A Simulation Study on Multi-Rate Mobile Ad hoc Networks[1]

Gang Ding
Purdue University
250 N. University Street
W. Lafayette, IN 47907, USA
+1 765-494-6702
dingg@purdue.edu

Xiaoxin Wu
Purdue University
250 N. University Street
W. Lafayette, IN 47907, USA
+1 765-496-2718
wu@cs.purdue.edu

Bharat Bhargava
Purdue University
250 N. University Street
W. Lafayette, IN 47907, USA
+1 765-494-6013
bb@cs.purdue.edu

ABSTRACT
This paper studies the performance of a multi-rate mobile ad hoc network (MANET) using an extended ns-2 simulator. A link adaptation algorithm is developed and tested. The multi-rate control algorithm is based on the channel access mechanism for IEEE 802.11 with modifications. Some realistic models for radio propagation, such as lognormal fading and Walfisch/Ikagami propagation model, are used. At transport and application layer, different kinds of data traffic, including constant bit rate, TCP, voice over IP, and video are tested. The effects due to position error and mobility are also examined. The simulation results show that link layer data rate control can greatly improve network performance. Components at different layers all contribute to the system performance of a MANET. It is also shown that multimedia data transmission over MANETs deserves future study.

Categories and Subject Descriptors
C.4 **[Performance of Systems]**: *performance attributes, design studies*; C.2.2 **[Computer-Communication Networks]**: Network Protocols – *protocol verification*; C.2.1 **[Computer-Communication Networks]**: Network Architecture and Design – *wireless communication, distributed networks*.

General Terms
Performance, Experimentation, Measurement.

Keywords
Mobile ad hoc network, Link adaptation, IEEE 802.11.

1. INTRODUCTION
Mobile ad hoc networks (MANETs) have been widely studied in the literature [10]. Due to the self-organized nature, the dynamic topology caused by node mobility, and the multi-hop connections in MANETS, it is difficult to build an analytical model for the network performance study. The real test bed is expensive. Therefore, simulation studies become very important. Ns-2 [1] has been the major tool for network simulation study. It is also the mostly used tool for simulation study on MANETs since it has the following features:

- At MAC layer, the IEEE 802.11 distributed coordination function (DCF) [5] is implemented, which uses Request to Send (RTS) / Clear to Send (CTS) / DATA / ACK.
- Radio propagation models are provided. The Friss-space model and the two-ray-ground model are used.
- CBR and TCP traffic generator and mobility generator using random waypoint model are provided.
- Some major ad hoc routing protocols, such as Distance Sequence Distance Vector (DSDV) [8], Ad hoc On-demand Distance Vector (AODV) [9], and Dynamic Source Routing (DSR) [7], are implemented.

In this study, we investigate the system performance of a multi-rate MANET in the urban areas. Specifically, we explore the data delivery ratio, throughput and transmission delay in a MANET. A more realistic urban area radio propagation model, along with lognormal fading, is implemented. To efficiently utilize the network bandwidth, link adaptation based on the signal-noise-ratio (SNR) is applied. Other than Constant Bit Rate (CBR) and TCP traffic, Voice over IP (VoIP) and video will be used at the application layer. The experiments are conducted in an enhanced ns-2 simulator, including following functions:

- Urban radio propagation models.
- Implementation of link adaptation algorithm.
- Development of a position aware ad hoc routing protocol.
- VoIP and Video traffic generator.

This paper is organized as follows. Section 2 introduces the details for the extended functions in ns-2 and the simulation scenario. Section 3 presents the implementation details and simulation results for the link adaptation in MANETs. Performance for the networks involving different traffic patterns is studied in section 4. The impact of position error and moving speed is studied in section 5. Section 6 concludes the paper.

2. SIMULATION SET UP

2.1 Radio Propagation Model
Walfisch/Ikagami model [2] is a radio propagation model that fits to the real measurement in urban area. In contrast to only consider

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a flat area in the traditional two-ray-ground model, the Walfisch/Ikegami model considers the effect due to dense streets and buildings. We implemented the Walfisch/Ikegami model in the physical layer in ns-2. Some parameters are given in Table 1.

Table 1 Parameters for Walfisch/Ikagami model

Parameter	Value
Street width	25 m
Building height	30 m
Building distance	150 m
Propagation angle	55 degrees
Frequency	2.4 GHz
Channel width	22 MHz

The current implementation of ns-2 does not consider the path loss and fading during radio propagation. We added a path loss model including the lognormal fading. The lognormal fading has a standard deviation of 6 dB and 10 dB corresponding to suburban and urban environments, respectively, and a correlation coefficient of 0.5.

2.2 Link Adaptation

Since Carrier Sense Multiple Access with Collision Avoidance (CSMA-CA) and RTS/CTS are used as the random channel access mechanism, the receiver-based link adaptation [4] is used to decide the link data rate. Upon receiving the RTS, the receiver makes the estimation on SNR and selects the data rate based on Table 2. We adopt IEEE 802.11g standard [6] which provides different data rates by various Extended Rate PHY (ERP) modulation modes at physical layer. The selected rate information is inserted in CTS and sent back to the sender. The sender then uses this rate for data transmission. RTS and CTS packets themselves are transmitted at the basic data rate of 1Mb/s.

Table 2. Data rate selection

SNR (dB)	Data rate (Mb/s)	Modulation mode
> 30	48	ERP-OFDM
26 – 30	36	
21 – 26	24	
18 – 21	11	ERP-CCK
16 – 18	5.5	
14 – 16	2	ERP-DSSS
11 – 14	1	
<=11	Frame loss	

One problem on such a rate selection scheme is that it is impossible for the sender to include the precise transmission duration time in RTS, as required by the IEEE 802.11 standard [5], because when sending RTS, the sender does not know exactly what data rate will be used. Yet the duration time determines Network Allocation Vector (NAV), which plays an important role for better CSMA-CA performance. We propose a few ways to determine NAV, which will be explained in the next section.

2.3 Position-Aware Routing Protocol

The routing protocols have been the major topic of previous study of MANETs, but in this paper we are more interested in the effects due to other system components. To reduce the effect due to a particular routing protocol, we implemented a position-aware routing protocol [11] which assumes the position information of each mobile node is available so that the routing decision is based on the positions of source, destination and neighbor nodes.

2.4 VoIP and Video Traffic Generator

For VoIP, the constant rate data flow with an interval of 20ms between any two consecutive data packets is used. The packet size is 20 bytes. For video traffic, MPEG-4 video trace data is obtained from a 60-minute movie "the Jurassic Park", which is publicly available for the test of video transmission performance [12], especially for wireless networks [3]. A video traffic generator is developed so that the video trace data are fragmented in a given packet unit size and transmitted at a given speed such as 25 frames/second. Some important parameters for the video traffic are given in Table 3.

Table 3. Video parameters

Parameter	Value
Resolution	QCIF 176*144
Frame rate	25 frames/sec
Frame sequence	IBBPBBPBBPBB
Compression ratio	YUV: 49.96
Video run time	3.6e+6 msec
Min frame size	26 bytes
Max frame size	8154 bytes

2.5 Network Scenario

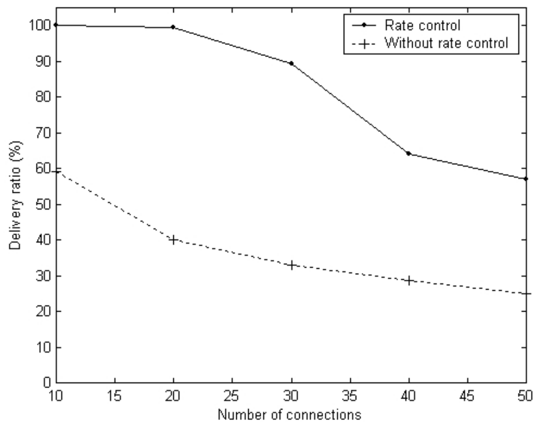
Without specifications, the simulated ad hoc network has 100 nodes uniformly distributed in an area of 1000m x 1000m. Each node moves within the area, with a random direction and a random velocity uniformly distributed between 0 and a maximum value of 10m/s. When data is transmitted at 1Mb/s, i.e., the basic data rate, a receiver can receive the data correctly when it is as far as 250m away from the sender. For a connection of a source and its destination, a constant bit rate (CBR) of 4 packets per second is used. Without any specification, the packet size is 512 bytes.

3. SIMULATION ON LINK ADAPTATION

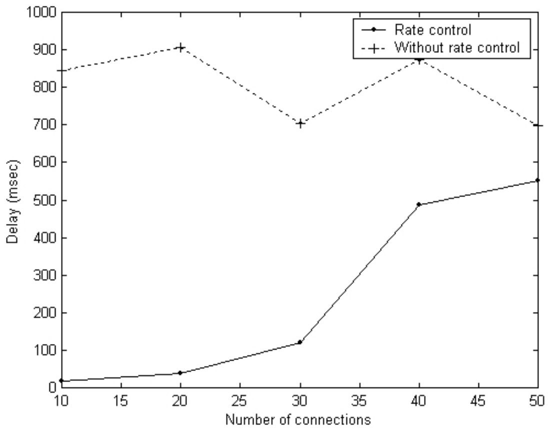
In this section, we first show the performance improvement when data rate control algorithm is applied in the link layer. Then two implementation issues, NAV and carrier sense threshold, are addressed.

3.1 Performance Improvement

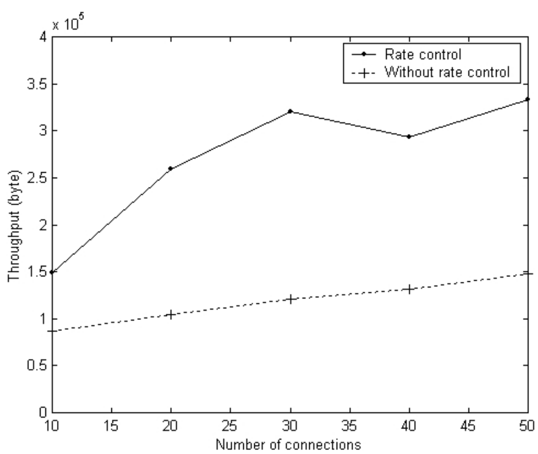
Figure 1 compares the packet delivery ratio, transmission delay, and throughput before and after the rate control algorithm is applied. The load of network traffic is changing from 10 to 50 connections. It is shown that, after the rate control algorithm is applied, the delivery ratio is improved from 69% up to 171%, the transmission delay is reduced from 21% to 96%, and the throughput is improved from 71% to 165%.



a. Delivery ratio



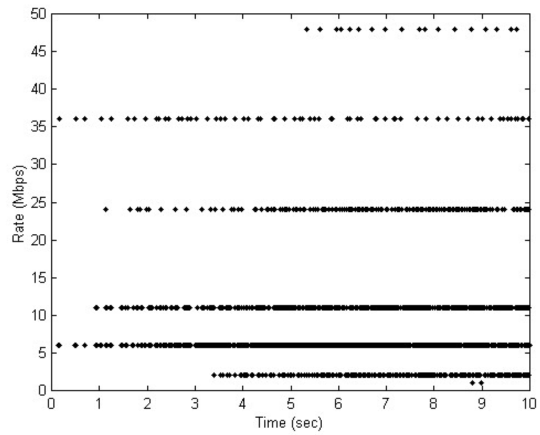
b. Transmission delay



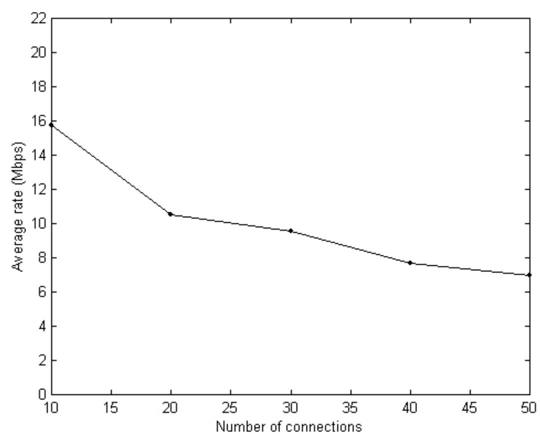
c. Throughput

Figure 1. Results of rate control

Figure 2.a shows the data rate dynamically selected by each sent packet during the first 10 seconds of simulation. The density of points for each rate represents how often this rate is selected. The average selected data rate is shown in Figure 2.b. Which rate is selected largely depends on the network traffic and topology. It is clearly shown in Figure 2.b that when more connections, that is more network traffic, are involved, a smaller data rate tends to be selected because the interference from mobile nodes in the neighborhood significantly affects the quality of received data. This also explains why the transmission delay increases when the number of connections increases because, in this case, smaller data rates imply longer transmission time.



a. Data rate histogram



b. Average data rate

Figure 2. Statistics of selected data rate

3.2 NAV estimation and update

At the IEEE 802.11 MAC layer, the NAV used in RTS and CTS packets indicates the time period during which other mobiles nodes can not take the shared wireless channel. NAV calculated in RTS and CTS is shown Figure 3, where SIFS means the Short InterFrame Space that a packet must wait before transmission. Since RTS, CTS and ACK are all transmitted at the basic rate, NAV can be easily calculated if the transmission rate of the DATA packet is fixed and known before RTS and CTS are sent. But when the above adaptive rate control algorithm is applied, the actual data rate is selected by the receiver upon receiving the RTS.

This implies that the NAV can only be estimated in RTS by the sender. It can be updated later in CTS by the receiver, based on the actual selected data rate. But in the real system, a neighbor who receives an RTS and sets an NAV accordingly may not be able to receive the CTS with an updated NAV. So it may not always benefit to take extra effort to update the NAV in CTS.

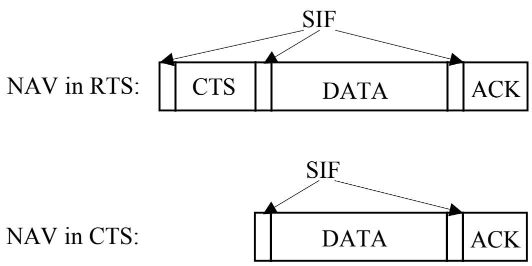
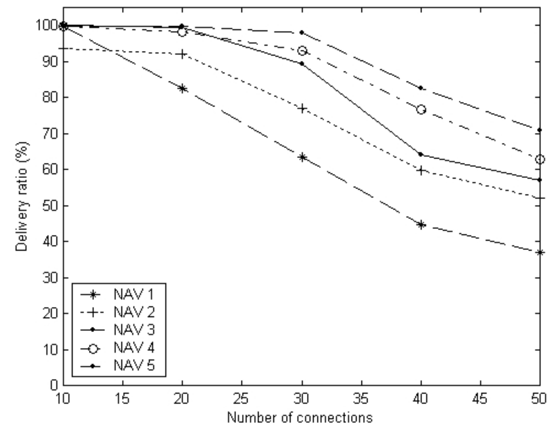


Figure 3. NAV in RTS and CTS

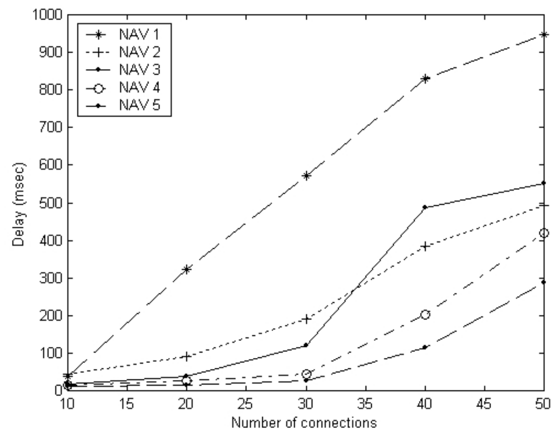
There are many ways to estimate NAV in RTS and update it in CTS. We here compare five possible schemes.

- NAV 1: Uses the maximal transmission rate, 54 Mb/s, to estimate NAV in RTS, but does not update NAV in CTS.
- NAV 2: Uses the basic transmission rate, 1 Mb/s, to estimate NAV in RTS, but does not update NAV in CTS.
- NAV 3: Uses 54 Mb/s to estimate NAV in RTS and update NAV in CTS based on the selected new data rate.
- NAV 4: Uses the last data sending rate to estimate NAV in RTS, but does not update NAV in CTS.
- NAV 5: Assuming we know the actual data rate before sending the RTS, the appropriate NAV can be obtained in RTS so that it is not necessary to update it in CTS. This is the best but not realistic case because the actual data rate selected by the receiver can not be available to the sender when it is sending the RTS. We only use this scheme for the comparison with other schemes.

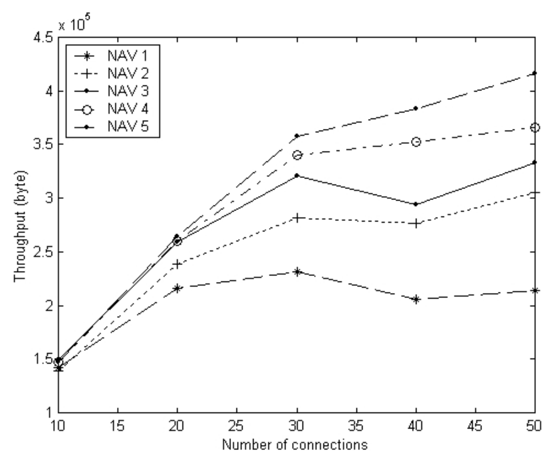
For all the above schemes, Figure 4 shows the delivery ratio, transmission delay, and throughput. It is evident that, for most traffic loads of connections from 10 to 50, the performance in the increasing order is NAV 1 < NAV 2 < NAV 3 < NAV 4 < NAV 5. The only exceptions are that the delivery ratio of NAV 3 and NAV 1 outperform NAV 4 and NAV 2, respectively, at low traffic loads; and the transmission delay of NAV 3 is a little larger than NAV 2 for high traffic loads. According to these simulation results, we observe that small transmission rate is preferred for NAV estimation in RTS because this will give longer NAV so that less collision and interference will happen. We also observe that, when NAV in CTS is updated (as in NAV 3), it works better when the traffic load is low. Since NAV 3 and NAV 4 perform better than NAV 1 and NAV 2 in most cases, we suggest using NAV 3 and NAV 4. NAV 4 performs better for high traffic loads and is simpler than NAV 3 because it does not need NAV update in CTS. But NAV 3 performs better for low network loads. We will use NAV 3 for the following simulations.



a. Delivery ratio



b. Transmission delay

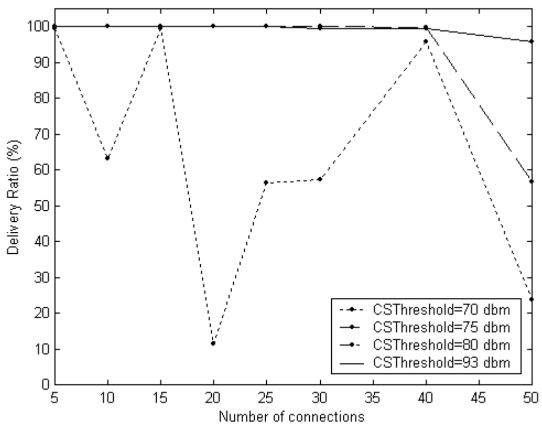


c. Throughput

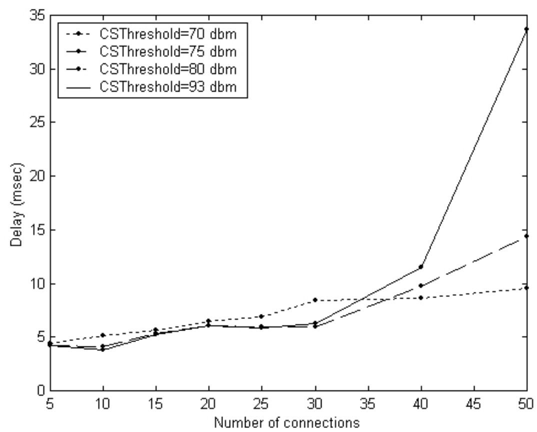
Figure 4. Results of NAV estimation and update

3.3 Carrier Sense Threshold

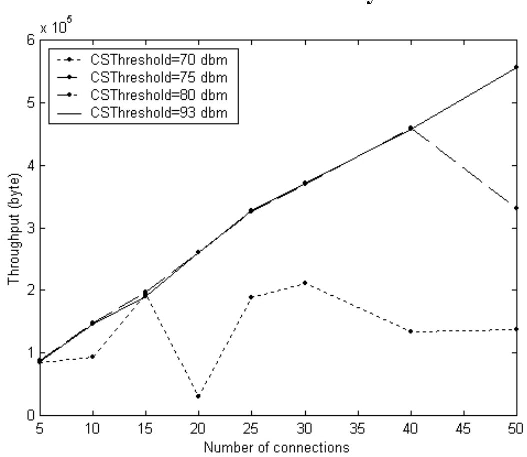
Carrier Sense Threshold (CSThresh) decides if an arriving packet has enough power to be detected. It is an important parameter at physical layer that affects the upper layer performance. According to Figure 5, it is obvious that higher CSThresh maintains higher delivery ratio and throughput. But if it is as low as 70 dBm, a large percent of packets can be lost.



a. Delivery ratio



b. Transmission delay



c. Throughput

Figure 5. Results of carrier sense threshold

For low traffic loads, the transmission delays are very close for different CSThreshs. But when the number of connections increases, higher CSThresh shows higher transmission delay. This is due to the fact that only the received data packets are employed to collect the average transmission delay. Since the delivery ratio for lower CSThresh is low at high network loads, there is actually smaller number of packets being used for calculation of transmission delay. So we could get a smaller average delay for lower CSThresh. If we set the transmission delay of all the lost packets to be a very large value, then the corresponding average delay will be actually very high.

4. SIMULATION ON DATA TRAFFIC

Last section uses CBR traffic over MAC and physical layers, the impact of three other types of data traffic at upper layers will be tested in this section.

4.1 TCP Traffic

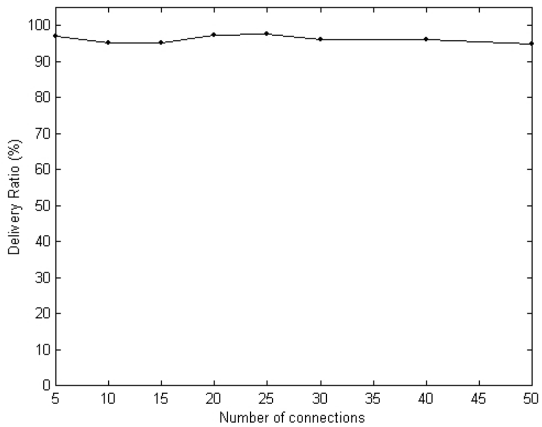
We first test the impact of TCP data traffic. TCP traffic sends continuous data packets controlled by the flow and congestion control algorithms provided by TCP transport layer. The number of connections is 5, 10, 15, 20, 25, 30, 40 and 50. The simulation results are given in Figure 6. TCP traffic can maintain a very good data delivery ratio: at least 95% of sent packets can be successfully received. The delivery ratio only drops a little bit for higher number of connections. But the transmission delay of TCP traffic is quite long. This is due to the flow and congestion control in TCP that scarifies the transmission time to the reliable transport. It is a little bit surprised that the transmission delay for TCP traffic is long even when the traffic load is low. This can be explained as that when high data rate is chosen at low loads, it may introduce congestion in the intermediate mobile nodes so that the congestion control mechanism of TCP is triggered, which will delay the data transmission.

The high throughput for connections more than 15 is due to the fact that more data packets are continuously being sent by the TCP traffic generator and most of the send packets are successfully received because of the high delivery ratio.

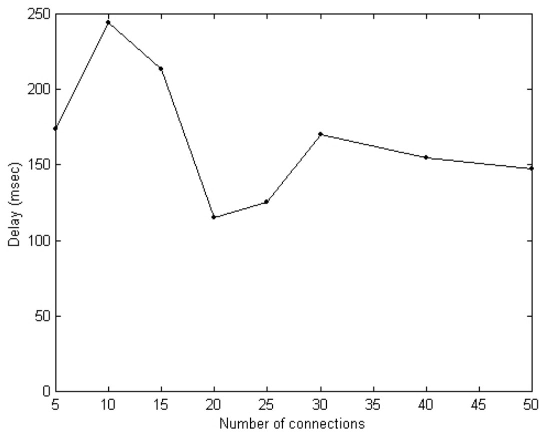
4.2 Real-Time Data Traffic

The real-time data traffic, such as voice and video, is increasing dramatically in internet and mobile wireless networks. It is characterized as transmitting a large volume of data in a time sensitive fashion. Due to the limited bandwidth and the error-prone link in wireless networks, it is very important to test if a MANET can support such kind of real-time traffic.

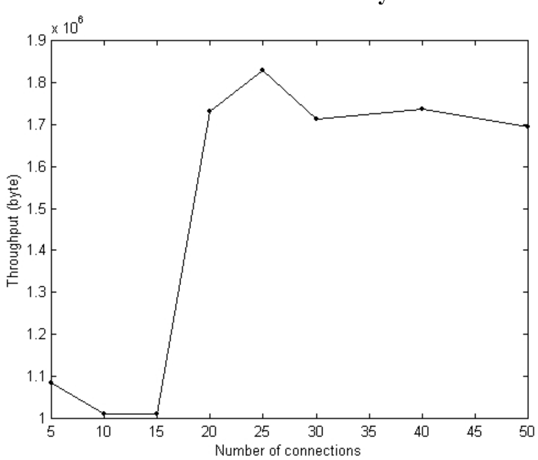
We have applied the VoIP and video traffic to various simulation scenarios. In the same scenario as that for CBR and TCP traffic, it has been observed that video traffic performs very poor. Acceptable performance can only be obtained when the number of connections is less than 10. Figure 7 shows the simulation results of VoIP and Video, when the number of connections is changing from 1 to 8. This is mainly due to the bandwidth limit, high application data rate, and the mobility in MANETs. It is shown that almost all VoIP packets can be successfully received in a timely way. The delivery ratio only drops a little bit for higher number of connections. But for video traffic, when the number of connections increases from 3 to 8, the delivery ratio significantly drops from 99% to 50%, and the transmission delay increases from 45 ms to 760 ms.



a. Delivery ratio



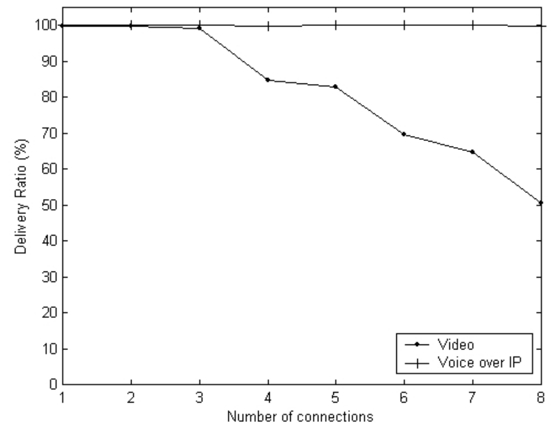
b. Transmission delay



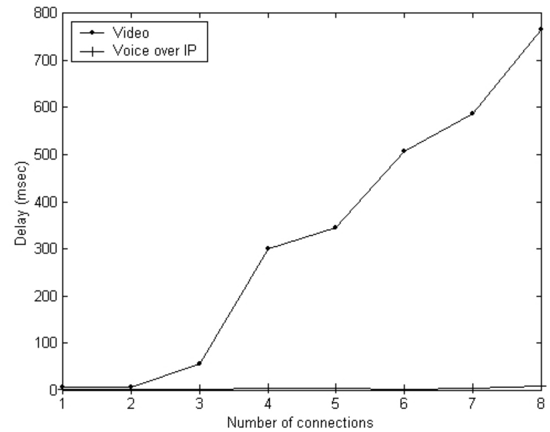
c. Throughput

Figure 6. Results of TCP traffic

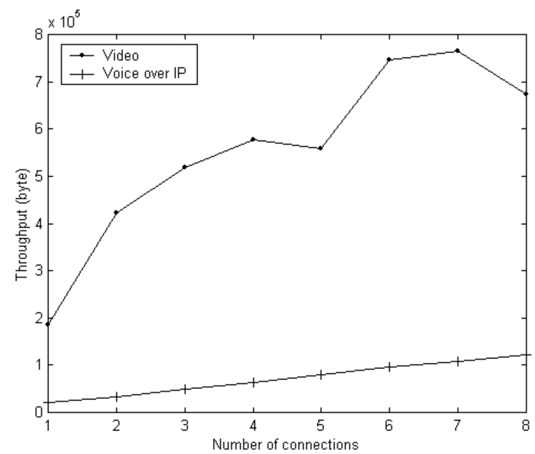
Two reasons for why the performance of video transmission for higher loads is not as good as VoIP is that: 1) VoIP is a particular case of constant bit rate traffic, while the packet size for video traffic is varying in a large range as shown in Table 3. 2) The actual transmitted video data is much larger than that of VoIP, as shown in Figure 7.c, which implies that more network interference and congestion can be introduced by the video traffic.



a. Delivery ratio



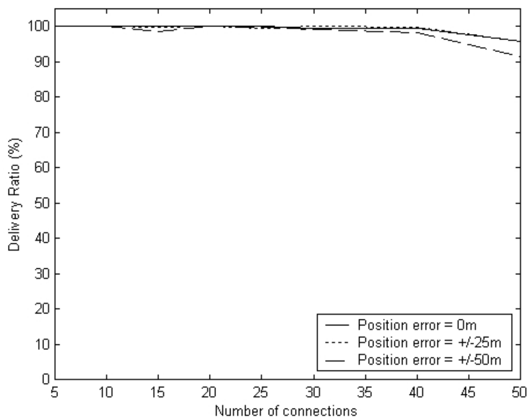
b. Transmission delay



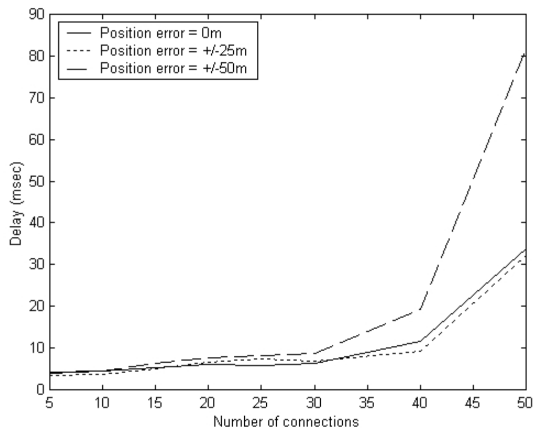
c. Throughput

Figure 7. Results of VoIP and video traffic

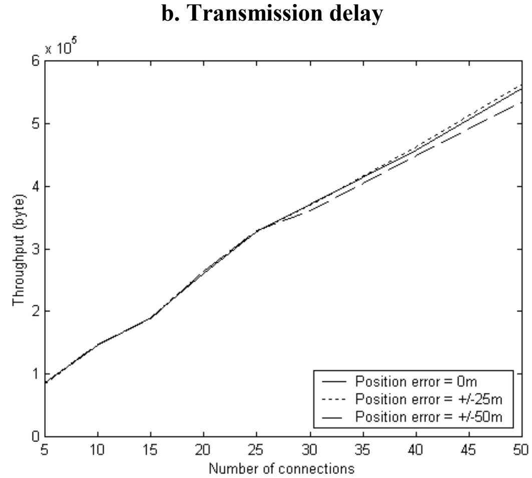
In most cases, throughput increases for higher loads because there are more data being sent and received. But since the packet size for video is varying in a large range, there might be the case when the throughput decreases although the number of packets increases, such as 5-connection case in the Figure 7.c.



a. Delivery ratio



b. Transmission delay



c. Throughput

Figure 8. Results of using different position errors

5. SIMULATION ON MOBILITY

The mobility is an important character of MANETs. It can affect the system performance in various ways. This section studies the impact of two factors due to mobility: the position measurement error and the moving speed of mobile nodes.

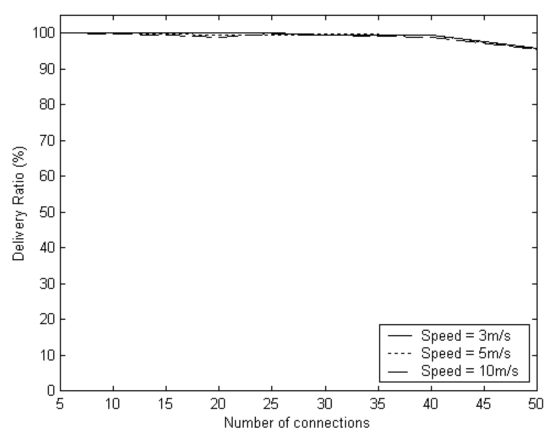
5.1 Position Error

We compare the system performance using different position measurement errors, which are randomly generated within the range: 0m, ±25 m, and ±50 m. The moving speed of mobile nodes is at most 3 m/s.

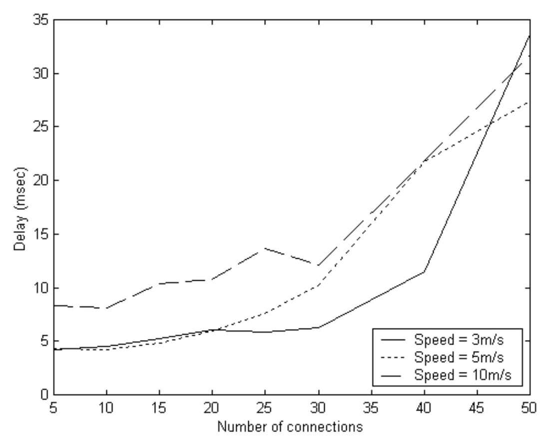
The simulation results are given in Figure 8. It is shown that the system works pretty well for position errors within ±25m, but ±50m position error does deteriorate the performance: the transmission delay significantly increases when the inaccuracy of position information increases.

5.2 Moving Speed

The system performance is tested when mobile nodes are moving at different maximum speeds of 3m/s, 5m/s, and 10m/s. The rate of position updates for these simulations is not fixed so that it can be updated whenever necessary in the routing protocol. It is shown from Figure 9 that the system works well for all these speeds. So when the maximal moving speed is less than 10 m/s, different moving speed does not introduce significant changes of the delivery ratio, transmission delay and throughput.

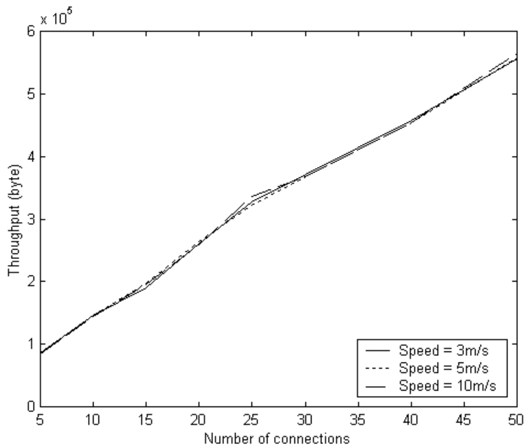


a. Delivery ratio



b. Transmission delay

Figure 9. Results of using different moving speeds (continued)



c. Throughput

Figure 9. Results of using different moving speeds

6. CONCLUSION

This paper proposes a simple link layer rate control algorithm for IEEE 802.11 in a MANET. Extensive simulations are conducted to test the impact of various system components. Several conclusions are drawn as follows.

- More accurate radio propagation and packet loss models must be used for simulation studies of MANETs.
- It is found that the NAV should be estimated in RTS by small or the last used transmission rate. The update of NAV in CTS is preferred when the network load is low.
- The carrier sense threshold should be no less than 70 dBm. But when it is large enough, larger threshold does not significantly improve the system performance.
- CBR traffic performs very well. TCP traffic provides similar delivery ratio to that of CBR, but it introduces longer delay. VoIP traffic performs much better than video traffic. But both VoIP and video do not work well when the network load is higher than 10 connections. So the real-time data transmission over MANETs deserves further studies.
- The large range of position error due to mobility may introduce longer transmission delay. The moving speeds less than 10 m/s do not introduce any significant impact on the system performance.
- A general conclusion is that simulations have shown that many kinds of system components contribute to the overall performance of a MANET. So it is not enough to study a MANET in a particular scenario.

A last note about the above conclusions is that, since they depend on the chosen system parameters and scenarios, the exact number may not apply to the real system. For example, the 70 dBm for CSThresh may not necessarily be the same for various environments; the system performance might be significantly affected when the moving speed is much larger than 10 m/s.

7. ACKNOWLEDGEMENTS

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