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Abstract

Multi-modulation-mode mobile devices can transmit or receive data at different rates. Link adaptation is widely used for improving throughput, through which link layer chooses the data rate based on signal-to-noise ratio (SNR). In most routing protocols for ad hoc networks, the route with the minimum hop count will be selected. The end-to-end data rate can be low since the small hop number implies the large geographic distance (*link distance*) between nodes. The SNR at the receiving node is low due to the large radio signal attenuation, which results in low data rate. On the other hand, making routes with large number of hops means more nodes will contend for the channel. This will reduce the channel occupation time for each node and decrease the end-to-end throughput. In this work, analysis shows that changing link distance affects the saturated throughput, and an optimal link distance exists so that the highest network end-to-end throughput can be achieved. Based on this observation, an Adaptive-Searching-Range Routing Protocol (ASRP) is proposed, in which the searching range for next hop is adaptive according to the network load, so as to get better network throughput. Link layer and routing layer protocol issues are addressed. Simulation results show that ASRP can improve network end-to-end delivery ratio.

Keywords

network throughput, link adaptation, data rate control, ad hoc networks, routing protocol design

I. Introduction

Mobile computing has been visioned wide applications in the near future [8]. Playing a key role in mobile services, the wireless network has a few major challenges, one of which is the upper-bounded wireless bandwidth. This limits the network capacity especially for high-data-rate applications. Multi-data-rate scheme has been proposed to improve bandwidth efficiency, so as to improve network capacity.

The basic idea of multi-data-rate mechanism is that, given a quality of service requirement (e.g., a tolerable bit error rate), a mobile user should apply the modulation scheme that gives the maximum transmission data rate, since higher data rate generally results in higher throughput. The signal-to-noise ratio (SNR) at the receiving end is the major criteria to determine which modulation to be used. The higher the SNR is, the modulation of a higher data rate can be used, and vice versa. The availability of the low-cost, multi-modulation device and the technique for SNR estimation [17] on a received signal makes multi-rate control feasible. Currently, four different modulation schemes (DBPSK, DQPSK, CCK, and MBOK) [2] are available in 802.11b WLAN and data rates of $1Mb/s$, $2Mb/s$, $5.5Mb/s$, and $11Mb/s$ are supported.

Data rate selection (or modulation selection) in infrastructured wireless networks is also called link adaptation [12][14][16] since only link layer will involve. In networks such as a cellular network or a WLAN, a mobile user only has to connect to its base station or the access point through a one-hop wireless link. There is no alternative route. The maximum data rate for this mobile user is determined by the radio propagation distance and the quality of the wireless channel. Routing thus can not help with network throughput improvement.

In an infrastructureless network, such as an ad hoc network, a source connects to its destination through a multi-hop route. The route is determined in the stage of routing discovery. In most on-demand ad hoc routing protocols, such as Ad Hoc Distance Vector (AODV) routing protocol [18] and Dynamic Source Routing (DSR) [11], or in the position-aided routing protocols, such as Greedy Perimeters Stateless Routing(GPSR) [5], the route with the minimum hop count will be used. This works well if there is only a fixed transmission data rate, as the route with the minimum number of hops has the minimum number of transmitting nodes. Fewer nodes will contend for the shared wireless channel when using the contention-mode Media Access Control (MAC). When multiple data rates are available, the route with minimum hop count may not bring high end-to-end throughput. The reason is that a route with a small number of hops normally implies the large geographic distance for each hop. According to the radio propagation theory [20], the radio signal attenuates fast when the propagation distance increases. The SNR at the receiving end is low, and only low data rate can be used. This results in low end-to-end throughput. On the other hand, achieving high data rate by reducing the geographic distance for links generates more contending nodes in the network because more intermediate nodes are needed to build routes. Each node then has less opportunity to send out data. The available data rate and the number of active nodes are two contradictory factors that affect the network throughput. Since these two factors depend on route selection, routing can play a major role in throughput improvement.

In this work, the impact of routing hop counts on the system end-to-end throughput will be investigated. The number of hops for the routes is adjusted by changing the maximum geographical distance for each hop in the routing discovery stage. In the rest of the paper, we call this geographical distance **link distance**. Link distance can be changed by using geographic information, or by using different receiving threshold values for the broadcast messages or "hello" messages that are used for network connectivity. Analysis and simulation results show that in a multi-rate ad hoc network, an optimum link distance exists so that the highest network end-to-end throughput can be achieved. Based on this, an Adapted Searching Range Protocol (ASRP) is proposed, which controls the hop counts by changing the maximum searching range for the next hop. The protocol is simple, and easy to be adopted in the existing ad hoc routing protocols. The Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) will be considered for the MAC mechanism in the link layer, and the corresponding link adaption schemes will be defined for multiple data rate selection.

The paper is organized as follows. Section II goes over the related research. In section III, the relationship between the link distance and the saturated end-to-end throughput is analyzed. Section IV describes the details of ASRP, the proposed protocol for hop count control. Simulation results are shown in Section V. Section VI is the conclusion.

II. Related Works

Link adaptation in 802.11a WLAN is studied in [19]. The data rate is selected by the sender. A novel MPDU (MAC protocol Data Unit)-based link adaptation scheme is proposed to help the sender to make more accurate decision according to the data payload length, the wireless channel condition, and the frame retry account. The sender determine the transmission mode for the next transmission attempt by a table lookup, using the most up-to-date system status as the index. In [9], the Receiver-Based Auto Rate (RBAR) is proposed, in which the receiver determines the data rate for the next transmission based on the received RTS when RTS/CTS handshake is used. The sender gets to know the data rate by CTS. The paper also describes the details for protocol modification in the existing standard to adopt RBAR. An enhanced protocol called Opportunistically Auto Rate (OAR) is described in [21] for high-quality channel conditions. The key mechanism of the OAR protocol is to opportunistically send multiple back-to-back data packets whenever the channel quality is good while over longer time scales, it is ensured that all nodes are granted channel access for the same time-shares as achieved by single-rate IEEE 802.11.

The impact of multi-hop wireless connection on network throughput can be found in [15]. The paper examines the network size, traffic pattern, and radio interaction along and in combination. It shows that the end-to-end throughput is related to how many nodes in the route interfere with each other. In another paper [6], the multi-data-rate is considered along with the multi-hop effect in the ad hoc network, and it shows the routing based on generating shortest hop counts is not always good to achieve optimum system throughput.

To improve network throughput, the metric other than hop count are proposed for routing selection. In both [10],[22], link delay is used as the major criteria to determine the route in a network with multiple data rates. The corresponding routing protocols are also proposed. The route with the minimum end-to-end delay will be used. In [22], the delay includes the MAC delay for channel contention and the packet queuing delay. A Poisson distributed packet arrival to each node is used for delay estimation. In [10], route is selected considering link delay, which is the overall time needed for a packet transmission during a successful attempt. This requires that the link data rate be known in advance. The results in both paper show that when using end-to-end delay for routing selection, network throughput can be improved. In another work [7], a so-called expected transmission count (ETX) metric is used for routing discovery. ETX is the predicted data transmissions (including re-transmissions) required to send a packet over the link. Higher throughput can be achieved compared with the minimum hop count routing.

Some analytical models for performance study can be used in ad hoc networks. Analysis for CSMA/CA is present in [4]. An explicit formula is derived for the saturated throughput by using a Markov state transitions. The throughput is defined as the time fraction that the shared channel is occupied for data transmission. Analysis is verified to be accurate by simulation when there are a relatively large number of contending nodes. Analysis for goodput of a WLAN with link adaptation can be found in [19]. It covers the effect of the physical layer, and the retransmission caused by the error in a wireless channel is considered. In [22], analytical model for MAC delay is given based on the assumption that the packet arrival to each node is Poisson distributed.

III. Analyisis for Impact of Link Distance on Throughput

A. Motivation: Data Rate vs. Number of Contending Nodes

According to Shannon's theory, the maximum data rate C that can be achieved in the channel with a bandwidth of B is:

$$C = B\log_2(1+\frac{p_r}{n_r}), \tag{1}$$

where p_r is the receiving power and n_r is the noise. When $p_r/n_r \leq 1$, C is approximately linearly related to p_r/n_r. When p_r/n_r is large, the data rate increases slowly when p_r/n_r increases. For consistency, in this paper, p_r/n_r is called the receiving SNR.

In a Two-Ray Ground propagation model, the radio propagation model often used for the open field, the signal power attenuates fast as the transmission distance increases. Define p_t as the transmitting power and d as the distance between a transmitter and its receiver, in such a model, the receiving power p_r is:

$$p_r = \frac{p_t G_t G_r h_t^2 h_r^2}{d^4} = \frac{kP_t}{d^4}, \tag{2}$$

where G_t and G_r are antenna gains at the transmitter and the receiver, and h_t and h_r are the antenna heights. Since the gains and the heights are fixed, k is a constant.

In a multi-rate ad hoc network, routing selection based on hops with different link distance may result in different end-to-end throughput. When considering the Two-Ray Ground model, for a wireless link that originally has a small SNR at the receiving end, reducing the link distance for this hop can result in a large SNR gain. The data rate can increase significantly. This will overwhelm the fact that more nodes have to contend for the wireless channel. The overall network end-to-end throughput may be enhanced. When the link distance is short so that the receiving SNR has been high, reducing the distance further does not generate much data rate improvement. In this case, the overall system throughput may decrease since the increase of number of contending nodes is more dominant. For a network, a high throughput can be achieved when a proper link distance is used.

The illustrated example in Fig. 1 shows the change of the data rate and the number of the contending nodes in a evenly distributed ad hoc network when link distance changes. Here every node in a row receives data from the node to its left and forwards the data to its right. Compared with that in Fig. 1(A), the link distance between any two consecutive nodes in Fig. 1(B) is the half. Assume the bandwidth is $1Mb/s$, and in Fig. 1(A) the receiving SNR for a node is 1, so the data rate in each link is $1Mb/s$ (Eqn. 1). In Fig. 1(B), the data rate then is approximately $4Mb/s$ (by Eqn. 1 and Eqn. 2). However, for any node such as the node n, it has to contend with approximately twice as many as nodes within its coverage of r_c. r_c is the radius for the carrier sense zone for CSMA.

B. Overview for Distributed Coordination Function with CSMA/CA

Distributed Coordination Function (DCF) using CSMA/CA as the MAC technique is considered to be used in the link layer of ad hoc networks. In DCF, a node with a new packet to transmit monitors the channel activity. If the channel is idle for a period of time equal to a distributed inter-frame space (DIFS), the node transmits. Otherwise, if the channel is sensed busy (either immediately or during the DIFS), the node persists to monitor the channel until it is measured idle for a DIFS. At this point, the node generates a random backoff interval before transmitting to minimize the probability of collision with packets being transmitted by other nodes.

For efficiency reasons, DCF employs a discrete-time backoff scale. The time immediately following an idle DIFS is slotted, and a node is allowed to transmit only at the beginning of each slot time. The slot time size, δ, is set equal to the time needed at any node to detect the transmission of a packet from any other node. DCF adopts an exponential backoff scheme. At each packet transmission, the backoff time is uniformly chosen in the range $(0, w-1)$. The value w is called contention window, and depends on the number of transmissions failed for the packet. At the first transmission attempt, w is set equal to a value called minimum contention window. After each unsuccessful transmission, w is doubled, up to a maximum value.

Since CSMA/CA does not rely on the capability of the nodes to detect a collision by hearing their own transmission, an ACK is transmitted by the receiving node to signal the successful packet reception. The ACK is immediately transmitted at the end of the packet, after a period of time called short inter-frame space (SIFS). As the SIFS (plus the propagation delay) is shorter than a DIFS, no other node is able to detect the channel idle for a DIFS until the end of the ACK. If the transmitting node does not receive the ACK within a specified ACK Timeout, or it detects the transmission of a different packet on the channel, it reschedules the packet transmission according to the given backoff rules.

The above described two-way handshaking technique for the packet transmission is called basic access mechanism. To solve the so-called hidden nodes problem [13], DCF defines an additional four-way handshaking technique to be optionally used for a packet transmission. This mechanism is known with the name RTS/CTS. A node that wants to transmit a packet, waits until the channel is sensed idle for a DIFS, follows the backoff rules explained above, and then, instead of the packet, transmits a special short frame called request to send (RTS). When the receiving node detects an RTS frame, it responds, after a SIFS, with a clear to send (CTS) frame. The transmitting node is allowed to transmit its packet only if the CTS frame is correctly received.

A more complete and detailed presentation is the 802.11 standard [1].

C. *Saturation Throughput for CSMA/CA*

The saturation throughput is defined as the limit reached by the system throughput as load increases, and represents the maximum load that the system can carry in stable conditions. In a saturated network, active nodes always have packets to send. The analytical model for saturation throughput in WLAN using CSMA/CA is presented in [4]. Here we list the major results.

Define τ as the stationary probability that a node transmits a packet in a generic (i.e., randomly chosen) slot time, and p the constant and independent probability that a packet will collide with other packets. τ and p can be solved by the following equation group:

$$\begin{cases} \tau = \frac{2(1-2p)}{(1-2p)(W+1)+pW(1-(2p)^m)}, \\ p = 1 - (1 - \tau)^{n-1}, \end{cases} \tag{3}$$

where W is the minimum contention window size, m is the maximum backoff stage so that the maximum window size W_{max} is $2^m W$, and n is the number of contending nodes in the system.

Define P_{tr} as the probability that there is at least one transmission in the considered slot time, and P_s as the probability that exactly one node transmits on the channel so that a transmission occurring on the channel is successful. P_{tr} and P_s can be calculated by:

$$P_{tr} = 1 - (1 - \tau)^n. \tag{4}$$

$$P_s = \frac{n\tau(1 - \tau)^{n-1}}{1 - (1 - \tau)^n}. \tag{5}$$

Let S be the normalized system throughput, defined as the fraction of time the channel is used to successfully transmit pay-load bits, which is:

$$S = \frac{P_s P_{tr} E[P]}{(1-P_{tr})\sigma + P_{tr} P_s T_s + P_{tr}(1-P_s)T_c}, \tag{6}$$

where $E[P]$ is the average packet transmitting time, T_s is the average time the channel is sensed busy because of a successful transmission, and T_c is the average time the channel is sensed busy by each node during a collision. For the basic DCF, T_s and T_c can be calculated by:

$$\begin{cases} T_s = H + E[P] + SIFS + \delta + ACK + DIFS + \delta, \\ T_c = H + E[P^*] + DIFS + \delta, \end{cases} \tag{7}$$

where $E[P^*]$ is the average length of the longest packet transmission time in a collision. Define $F(x)$ the probability for the packet transmission time, then:

$$E[P^*] \approx \int_0^{P_{max}} (1 - F(x)^2) dx, \tag{8}$$

where P_{max} is the maximum transmitting time for the packets.

For the four-way handshake with RTS/CTS scheme, T_s and T_c can be calculated by:

$$\begin{cases} T_s = RTS + CTS + \delta + CTS + SIFS + \delta + H + E[P] + SIFS + \delta + ACK + DIFS + \delta, \\ T_c = RTS + DIFS + \delta, \end{cases} \tag{9}$$

where δ is the duration of an empty slot time, $H = PHY_{hdr} + MAC_{hdr}$ is the packet header, and σ is the propagation delay.

D. Saturation Throughput in a High-Density Ad Hoc network

In this subsection, an analytical model is built for end-to-end saturation throughput calculation in an ad hoc network. CSMA/CA is considered to be the MAC mechanism.

To simplify analysis, it is assumed that the network has large density of nodes, which means for a pair of source and destination, routes made up by different number of hops can always be found, and the intermediate nodes of the routes can be on or close to the straight line connecting the source and the destination. It is also assumed that a node is not included in more than one connection. Without losing generality, we analyze a network with a small size in which the transmission of any node will interfere all the other nodes. This size is determined by the radio propagation model and the carrier sense threshold value for CSMA/CA. The analysis results hold for large-size networks when nodes and traffic are uniformly distributed.

We study a network with i connections, and the geographical distance between the sources and their destinations for these i connections are $\mathbf{D} = \{D_1, D_2, ...D_i\}$. Define d_{max} to be the maximum link distance for a hop. Let $\mathbf{h} = \{h_1, h_2, ...h_i\}$ be the number of hops in each connection, which is also the number of transmitting (contending) nodes. For any connection j, $h_j = \lceil \frac{D_j}{d_{max}} \rceil$. Here symbol $\lceil \cdot \rceil$ stands for rounding a real number towards plus infinity. The overall number of the nodes that will contend for the channel, N, is:

$$N = \sum_{j=1}^{i} \lceil \frac{D_j}{d_{max}} \rceil = \sum_{j=1}^{i} h_j. \tag{10}$$

The end-to-end throughput for a connection is normally determined by the worst link in the route. Considering an ideal radio environment, to achieve high throughput, hops in a connection should have equal link distance, so that the data rate in the links along the route is the same. Let $\bar{\mathbf{d}} = \{\bar{d}_1, \bar{d}_2, ...\bar{d}_i\}$, of which $\bar{d}_j$ is the link distance for each hop in connection j, and $\bar{d}_j = D_j/h_j$. Let $\mathbf{c} = \{c_1, c_2, ...c_i\}$ be the set for all the data rates for these i connections. Using Shannon's Theory (Eqn. 1) and the Two-Ray ground radio propagation model (Eqn. 2), the highest available data rate in every link for this connection, defined as c_j, is:

$$c_j = Wlog_2(1 + \frac{kP_{tr}}{n_j \bar{d}_j^{\,4}}), \tag{11}$$

where n_j is the background noise for connection j. In our network scenario, during the time that a node is transmitting a packet, all the other nodes can sense it and will not transmit anything. n_j is approximately the same as the white noise n_0.

For analysis tractability, we assume all the packets have the same size P_{len}. In the links with different transmitting rates, packet transmitting time is different. In a saturated ad hoc network, all the nodes have the equal opportunity to send a data packet. For a long period of time, every node will successfully send the same number of data packets. Thus the average transmitting time of a packet in the network, $E[P]$, is:

$$E[P] = \frac{1}{N} \sum_{j=1}^{i} h_j \frac{P_{len}}{c_j}. \tag{12}$$

With the knowledge of the overall number of contending nodes N and the average transmitting time $E[P]$, the time fraction for data transmission with RTS/CTS handshake, S, can be calculated. For S in the basic DCF, $F(x)$ can be found numerically so that $E[P^*]$ can be found.

The time fraction that a node occupies the channel depends on the data rate in the link. The nodes in the same connection have the same time fraction to occupy the channel since these nodes use the same data rate for packet transmission. Let $\mathbf{t} = \{t_1, t_2, ...t_i\}$ be the time fraction that a node in each of these i connections occupies the wireless channel. Specifically, t_1 is the time fraction that a node in connection 1 occupies the channel, and t_1 can be solved from

$$t_1 N_1 + \sum_{j=2}^{i} t_1 \frac{c_1}{c_j} N_j = S. \tag{13}$$

For any connection j, $t_j = t_1 c_1 / c_j$.

The overall system throughput η is defined as the number of bits that can be transmitted in all i connections per second. For any connection, its end-to-end throughput is the same as the bits successfully transmitted by the node which is the previous hop of the destination. In our network scenario, where every node has the same time fraction for data transmission, η can be calculated by:

$$\eta = \sum_{j=1}^{i} t_j c_j. \tag{14}$$

E. Numerical Results

If not specified, the major system parameters are those listed in Table I. The bandwidth for the ad hoc wireless channel is assigned to be $1Mb/s$. We assume that in a link, if a receiver is a normalized geographic distance of 1 away from its transmitter, the SNR value at the receiver is 1. The corresponding achievable data rate in this link then is $1Mb/s$. There are a number of connections in the network, and distance between a source and its destination is uniformly distributed.

Figure 2 and Fig. 3 show the system end-to-end throughput against the maximum link distance with/without using RTS/CTS handshake. There are three major observations.

1. There exists an optimum link distance, by using which the network end-to-end saturation throughput is the maximum. When the link distance changes from a large value towards the optimum one, the throughput gain brought by the increased data rate is larger than the loss caused by the increasing number of contending nodes. The system throughput gets higher. When this distance becomes shorter, the loss caused by contending nodes is more dominant, and the system throughput decreases.
2. The optimum link distance changes based on the number of connections, i.e., the number of contending nodes. In a network with a small number of connections, this optimum distance is relatively short. Otherwise, it is long. This means in a lightly loaded network, it is possible to improve the throughput by reducing the link distance and getting a higher data rate along the route. In a heavily loaded network, especially the network using the basic DCF, it is more important to keep the number of contending nodes small, and routes with long link distance

should be used.

3. The network saturation throughput decreases as the number of connections increases. This is due to the increased number of contending nodes, which results in more packet collisions caused by the hidden node problem. The problem is more serious in the network using the basic DCF, in which the throughput decreases fast when the number of connection increases. The reason is that the size for a data packet is normally much larger than that of RTS. In a network with basic DCF, the probability of a data packet collision caused by the hidden nodes is much higher than that of a RTS collision in a network using RTS/CTS handshake.

If the data rate is high, data can be transmitted in large-size packets even in the fast-fading channel. Increasing the size of the packets can also improve the efficiency of DCF. Figure 4 shows the throughput improvement in a network using RTS/CTS when the packet size increases. The increased packet size also results in the reduction of the optimum link distance. It should be pointed out that, in this analysis, an ideal radio environment is considered and there is no packet re-transmission caused by the channel error. Figure 5 shows the results in the network using the basic DCF. The throughput improvement due to the increased packet size is not as significant as that in a network using RTS/CTS handshake, because the larger size of the packet also causes more serious hidden nodes problem.

IV. ASRP: Link Distance Control by Searching Range Adaptation

A. *Adaptive Searching Range Routing Protocol (ASRP)*

Analysis results show that an optimum link distance exists so that the highest throughput can be achieved. Adaptive Searching Range Routing Protocol (ASPR) is designed to find the routes made up by the links with optimum link distance. Specifically, in the routing discovery stage, the searching range for the next hop is adapted according to the variance of the link distance for better network throughput. For a source node or an intermediate node, only a node within its searching range can be its next hop. When the searching range is small, routes are made up by the links with short link distance. The corresponding data rate in the links can be high. On the other hand, if the searching range is large, routes can be built by links with long link distance but likely low data rate. ASRP is not a best-effort routing discovery protocol. Instead, it finds out the routes with the required link distance, and in subsequence, finds out the route with a certain data rate. If such a route cannot be found, a larger searching range can be used.

For ASRP routing maintenance, when the receiving signal quality degrades, the receiving end of the link will inform the transmitting end. Based on different routing recovery strategies, the transmitting end will either find another route locally which satisfies the link distance requirement, or send back a routing error message to the source node. However, this link can degrade its data rate and successfully transmit the outstanding data packets in the route. This makes ASRP more robust since originally a data rate higher than the basic rate will probably be used. When that data rate cannot be used due to the degraded signal quality, a lower data rate can be used.

The searching range can be a distance in a network with position information, or can be a threshold value for receiving SNR when no position information is available. To get better network throughput, learned from analysis, the searching range for a node should be small if a small number of active nodes (i.e., low load) are within its radio coverage (i.e., the carrier sense range), and the searching range should be large if this number is large. The more precise relationship between searching range and load in real networks can be learned by extensive simulations.

The load around a node can be estimated by how frequently it senses a transmission. It then uses the estimated load to decide the searching range. Exact load information can also be collected by a two-hop local information exchange as the illustrated example in Fig. 7. The traffic information is broadcast two hops by setting the TTL (time-to-live) value as two. In the example, the traffic in A and B is known to D through C. After D processes the two-hop information exchange with E and F, the load for A and B is learned by E and F. A node thus can know the traffic for another node which is no more than four hops away. Since the information is exchanged at the basic rate, the distance of four hops is approximately the carrier sense range [15].

ASRP is designed to improve network throughput. It is also suitable for applications with data rate requirements, and the searching range can be determined by the minimum required data rate. However, it is not trivial to estimate the efficient end-to-end data rate even the data rate in each link is known.

B. Searching Range Adaptation in Position-Aided Routing Protocols

The searching range for ASRP in position-aided routing is simply a distance. A node judges whether another node is within its searching range by finding out whether the distance between them is smaller than a value. An illustrated ASRP with greedy position routing is in Fig. 6. A source S needs to find a route to the destination D. When using a large searching range r_1, node n_1 will be the next hop of S since among the nodes that are within the source's searching range, n_1 is the closest to the destination. Through the same way, node n_1 will find n_2 as its next hop and n_2 will reach the destination. A 3-hop route $S \rightarrow n_1 \rightarrow n_2 \rightarrow D$ can can be found. When using smaller searching range r_2, a 4-hop route $S \rightarrow n_3 \rightarrow n_4 \rightarrow n_5 \rightarrow D$ is found, in which the link distance is smaller.

C. Searching Range Adaptation in On-Demand Routing Protocols

ASRP can be jointly applied with on-demand ad hoc routing protocols. The existing on-demand ad hoc routing protocols such as AODV and DSR depend on the local connectivity management in routing discovery, maintenance, and recovery. Local connectivity is attained by a node learning from its neighbor's existence. For example, in AODV, nodes learn of their neighbors by receiving a broadcast message (e.g., a routing request) or by a "hello" message. This connectivity between neighbors will be used for routing decision in the routing discovery stage. A node receiving a routing request from one of its neighbors will consider itself as the possible next hop for this neighbor. At the mean time, it adds its identification to the message and broadcast it again. This process is repeated until the destination node is reached.

When using ASRP, the searching range can be adjusted by changing the receiving SNR threshold value for the broadcast messages or "hello" messages, which virtually changes the maximum link distance. The threshold value is derived from the transmitting power, the propagation model in the wireless channel, and the link distance requirement. The value is carried in the routing request message. On receiving a routing request, only the nodes with the received SNR above that threshold value consider themselves to be the possible next hops and re-broadcast. Even in a real network, a high SNR threshold value for the link generally results in routes with short link distance, more contending nodes, and high data rate.

The searching range can also be adjusted by changing the transmitting power for the broadcast message or the "hello" message. For example, the link distance can be reduced by using a low-power broadcast message during routing discovery. This also reduces the co-channel interference caused by the broadcast messages.

ASRP can be well adopted by the on-demand routing protocols. First, it is simple. Only routing request needs to be modified by inserting the receiving SNR threshold value. Second, ASRP does not need link information (e.g., data rate, delay) in advance. The link condition is collected only during routing discovery by sending out broadcast messages. No historic information needs to be stored. Finally, decision of connectivity is made by the receiving end, which is more precise.

D. MAC Layer Modification

ASRP determines the routes and gives the lower bound for link data rates. It is possible that a higher data rate is available with link adaptation. The receiver-based link adaptation can be used when there is a RTS/CTS handshake. On receiving the RTS, the receiver makes the estimation on SNR and selects the data rate. The rate information is inserted in CTS and sent back to the sender. The sender then uses the rate for data transmission. RTS and CTS are transmitted at the basic data rate. Since there is little time between RTS and data transmission, the radio environment can hardly change. The selected data rate based on RTS should be proper for the data packet.

One problem on such a rate selection scheme is that it is not possible for the sender to include the precise transmission duration time in RTS, as required in the standard, because at this moment, the sender does not know exactly what data rate will be used. Yet the duration time determines network allocation vector, which is an important factor for CSMA/ÇA performance. However, the nodes around the receiver can also retrieve the accurate duration information from the data packet in the trade off of more power consumption. The transmitting node can also estimate the duration time using the data rate for last transmission when channel variation is not fast.

The receiver-based link adaptation scheme can be implemented into the 802.11 standard with minor modification. For more details on implementation issues, refer to [9].

In case there is no RTS/CTS handshake, the sender can use the minimum data rate directly. To utilize bandwidth more efficiently, the sender can also make the estimation based on the reverse channel. In on-demand routing protocols, this works when there is a symmetric channel propagation condition, since the signal reception quality can be calculated on the received routing reply.

V. Simulation Results

The most recent version (2.26) of the network simulator ns2 is used for the experimental study. We simulate an ad hoc network with 100 nodes residing in an area of $1000m \times 1000m$. Each node moves within the area, with a random direction and a random velocity uniformly distributed between 0 and a maximum value. Without any specification, this maximum value is $10m/s$. The wireless interface works like the 914 MHz Lucent WaveLAN, with a nominal radio range of $250m$ when transmitting at the rate of $1Mb/s$. Link data rates and the corresponding SNR values follow the standard of 802.11g, which are listed in Table II. For a connection of a source and its destination, a constant bit rate (CBR) of 4 packet per second is used, with the packet size of $4096bits$.

To get more precise simulation results, we modified the simulator by using SNR value to decide whether a data packet can be received correctly or not. The simulation environment is the cellular-aided mobile ad hoc network (CAMA) [3], in which a greedy routing with full position information is used. The searching range in ASRP then is the same as the maximum link distance for a hop.

Figure 8 and Fig. 9 show the delivery ratio and delay when using different schemes to set transmission duration time in RTS for receiver-based link adaptation. In schemes NAV_1 and NAV_2, the duration time is estimated based on the lowest data rate (i.e., $1Mb/s$) and the highest data rate (i.e., $48Mb/s$). In scheme NAV_3, the data rate for last transmission is used to calculate the duration time. In scheme NAV_4, the exact time is learned by listening to the data packet header. It shows that NAV_1 has the lowest delivery ratio and the longest delay since it unnecessarily turns off the nodes around the transmitting node for a long time. NAV_3 is close to NAV_4 which has the best performance. In the rest of the simulation, NAV_4 is used.

Figure 10 shows the delivery ratio against different searching range (the maximum link distances). The network with a light load of 10 connections, a medium load of 30 connections, and a heavy load of 50 connections are simulated. Although the network may not be saturated, the result is similar to that of analysis (refer to III-E). The figure shows that there is an optimum link distance at which the delivery ratio is the highest, especially for the low-load and medium-load networks. When the number of connections is small, fewer nodes will contend for the channel, and the optimum link distance can be short. When the load is high, the optimum link distance is longer. Table III shows the simulated optimum link distance at different network loads.

Figure 11 shows the end-to-end delay against different link distances. When the link distance decreases, the hop counts increases (as shown in Table. IV) and for a packet successfully delivered to the destination, it needs more transmissions. However, delay does not necessarily increase since when the link distance gets shorter, the data rate in the link can be higher. The packet transmission time then can be smaller. The figure shows that delay is closely related to the delivery ratio. If the optimum link distance is used so that the delivery ratio is the best, the corresponding delay will be the minimum.

Figure 12 shows the histogram for the data rate applied in the links when the optimum link distance is used as the searching range. For the networks with different loads, the mostly used data rates are $2Mb/s$, $5.5Mb/s$, and $11Mb/s$. The mostly used data rate changes from high to low as the network load increases. Figure 13 shows the data rate during a 10 second simulation for the network of 30 connections.

Figure 14 shows the optimum delivery ratios at different node mobility. The delivery ratio does not decrease significantly when mobility increases. This is partly because the strong routing recovery capability in the CAMA environment.

Figure 15 shows the improvement of the delivery ratio with the increased packet size. We simulate a network with large load (50 connections). CBR rate is kept the same by adjusting the packet transmitting rate when the packet size changes. The simulation results show that when packet size increases, the delivery ratio increases. It also shows the optimum link distance gets smaller when packet size gets larger [1], since the larger size of packet means the fewer transmission attempts. The optimum link distance depends on the number of contending nodes and how often a node sends packets.

When applying ASRP in a real network, it is difficult to set the searching range exactly the same as the optimum link distance at the moment a route needs to be found. A sub-optimal searching range scheme can be used, in which the network load is categorized as three classes: low load (e.g., fewer than 25 connections), medium load (e.g., $25 - 40$ connections), and high load (e.g., more than 40 connections). The corresponding searching ranges for the sub-optimum ASRP in our simulation are set as $140m$, $180m$, and $220m$ based on the result in Table. III. Figure 16 shows the delivery ratio when using protocols with the optimum searching range, the sub-optimum searching range, and the fixed searching range of $250m$. The sub-optimum scheme has a close performance to the optimum scheme and can improve network throughput.

VI. Conclusion

In this work, we investigate the impact of routing on the network end-to-end throughput of a multi-rate ad hoc network. Reducing the geographical distance for the hops, named as the link distance in this work, generates routes with larger hop counts (i.e., more contending nodes) but higher link data rate. This two-fold effect has been studied by analysis. The results show that

[1] The optimum link distance is $200m$ when the packet size is $16384bit$.

by changing the link distance, an optimum network throughput can be reached. The adaptive searching range routing protocol (ASRP) is proposed. ASRP achieve a better throughput over the existing routing protocols by changing link distance. The protocol is evaluated by simulation.

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TABLE I

SYSTEM PARAMETERS IN ANALYSIS.

Packet Length	8196 bits
Packet Header (H)	400 bits
ACK	240 bits
RTS	288 bits
CTS	240 bits
Bandwidth	1 Mb
Basic data rate	1 Mb/s
Propagation Delay (σ)	1 μs
Slot Time (δ)	50 μs
SIFS	28 μs
DIFS	128 μs

TABLE II

DATA RATES AND SNR THRESHOLD VALUES FOR 802.11G.

Data rate (Mb/s)	1	2	5.5	11	24	36	48
SNR threshold value (dB)	11	14	16	18	21	26	30

TABLE III

OPTIMUM LINK DISTANCE VS. NETWORK LOAD.

Number of connections	10	20	30	40	50
Optimum link distance (m)	150	175	185	200	225

TABLE IV

AVERAGE HOP COUNTS VS. MAXIMUM LINK DISTANCE.

Link distance (m)	75	100	125	150	175	200	225	250
Average hop counts	9.4	7.3	6.8	6.1	5.2	4.4	4	3.8

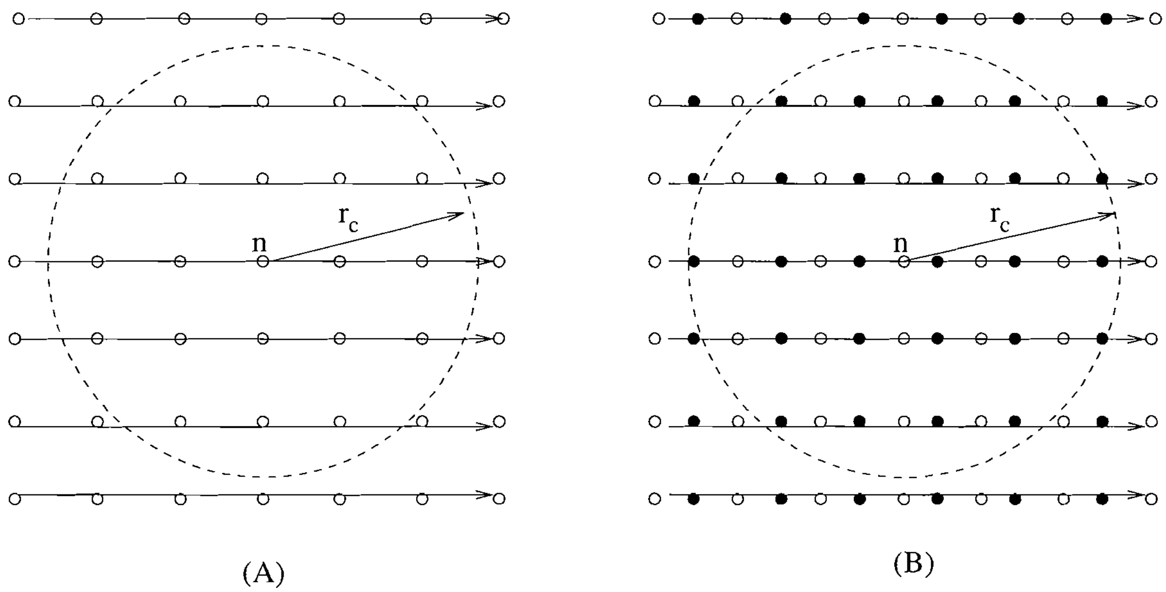

Fig. 1. Link distance vs. contending nodes: an example

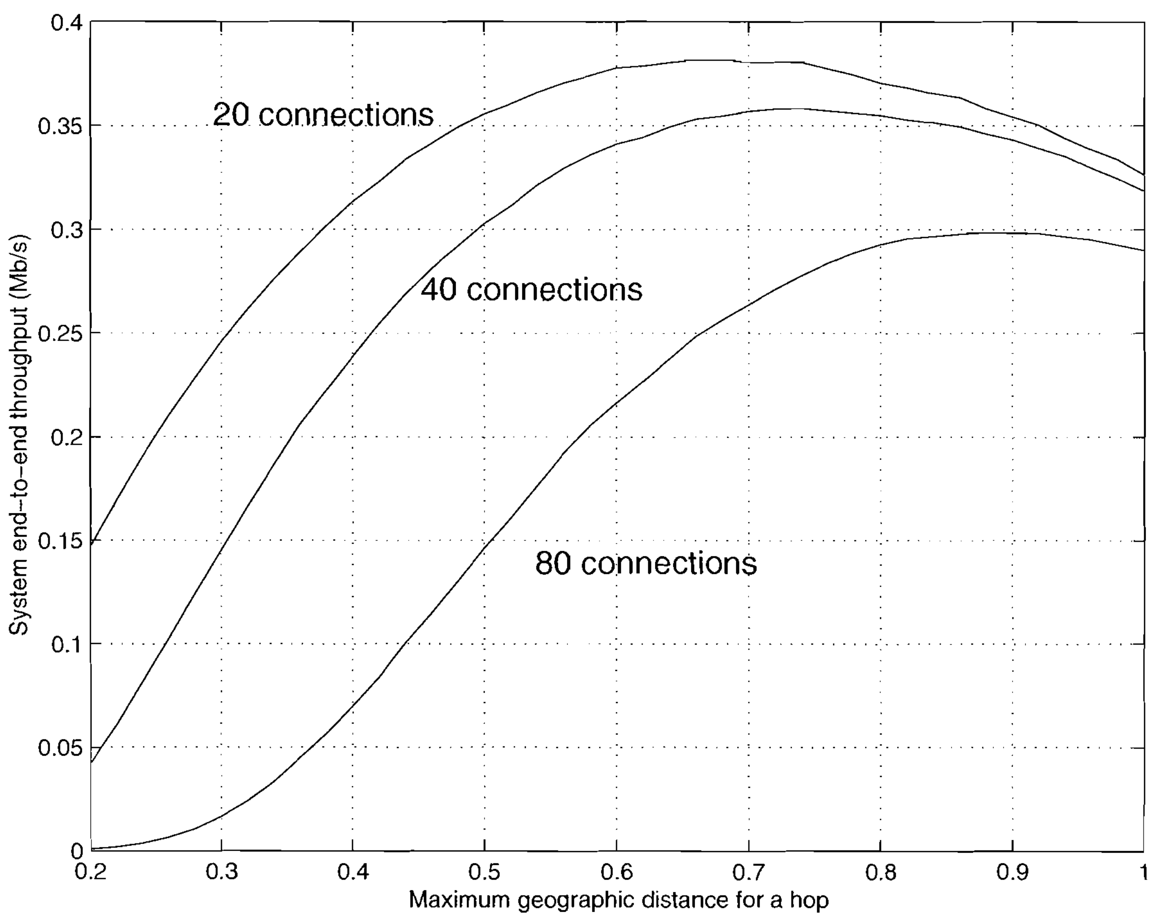

Fig. 2. Throughput vs. maximum link distance with RTS/CTS handshake.

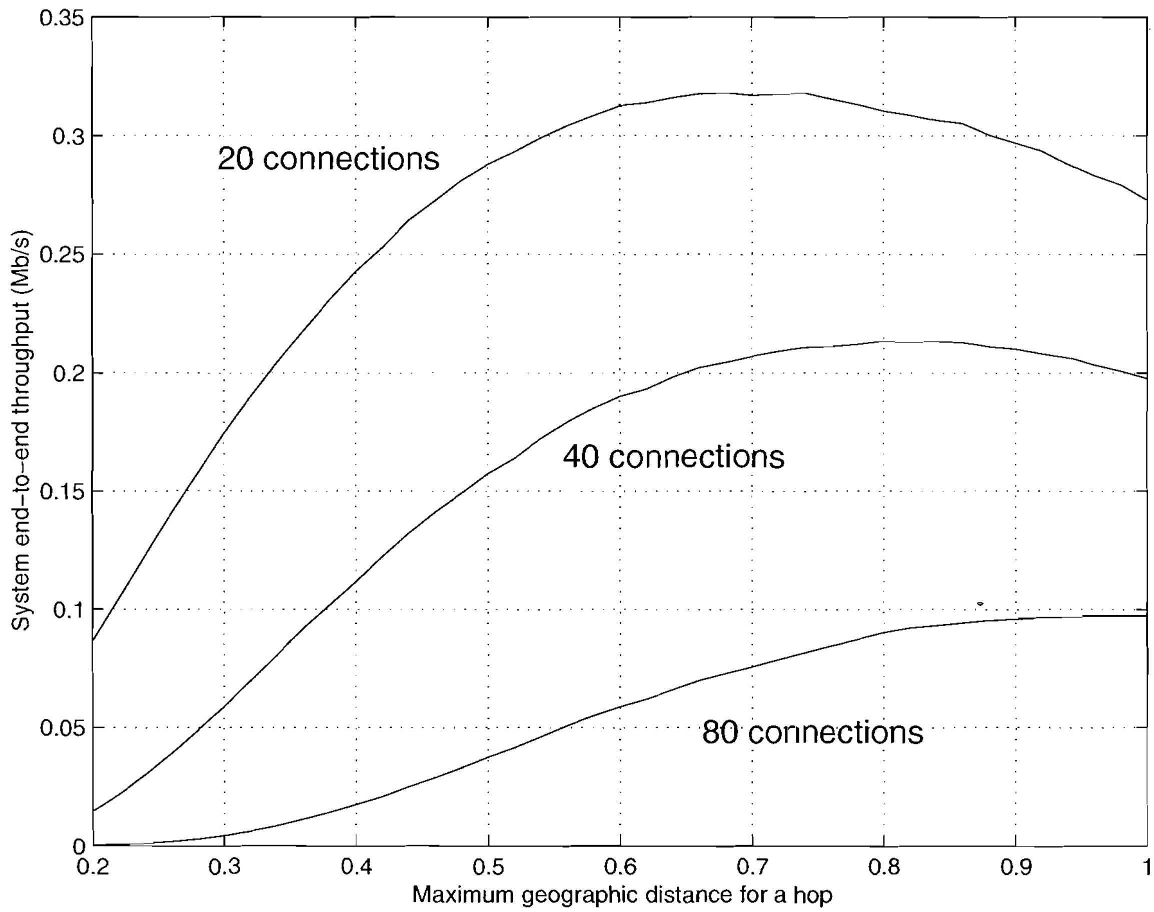

Fig. 3. Throughput vs. link distance without RTS/CTS handshake.

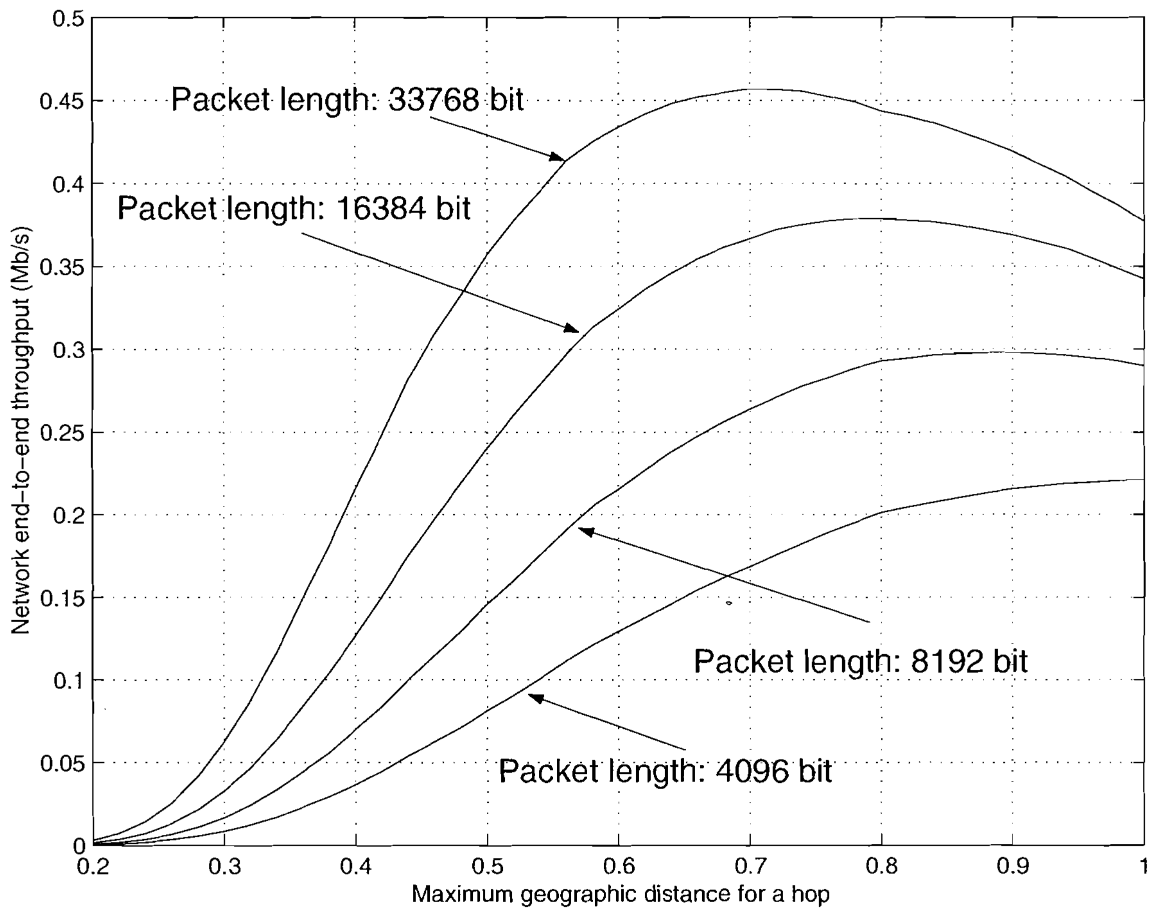

Fig. 4. Throughput vs. different packet lengths with RTS/CTS.

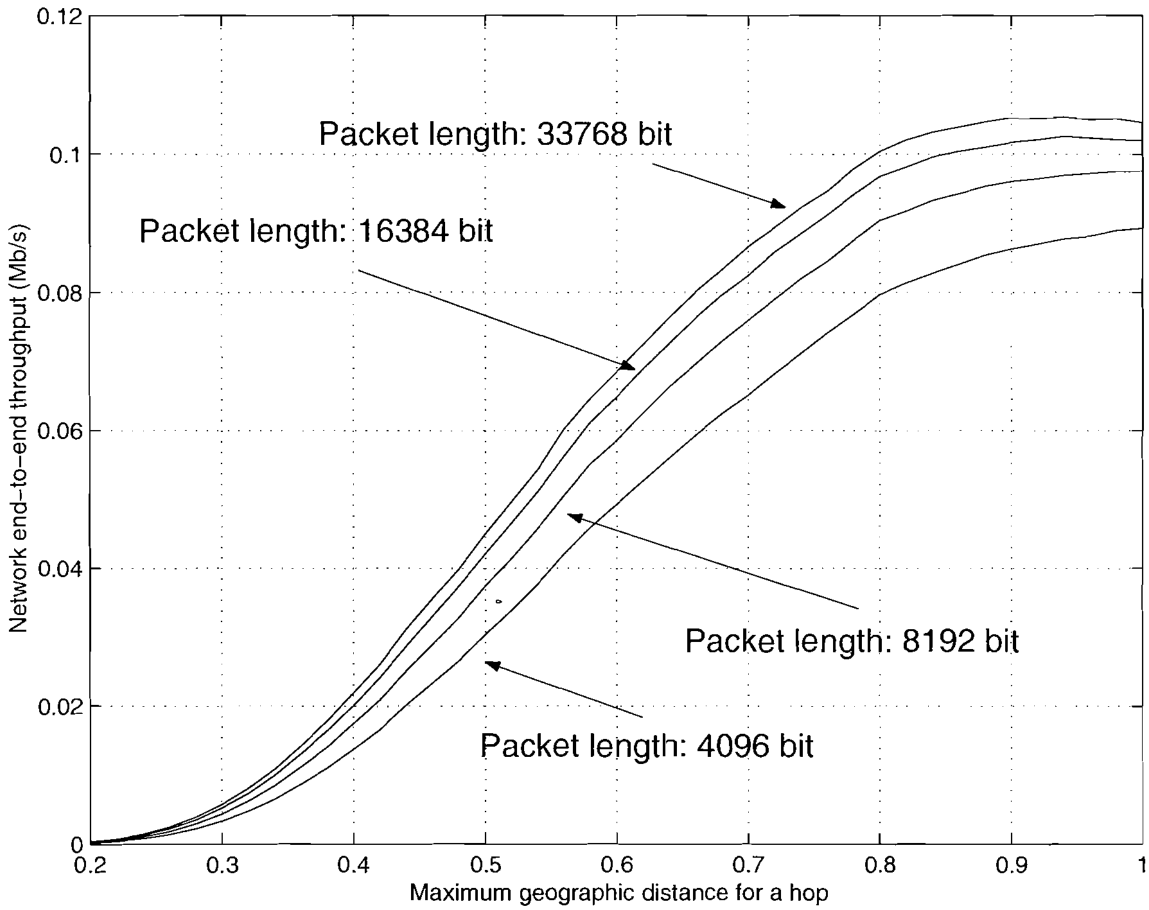

Fig. 5. Throughput vs. different packet lengths without RTS/CTS.

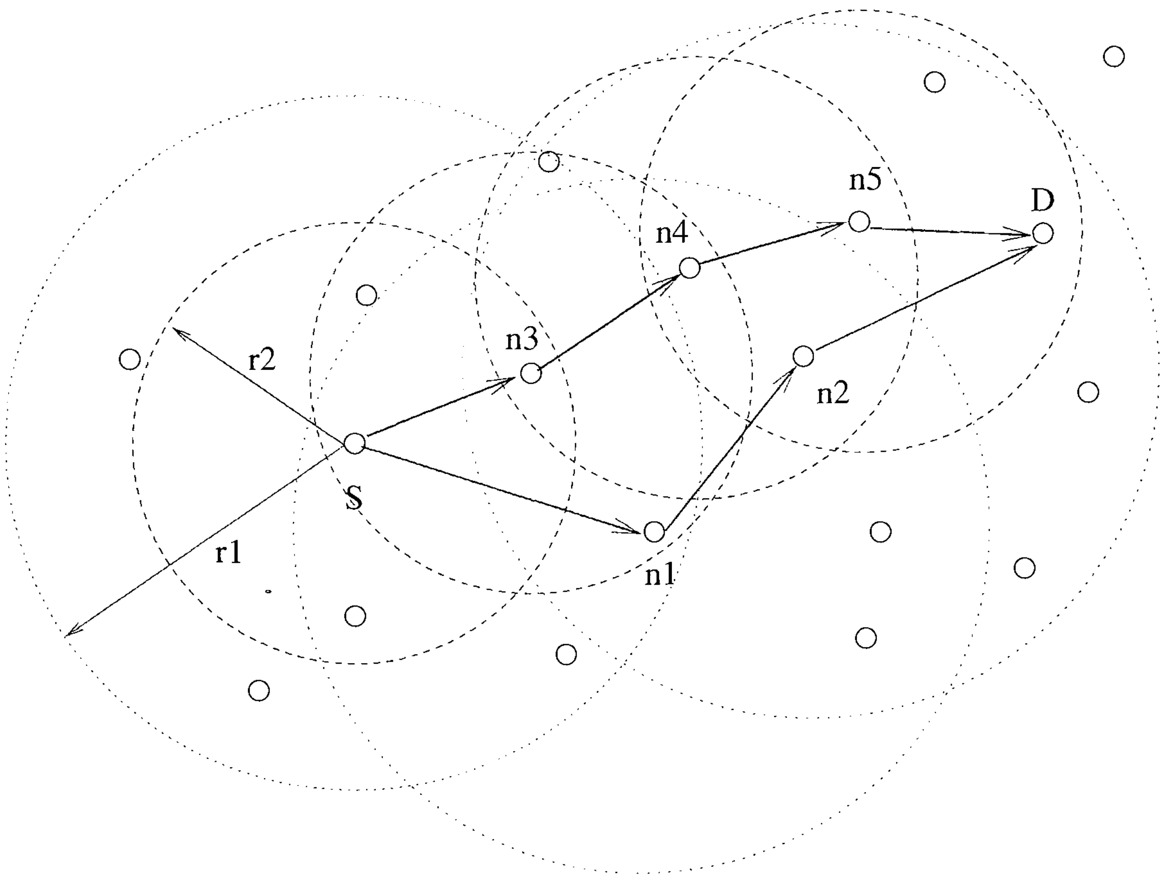

Fig. 6. Adaptive Searching Range Routing (ASRP) with positioning information.

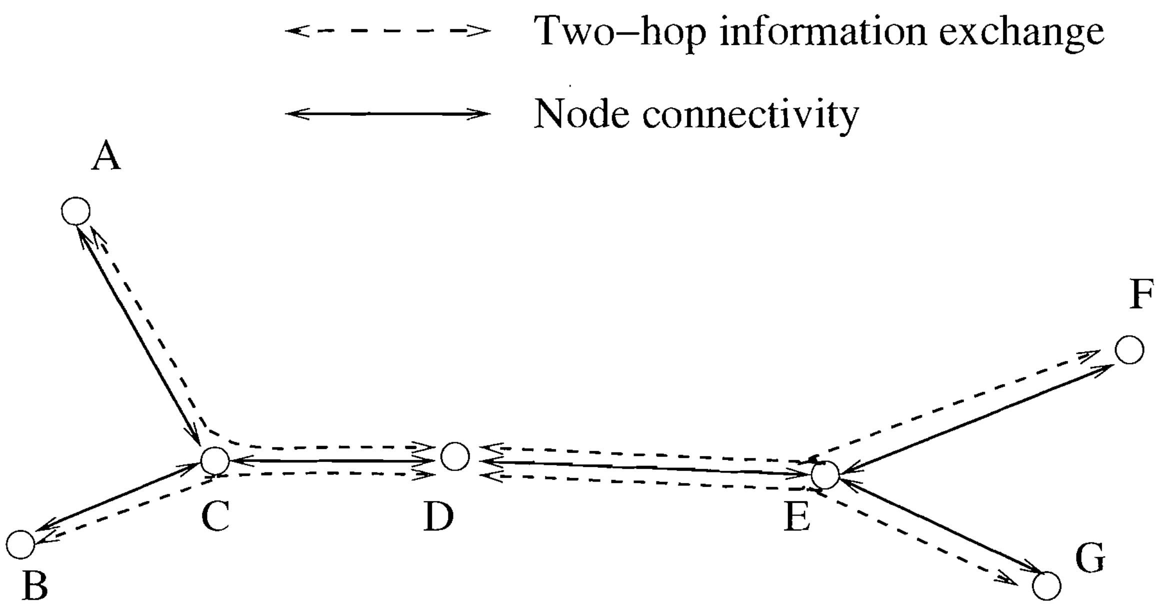

Fig. 7. A two-hop information exchange.

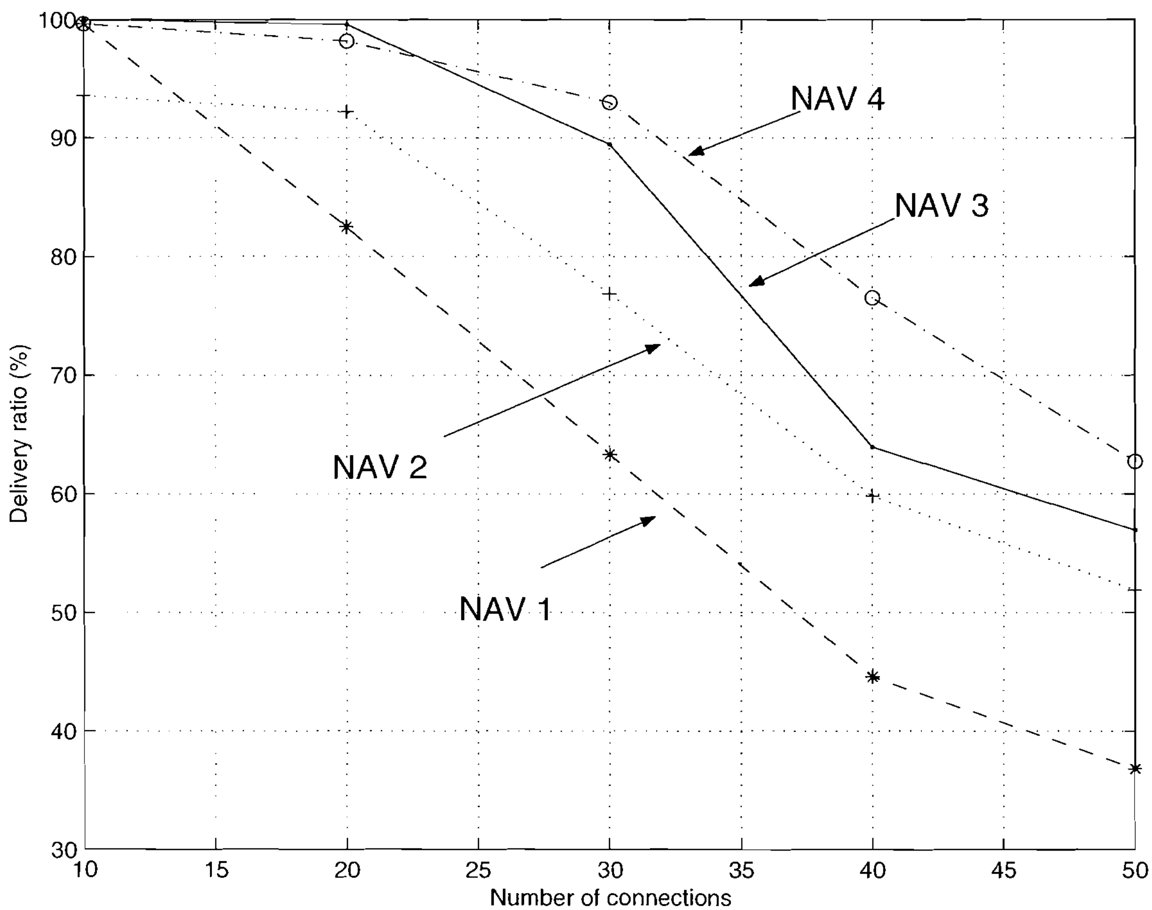

Fig. 8. Delivery ratio for different duration time estimation schemes.

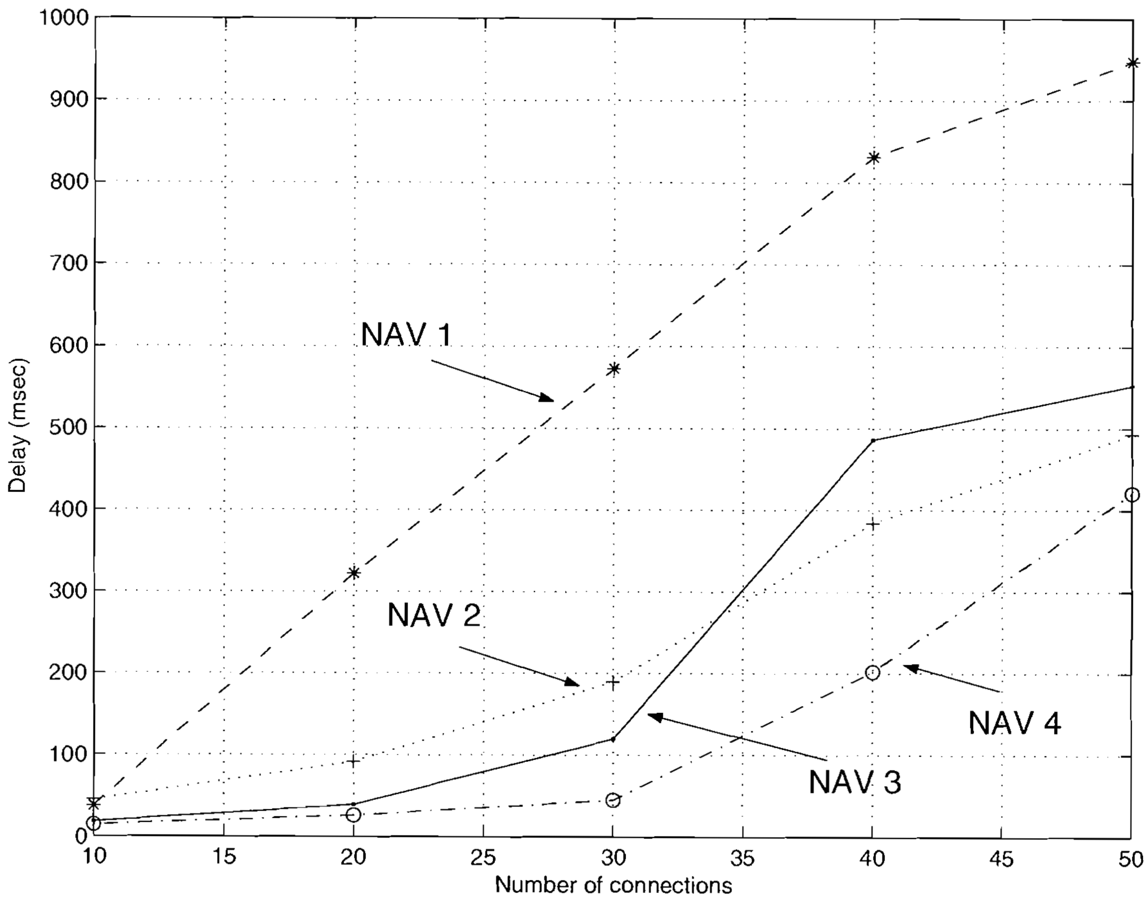

Fig. 9. Delay for different duration time estimation schemes.

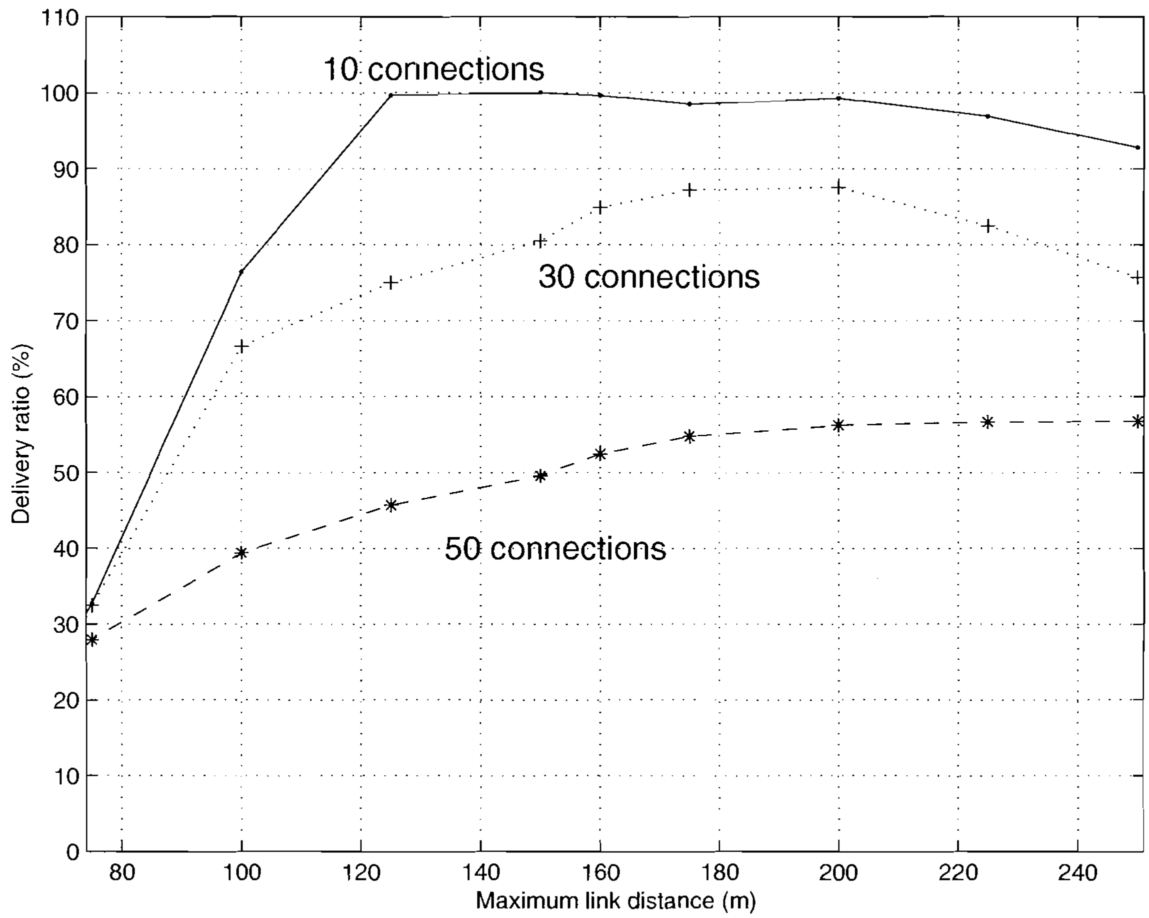

Fig. 10. Delivery ratio vs. link distance.

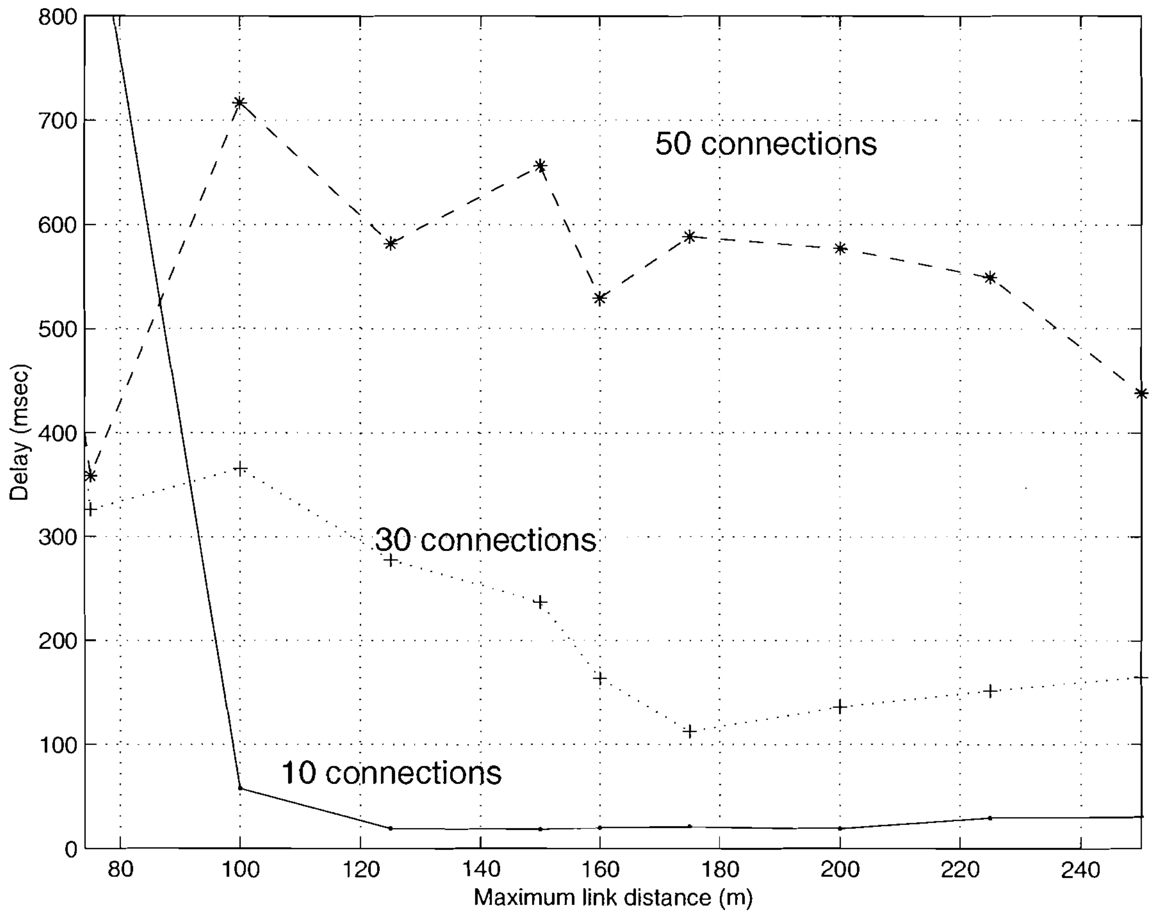

Fig. 11. Delay vs. link distance.

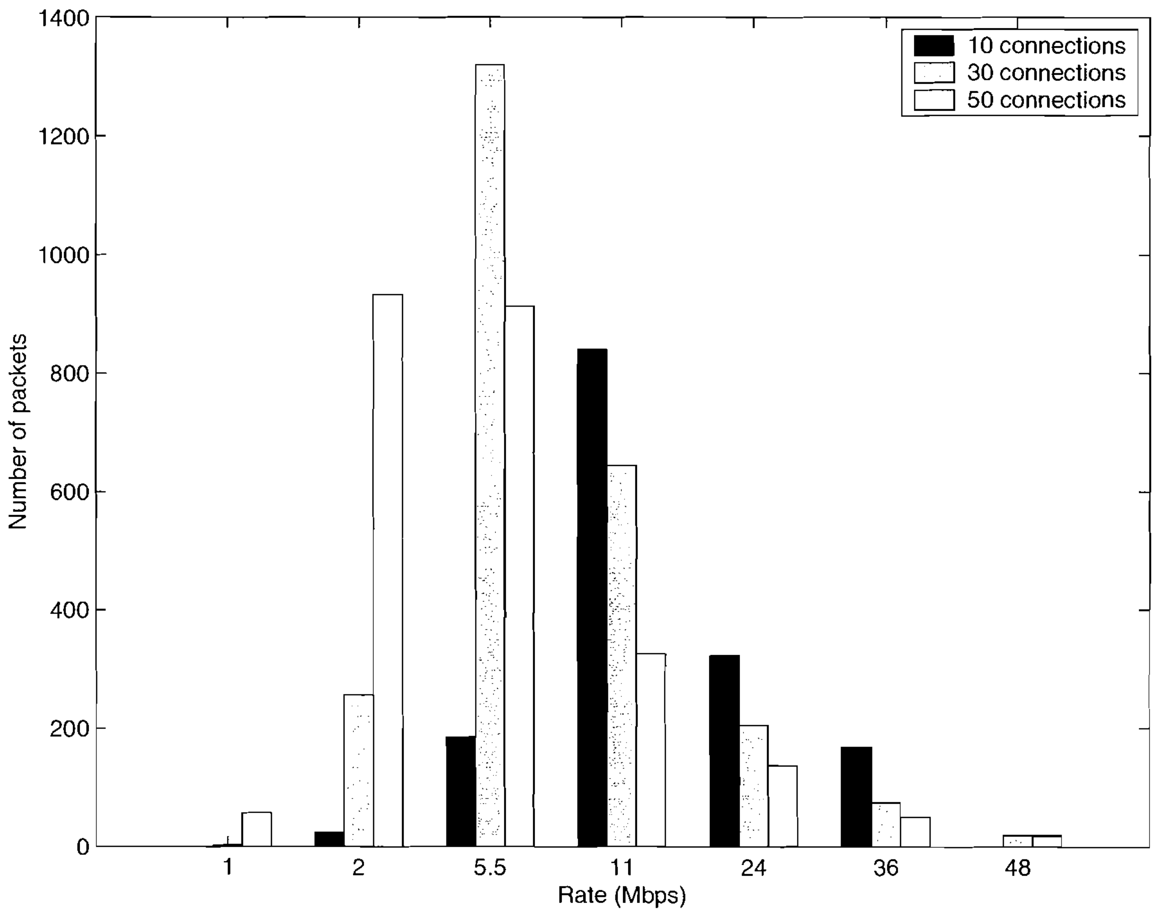

Fig. 12. Histogram of applied data rate.

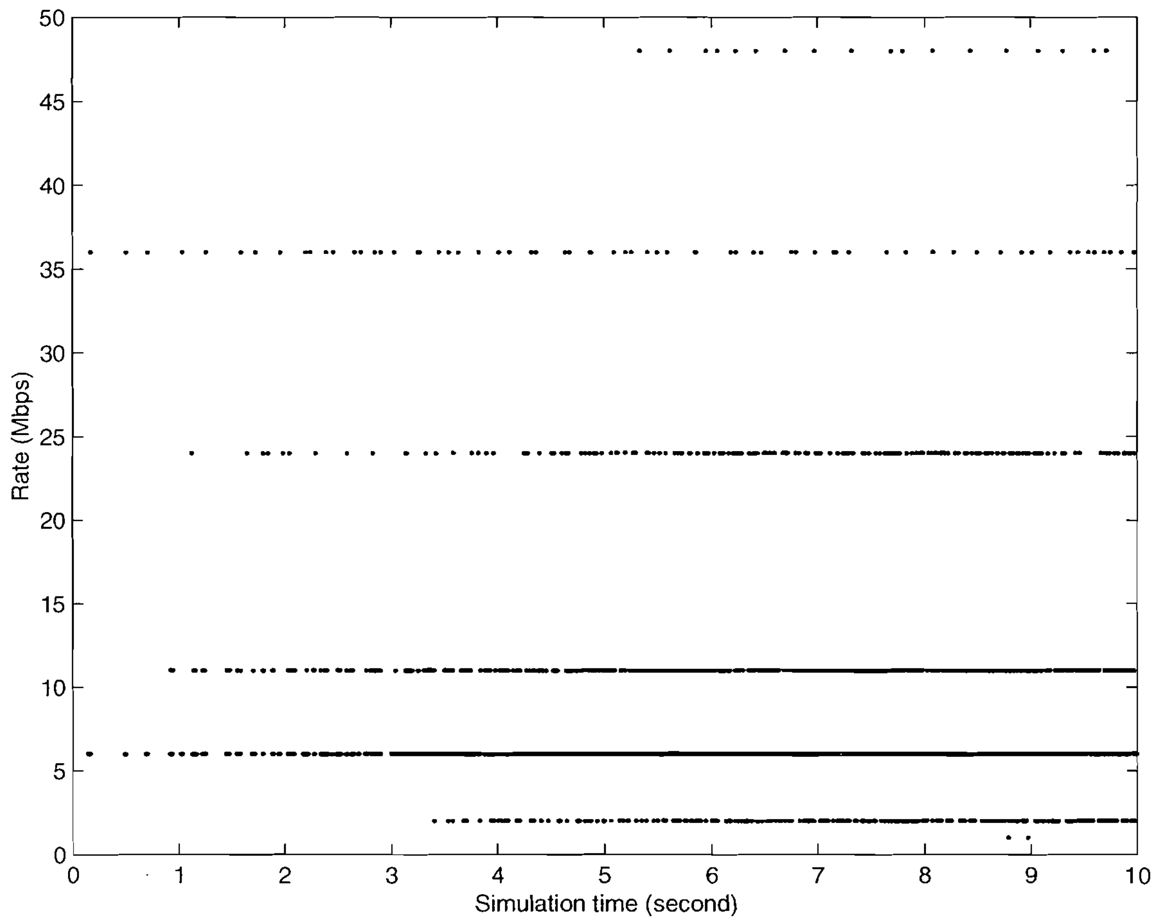

Fig. 13. Data rate in a 10-second run.

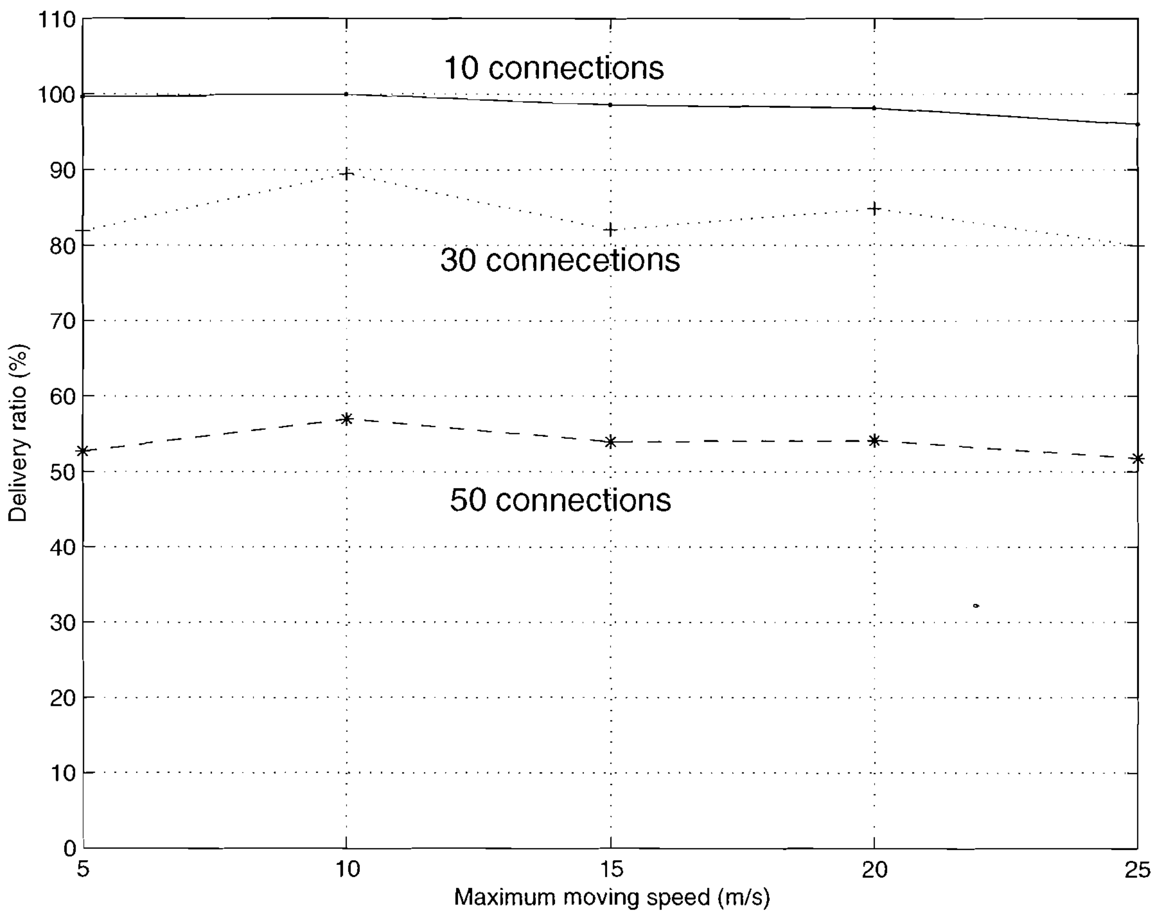

Fig. 14. Delivery ratio vs. node mobility.

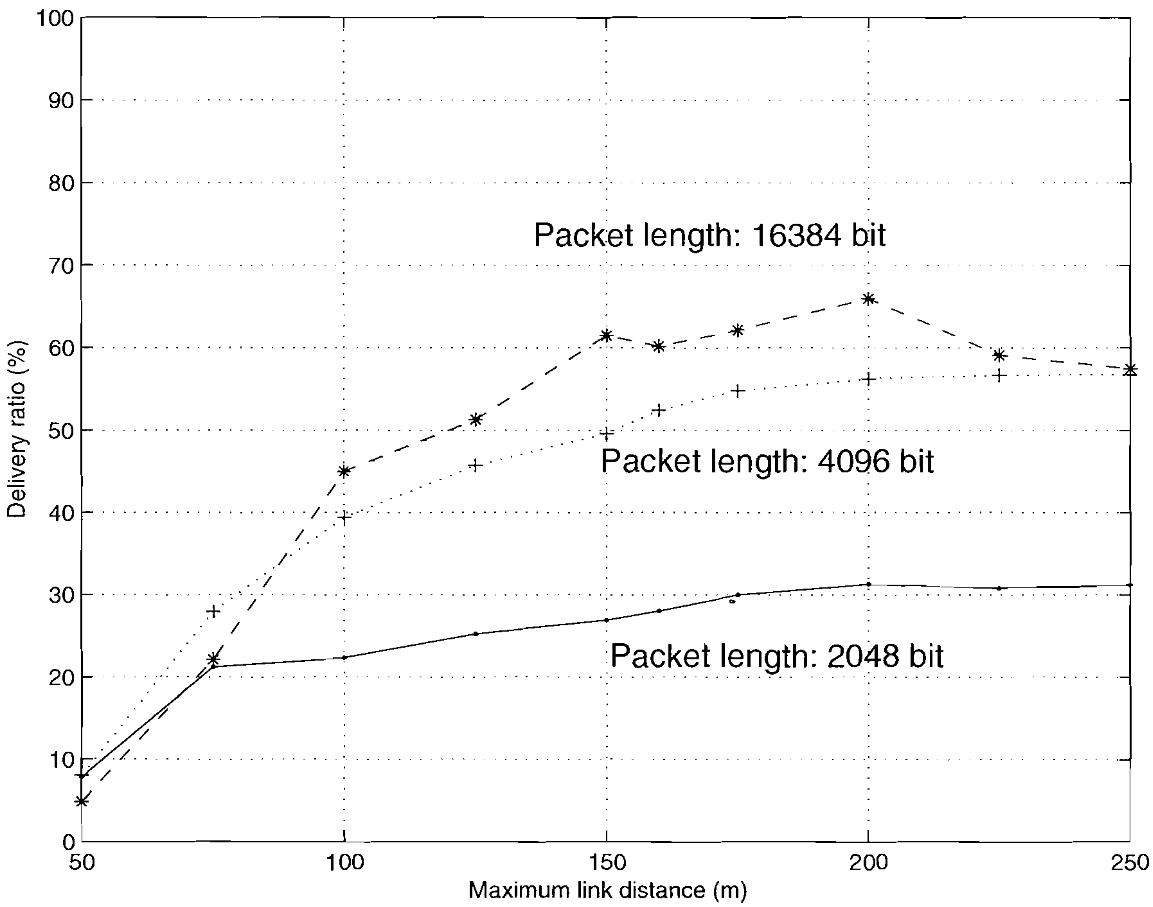

Fig. 15. Delivery ratio vs. data packet size.

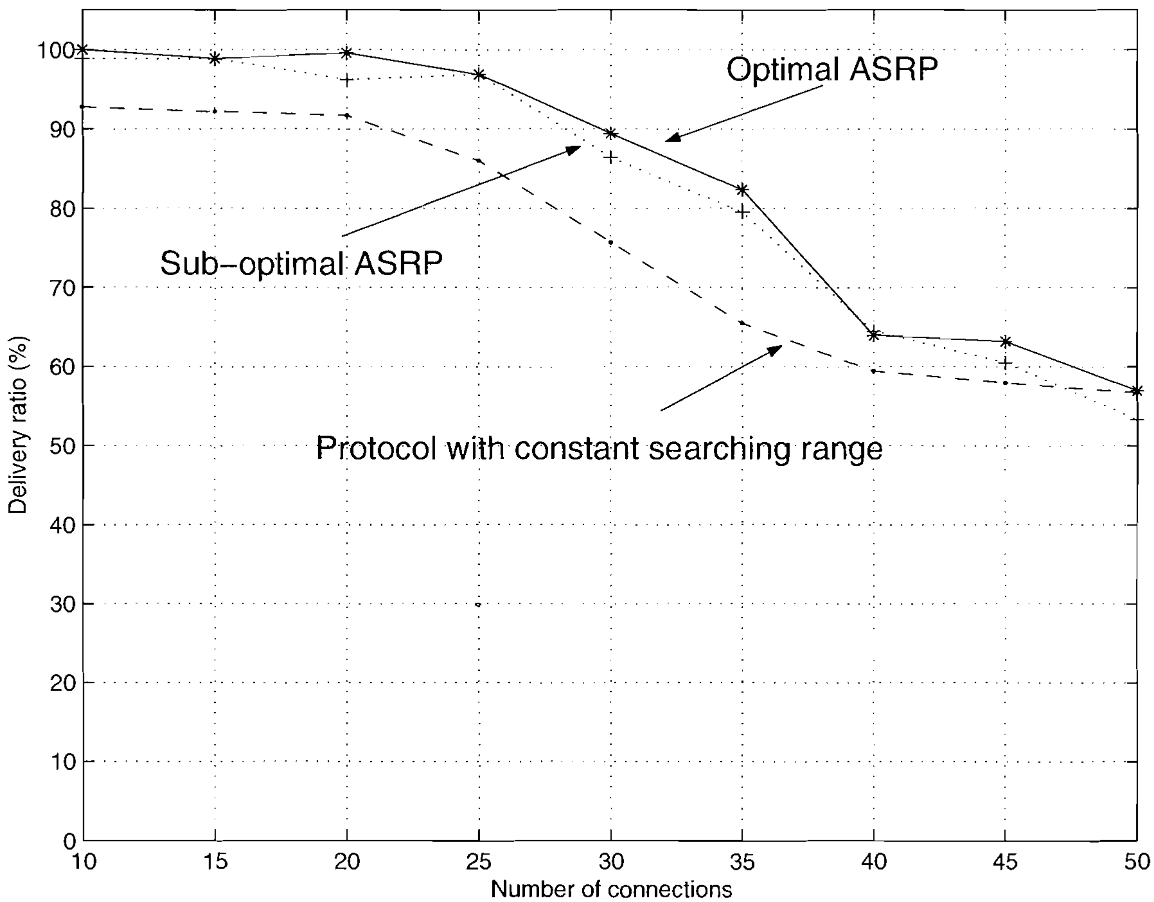

Fig. 16. Delivery ratio for optimum, sub-optimum, and normal schemes.