Appl. Math. Inf. Sci. 6 No. 1S pp. 271S-283S (2012)

Applied Mathematics & Information Sciences
An International Journal

A Queuing Network Model Based on Ad Hoc Routing Networks for Multimedia Communications

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Received: Received May 02, 2011; Revised July 25, 2011; Accepted September 12, 2011
Published online: 1 January 2012

Abstract: In real-time multimedia applications, the delivery of multimedia information over ad hoc wireless networks has presented difficult challenges requiring considerable research efforts to overcome. To analyze the delivering multimedia packets between mobile nodes with low end-to-end delay and less bandwidth overhead while ensuring high throughput, we propose a queuing network model based on our adaptive-gossip algorithm with probability p_n that conserves network bandwidth at each node by reducing the routing overhead. We also analyze the queuing delay in regard to the number of nodes, the transmission range of a node, the behavior of routing, and MAC protocols. We present both analytical and experimental results to thoroughly evaluate our proposed queuing network model, which demonstrates the advantages of an adaptive-gossiping routing method over flooding routing.

Keywords: Ad hoc wireless networks; Adaptive-gossip routing; Queuing networks; Modeling and performance evaluation; Multimedia communications

1 Introduction

An ad hoc wireless mobile network consists of a set of mobile nodes connected by shared wireless channels to form an arbitrary topology without the use of any established infrastructure or centralized control. In such an environment, each node performs as a host and a router, and forwards packets to other neighbor nodes by discovering multiple hop paths. Thus, this type of network is a useful design for low-cost networking and quick-deployed networking where any fixed network infrastructures such as base stations are not feasible. Advances in wireless local-area networks (WLANs) based on IEEE 802.11 technologies and growing mobile device uses allow consideration of ad hoc wireless networks for delivery of multimedia information services such as emergency response, search and rescue, group communications, etc.

However, there are many challenges for providing real-time transmission of multimedia information (including voice and video packets) from source to destination over low-bandwidth ad hoc wireless networks with limited resources and under dynamic topology. As all nodes are mobile and packets are broadcasting over the shared wireless channel with the limited transmission power of a node, packets may have to be forwarded by several intermediate nodes before they reach their destinations, if the destination nodes are not directly in their transmission ranges. Moreover, packets may have to be retransmitted several times if mobile nodes leave or join the transmission range, which causes significant end-to-end delay and a high level of packet losses that can deteriorate the network performance with fewer throughputs for providing real-time multimedia applications.

Thus, ad hoc nodes should be deployed densely to maintain a high degree of interaction between mobile nodes because of their limited transmission power, and many ad hoc routing protocols have been developed based on a simple flooding routing

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method of periodically broadcasting routing packets to all other nodes to provide the shortest-path routing and achieve a high degree of availability to efficiently establish routes. However, there are some critical problems with flooding overhead causing broadcast storms [1] because many routing messages are propagated unnecessarily. Furthermore, there exists a high packet level of collisions due to rebroadcasting mobile nodes that are close to each other and periodical rebroadcasts. Therefore, routing protocols based on the use of flooding are inefficient since they require high bandwidth of multimedia communication over multi-hop wireless ad hoc networks.

In this paper, we propose a queuing network model that extends our research work in our previous work [2] with an analysis of queuing delay that is one of the most important parameters in the end-to-end delay. The queuing delay in ad hoc wireless networks depends on the number of nodes, the transmission range of a node, the network traffic pattern with the behavior of routing, and MAC protocols. Our primary purpose in this study is to determine how to deliver packets between mobile nodes with low end-to-end delay and less bandwidth overhead while ensuring high throughput in multimedia over ad hoc wireless mobile networks. To accomplish our goals, we have designed a routing protocol based on our adaptive-gossip algorithm that is a probabilistic broadcast mechanism [3].

Routing packets are broadcast with the adaptive-gossip probability p_n assigned according to the number of neighbor nodes n when routing packets are broadcasting. It is scalable because it can significantly reduce the communication overhead compared to flooding and other gossiping approaches for dense networks. Most of the published gossip-based routing protocols are static [4-6]; that is, all nodes have the same gossip probability p for all gossip packets during executions of the whole network, which is unnecessary. Therefore, our proposed queuing network model based on adaptive-gossip probability p_n can provide an analysis of low end-to-end delay compared to other broadcast mechanisms. Subsequently, the low end-to-end delay that results from applying the adaptive-gossiping algorithm will improve the network performances to achieve a high throughput.

Our analytical results are based on a queuing model with finite buffers since network resources are limited in ad hoc wireless networks. The results of reduced end-to-end delay can be obtained from a new analysis model of the packet arrival rate and the packet drop probability derived by our adaptive-gossip probability p_n. We use the IEEE 802.11b MAC-based WLAN into our queuing network model for the service rate of a node regarding low-cost and low-bandwidth networking. We also present the simulation results to prove that the adaptive-gossip based routing protocol outperforms and uses less network bandwidth compared to flooding-based routing for multimedia packets over ad hoc wireless networks.

2 Queuing System Model

This section presents a network model including flooding and adaptive-gossip based routings, the traffic model, and the queuing network model in an ad hoc wireless network.

2.1 The Network Model

The network denoted by $G^2(N, r_0(N))$ consists of N nodes $(1,2,\ldots,N)$ to form an arbitrary "*ad hoc*" network topology, such that the nodes are independently randomly placed on a two-dimensional area A. Each node is assumed to have the same transmission range, denoted by $r_0(N)$. Let r_{ij} denote the distance between nodes i and j. Nodes i and j are said to be neighbors if they can directly communicate with each other, that is if $r_{ij} \leq r_0(N)$. Hence, a circle of area $\pi r_0^2(N)$ is termed the "*communication area*" of a node. We assume the number of nodes that are neighbors of node i; they lie on the communication area of $\pi r_0^2(N)$ in the node deploy area of two-dimension A for $\pi r_0^2(N) \ll A$. We define that A is a rectangular area of size $a\times b$ for $a\geq b$, and the node density is $(1/A)N$. This network model is depicted in Fig.1.

2.1.1 Flooding-Based Ad Hoc Routing

Most ad hoc routing protocols utilize three types of control packets such as *request*, *reply* and *failure-request* for a route discovery and maintenance operation [8]. These routing control packets may be broadcast by means of different broadcasting mechanisms. Flooding-based routing does not require *a priori* knowledge of network topology; it simply broadcasts packets to all nodes. For example, to find a route to the destination node, the source node sends a request packet to all neighbors of a node. When a node receives the packet and it is not the destination, it simply rebroadcasts the packet once to all its neighbors within its transmission area of $\pi r_0^2(N)$. In a flooding mechanism, the expected number of neighbors of a node within

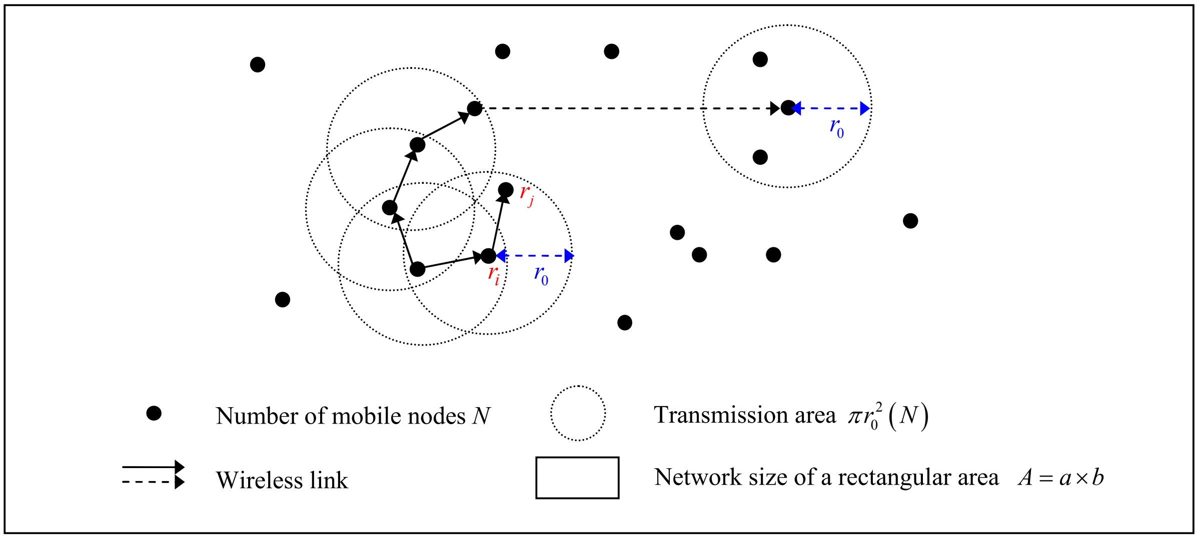


Fig.1 The model of network topology is an undirected random geometric graph. Nodes are randomly distributed in a network of size A=a×b and each node has the same radio transmission range r_0.

its transmission range $r_0(N)$ that receives the packet is given by

$$E(n) = \frac{\pi r_0^2(N)}{A} N = \log(N), \qquad (1)$$

where $\frac{\pi r_0^2(N)}{A} N$ is expressed as log(N) in Penrose's definition of a high connectivity [9]. Hence, we can obtain the critical transmission range of a node for flooding routing from (1), which is defined by

$$r_0(N) = \sqrt{\frac{A \log(N)}{\pi N}} . \qquad (2)$$

However, the flooding is unsuitable for frequently and rapidly changing networks that may require rediscovery of a route. Because mobile nodes are randomly and independently distributed, these nodes can send and receive packets nearly simultaneously to their neighbors without any information about available bandwidth or their buffer capacity, which results in collisions and retransmissions possibly leading to network overhead and collapse [8]. Moreover, in the large transmission range, $r_0(N)$ has bursts of flooding packets that can significantly increase both packet loss rate and end-to-end delay. Therefore, we present an ad hoc routing based on adaptive-gossiping that is described in the following section.

2.1.2 Adaptive-Gossip Based Ad Hoc Routing

We have proposed an adaptive-gossip routing algorithm to reduce the redundant routing packets broadcast throughout the ad hoc wireless network, which can save bandwidth usage by reducing network overall overhead. Our adaptive-gossip algorithm has been developed based on GOSSIP3 with gossip probability p by Hass et al. [4]. Their gossip probability p is static, assigned by selecting a fixed heuristic value (such as p=0.65). However, this static-gossip algorithm does not consider the number of neighboring nodes for a node to broadcast a message with p to its neighbors. For example, a node with too many neighbors could yield high overhead and collisions, while too few neighbors could result in network unreliability even with a good heuristic gossip probability p. Thus, this static value of the gossip probability p might not improve overall performances in an ad hoc wireless network where nodes are densely deployed with independent and random distribution.

We define an adaptive-gossip probability p_n for determining a gossip probability value that is an essential element of gossiping. The adaptive gossiping is a broadcasting mechanism whereby forwarding nodes are selected with a probability assigned by the number of neighbors of a node, instead of the gossiping with a static probability p and flooding that simply broadcasts to all of its neighbors as probability 1. As an example of a route discovery, when each node receives a request packet from a source node, it broadcasts the packet to its neighbors with probability $p_n = (1/n)\log(N)$ where n is its actual neighbor nodes and $\log(N)$ is the expected number of neighbors for $n > \log(N)$, and discards the packet with probability $1-p_n$. Thus, the expected number of neighbors of a node within its transmission range $r_0(N)$ that receives the packet in the adaptive-gossiping mechanism is given by

$$E(n) = np_n = \frac{\pi r_0^2(N)}{A} Np_n = \log(N). \qquad (3)$$

From (3), we can obtain the critical transmission range of a node in adaptive-gossiping with high connectivity, which is defined by

$$r_0(N) = \sqrt{\frac{A\log(N)}{\pi N p_n}}. \qquad (4)$$

To obtain the hop count for the distance L between a random source to destination pair, we adopted the result of the probability distribution of the distance between two random points derived by Ghosh [10], and the result from Bettstetter et al. [11] for the probability density function of the transition length L of nodes moving according to the random waypoint mobility model in a rectangular area of size $a \times b$ for $a \geq b$, and the expected distance within a rectangle of size of $a \times (a/2)$ yields $E(L)=0.402 \times a$ in [11]. Therefore, H denotes the number of hop counts and the expected hop count between random source-destination pairs is defined by

$$E(H) = \frac{E(L)}{r_0(N)}, \qquad (5)$$

where $r_0(N)$ can be used for a shortest one hop distance defined by (2) and (4) for flooding and adaptive-gossiping respectively. For the example of flooding, $r_0(N) \approx 85m$ with $N=100$ nodes in the network size of $A=a \times b = (1000 \times 500)m^2$, hence the $E(H)=(0.4020 \times a)/85 = (0.4020 \times 1000)/85 = 4$ hop counts between random source and destination pairs, and for adaptive-gossiping, $r_0(N) \approx 128m$ and the $E(H)=(0.4020 \times 1000)/128=3$ hop counts between random source and destination pairs. Therefore, the adaptive-gossiping results in fewer hop counts in small-sized networks and much fewer hop counts in large-sized networks than flooding routing algorithm (see Fig.2).

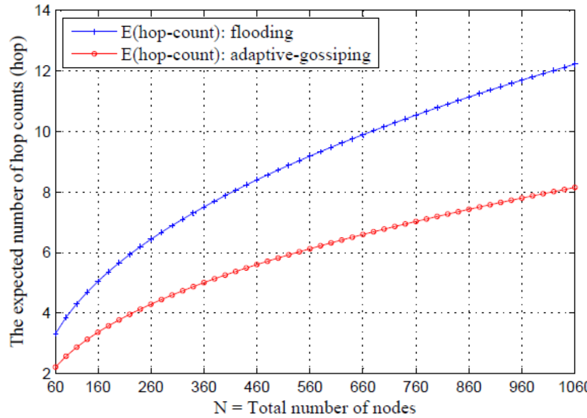


Fig.2 The expected hop count between a random source and destination pair. The analysis results are compared with flooding and adaptive-gossiping according to question (5) in a given network size of $(1000 \times 500)m^2$ with the total number of nodes from 60 to 1060.

2.2 The Traffic Model

The traffic model for ad hoc wireless networks based on adaptive-gossip routing is described as follows. Each node could be a source, destination and intermediate node. Each node generates packets with rate λ packets/s on average, where the packet generation process at each node is an independent and identically distributed Poisson process. Each data packet size is constant and equals L bits (e.g., 512 bytes × 8 bits) that is longer than other packets such as routing control packets (about 32 bytes). Before sending a data packet, a route from a source node to a destination node has to be established by a routing protocol using routing control packets. The priority scheduling policy used is that routing control packets get higher priority than data packets.

The route is connected by multi-hops between the source and the destination nodes. In this ad hoc wireless network, we implement the adaptive-gossip routing protocol to reduce the redundant routing packets, which can save network resources by reducing network overall overhead. The packet delivery scheme has the following simplifications. When a source has data to transmit to an unknown destination, it broadcasts a route request packet for that destination to its neighbors with adaptive-gossip probability p_n, hence np_n is the selected number of neighbor nodes to forward packets, and discards the request packet with probability $(1-p_n)$; thus the number of $n(1-p_n)$ neighbors do not forward packets. When each intermediate node receives the request packet from any of its neighbors, checks if the receiving node has not received this request packet before and is not the destination, it rebroadcasts the request packet with probability (1−*absorption probability*). If the receiving node is the destination, it absorbs the request packet and it generates a route reply packet. The reply packet is unicast in a hop-by-hop fashion to the source. As the reply packet propagates, each intermediate node creates a route to the destination. When the source receives the reply packet, it records the route to the destination and can begin sending data.

2.3 The Queuing Network Model

We develop a queuing network model for ad hoc wireless networks with their underlying multi-hop packet forwarding. The stations of the queuing network correspond to the nodes of the network. The forwarding probabilities in the queuing network, denoted by p_{ij}, correspond to the probability that a packet transmitted by node i enters node j's queue, and the p_{ij} can be defined as $p_{ij} = \frac{1}{N-1}\left(1 - absorption\ probability\right)$. We apply an $M/M/1/B$ queue to our queuing network model. Thus, we assume that each node has a finite buffer B, which means that packets are dropped when the buffer is full in the network. The packets are served by the nodes on a first-come first-serve (FCFS) basis. We assume that the number of hops is a geometric random variable. The parameters of the queuing network model can be expressed as the following Assumptions.

Assumption 1 *The absorption probability is the probability that a node is the destination of a packet that traverses through a number of hops from its source to the destination node, which is denoted by AP as*

$$AP = \frac{1}{The\ expected\ number\ of\ hop\ count} = \frac{1}{E(H)}, \tag{6}$$

where the $E(H)$ is the expected hop count between a random source and destination pair.

Assumption 2 *The forwarding probability that a packet is forwarded from node i to node j, which is denoted by p_{ij} as*

$$p_{ij} = \begin{cases} \frac{1}{N-1}\left(1 - AP\right) & i \neq j \\ 0 & i = j \end{cases} \tag{7}$$

The end-to-end delay in an ad hoc wireless network equals the sum of queuing, transmission and propagation delays from a source to a destination node including intermediation nodes. It is denoted by D that is accumulated as

$$D = D_Q + D_T + D_P. \tag{8}$$

The queuing delay D_Q is the sum of waiting time at a source node and intermediate nodes due to the route establishment and network congestion; the transmission delay D_T is the sum of time required to push all of the packet's bits into the link from a source to a destination node, such that if the length of a packet is denoted by L bits and the transmission rate of a link by R bits/s, then $D_T = L/R$ for one hop; and the propagation delay D_p is the sum of time required to propagate a packet on each link from a source to a destination node and hence $D_p = d/s$ for one hop, where d is the distance between node i and node j, and s is the propagation speed of the link. In our delay analysis, the propagation delay is about $1\mu s$, which is insignificant compared to other delays in the network system. The transmission delay is a constant value and is ignored in our delay analysis. Unlike the other two delays, the queuing delay can vary from the number of nodes, the transmission range of a node, the network traffic pattern with the behavior of routing, and MAC protocols. Therefore, we consider the queuing analysis to estimate accurately the average end-to-end delay in the next section.

3 Analysis of End-to-End Delay

3.1 Packet Service Rate

Arriving packets at the queue are forwarded to the next hop through the Medium Access Control (MAC)/Physical (PHY) layer. To find out the service rate of a node regarding low-cost and low-bandwidth networking of ad hoc wireless networks, we choose the IEEE 802.11b MAC, which is widely adopted in the WLAN and offers the medium-bandwidth wireless connectivity well-suited for a variety of traffic types, including multimedia distribution [12].

IEEE 802.11b MAC specifies a primary mechanism called Distributed Coordination Function (DCF) to access the medium. DCF uses the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) scheme to transmit data packets using the Request-To-Send (RTS) and Clear-To-Send (CTS) access method [13]. In this random access MAC model, a node transmits its packet if it senses the channel idle for a period of Distributed Inter Frame Space (DIFS). If the channel is sensed busy, the node defers its transmission until an idle DIFS is detected and then generates a random backoff interval before transmitting. The value of contention window CW depends on the number of failed transmissions for a packet. It is initially set to $CW_{\min}$ for each packet transmitted successfully. After an unsuccessful transmission m times, the CW is doubled up to a maximum value such as $CW_{\max}=2^m CW_{\min}$. The backoff time counter is decreased by one at each time slot as long as the channel is sensed idle. It is stopped when the channel is busy and resumes when the channel is sensed idle again for more than DIFS. With the RTS/CTS mechanism, a node transmits a short RTS packet first instead of the data packet when its backoff timer reaches zero. The receiving node responds with a CTS packet after a Short Inter Frame Space (SIFS) time interval. The sender is allowed to transmit the data packet only if it receives a valid CTS. The receiver transmits an ACK when the data packet is successfully received. If the transmitting node does not receive an ACK, the data packet is assumed to have been lost and a retransmission is scheduled.

To identify an average packet service rate μ, we use the models by Bianchi [14] and Carvalho et al. [15] to determine an average virtual slot. Let T_s denote the average time the channel is sensed busy due to a successful transmission, and denote by T_c the average time the channel is sensed busy due to a collision. In [15], the packet transmission time T_s is given by

$$T_s = \frac{RTS}{R} + \frac{CTS}{R} + 3SIFS + \frac{ACK}{R} + \frac{L}{R} + DIFS \tag{9}$$

and the packet collision time T_c is given by

$$T_c = \frac{RTS}{R} + DIFS , \tag{10}$$

where the L (bits) is the data packet size and the R (bits/s) is the transmission rate of a link. Assume that a fixed duration of an empty slot time denoted by δ is given the value $20\mu s$. Denote by P_{tr} the probability that at least one transmission is in the considered slot time, and denote by P_s the probability that the channel at a given node has a successful transmission. Thus in [14], the probability P_{tr} that N nodes contend in the channel and that each transmits with probability τ is given by

$$P_{tr} = 1-\left(1-\tau\right)^N . \tag{11}$$

The probability P_s that exactly one node transmits packets over the channel, under the condition that at least one node transmits, is given by

$$P_s = \frac{N\tau\left(1-\tau\right)^{N-1}}{P_{tr}} = \frac{N\tau\left(1-\tau\right)^{N-1}}{1-\left(1-\tau\right)^N} , \tag{12}$$

where τ is obtained by providing two equations. First, the probability τ that a node attempts a transmission in a randomly chosen slot time, which is given by

$$\tau = \frac{2(1-2p)}{(1-2p)(CW_{\min}+1) + pCW_{\min}(1-(2p)^m)} , \tag{13}$$

where p is a function of the conditional collision probability. Second, the probability p that a transmitted packet encounters a collision is the probability that at least one of the N–1 remaining nodes transmits in the same time slot. If all nodes transmit with probability τ, the collision probability p is given by

$$p = 1-\left(1-\tau\right)^{N-1} . \tag{14}$$

The p solved using numerical techniques for the two unknowns of τ and p from (13) and (14) respectively in [15] is given by

$$p = \frac{2CW_{\min}(N-1)}{(CW_{\min}+1)^2 + 2CW_{\min}(N-1)} . \tag{15}$$

In [15], the average length of a slot time in IEEE 802.11b single-hop ad-hoc wireless networks is

$$E\left(slot\right) = \left(1-P_{tr}\right)\delta + P_{tr}P_sT_s + P_{tr}\left(1-P_s\right)T_c . \tag{16}$$

Eventually, we can obtain that the packet service rate μ for all nodes N randomly distributed in a network is

$$\mu = \frac{1}{E\left(slot\right)} . \tag{17}$$

3.2 Packet Arrival Rate

Mobile nodes are randomly and independently deployed in an ad hoc wireless network, which can be applied to the *Jackson network* [16] with the following assumptions. Packets from one node i proceed to an arbitrary node and new packets may join a node from outside. Suppose that there are N nodes, where the ith node (i=1,…,N) consists of a single server queue with an exponential service rate μ_i. Packets arrive at node i from the outside of the network according to Poisson processes with rate λ_i, and all arrival packets are independent of each other. Packets after receiving service at the ith node proceed to the jth node with forwarding probability p_{ij}.

Let us consider the ith node, where packets arrive at the queuing system of node i from the outside in accordance with independent Poisson processes at rate λ_i. In this network, λ_i is a packet generating rate at the application layer of node i. Also, packets arrive at this node from other nodes within the network; it is called internal arrivals. Let b_i represent the rate of internal arrivals at node i. Then, for each node i, the total arrival rate at node i is represented by a_i, given by

$$a_i = \lambda_i + b_i, \qquad 1 \le i \le N. \tag{18}$$

Now, if the service nodes are all stable, the departure rate of packets from the jth node will be same as the total arrival rate to node j, namely, a_j. A fraction p_{ij} of these departing packets go to node i. Hence the arrival rate of internal packets from node j to node i is $a_j p_{ji}$, so that $\sum_{j=1}^{N} p_{ji} \le 1$, and $1-\sum_{j=1}^{N} p_{ji}$ represent the probability that a packet departs the system after being served by node i. Thus, the internal arrival rate at node i from all the nodes in the network is given by

$$b_i = \sum_{j=1}^{N} a_j p_{ji} \qquad 1 \le i \le N. \tag{19}$$

Substituting in the previous equations, the total arrival rate of packets at node i is given by

$$a_i = \lambda_i + \sum_{j=1}^{N} a_j p_{ji} \qquad 1 \le i \le N. \tag{20}$$

Since p_{ij} is the forwarding probability from Assumption 2, node j must be not the destination, thus we have an additional condition for $j \neq i$,

$$a_i = \lambda_i + \sum_{\substack{j=1 \\ j\neq i}}^{N} a_j \frac{1}{N-1}\left(1-AP\right) \qquad 1 \le i \le N. \tag{21}$$

Since the absorption probability is $AP = \frac{1}{E(H)}$ from Assumption 1,

$$a_i = \lambda_i + \sum_{\substack{j=1 \\ j\neq i}}^{N} a_j \frac{1}{N-1}\left(1-\frac{1}{E(H)}\right) \qquad 1 \le i \le N, \tag{22}$$

where $\frac{1}{N-1}$ is the probability of packets forwarded to node j given node i is not a destination and where $\left(1-\frac{1}{E(H)}\right)$ is the probability that node i is not the destination to pick randomly in any nodes.

The end-to-end delay equals the sum of queuing and transmission delays at source and intermediate nodes in an ad hoc wireless network. In reality, there is always a limited system capacity due to a finite buffer size B, in the sense that there can be no more than B packets in the system of a node at any time. If an arriving packet finds that there are already B packets present, then it does not enter the queue and is lost. Hence, let P_k for $0 \le k \le B$, denote the probability that there are k packets in the queue by the formula definition of a single-server exponential queuing system having finite capacity [19],

$$P_k = \frac{\left(\frac{a}{\mu}\right)^k}{\sum_{k=0}^{B}\left(\frac{a}{\mu}\right)^k} = \frac{\left(\frac{a}{\mu}\right)^k\left(1-\frac{a}{\mu}\right)}{1-\left(\frac{a}{\mu}\right)^{B+1}}. \tag{23}$$

The average number of packets in a queue with finite buffer size B at a node is given by

$$\bar{Q} = \sum_{k=0}^{B} kP_k = \frac{a\left[1+B\left(\frac{a}{\mu}\right)^{B+1} - \left(B+1\right)\left(\frac{a}{\mu}\right)^{B}\right]}{\left(\mu-a\right)\left(1-\left(\frac{a}{\mu}\right)^{B+1}\right)}. \tag{24}$$

By symmetry, the average queuing delays at all stations are the same, and the average queuing delay at a node is given by

$$\bar{W} = \frac{\bar{Q}}{a}. \tag{25}$$

Therefore, the expected queuing delay that a packet waiting for transmission in the entire network from source to destination is given by

$$E\left(D_Q\right) = \bar{W}E\left(H\right) = \frac{1}{a}\bar{Q}E\left(H\right), \tag{26}$$

where a is the arrival rate, $\bar{Q}$ is the mean number of packets in the queue of a node and $E(H)$ is the expected hop count between a random source and destination pair.

3.3 Packet Drop Probability

The probability of packet drop is denoted by $P(drop)$ when the queue of a node is full. Since each node has a single queue of the finite buffer size B, the queuing system has a capacity to hold B packets that are served in an FCFS fashion. Thus, an arriving packet that sees the queue full with probability P_B is dropped, represented as

$$P(drop) = P_B = \frac{\left(\frac{a}{\mu}\right)^B \left(1-\frac{a}{\mu}\right)}{1-\left(\frac{a}{\mu}\right)^{B+1}}. \quad (27)$$

Since we assume that the number of hop counts H between a random source and destination pair is a geometric random variable with expectation of $E(H)$, it follows that the parameter p of this geometric random variable is $\frac{1}{E(H)}$. If the number of hop counts H equals to k, it is necessary and sufficient that the first k–1 trials is failures and the kth trial a success with parameter p, expressed as

$$P(H=k) = (1-p)^{k-1}\, p,\; k = 1,2,\cdots. \quad (28)$$

We consider the probability that there are no packet drops at any node between a source and a destination in the network as

$$\overline{P}(no\ drop) = E\left[\prod_{i=1}^{H}(1-P_i)\right] = \sum_{k=1}^{\infty}(1-P_i)^k\, P(H=k) \quad (29)$$

$$= \sum_{k=1}^{\infty}\left(1-P(drop)\right)^k \left(1-\frac{1}{E(H)}\right)^{k-1} \frac{1}{E(H)} \quad (30)$$

$$= \frac{1-P(drop)}{E(H)P(drop)-P(drop)+1}. \quad (31)$$

Hence, the expected end-to-end delay

$$E(D) = \left[E(D_Q) + D_T E(H) + D_P E(H)\right]\frac{1}{\overline{P}(no\ drop)} \quad (32)$$

where again $E(D_Q)$ is the queuing delay; $D_T E(H) = \frac{L}{R}E(H)$ is the transmission delay; and $D_P E(H) = \frac{d}{s}E(H)$ is the propagation delay from source to destination.

Parameter	Value
Packet size	32 (voice) and 512 (video) bytes
PHY header	34 bytes
RTS	20 bytes
CTS	14 bytes
ACK	14 bytes
Slot Time, δ	20 μs
SIFS	10 μs
DIFS	50 μs
ACK-Timeout	212 μs
CTS-Timeout	348 μs
Initial backoff window (CW_{min})	32
Max backoff window (CW_{max})	1024 ($CW_{max}=2^m \times CW_{min}$)
Backoff stages, m	7
Max channel bit rate (Bandwidth)	11.0 Mbit/s
Propagation delay	1 μs

Table.1 Physical Layer Parameters: Protocol and channel parameters are specified by the IEEE 802.11 DCF standard with Direct-Sequence Spread Spectrum (DSSS) [12].

4 Performance Evaluations

To compare the performance of ad hoc routing protocols based on two different broadcasting mechanisms with flooding and adaptive-gossiping, we choose the On-Demand Distance Vector Routing protocol (AODV) [17] to implement the adaptive-gossiping algorithm. It is designed to use bandwidth efficiently and to be capable of supporting large populations of nodes in dynamically changing networks. According to our previous study [18]: reactive (on-demand) ad hoc routing protocols as AODV are better able to reduce routing overheads than proactive protocols, and the simulation results showed that AODV outperforms others at high mobility in a large network.

To evaluate the performance of each broadcasting method, our queuing system model was implemented using Matlab-8(b) and simulated using the ns-2 network simulator with version ns-2.32 [22]. To demonstrate that our adaptive-gossip based routing protocol outperforms the flooding-based routing protocol for multimedia applications, we tested voice/video packets that are transmitted over the IEEE 802.11b DCF, which is the most commonly used wireless medium with maximum bandwidth capacity limited to 11 Mbps (see Table 1). Therefore, we present the performance results in an ad hoc routing of multimedia over the low-bandwidth of large dense networks.

4.1 Analytical Results

For numerical analysis using Matlab, the queuing system was performed under the following settings. The network topology consists of N nodes (between 100 and 1000 nodes) distributed randomly over the two-dimensional area of $(1000\times500)m^2$. Since ad hoc routing protocols are designed effectively for providing the shortest-path routing, we assume that the critical transmission range $r_0(N)$ is a one-hop distance for direct communications without intermediate nodes, such that a one-hop distance of $r_0(N)=\sqrt{\frac{A\log(N)}{\pi N}}$ and $r_0(N)=\sqrt{\frac{A\log(N)}{\pi N p_n}}$ for flooding from (2) and adaptive-gossiping from (4) respectively. Each node generates packets of size L=32 bytes for voice traffic and size L=512 bytes for video traffic at the rate of λ packets/s increasing from 1 to 100 packets per second. The transmission rate of each node is $R=11\times10^6$ bits/s. Other basic parameters are presented in Table 1.

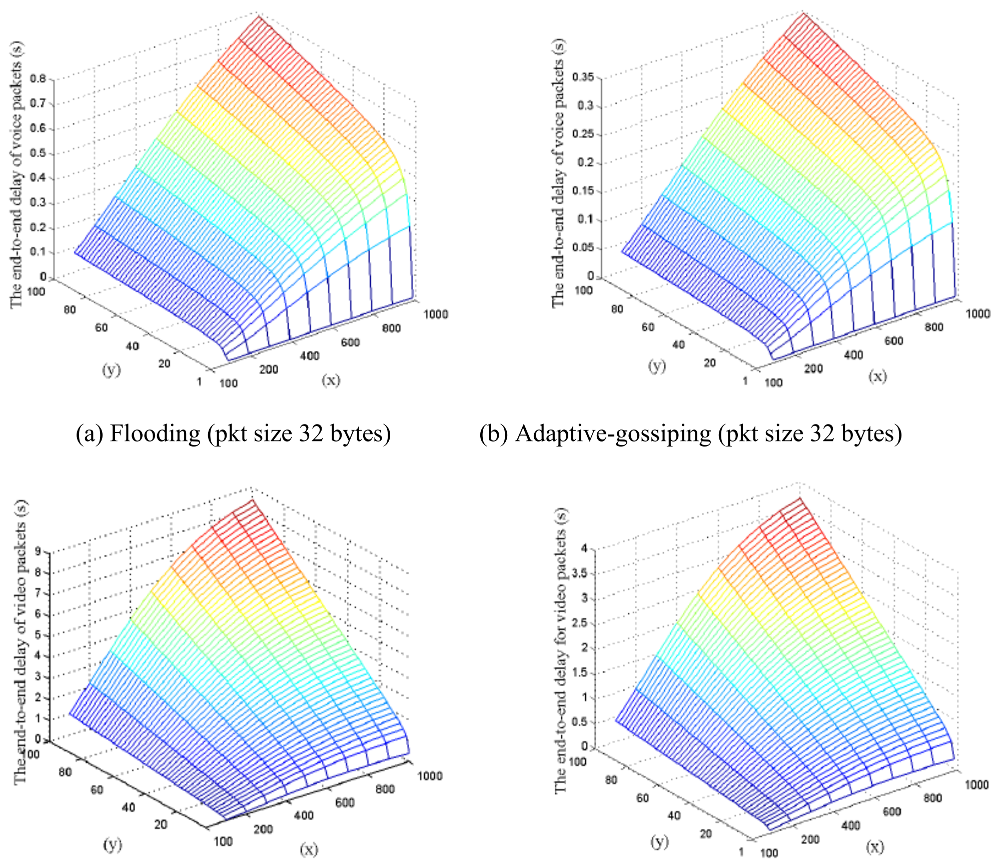
Fig.3 shows the expected end-to-end delay, $E(D)$, between a random source and destination pair, comparing flooding and adaptive-gossip algorithms according to equation (32). First, the analysis results are presented for voice traffic with packet size of 32 bytes; the $E(D)$ is between $5\times10^{-4}s$ and $0.78s$ for flooding in (a) and the $E(D)$ is between $3\times10^{-4}s$ and $0.34s$ for adaptive-gossiping in (b) at points between 100 nodes with λ=1 representing a small low-traffic network and 1000 nodes with λ=100 representing a large high-traffic network. Our adaptive-gossiping takes about 40% less delay than flooding at the small low-traffic network and about 56% less delay than flooding at the large high-traffic network. According to the quality of service (QoS) requirements for voice/video in Cisco's articles [21], $0.15s$ of end-to-end one-way delay does not impact a perceivable degradation in voice quality. Thus, our adaptive-gossiping outperforms $E(D)\leq0.15s$ when nodes are up to 500 with λ between 1 and 100 packet/s, while flooding does less delay within $0.15s$ only up to 200 nodes with λ between 1 and 100 packet/s. Second, the analysis results of video traffic with packet size of 512 bytes are presented; the $E(D)$ is between $3\times10^{-2}s$ and $8.33s$ for flooding in (c), and the $E(D)$ is between $9\times10^{-3}s$ and $3.65s$ for adaptive-gossiping in (d). Our adaptive-gossiping has significantly less delay, about 70%, than flooding at the small low-traffic network, and about 56% less delay than flooding at the large high-traffic network for a large packet size such as a video traffic. Therefore, our adaptive-gossip routing algorithm can be sufficient to satisfy QoS requirements for voice/video communications compared with flooding routing algorithm.

4.2 Simulation Results

For actual simulation tests, we implemented ad hoc routing protocols with flooding-based and adaptive-gossip based routing methods over NS simulator. These routing protocols are simulated with two types of multimedia application traffics: voice and video packets transmitted over the IEEE 802.11b channel. The multimedia traffics are CBR (constant bit rate) with voice packet size of 32 bytes and video packet size of 512 bytes. The source-destination pairs are distributed randomly in the network between 100 nodes modeling a small network and 1000 nodes modeling a large network. There are maximum source-destination pairs of 30 nodes with a sending rate of 1 packet per second. All nodes are mobile with the speeds of nodes randomly distributed between 0 to $2m/s$ in the two-dimensional area of $(1000\times500)m^2$.

For performance evaluations of the ad hoc routing protocols, we considered the quality of service (QoS) that most includes bandwidth and delay management; it is a set of service requirements to be met by the network while transmitting a packet flow from source to destination [26]. However, in the dynamic topology of an ad wireless hoc network, it should be considered a soft QoS rather than a hard QoS that guarantees quality of services such as packet delay or packet delivery ratio during a session holding time. Thus, our performance evaluations were based on the four performance metrics regarding QoS requirements [21] as follow.

- *Packet drop fraction*: The ratio of data packets dropped to the destinations to those generated by the CBR sources. (Voice/Video packet loss ratio ≤ 1%).
- *Average end-to-end delay*: This includes all possible delays caused by buffering during route discovery latency, queuing at the interface queue, retransmission delays at the MAC layer, propagation time, and transfer time. (Voice/Video traffic latency ≤ $0.15s$).



(a) Flooding (pkt size 32 bytes) (b) Adaptive-gossiping (pkt size 32 bytes)

(c) Flooding (pkt size 512 bytes) (d) Adaptive-gossiping (pkt size 512 bytes)

Fig.3 The expected end-to-end delay: $E(D)$ is between a random source and destination pair. The analysis results are compared with flooding and adaptive-gossiping for 32 and 512 bytes for voice and video data respectively. Where (x) is the total number of nodes from 100 to 1000 and (y) is the packet generation rate per second from 1 to 100 in a given network size of $(1000\times500)m^2$.

- *Normalized routing load*: The number of routing packets transmitted per data packet delivered at the destination, which is an important metric for evaluating less routing overhead.
- *Throughput*: The total number of delivered data packets divided by the total duration of simulation time, which gives the fraction of the channel capacity used for useful transmission.

Fig.4 shows the simulation results of multimedia communications with packet size of 32 bytes for voice traffic and 512 bytes for video traffic according to the above performance metrics. The packet drop fractions (PDFs) are presented between 100 and 1000 nodes in (a). First, the comparison of voice packet size of 32 bytes: the PDF of adaptive-gossiping is about 0.3% to 1.9% from 100 to 1000 nodes and PDFs $\leq$ 1% up to 700 nodes; however, the PDF of flooding is about 1.1% to 21.3% from 100 to 1000 nodes and PDFs $\leq$ 1% only at 100 nodes. Our adaptive-gossip routing method has significantly lower PDFs than a flooding-based routing method for voice traffic. Next, for the comparison of video packet size of 512 bytes: the PDFs of our adaptive-gossip routing method are about 0.6% to 8.6% from 100 to 1000 nodes and PDFs $\leq$ 1% up to 500 nodes, but the PDFs of flooding are about 1.1% to 33.2% from 100 to 1000 nodes and for PDFs $\leq$ 1% only at 100 nodes, similar to voice traffic. Our adaptive-gossip routing method has about 90% lower PDFs for voice traffic and 73% lower PDFs for video traffic compared to flooding routing methods, especially in the large network at 1000 nodes. The average end-to-end delays are presented in Fig.4 (b). Our adaptive-gossip routing method has significantly less delay, about 90%, as compared to flooding for both voice and video traffics between 100 and 1000 nodes. Thus, our routing method has the delay ≤ 0.15*s* at all network sizes from small to large networks as 100 to 1000 nodes for both voice and video traffic. However, flooding has the delay ≤ 0.15*s* only at 100 nodes for both voice and video traffic. Fig.4 (c) shows the throughputs; our adaptive-gossip routing algorithm outperforms flooding at all network sizes. Our proposed algorithm has high throughputs fairly consistently between 99% and 96% for voice traffic and between 99% and 91% for video traffic from 100 to 1000 nodes. However, the flooding method has low throughputs, about 70%, especially in a large network at 1000 nodes for both voice and video traffics. The normalized routing loads are presented in (d). Our adaptive gossip routing also outperforms flooding - about 50% less routing overheads for both voice and video traffics - over all networks. Hence, we can conclude that an ad hoc routing protocol based on adaptive-gossip routing algorithm may be the most efficient routing protocol in large-scale networks for multimedia communications.

5 Related Works

The most related work to our study is the queuing network models for delay performance of a multi-hop wireless ad hoc network studied by N. Bisnik et al. [23] using diffusion approximation to estimate the average end-to-end delay. However, their approach was more theoretical work and their assumption was that the queuing system has infinite buffers; however, each node in a real-system of ad hoc wireless networks has a limited buffer, which make us evaluate the packet drop probability. For a study of real-time streaming media issues in ad hoc wireless network, R. Ell-Khoury et al. [24] proposed a cross-layer scheme of ad hoc network for improving the end-to-end delay of real-time traffics by decreasing the packets that arrive after their schedule deadline; they focused on theoretical work. Other research work relevant to our study for conserving network resources can be found in the area of designing energy-efficient sensor networks for reducing the power consumption by setting unused sensors to idle, by H. Jabbar et al. [25]. However, those efforts were focused mostly on architectural and system features for improving performance of sensor networks. Therefore, in this paper, we try to provide an analytical model to prove that our adaptive-gossip routing protocol outperforms other approaches such as that of a flooding-based routing protocol.

6 Conclusions

We propose the queuing network model based on our adaptive-gossip routing algorithm to analyze low end-to-end delay and less bandwidth overhead with maximum achievable throughput for multimedia communications in an ad hoc wireless network. We evaluate the performances of ad hoc routing protocols based on our adaptive-gossiping algorithm compared to a flooding algorithm by using the proposed queuing network model. The principal method in our queuing network model is the absorption probability defined by the critical transmission range of adaptive-gossiping for a high connectivity with probability p_n, which improves routing performance due to fewer hop counts between source and destination nodes as compared to flooding-based routing. The results of both analysis and simulation tests are based on the IEEE 802.11b due to the limited resources and low-cost networking of ad hoc wireless networks. The most important observation of our queuing analysis results with adaptive-gossip probability p_n is the reduced routing overheads that can provide a significantly lower end-to-end delay and less packet loss compared to the flooding-routing problems. All performance metrics suggest that our adaptive-gossip routing protocol has high throughput compared to flooding, especially in large-scale networks.

Acknowledgements

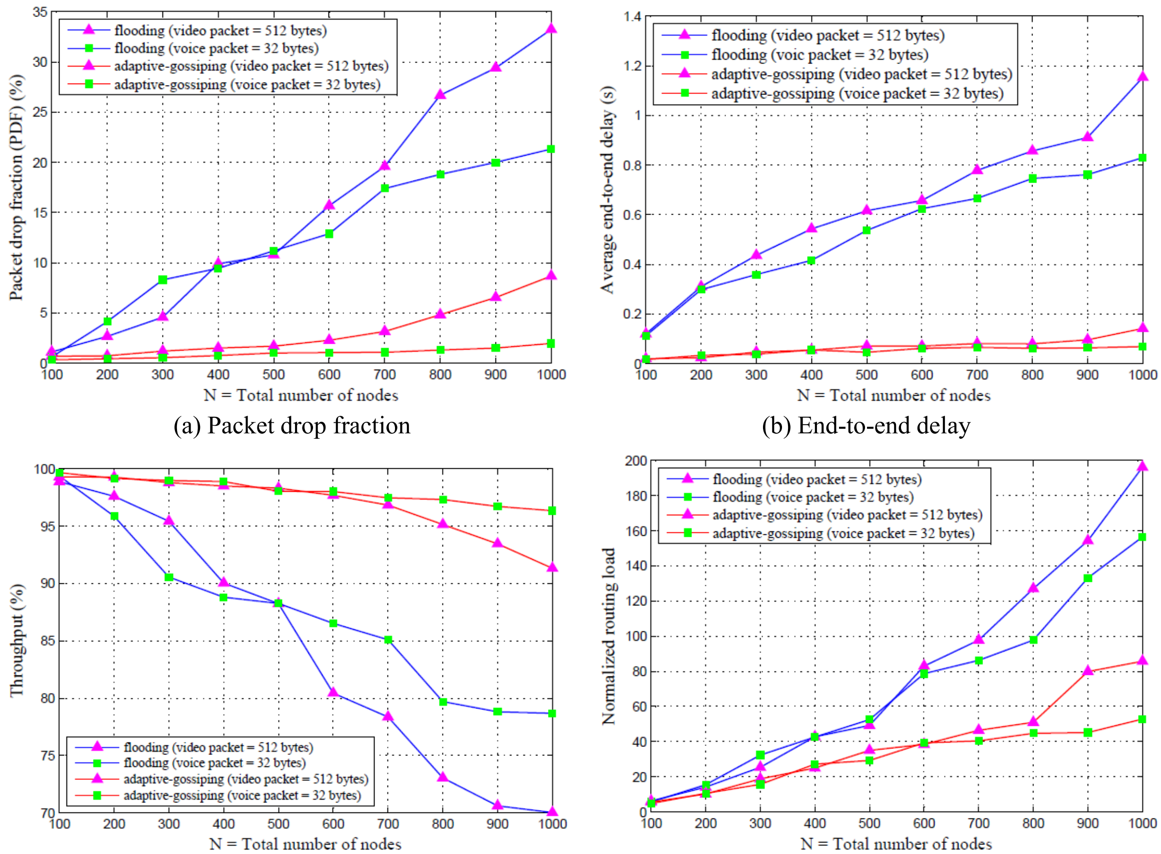
The authors would like to thank Anatolii A. Puhalskii, an associate professor of mathematics at the University of Colorado Denver, for explaining the relevant analysis of queuing delay based on adaptive-gossiping routing algorithm.

References

[1] S.-Y. Ni, Y-C. Tseng, Y.-S. Chen, and J.-P. Sheu. The Broadcast Storm Problem in a Mobile Ad Hoc Network. In *Proc.* of(MobiCom'99,) 151-162, (1999).

[2] A. Lee and I. Ra, Adaptive-Gossiping for An Energy-Aware Routing Protocol in Wireless Sensor Networks. In Proc. (IWCMC10), pp. 1131-1135, (2010).

[3] P.Th. Eugster, R. Guerraoui, S.B. Handurukande, P. Kouznetsov, and A.-M. Kermarrec. Lightweight Probabilistic Broadcast. *ACM Transactions on Computer Systems*, 21(4):341-374, (2003).



(a) Packet drop fraction (b) End-to-end delay

(c) Throughput (d) Normalized routing load

Fig.4 The simulation results of multimedia communications with packet sizes of 32 and 512 bytes for voice and video data, respectively. The comparison of flooding and adaptive-gossiping is between 100 and 1000 nodes with 1 packet sent per second in a given network size of $(1000\times500)m^2$.

[4] Z. Hass, J.Y. Halpern, and L. Li. Gossip-Based Ad Hoc Routing. *IEEE/ACM Transaction on Networking*, 14(3):479-491, (2006).

[5] A.-M. Kermarrec and M.V. Steen. Gossiping in Distributed Systems. *ACM Operating System Review*, 41(5):2-7, (2007).

[6] R. Chandra, V. Ramasubramanian, and K.P. Birman. Anonymous Gossip: Improving Multicast Reliability in Mobile Ad Hoc Networks. In *Proc.* of (ICDCS01), 275-283, (2001).

[7] B. Bollobas. *Modern Graph Theory*. Springer, (1998).

[8] M.S. Corson and A. Ephremides. A distributed routing algorithm for mobile wireless networks. *ACM/Baltzer Wireless Networks*, 1:61-81, (1995).

[9] M. Penrose. *Random Geometric Graphs*. Oxford University Press, (2003).

[10] B. Ghosh. Random distances within a rectangle and between two rectangles. Bulletin of the Calcutta Mathematical Society, 43:17-24, (1951).

[11] C. Bettstetter, H. Hartenstein, and X. Perez-Costa. Stochastic Properties of the Random Waypoint Mobility Model. *Wireless Networks*, 10(5):555-567, (2004).

[12] IEEE standard for Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specif cations, IEEE Std: 802.11b-1999/Cor 1-2001, (2001).

[13] P. Raptis, A. Banchs, V. Vitsas, K. Paparrizor, and P. Chatzimisios. Delay Distribution Analysis of the RTS/CTS mechanism of IEEE 802.11. In *Proc.* of the 31st IEEE Conference on Local Computer Netowrks, 404-410, (2006).

[14] G. Bianchi. Performance Analysis of the IEEE 802.11 Distributed Coordination Function. *IEEE Journal on Selected Area in Communications*, 18(3):535-547, (2000).

[15] M.M. Carvalho and J.J. Garcial-Luna-Aceves. Delay Analysis of IEEE 802.11 in Single-Hop Networks. In *Proc.* Of 11th IEEE International Conference on Network Protocols (ICNP 2003), Atlanta, USA, (2003).

[16] V.G. Kulkarni. Modeling, *Analysis, Design, and Control of Stochastic Systems*. Springer-Verlag Berlin Heidelberg New York, (1999).

[17] C.E. Perkins, and E.M. Royer. Ad Hoc On-Demand Distance Vector Routing. In *Proc.*of (WMCSA99), 90-100, (1999).
[18] A. Lee, I. Ra, and H. Kim. Performance Study of Ad Hoc Routing Protocols with Gossip-Based Approach. In *Proc.* of (CNS09), (2009).
[19] S.M. Ross. *Introduction to Probability Models*. Elsevier Inc. 9th ed., (2007).
[20] S. Corson, and J. Macker. Mobile Ad hoc Networking (MANET): Routing Protocol Performance Issues and Evaluating Consideratins. RFC Editor, (1999).
[21] T. Wagner. Campus QoS for Voice and Video. Cisco Systems, (2011). http://www.sdcug.com/wp-content/uploads/2011/04/Campus-QoS-for-Voice-and-Video.pdf.
[22] K. Fall and K. Varadhan. *ns-2*, (2010).
[23] N. Bisnik and A. Abouzeid. Queueing Network Models for Delay Analysis of Multihop Wireless Ad Hoc Networks. In *Proc*. of International Conference on Wireless Communications and Mobile Computing, Vancouver, Canada, 773-778, (2006).
[24] R. Ell-Khoury, R. El-Axouzi and E. Altman. Delay analysis for real-time streaming media in multi-hop ad-hoc networks. In Proc. of the International Symposium on Modeling and Optimization in Mobile, Ad Hoc, Wireless Networks (WiOPT), Germany, (2008).
[25] H. Jabbar, S. Lee, S. Choi, S. Baek, S. Yu, and T. Jeoung. A Novel Sensing Method and Sensing Algorithm Development for a Ubiquitous Network. Journal of MDPI Sensors, 10:8129-8139, (2010).
[26] P. Mohaparta, J. Li and C. Gui. QoS in Mobile Ad Hoc Networks. IEEE Wireless Communication, Mag. (Special Issue on QoS in Next-Generation Wireless Multimedia Communications systems), pp. 44-52, (2003).

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