the methods to improve end-to-end connectivity. However, almost all researchers consider the lower layer performance, not the upper layer performance. Moreover, the criteria for selecting an adequate transmission rate are unclear. In this research, our adaptive rate control scheme intends to improve TCP performance in fading channels over ad-hoc networks. Therefore, our scheme takes TCP performance into account to select the adequate transmission rate for each link. In the numerical results, we evaluate our scheme with AODV and OLSR protocols by using the network simulator QualNet. Finally, we show that our scheme can improve TCP performance greatly.**

I. INTRODUCTION

The wireless channel quality may change due to noise, fading, and the mobility of nodes. Therefore, a fixed modulation scheme is not enough to adapt these fluctuations of wireless channel status. IEEE 802.11 is the well-known wireless system, which is used for wireless LAN, and is one of the candidate devices for ad-hoc networks. IEEE 802.11a, 802.11b, and 802.11g have multi-rate capability. With the multi-rate mechanisms, transmission takes place at various transmission rates according to channel conditions.

Several researches about an adaptive rate control method have been proposed[1]-[3]. Auto rate fallback (ARF) is the well-known adaptive rate control method for IEEE 802.11, and is supported by a lot of devices[4]. It is a sender based media access control (MAC) protocol that controls the transmission rate according to the reception of acknowledgement (ACK) packets. In ARF, if ten consecutive acknowledgment 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 method cannot handle fast channel fluctuation. Receiver based auto rate (RBAR) is receiver based MAC protocol, and is a more effective method that uses multi-rate capability[5]. In RBAR, after receiving a request to send (RTS) packet, the receiver calculates the adequate 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 informed to the sender through a clear to send (CTS) packet. Therefore, RBAR can select the transmission rate packet by packet. However, the RBAR requires modifications of packet structure because the selected transmission rate should be included in the packet. Therefore, it is incompatible with IEEE 802.11 standards. OAR is a similar scheme to RBAR. The main idea of OAR is to utilize high quality channels effectively[6]. Therefore, a node can be allowed to transmit multiple packets under the good channel condition. However, it is also incompatible with IEEE 802.11 standards. Finally, although almost all methods can improve the communication performance, more discussions about a criterion for selection of the transmission rate are required[7]. As a result, it is difficult to set adequate parameters to improve the performance.

In this paper, we evaluate our adaptive rate control mechanisms with ad-hoc on-demand distance vector (AODV)[8] and optimized link state routing (OLSR)[9] in fading environments. Our mechanisms are based on the RTS/CTS mechanisms similar to the RBAR. However, the packet structure of IEEE 802.11 standard can be used. Therefore, our scheme is compatible with IEEE 802.11 standard, and is used with IEEE 802.11 devices at the same time. In order to convey the required information for the selected transmission rate, our scheme makes active use of a Network Allocation Vector (NAV) in RTS and CTS packets. Moreover, we employ the estimated congestion window size of transmission control protocol (TCP) as criteria for selection of the transmission rate, because it is known that the TCP suffers from the fluctuation of wireless channel quality[10], [11]. From simulation results, our mechanisms can achieve the high throughput performance and improve TCP performance if OLSR or AODV is employed in ad-hoc networks.

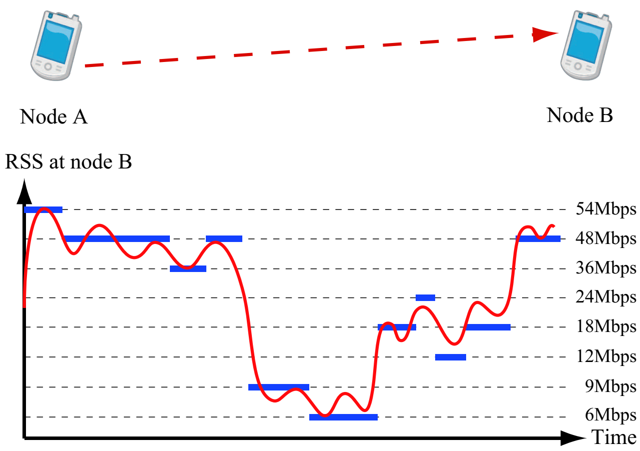

Fig. 1. Adaptive rate control according to RSS.

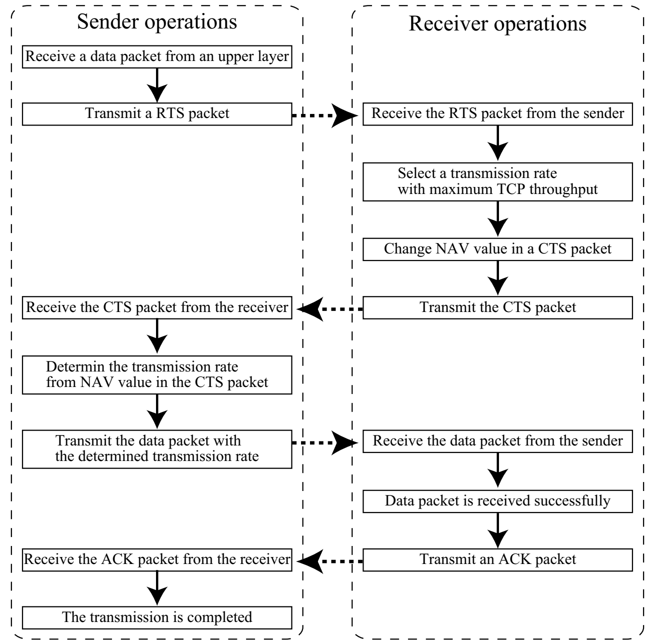

Fig. 2. Flow charts of the adaptive rate scheme.

II. Adaptive Rate Control Scheme

IEEE 802.11 supports multi-rates transmission mechanisms. Figure 1 is example communication between a node A and a node B with the adaptive rate control. In this figure, the node A selects the transmission rate from 6M [bps] to 54M [bps] according to the received signal strength (RSS).

Figure 2 shows flow charts of our adaptive rate control scheme. Our scheme is modified based on the RTS/CTS mechanisms in IEEE 802.11. A first key idea is utilizing a Network Allocation Vector (NAV) in RTS/CTS packets to convey an adequate transmission rate for channel status. The NAV is originally used to suppress transmission of neighbor nodes. A sender node can determine a selected transmission rate from NAV in the CTS packet because it knows the packet length and obtains the transmission period from the NAV in the CTS packet. Second key idea is employing TCP performance to select the adequate transmission rate. Therefore, each node has a special table which includes a relation between a packet error ratio and a congestion window size of TCP. In this research, we assume that this table should be calculated by simulations or analytical models[12], [13] beforehand.

A. Estimation of packet error rate

A packet error rate is effected by a bit error rate (BER) and a packet length. The BER is effected by a signal to noise ratio (SNR). Therefore, a node estimates the SNR of a RTS packet when it receives this RTS packet. Then, it calculates the BER $P_b(R)$ with transmission rate R by the equations derived from an analytical model of modulation schemes.

Generally, the RTS packet does not include information about the packet length of an upcoming data packet. In the proposed scheme, we employ a network allocation vector (NAV), which indicates the transmission period, to estimate the packet length of the upcoming data packet. Since the initial NAV value NAV_i includes the transmission period of a CTS packet, the data packet and an ACK packet, the initial NAV value is obtained as follows.

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

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

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

B. Selection of transmission rate

TCP is the most famous protocol to achieve reliable communications in Internet. However, it is known that the TCP performance deteriorates sharply over the wireless channel. This is because, TCP is designed on the transmission characteristics of wired networks.

In this paper, we employ the estimated TCP performance as the criteria for the selection of the transmission rate. The TCP performance depends on the segment error rate. Therefore, we prepare the relation table between the congestion window size of TCP and the segment error rate by using an analytical model or simulation results in Fig. 3. Then, the estimated congestion window size can be expressed as follows.

$$CW_R = Func(P_f(R)) \quad (3)$$

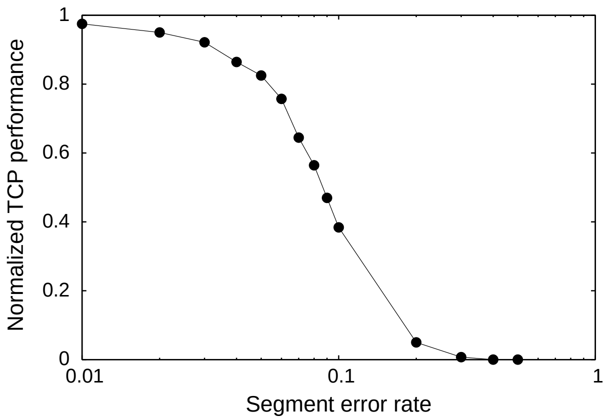

Fig. 3. TCP Performance versus segment error rate.

TABLE I
SIMULATION PARAMETERS

Simulator	QualNet
Communication period	600 [s]
Simulation trials	100 times
Simulation area	1000 x 1000 [m]
Number of nodes	50
Node placement	Random
Node mobility	None
Communication system	IEEE 802.11g
Transmission rates	6, 9, 12, 18, 24, 36, 48 and 54 [Mbps]
Adaptive rate control	ARFB, Proposed
Propagation pathloss model	Free space
Wireless environment	Rayleigh fading
Doppler frequency	0.1 [Hz]
Routing protocol	AODV, OLSR
Application	FTP
Data packet size	1 [KB]
Number of connections	1 - 5

The transmission period depends on the transmission rate. Therefore, the estimated congestion window size considering the transmission period is

$$\hat{CW}_R = CW_R \frac{R}{R_{MAX}} \tag{4}$$

where R_{MAX} is the maximum transmission rate. Consequently, we can select the transmission rate R_s according to the congestion window size of TCP and the transmission period.

$$R_s = \max_R(\hat{CW}_R) \tag{5}$$

C. Estimation of selected transmission rate

The proposed scheme is compatible with IEEE 802.11 standard. However, a standard packet format of IEEE 802.11 does not have a special field to convey a transmission rate. In this paper, we employ a NAV field to convey the selected transmission rate from a receiver to a sender. If the transmission rate is selected as R_s, the transmission period of the data packet will be shown as follows.

$$D_{DATA(R_S)} = L/R_s \tag{6}$$

This data transmission period is included into a new NAV value in a CTS packet. Then, the new NAV value NAV_{new} is obtained as follows.

$$NAV_{new} = 2D_{SIFS} + D_{DATA(R_S)} + D_{ACK} \tag{7}$$

The sender estimates the selected transmission rate with the new NAV value from the receiver. The estimated transmission rate $\hat{R}_s$ is as follows.

$$\hat{R}_s = L/(NAV_{new} - 2D_{SIFS} - D_{ACK}) \tag{8}$$

III. NUMERICAL RESULTS

We evaluate the proposed scheme with AODV[8] and OLSR[9], and compare the basic RTS/CTS mechanisms for the fixed transmission rates 6, 9, 12, 18, 24, 36, 48, and 54 [Mbps] and ARF method which supports the adaptive rate mechanism. The simulations are performed by the network simulator QualNet[14]. In the simulations, 50 nodes are placed randomly in the 1000 [m] square area, node mobility is not considered. The wireless channels are assumed to be Rayleigh fading channel with a Doppler frequency equals to 0.1 [Hz]. The application is considered as file transfer protocol (FTP) and data packets with the length of 1 [KB] are transfered for 600 [s]. Source and destination nodes are selected randomly. Figures show the average of 100 simulation results. Detail simulation parameters are shown in Table I.

Figure 4 shows the TCP throughput performance versus the number of TCP connections with AODV. From results, we can find that the proposed protocol can achieve the highest throughput. Moreover, the throughput of the proposed protocol is higher than that of ARFB, which is the famous adaptive rate mechanism. This is because our scheme employs the TCP performance as criteria for selection of the transmission rate. Additionally, our protocol can select the transmission rate packet by packet. On the contrary, ARFB cannot handle the fast variation of the channel. Therefore, the adaptive rate control mechanism of ARFB is not enough to adapt the change of channel even if the Doppler frequency is low. Finally, the fixed rate with 18 [Mbps] has better throughput than those with another fixed rates. However, it is difficult to select the adequate transmission rate for different network situations.

Figures 5 and 6 represent the number of fast retransmit occurrence and the number of timeout occurrence with AODV. If the fast retransmit occurs, the sender retransmits a data packet and decrease the transmission rate by reducing the congestion window size. If the timeout occurs, the sender

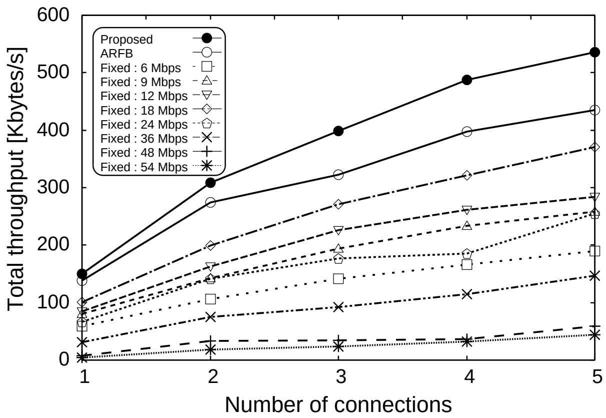

Fig. 4. Total throughput (AODV).

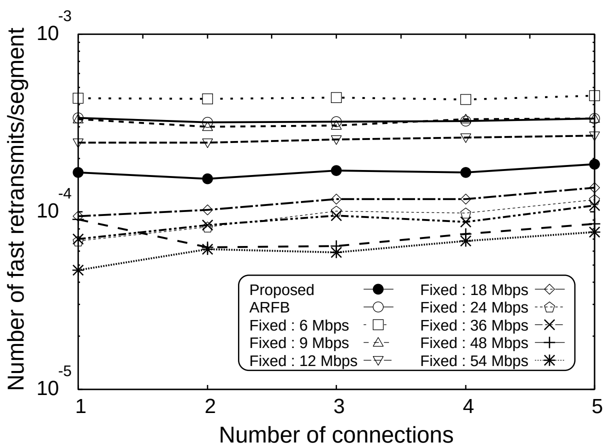

Fig. 5. Number of fast retransmit occurrences per segment (AODV).

retransmits a data packet and decreases the transmission rate to the minimum rate. Therefore, the occurrence of timeout affects the communication performance larger than the occurrence of fast retransmit. Results show that the adaptive rate control mechanisms reduce the number of timeout occurrence. Especially, our proposed protocol can reduce it greatly. This is because, our protocol can select a lower transmission rate if the channel condition is not good for TCP communication. Therefore, almost all data packets can be transfered successfully even if the channel condition becomes bad.

Figure 7 shows the number of link losses per segment with AODV. A source node will perform a route reconstruction in AODV if the link losses occur. Therefore, the number of link losses is one of the indicators for route stability. From results, we can find that the proposed protocol can keep the lowest value even if the number of connections is increased. This is because, the link loss is detected in IEEE 802.11, when the node failed to retransmit the packet. Hence, the selection of the adequate transmission rate that satisfies the link quality is especially important. In the proposed protocol, the sender can select the transmission rate adaptively according to the

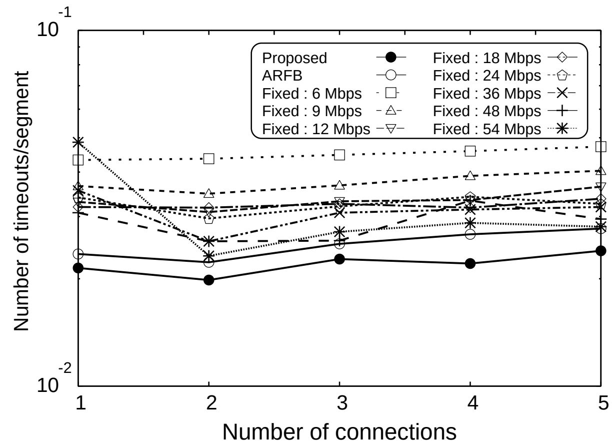

Fig. 6. Number of timeout occurrences per segment (AODV).

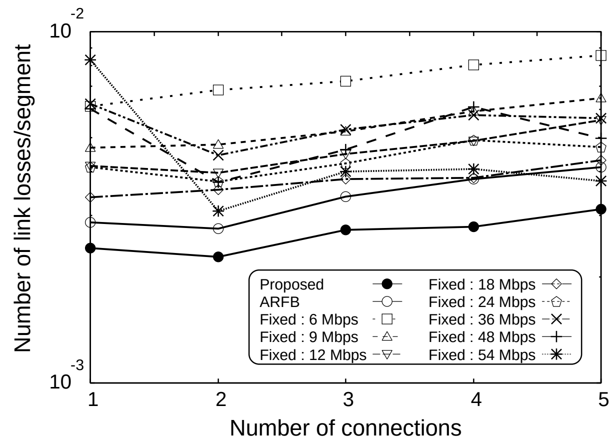

Fig. 7. Number of link losses per segment (AODV).

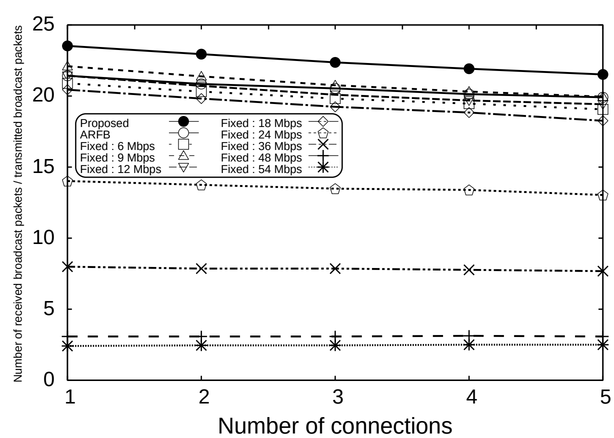

Fig. 8. Number of received broadcast packets per transmitted broadcast packet (AODV).

channel condition. So the effect from the link detection is also improved.

Figure 8 shows the number of received broadcast packets per transmitted broadcast packets with AODV. This value

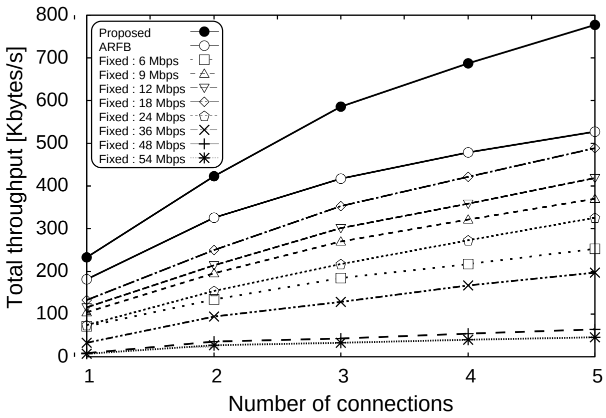

Fig. 9. Total throughput (OLSR).

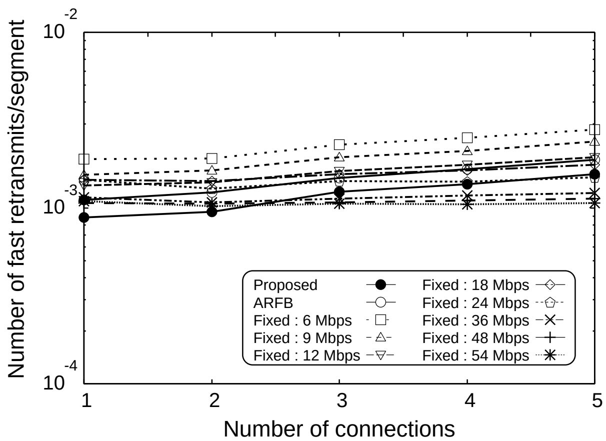

Fig. 10. Number of fast retransmit occurrences per segment (OLSR).

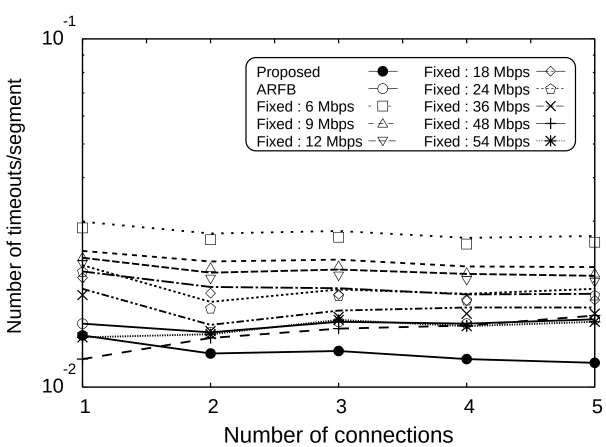

Fig. 11. Number of timeout occurrences per segment (OLSR).

means that how many nodes can receives the same broadcast packet from one node. Results show that our proposed protocol can keep the highest value. Because, employing the adaptive transmission rate is effective to improve the wireless resource utilization. Then, more faraway nodes can receive the packet due to the decreasing of interference.

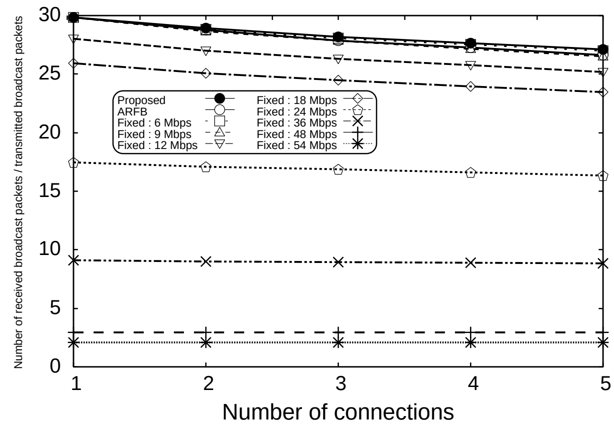

Fig. 12. Number of received broadcast packets per transmitted broadcast packet (OLSR).

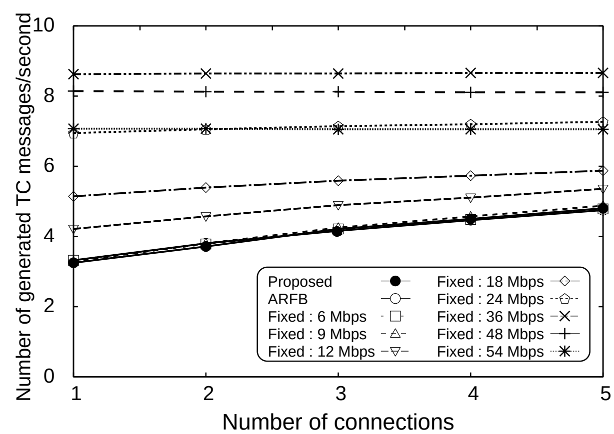

Fig. 13. Number of generated TC messages per second (OLSR).

Figure 9 shows the TCP throughput performance versus the number of TCP connections with OLSR. From results, it is evident that our scheme can maintain the highest throughput. Moreover, the throughput difference between the proposed protocol and ARFB increases according to the increasing of connections. The MPR node has to perform the forwarding of control messages and data packets. Therefore, it has to select the adequate transmission rate to the neighbor nodes. However, the transmission rate is controlled for all neighbor nodes in ARFB. Therefore, the optimum transmission rate is not selected when the node communicates with some neighbor nodes. In the proposed protocol, the transmission rate is controlled for each neighbor node. Hence, the optimum transmission rate can be selected even if some neighbor nodes exist and communicate with the node simultaneously. Moreover, the TCP throughput performance of OLSR is larger than that of AODV. This is because, OLSR can construct the route beforehand and the route is always maintained. Therefore, TCP can keep a stable communication.

Figures 10 and 11 show the the number of fast retransmit occurrence and the number of timeout occurrence with OLSR.

From results, we can find that the proposed protocol can reduce the number of fast retransmit occurrences. However, the number of fast retransmit with OLSR is larger than that with AODV. This means that small number of packet losses occurred more frequently. This is because, the number of transfered data packets also increases with OLSR. Then, the interference level also increases. As a result, some packets are corrupted due to the interference from neighbor communications. On the contrary, the results show that the proposed protocol can reduce the number of timeout occurrences. Moreover, the number of timeout with OLSR is smaller than that with AODV. This means that interruption of communication can be reduced. In AODV, the source node starts a route reconstruction according to the link losses, which are caused by the packet transmission failure. Therefore, some packets transmission failures cause the interruption of communication. As a result, timeout will be occurred due to this interruption. On the contrary, the route can be maintained in OLSR even if some packets transmissions are failed. Consequently, OLSR can maintain a stable route, and the proposed protocol can improve the communication performance effectively.

Figure 12 shows the number of received broadcast packets per transmitted broadcast packets with OLSR. Results show that OLSR can keep the higher value than that with AODV. This is because, almost all broadcast packets in OLSR are transmitted independently from each node. Moreover, the transmission rate for broadcast packets is also changed according to the transmission rate selection for the neighbor nodes. Hence, our proposed protocol can achieve the high reception ratio and high wireless resource utilization ratio concurrently. On the contrary, almost all broadcast packets in AODV are transmitted simultaneously when the route discovery is performed. Because the route request packet is delivered to the whole network by flooding. As a result, some packet collisions tend to occur in AODV. Finally, the adaptive rate mechanism is effective to improve the reception rate of broadcast packets. However, the fast handling performance to the channel condition is not so important. Because, the performance of the proposed protocol is similar to that of ARFB.

Figure 13 shows the number of topology control (TC) messages per seconds versus the number of connections. From this results, we can find that our scheme suppresses the generation of TC messages. This means that our scheme can maintain a stable route for a long time. This is because, our scheme can improve the utilization performance of wireless channel by selecting the adequate transmission rate. Therefore, the reception probability of broadcast packets also improved by reducing the probability of collisions. As a result, our scheme also improves the transmission performance of control messages for OLSR protocol. Therefore, redundant control messages cannot be transmitted a lot.

IV. Conclusion

We have evaluated our MAC protocol that improves the TCP performance with AODV and OLSR protocols in the fading environments. By estimating the packet error ratio for the upcoming data packet and the TCP throughput performance, the receiver can select an adequate transmission rate. Our scheme is compatible with the IEEE 802.11 standards since packet structure is kept as standards. Finally, if an IEEE 802.11 standard node exists, it wait for a little longer than the proposed scheme. However, the performance is not so deteriorated. Therefore, we consider that our scheme can coexistent with the IEEE 802.11 standard methods.

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