Delay and Capacity in Ad Hoc Mobile Networks with f-cast Relay Algorithms

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Delay and Capacity in Ad Hoc Mobile Networks with f-cast Relay Algorithms

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***Abstract*—The two-hop relay algorithm and its variants have been attractive for ad hoc mobile networks, because they are simple yet efficient, and more importantly, they enable the capacity and delay to be studied analytically. This paper considers a general two-hop relay with f-cast (2HR-f), where each packet is delivered to at most f distinct relay nodes and should be received in order at its destination. We derive the closed-form theoretical models rather than order sense ones for the 2HR-f algorithm with a careful consideration of the important interference, medium contention, traffic contention and queuing delay issues, which enable an accurate delay and capacity analysis to be performed for an ad hoc mobile network employing the 2HR-f. Based on our models, one can directly get the corresponding order sense results. Extensive simulation studies are also conducted to demonstrate the efficiency of these new models.**

***Index Terms*—Ad hoc mobile networks, two-hop relay, packet redundancy, capacity, delay.**

I. Introduction

An ad hoc mobile network is a self-configuring network, where mobile users can communicate with each other via wireless links without the aid of any existing infrastructure and centralized administration [1]–[3]. In such an autonomous network, each user acts not only as a host but also as a relay, storing and forwarding packets for other nodes in the network.

Since the seminal work of Grossglauser and Tse (2001) [4], the two-hop relay algorithm and its variants have become a class of attractive routing algorithms for ad hoc mobile networks, because they are simple yet efficient, and more importantly, they enable the capacity and delay to be studied analytically. As illustrated in Fig. 1, the two-hop relay algorithm defines two phases for packet transmission, where a packet is first transmitted from its source node to an intermediate node (relay node) in phase 1, and then in phase 2 the packet is transmitted from the relay node to its destination node. Since the source node can directly transmit a packet to its destination node once such transmission opportunity arises, every packet goes through at most two hops to reach its destination in a two-hop relay network.

By now, extensive order sense results of delay and capacity have been reported for the two-hop relay in ad hoc mobile networks (see Section VI for related works). The term of

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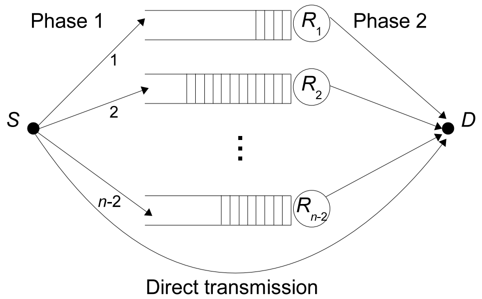


Fig. 1. Illustration of the two-hop relay.

"order sense" corresponds to "asymptotic" in this paper, which usually appears together with notations $(O, \Omega, \Theta, o, \omega)$ to describe the growth rate of network throughput or delay when the network size n tends towards a particular value or infinity. In the case of no packet redundancy (i.e., no redundant copies for each packet), the order sense scaling laws of two-hop relay have been explored under various mobility models, like the i.i.d. mobility model [4], [5], the Brownian mobility model [6], [7], the random walk model [8], the random waypoint model [9], the restricted mobility model [10], [11], and the correlated mobility model [12], etc. Notice that by allowing proper packet redundancy in the two-hop relay (i.e, each packet can have more than one copies in its delivery process), we may achieve more flexible trade-off between delay and capacity. Actually, the idea of using packet redundancy has been adopted in intermittently connected mobile networks (ICMNs) to reduce average packet delivery delay there [13]–[17], where nodes are sparsely distributed, and the interference and medium contention are of no concern. The order sense results of delay and capacity of two-hop relay with packet redundancy have also been explored recently, see, for example, [18]–[21].

Although the order sense results are helpful for us to understand the general scaling laws of delay and capacity in two-hop relay ad hoc mobile networks, but they tell us a little about the actual end-to-end delay and capacity of such networks. In practice, however, the actual delay and capacity results are of great interest for network designers. Also, some recent work [22], [23] indicate that even for sparse ad hoc mobile networks (like the ICMNs), ignoring the interference and medium contention in delay and capacity analysis may lead to inaccurate and even misleading results. In this paper we focus on the analysis of a general two-hop relay with f-cast (2HR-f), where each packet can be delivered to at most f distinct relay nodes and should be received in order at its destination. The main contributions of this paper are summarized as follows:

- We develop closed-form rather than order sense models for 2HR-f relay with a careful consideration of the important interference, medium contention, traffic contention and queuing delay issues, which enable an accurate delay and capacity analysis to be performed for the 2HR-f-based ad hoc mobile networks.
- With the new closed-form models, one can explore the trade-off between packet redundancy f and delay/capacity, and can also easily derive the corresponding order sense results for delay and capacity.
- Extensive simulation studies are also conducted to demonstrate the efficiency of these new theoretical models in capturing the behaviors of network throughput and delay performance under the 2HR-f relay.

The rest of the paper is organized as follows. In Section II, we provide the network model, interference model and mobility model considered in our analysis. Section III introduces the 2HR-f algorithm and the corresponding transmission scheduling scheme. We develop the closed-form models to analyze the per node throughput capacity and the expected end-to-end delay in Section IV, and present numerical results to validate these theoretical models in Section V. We introduce the related works in Section VI, and finally conclude the paper in Section VII.

II. System Models

A. Network Model

The network we consider in this paper consists of n mobile nodes inside a square region of unit area, which is evenly divided into $m \times m$ cells. We focus on a slotted system and a fast mobility scenario [11], where only one-hop transmissions are possible within each time slot, and the total number of bits transmitted per slot is fixed and normalized to 1 packet here.

The nodes independently roam from cell to cell and follow the bi-dimensional i.i.d. mobility model (or so-called reshuffling model) [18]. At the beginning of each time slot, each node independently and uniformly selects a destination cell among all m^2 cells and stays in it for the whole time slot. Since the destination cell of a node is randomly chosen among all m^2 cells, each cell has the probability $1/m^2$ to become the destination cell of the node.

Notice that due to the node mobility in mobile ad hoc networks (MANETs), the meeting duration and thus the available data transmission time between any node pair are actually very limited, and no contemporaneous end-to-end path may ever exist at any given time instant [13], [24], [25]. The number of available transmission hops during a time slot depends on the slot length and the relative mobility pattern. Here we consider such a scenario where the time slot length is defined as the available data transmission time during each node meeting, and a whole time slot will be allocated only for data transmissions in one hop range.

B. Interference Model

We assume an uniform communication range r for all nodes, and adopt the model introduced in [26] to account for the interference among simultaneous transmissions. Suppose that at some time slot t a node i is trying to transmit to another node j, and their Euclidean distance is $d_{ij}(t)$. According to the interference model, this transmission can be successful if and only if the following two conditions hold:

(1) $d_{ij}(t) \leq r$;
(2) $d_{kj}(t) \geq (1+\Delta)r$ for every other node k that simultaneously transmits with the node i, where Δ is a specified guard-factor for interference prevention.

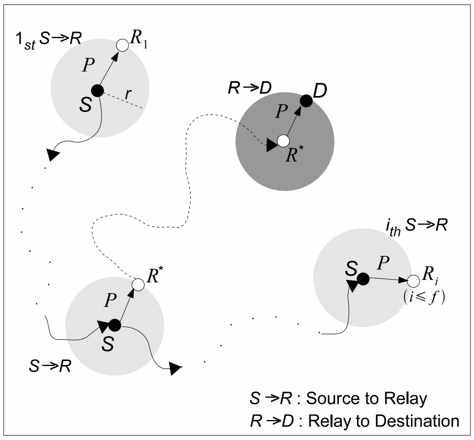


Fig. 2. Illustration of the 2HR-f relay for a tagged flow, where the source node S is transmitting packet P to the destination node D. The movement of all the remaining nodes in the unit square is not shown for simplicity.

C. Traffic Model

Similar to [4], [11], [18], [27]–[29], we consider in this paper the permutation traffic pattern, in which there are in total n distinct flows (source-destination pairs), and each node is the source of its locally generated traffic flow and at the same time the destination of a flow originated from some other node.

We further assume that the traffic originated from each node is a Poisson stream with rate λ (packets/slot), a packet arrives at the beginning of time slots, and the arrival process at each node is independent of its mobility process.

Remark 1: The permutation traffic pattern can be regarded as the worst-case uni-cast scenario, under which each node has a local outgoing traffic to deliver and also an incoming traffic to receive. According to the 2HR-f algorithm, therefore, a node will choose to forward traffic for other flows only when the node does not meet the destination of its own outgoing flow. In light of the fact that in the real-world MANETs some nodes may have no traffic to deliver or receive, i.e., may serve as pure relays, the throughput capacity derived under the permutation traffic pattern may serve as an achievable lower bound.

III. 2HR-f Relay and Transmission Scheduling

A. 2HR-f Algorithm

Without loss of generality, we focus on a tagged flow and denote its source node and destination node as S and D, respectively. We consider a generalization of the two-hop relay algorithm [18] with *f-cast* (2HR-f), $0 \leq f \leq \lfloor\sqrt{n}\rfloor$. As illustrated in Fig. 2 that with the 2HR-f algorithm, the source node S will deliver at most f copies of a packet P to distinct relay nodes, while the destination D may finally receive the packet from one relay node R^*. Thus, each packet in a 2HR-f network will have at most $f+1$ copies (including the one in its source node).

Since each node can be a potential relay for other $n-2$ flows (except the two flows originated from and destined for itself), we assume that each node maintains n individual queues at its buffer: one local-queue for storing the packets that are locally generated at the node and waiting for their copies (up to f copies for each packet) to be distributed, one already-sent-queue for storing packets whose f replicas have already been distributed but reception status are not confirmed yet (from destination node), and $n-2$ parallel relay-queues for storing packets of other flows (one queue per flow). Notice that all these n queues are FIFO queues, and each queue is assumed to have enough buffer space and thus no packet overflow will happen.

Notice that one common complication of designing relay algorithms with packet redundancy is that remnant copies of a packet that has already been accepted at its destination may create excess congestion and must somehow be removed [18]. To overcome this complication, we adopt a mechanism based packet sequence number for the 2HR-f algorithm. For the tagged flow, the source node S labels each packet P waiting at the local-queue with a *send number* $SN(P)$, such that a packet can be efficiently retrieved from the queue buffers of its source node or relay node(s) using its send number. Similarly, the destination node D also maintains a *request number* $RN(D)$ which indicates the send number of the packet it is currently requesting, such that each packet is received in order at the node D.

Based on the above definitions, the 2HR-f algorithm can be formally summarized as the following Algorithm 1.

Algorithm 1 2HR-f Algorithm:

if the node S gets a transmission opportunity **then**
    if the node D is among the one-hop neighbors of node S **then**
        S executes Procedure 1 with D; {source-to-destination transmission}
    else
        S randomly selects one node (say V) from its one-hop neighbors;
        S flips an unbiased coin;
        if it is the head **then**
            S executes Procedure 2 with V; {source-to-relay transmission}
        else
            S executes Procedure 3 with V; {relay-to-destination transmission}
        end if
    end if
end if

Procedure 1 Source-to-destination transmission:

S initiates a handshake to obtain the $RN(D)$ from node D;
if $SN(P_h) > RN(D)$ **then** {P_h is the head-of-line packet at the local-queue of S}
    S retrieves from its already-sent-queue the packet P with $SN(P) = RN(D)$;
    S sends the P to node D;
else if $SN(P_h) == RN(D)$ **then**
    S sends P_h directly to node D;
else $\{RN(D) = SN(P_h) + 1\}$
    S sends to node D the packet waiting right behind P_h in the local-queue;
end if
S deletes all packets with $SN \leq RN(D)$ inside the already-sent-queue and local-queue;
S moves ahead remaining packets waiting at its local-queue;

Procedure 2 Source-to-relay transmission:

S initiates a handshake with node V;
if V has one copy of P_h **then**
    S remains idle;
else
    S sends a copy of packet P_h to V;
    if f copies have been distributed for packet P_h **then**
        S puts P_h to the end of its already-sent-queue;
        S moves ahead the remaining packets in its local-queue;
    end if
    V puts P_h at the end of its relay-queue dedicated to node D;
end if

Procedure 3 Relay-to-destination transmission:

S initiates a handshake to obtain the $RN(V)$ from node V;
if S has a packet P in the relay-queue dedicated to V with $SN(P) = RN(V)$ **then**
    S sends packet P to node V;
else
    S remains idle;
end if
S deletes all packets with $SN \leq RN(V)$ from its relay-queue dedicated to V;

Notice that in Procedures 1 and 2, every time the node S (resp. node D) moves ahead its local-queue by one packet (resp. receives a packet), it increases its *send number* (resp. *request number*) by one.

Remark 2: Notice that the setting of $f = 0$ corresponds to the case that only source-to-destination transmission is allowed, so every packet takes exact one transmission opportunity to reach its destination there. Thus, in the case of $f = 0$ only the Procedure 1 will be executed.

Remark 3: Notice that in the 2HR-f Algorithm, the desti-

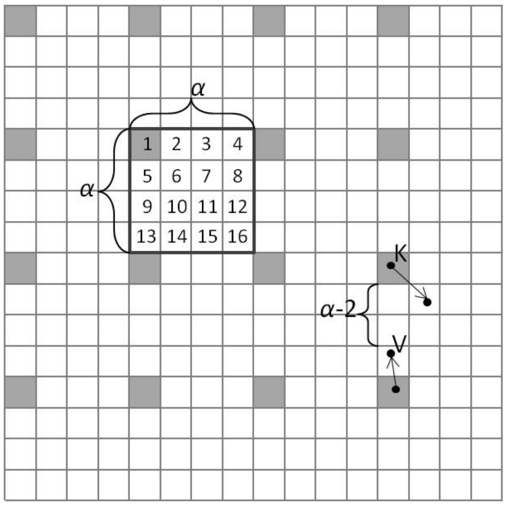


Fig. 3. An example of a transmission-group of cells with $\alpha = 4$. The cells are divided into 16 different transmission-groups and all the shaded cells belong to the same transmission-group. The distribution of all the remaining nodes in the unit square is not shown for simplicity.

nation node D receives the packets destined for it according to their request numbers, so all the packets will be received *in order* by their destinations.

Remark 4: The 2HR-f Algorithm indicates that each packet takes at most $f + 1$ transmissions to reach its destination, and there are two scenarios under which a packet will take less than $f+1$ transmissions: 1) before the S finishes the distribution of all f copies of the packet, the node D receives this packet directly from the S; 2) before the S finishes the distribution of all f copies, the D first receives this packet from one of its relay nodes and then meets the S notifying it the reception of this packet.

B. *Transmission Scheduling*

We consider a local transmission scenario [4], in which a node in some cell can only send packets to the nodes in the same cell or its eight adjacent cells. Two cells are called adjacent if they share a common point. Thus, the maximum distance between a transmitting node (transmitter) and a receiving node (receiver) is $\sqrt{8}/m$, so we set the communication range as $r = \sqrt{8}/m$. Due to the wireless interference, only cells that are sufficiently far away could simultaneously transmit without interfering with each other. To support as many simultaneous transmissions as possible, similar to the "equivalence class" in the [27], [30], [31] we define here the "transmission-group".

Transmission-group: As illustrated by the shaded cells in Fig. 3, a transmission-group is a subset of cells in which any two cells have a vertical and horizontal distance of some multiple of α cells, and all the cells there can transmit simultaneously without interfering with each other.

To guarantee the simultaneous transmissions in a transmission-group without interfering with each other, the parameter α should be set properly. As shown in the Fig. 3, suppose that during some time slot, the node V is scheduled to receive a packet. According to the definition of "transmission-group", we know that except the transmitting node of V, another transmitting node (say node K) in the same transmission-group is at least $(\alpha-2)/m$ away from V. The condition that K will not interfere with the reception at V is that,

$$(\alpha-2)/m \geq (1+\Delta)\cdot r$$

By substituting $r = \sqrt{8}/m$, we obtain that

$$\alpha \geq (1+\Delta)\sqrt{8}+2$$

As α is an integer and $\alpha \leq m$, we set

$$\alpha = \min\left\{\lceil(1+\Delta)\sqrt{8}\rceil + 2, m\right\} \tag{1}$$

where $\lceil x \rceil$ returns the smallest integer not smaller than x.

Notice that there are only α^2 transmission-groups, and each cell belongs to an individual transmission-group. If transmission-groups alternatively become active (i.e., get transmission opportunity), then each transmission-group will be active in every α^2 time slots. Therefore, each cell will also be active in every α^2 time slots. If there are more than one node inside an active cell, a transmitting node is selected randomly from them. The selected node then follows the 2HR-f algorithm for packet transmission.

Remark 5: Since at the beginning of each time slot each node can easily obtain the cell id where it resides inside, it can then judge whether it is inside an active cell or not for the current time slot. Thus, we can adopt a DCF-style mechanism to randomly select a transmitter from an active cell. If a node is inside an active cell, it randomly selects an initial value from $[0, CW]$ (CW represents the contention window) and starts to count down. If it hears no broadcasting message (regarding transmitter) until its back-off counter becomes 0, it broadcasts out a message denoting itself as the transmitter; otherwise it stops its back-off counter as some other node has claimed as the transmitter. The back-off counting mechanism is uniform for all network nodes, and the value of CW depends on the implementation details.

IV. Throughput Capacity and Expected End-to-End Delay

In this section, we first provide the analysis of some basic probabilities and introduce the service time at the source S and the destination D, then proceed to derive the per node throughput capacity and expected packet delay.

A. *Some Basic Probabilities*

For a given active cell, we first formally define the contention probability for transmitting opportunity and the contention probability for receiving opportunity.

Definition 1: For an active cell in any time slot, its contention probability for transmitting opportunity is defined as the probability that there are at least two nodes inside it.

Definition 2: For an active cell in any time slot, its contention probability for receiving opportunity is defined as the probability that aside from the selected transmitter, it has at least two other nodes inside its one-hop neighborhood (i.e., the cell itself and its eight adjacent cells).

Based on the above definition, we establish the following lemmas about some basic probabilities (See Appendix A for proofs).

Lemma 1: If we divide the network into $\sqrt{n} \times \sqrt{n}$ cells, i.e., $m = \sqrt{n}$, then for an active cell in any time slot, as n approaches infinity, its contention probability for transmitting opportunity approaches $1 - 2\mathrm{e}^{-1}$, while its contention probability for receiving opportunity approaches $1 - \mathrm{e}^{-1} - \frac{19}{2}\mathrm{e}^{-9}$.

Remark 6: The above results indicate that the medium contention happens with significant and non-neglectable probability, so ignoring it in the delay and capacity analysis may lead to inaccurate and even misleading results [22]. Thus, our following delay and capacity analysis will be conducted with a careful consideration of the important medium contention issue.

Lemma 2: For a given time slot and a tagged flow, we use p_1, p_2 and p_3 to denote the probability that the source node S conducts a packet transmission, the probability that the S conducts a source-to-destination transmission and the probability that the S conducts a source-to-relay or relay-to-destination transmission, respectively. Then we have

$$p_1 = \frac{1}{\alpha^2}\left\{\frac{m^2}{n}\left(1-\left(\frac{m^2-1}{m^2}\right)^n\right)-\left(\frac{m^2-9}{m^2}\right)^{n-1}\right\} \quad (2)$$

$$p_2 = \frac{1}{\alpha^2}\left\{\frac{9n-m^2}{n(n-1)}-\left(\frac{m^2-1}{m^2}\right)^{n-1}\frac{8n+1-m^2}{n(n-1)}\right\} \quad (3)$$

$$p_3 = \frac{1}{\alpha^2}\left\{\frac{m^2-9}{n-1}\left(1-\left(\frac{m^2-1}{m^2}\right)^{n-1}\right)-\left(\frac{m^2-9}{m^2}\right)^{n-1}\right\} \quad (4)$$

Remark 7: One can easily prove that $p_1 = p_2 + p_3$, and p_2 quickly approaches zero as n scales up.

Lemma 3: For a given time slot and a tagged flow, suppose that its source node S is distributing copies for the head-of-line packet P_h at its local-queue and that there are already j ($1 \le j \le f+1$) copies of P_h in the network at the current time slot and $SN(P_h) = RN(D)$. We use $P_r(j)$ and $P_d(j)$ to denote the probability that the destination node D will receive P_h and the probability that the S will successfully deliver out a new copy of P_h (if $j \le f$) in the next time slot, respectively. Then we have

$$P_r(j) = p_2 + \frac{j-1}{2(n-2)} \cdot p_3 \quad (5)$$

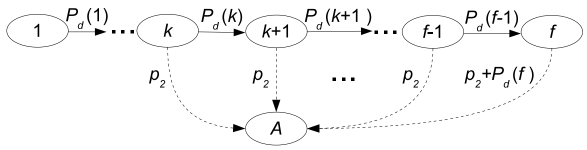
$$P_d(j) = \frac{n-j-1}{2(n-2)} \cdot p_3 \quad (6)$$

Remark 8: It is notable that the important medium contention, interference and traffic contention issues have been carefully incorporated into the derivations of the probabilities p_1, p_2, p_3, $P_r(j)$ and $P_d(j)$.

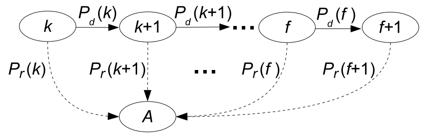
B. Service Time at Source S and Destination D

For a tagged flow, before formally defining the service time at the source S and the service time at the destination D, we introduce the following two queues.

The first queue is the local-queue at the source S. The local-queue stores the locally generated packets and operates as follows: every time a local packet P is generated, the P is put to the end of the queue; every time the S finishes the copy distribution for the head-of-line packet, S moves it out of the queue and moves ahead the remaining packets waiting behind it. The head-of-line packet of the local-queue indicates the packet for which the S is currently distributing copies.



(a) Absorbing Markov chain for the packet distribution process at the source node S.



(b) Absorbing Markov chain for the packet reception process at the destination node D.

Fig. 4. Absorbing Markov chains for a general packet P, given that the D starts to request for the P when there are already k copies of P in the network. For each transient state, the transition back to itself is not shown for simplicity.

The second queue is a virtual queue defined at the destination D. The virtual queue stores the *send numbers* of those packets not received yet by D, and the head-of-line entry of the virtual queue is the *send number* of the packet that the D is currently requesting for. The virtual queue operates as follows: every time a packet P is moved to the head-of-line of the local-queue at S, the corresponding packet send number $SN(P)$ is put to the end of the virtual queue; every time the D receives a packet whose *send number* equals to the head-of-line entry, the D moves the head-of-line entry out of the virtual queue and moves ahead the remaining entries.

Definition 3: For a packet P, the service time at the source S is the time elapsed between the time slot when the S moves the P into the head-of-line at the local-queue and the time slot when the S stops distributing copies for the P (i.e., when the S moves the P out of the local-queue).

Definition 4: For a packet P, the service time at the destination D is the time elapsed between the time slot when the D starts to request for the P (i.e., when the entry $SN(P)$ is moved to the head-of-line at the virtual queue), and the time slot when the D receives the P.

For a packet P, suppose that there are k copies of P in the network when its destination D starts to request for the packet, $1 \le k \le f+1$. If we use A to denote the absorbing state (i.e., the termination of the service process) for P, then the service processes for the packet at its source S and at its destination D can be defined by two finite-state absorbing Markov chains shown in Fig. 4a and Fig. 4b, respectively.

Given that there are k copies of P in the network when the D starts to request for the packet, we use $X_S(k)$ and $X_D(k)$ to the denote the corresponding service time of packet P at the S and the D, respectively [1]. From the theory of Markov chain [32], we can see that the $X_S(k)$ can be regarded as the time the Markov chain in the Fig. 4a takes to become absorbed given that the chain starts from the state 1, and the $X_D(k)$ can be regarded as the time the Markov chain in the Fig. 4b takes to become absorbed given that the chain starts from the state k.

Lemma 4: For a packet P of the tagged flow, suppose that there are k copies of P in the network when the destination node D starts to request for the P, $1 \le k \le f+1$, then we have

$$\mathbb{E}\{X_S(k)\} = \begin{cases} \sum_{i=1}^{k-1} \frac{1}{P_d(i)} + \frac{1}{p_2+P_d(k)} \\ \quad \cdot\big(1+\sum_{j=1}^{f-k}\phi_2(k,j)\big) & \text{if } 1 \le k \le f, \\ \sum_{i=1}^{f} \frac{1}{P_d(i)} & \text{if } k = f+1. \end{cases} \tag{7}$$

$$\mathbb{E}\{X_D(k)\} = \begin{cases} \frac{1}{p_2+p_3/2}\big(1+\sum_{j=1}^{f-k}\phi_3(k,j) \\ \quad +\frac{P_d(f)}{P_r(f+1)}\phi_3(k,f-k)\big) & \text{if } 1 \le k \le f-1, \\ \frac{1}{p_2+p_3/2}\big(1+\frac{P_d(f)}{P_r(f+1)}\big) & \text{if } k = f, \\ \frac{1}{P_r(f+1)} & \text{if } k = f+1. \end{cases} \tag{8}$$

where $\phi_2(k,j) = \prod_{t=1}^{j}\frac{P_d(k+t-1)}{p_2+P_d(k+t)}$ and $\phi_3(k,j) = \prod_{t=1}^{j}\frac{P_d(k+t-1)}{p_2+p_3/2}$.

Proof: We derive (7) first. For the absorbing Markov chain in the Fig. 4a and a given k there, $1 \le k \le f$, if we denote by a_i the mean time the Markov chain takes to become absorbed given that the chain starts from the state i, $1 \le i \le f$, and denote by q_{ij} the transition probability from state i to state j, $i,j \in [1,f]$, then we have

$$\mathbb{E}\{X_S(k)\} = a_1 \tag{9}$$

$$a_i = \frac{1+\sum_{j\in[1,f],j\ne i} q_{ij}\cdot a_j}{1-q_{ii}} \tag{10}$$

Notice that in the Markov chain of the Fig. 4a, except transiting back to itself and transiting to the absorbing state A, the state i can only transit to its next state, i.e., the state $i+1$. Thus, the a_i can be further determined as

$$a_i = \begin{cases} \frac{1}{P_d(i)} + a_{i+1} & \text{if } 1 \le i < k, \\ \frac{1+P_d(i)\cdot a_{i+1}}{p_2+P_d(i)} & \text{if } k \le i < f, \\ \frac{1}{p_2+P_d(f)} & \text{if } i = f. \end{cases} \tag{11}$$

The a_1 and thus the (7) can be derived from the (11) recursively.

Regarding the case that $k = f+1$, i.e., the destination D starts to request for the packet P after f copies of P have been distributed, it is easy to see that $\mathbb{E}\{X_S(f+1)\} = \sum_{i=1}^{f}\frac{1}{P_d(i)}$.

Now we proceed to derive (8). Similarly, for the Markov chain in the Fig. 4b, we have

$$\mathbb{E}\{X_D(i)\} = \begin{cases} \frac{1+P_d(i)\cdot\mathbb{E}\{X_D(i+1)\}}{p_2+p_3/2} & \text{if } k \le i \le f, \\ \frac{1}{P_r(f+1)} & \text{if } i = f+1. \end{cases} \tag{12}$$

The (8) can then be derived from the (12) recursively. ■

[1]The $X_S(f+1)$ corresponds to the case that the D starts to request for the packet P from the state that there are $f+1$ copies in the network, i.e., f copies of P have been distributed.

Lemma 5: For any $1 \le k \le f$, we have

$$\mathbb{E}\{X_S(k)\} < \mathbb{E}\{X_S(k+1)\} \tag{13}$$
$$\mathbb{E}\{X_D(k)\} > \mathbb{E}\{X_D(k+1)\} \tag{14}$$

Proof: We prove (13) first. When $k = f$, it is easy to see that

$$\mathbb{E}\{X_S(f+1)\} - \mathbb{E}\{X_S(f)\} = \frac{1}{P_d(f)} - \frac{1}{p_2+P_d(f)} > 0 \tag{15}$$

For the case that $1 \le k < f$, we have

$$\begin{aligned} &\mathbb{E}\{X_S(k+1)\} - \mathbb{E}\{X_S(k)\} \\ &= \frac{1}{P_d(k)} + \frac{1}{p_2+P_d(k+1)}\big(1+\sum_{j=1}^{f-k-1}\phi_2(k+1,j)\big) \\ &\quad - \frac{1}{p_2+P_d(k)}\big(1+\sum_{j=1}^{f-k}\phi_2(k,j)\big) \\ &= \frac{1}{P_d(k)} + \frac{1}{p_2+P_d(k+1)}\big(1+\sum_{j=1}^{f-k-1}\phi_2(k+1,j)\big) \\ &\quad - \frac{1+\frac{P_d(k)}{p_2+P_d(k+1)}\big(1+\sum_{j=1}^{f-k-1}\phi_2(k+1,j)\big)}{p_2+P_d(k)} \qquad (16) \\ &= \frac{1}{P_d(k)} - \frac{1}{p_2+P_d(k)} \\ &\quad + \frac{1+\sum_{j=1}^{f-k-1}\phi_2(k+1,j)}{p_2+P_d(k+1)}\big(1-\frac{P_d(k)}{p_2+P_d(k)}\big) \\ &> \frac{1}{P_d(k)} - \frac{1}{p_2+P_d(k)} > 0 \qquad (17) \end{aligned}$$

where the (16) follows after substituting

$$\sum_{j=1}^{f-k}\phi_2(k,j) = \frac{P_d(k)}{p_2+P_d(k+1)}\big(1+\sum_{j=1}^{f-k-1}\phi_2(k+1,j)\big).$$

Combining (15) and (17), the (13) follows.

Now we proceed to prove (14) using mathematical induction. The proof will now proceed in two steps:

Initial step: when $k = f$, we have that

$$\begin{aligned} \mathbb{E}\{X_D(f)\} &= \frac{1}{P_r(f+1)}\frac{P_d(f)+P_r(f+1)}{p_2+\frac{p_3}{2}} \\ &> \frac{1}{P_r(f+1)} = \mathbb{E}\{X_D(f+1)\} \qquad (18) \end{aligned}$$

Inductive step: our inductive assumption is: there is a t, $1 < t \le f$, such that $\mathbb{E}\{X_D(t)\} > \mathbb{E}\{X_D(t+1)\}$. We must prove the (14) is true for $k = t-1$.

Since

$$\begin{aligned} \mathbb{E}\{X_D(t-1)\} &= \frac{1+P_d(t-1)\cdot\mathbb{E}\{X_D(t)\}}{P_d(t-1)+P_r(t-1)} \\ &> \frac{1+P_d(t)\cdot\mathbb{E}\{X_D(t+1)\}}{P_d(t)+P_r(t)} \qquad (19) \\ &= \mathbb{E}\{X_D(t)\} \qquad (20) \end{aligned}$$

where the (19) follows because $P_d(t-1) > P_d(t)$, $\mathbb{E}\{X_D(t)\} > \mathbb{E}\{X_D(t+1)\}$ and $P_d(t-1)+P_r(t-1) = P_d(t)+P_r(t)$. The (20) completes the inductive step. ■

Although the throughput capacity and packet delay can be traded with each other by adopting different f in a large range of $[1, \lfloor\sqrt{n}\rfloor]$ [18]–[20], in the real-world MANETs, however, the redundancy f of each packet should be limited to a range of small values. This is because that each mobile node is not only buffer storage-limited but also energy-limited, so too many redundant copies will unavoidably waste these precious network resources [33], [34].

In light of this observation and the Lemma 5, the following lemma identifies such a range of f and also determines the corresponding property of average service time at the source S and destination D. Such property will be used later to derive the corresponding throughput capacity and end-to-end packet delay upper bound.

Lemma 6: For a 2HR-f MANET, if we denote by $\overline{X}_S$ the average service time at the source S taken over all locally generated packets, and denote by $\overline{X}_D$ the average service time at the destination D taken over all received packets, then the following property holds when $1 \le f \le f_0$,

$$\overline{X}_S \le \overline{X}_D \tag{21}$$

where f_0 is determined as

$$f_0 = \max\Big\{f \,|\, \mathbb{E}\{X_S(f+1)\} \le \mathbb{E}\{X_D(f+1)\}, \quad 1 \le f \le \lfloor\sqrt{n}\rfloor\Big\} \tag{22}$$

Proof: We first prove that the f_0 in the (22) does exist (i.e., the set there is not empty). According to the (3) and (4), it is trivial to see that $p_2 < \frac{p_3}{2} \cdot \frac{n-3}{n-2}$ in general [2]. Then we have

$$\mathbb{E}\{X_D(f+1)\}|_{f=1} > \mathbb{E}\{X_S(f+1)\}|_{f=1} \tag{23}$$

Notice that

$$\begin{aligned}\mathbb{E}\{X_S(f+1)\}|_{f=\sqrt{n}} &= \frac{2(n-2)}{p_3}\sum_{m=1}^{\sqrt{n}}\frac{1}{n-m-1}\\ &> \frac{2(n-2)}{p_3}\sum_{m=1}^{\sqrt{n}}\frac{1}{n}\\ &= \frac{2(n-2)}{\sqrt{n}}\cdot\frac{1}{p_3}\end{aligned}$$

and

$$\begin{aligned}\mathbb{E}\{X_D(f+1)\}|_{f=\sqrt{n}} &= \frac{1}{p_2 + \frac{\sqrt{n}}{2(n-2)}\cdot p_3}\\ &< \frac{2(n-2)}{\sqrt{n}}\cdot\frac{1}{p_3}\end{aligned}$$

Therefore, we have

$$\mathbb{E}\{X_S(f+1)\}|_{f=\sqrt{n}} > \mathbb{E}\{X_D(f+1)\}|_{f=\sqrt{n}} \tag{24}$$

It is further noticed that for a given setting of n and m, as f increases the $\mathbb{E}\{X_S(f+1)\}$ monotonically increases while the $\mathbb{E}\{X_D(f+1)\}$ monotonically decreases. Combining with the (23) and (24), we can see that the f_0 defined in (22) does exist.

Now we proceed to prove that whenever $f \in [1, f_0]$, we always have the (21). Since the $\overline{X}_S$ and $\overline{X}_D$ are taken over all locally generated packets and all received packets, respectively, together with the Lemma 5, we have

$$\mathbb{E}\{X_S(1)\} \le \overline{X}_S \le \mathbb{E}\{X_S(f+1)\} \tag{25}$$

$$\mathbb{E}\{X_D(f+1)\} \le \overline{X}_D \le \mathbb{E}\{X_D(1)\} \tag{26}$$

The (25) and (26) indicate clearly that in order to prove the (21), we just need to prove that whenever $f \in [1, f_0]$, we always have $\mathbb{E}\{X_S(f+1)\} \le \mathbb{E}\{X_D(f+1)\}$.

In light of the monotonicity property of $\mathbb{E}\{X_S(f+1)\}$ and $\mathbb{E}\{X_D(f+1)\}$ and the definition of f_0 in the (22), it is easy to see that whenever $f \in [1, f_0]$, we always have $\mathbb{E}\{X_S(f+1)\} \le \mathbb{E}\{X_D(f+1)\}$. Then the Lemma 6 follows. ■

C. Throughput Capacity and Packet Delay

For a MANET with 2HR-f and $0 \le f \le f_0$, we are now ready to derive its per node throughput capacity and overall expected end-to-end delay based on the basic property (21) of service time $\overline{X}_S$ and $\overline{X}_D$.

Theorem 1: For a network with the 2HR-f relay ($0 \le f \le f_0$), if we denote by μ the per node (flow) throughput capacity (i.e., the network can stably support any rate $\lambda < \mu$), then we have

$$\mu = \begin{cases} p_2 + \frac{f}{2(n-2)}\cdot p_3 & \text{if } 1 \le f \le f_0,\\ p_2 & \text{if } f = 0.\end{cases} \tag{27}$$

Proof: As indicated in the Lemma 6 that we always have $\overline{X}_S \le \overline{X}_D$ when $f \in [1, f_0]$, so the actual throughput for the tagged flow is $1/\overline{X}_D$. Then the per node (flow) throughput capacity can be determined as

$$\begin{aligned}\mu &= \max\{1/\overline{X}_D\}\\ &= \frac{1}{\mathbb{E}\{X_D(f+1)\}} \qquad (28)\\ &= p_2 + \frac{f}{2(n-2)}\cdot p_3 \qquad (29)\end{aligned}$$

where the (28) is due to (26) and the (29) is due to (8).

Regarding the case that $f = 0$, since only the Procedure 1, i.e., the source-to-destination transmission, will be executed, it is easy to see that $\mu = p_2$. ■

Lemma 7: For a network with the 2HR-f relay ($0 \le f \le f_0$), the maximum per node throughput capacity μ^* is achieved at $f = f_0$.

Proof: The Lemma 7 follows directly after the (27). ■

Under the setting that $\Delta = 1, m = \lfloor\sqrt{n}\rfloor$ and $36 \le n \le 1024$, the maximum per node throughput μ^* and the corresponding value of f (i.e., f_0) are summarized in the Fig. 7a and 7b, respectively.

Now we proceed to derive an upper bound for the expected end-to-end packet delay.

[2]The statement $p_2 < \frac{p_3}{2} \cdot \frac{n-3}{n-2}$ holds under most settings of m and n, except the settings that $\{m \le 5\}$, $\{m = 6, n \le 7\}$, $\{m = 7, n \le 6\}$ and $\{m \ge 8, n \le 5\}$. Notice that according to the (1) in the transmission-group based scheduling scheme, all these special settings correspond to very simple (if not impractical) network scenarios. We neglect these settings here and focus on other general settings in this paper.

Theorem 2: For a network with the 2HR-f relay ($0 \leq f \leq f_0$), if we denote by $\mathbb{E}\{T_e\}$ the expected end-to-end packet delay, then we have

$$\mathbb{E}\{T_e\} \leq \begin{cases} \frac{\mathbb{E}\{X_S(f+1)\}}{1-\rho_1} + \frac{\mathbb{E}\{X_D(f+1)\}}{1-\rho_2} & \text{if } 1 \leq f \leq f_0, \\ \frac{1}{1-\rho_2}\frac{1}{p_2} & \text{if } f = 0. \end{cases} \tag{30}$$

where $\lambda < \mu$, $\rho_1 = \lambda\mathbb{E}\{X_S(f+1)\}$ and $\rho_2 = \lambda\mathbb{E}\{X_D(f+1)\}$.

Proof: We first focus on the case $1 \leq f \leq f_0$ and consider a tagged packet P arriving to the local-queue of the source S at the beginning of a time slot. If we denote by $P_a(f+1)$ the probability that the P has not been received by the destination D yet when all its f copies are distributed out, denote by $X_1(P)$ and $X_2(P)$ the service time of P at the local-queue and the virtual queue, respectively, and further denote by $W_1(P)$ and $W_2(P)$ the waiting time of P at the local-queue and the virtual queue, respectively, then we have

$$\begin{aligned} &\mathbb{E}\{T_e\} \\ &= (1 - P_a(f+1))\Big(\mathbb{E}\{W_1(P)\} + \mathbb{E}\{X_1(P)\}\Big) + P_a(f+1) \\ &\cdot\Big(\mathbb{E}\{W_1(P)\} + \mathbb{E}\{X_1(P)\} + \mathbb{E}\{W_2(P)\} + \mathbb{E}\{X_2(P)\}\Big) \\ &= \mathbb{E}\{W_1(P)\} + \mathbb{E}\{X_1(P)\} \\ &\quad + P_a(f+1)\Big(\mathbb{E}\{W_2(P)\} + \mathbb{E}\{X_2(P)\}\Big) \\ &\leq \mathbb{E}\{W_1(P)\} + \mathbb{E}\{X_1(P)\} + \mathbb{E}\{W_2(P)\} + \mathbb{E}\{X_2(P)\} \end{aligned} \tag{31}$$

The following proof is similar to the derivation of the standard *Pollaczek-Khinchin* formula for mean waiting time in an $M/G/1$ queue. Regarding the waiting time $W_1(P)$ of P in the local-queue before getting service (i.e., before being replicated and delivered to f distinct relays), we have

$$W_1(P) = \sum_{i=1}^{L_q} X_1(P_i) + R \tag{32}$$

where variable R is the residual service time, L_q is the number of packets waiting in the queue, and $X_1(P_i)$ is the service time of the i_{th} packet.

The service times $\{X_1(P_i)\}$ are mutually independent, and as proved in the Lemma 5, their expected values are upper bounded by $\mathbb{E}\{X_S(f+1)\}$. Recall that $\overline{X}_S$ represents the actual mean time the node S takes to serve a generic packet, and if we let ρ_r represent the actual probability that the S is busy with delivering copies of some packet, then we have $\mathbb{E}\{R\} \leq \rho_r\overline{X}_S$ and $\rho_r = \lambda\overline{X}_S$.

As proved in the Lemma 6, $\overline{X}_S \leq \mathbb{E}\{X_S(f+1)\}$, then we have $\rho_r \leq \rho_1$ and thus $\mathbb{E}\{R\} \leq \rho_1\mathbb{E}\{X_S(f+1)\}$. Taking expectations of the both sides of (32) yields

$$\begin{aligned} &\mathbb{E}\{W_1(P)\} \\ &\leq \mathbb{E}\{L_q\}\mathbb{E}\{X_S(f+1)\} + \rho_1\mathbb{E}\{X_S(f+1)\} \\ &= \lambda\mathbb{E}\{W_1(P)\}\mathbb{E}\{X_S(f+1)\} + \rho_1\mathbb{E}\{X_S(f+1)\} \\ &= \rho_1\mathbb{E}\{W_1(P)\} + \rho_1\mathbb{E}\{X_S(f+1)\} \end{aligned} \tag{33}$$

We then have

$$\mathbb{E}\{W_1(P)\} \leq \frac{\rho_1\mathbb{E}\{X_S(f+1)\}}{1-\rho_1} \tag{34}$$

where $\rho_1 = \lambda\mathbb{E}\{X_S(f+1)\}$.

It is easy to see that the average input rate to the virtual queue is $P_a(f+1)\lambda$, and to simplify the analysis, we treat the input traffic as a Poisson stream here [3]. Notice that in the virtual queue, we have $\mathbb{E}\{X_2(P)\} = \mathbb{E}\{X_D(f+1)\}$. Using a similar derivation, we have

$$\mathbb{E}\{W_2(P)\} \leq \frac{P_a(f+1)\rho_2\mathbb{E}\{X_D(f+1)\}}{1 - P_a(f+1)\rho_2} \tag{35}$$

where $\rho_2 = \lambda\mathbb{E}\{X_D(f+1)\}$.

Substituting the (34) and (35) into (31) and combining that $\mathbb{E}\{X_1(P)\} \leq \mathbb{E}\{X_S(f+1)\}$ and $\mathbb{E}\{X_2(P)\} = \mathbb{E}\{X_D(f+1)\}$, we have

$$\begin{aligned} \mathbb{E}\{T_e\} &\leq \frac{\mathbb{E}\{X_S(f+1)\}}{1-\rho_1} + \frac{\mathbb{E}\{X_D(f+1)\}}{1 - P_a(f+1)\rho_2} \\ &\leq \frac{\mathbb{E}\{X_S(f+1)\}}{1-\rho_1} + \frac{\mathbb{E}\{X_D(f+1)\}}{1-\rho_2} \end{aligned} \tag{36}$$

Regarding the case of $f = 0$, using a derivation similar to the above, one can easily see that $\mathbb{E}\{T_e\} \leq \frac{1}{1-\rho_2}\frac{1}{p_2}$. Together with the (36), the Theorem 2 follows. ∎

Remark 9: The Theorems 1 and 2 provide closed-form (rather than order sense) results for the per node throughput capacity and the expected end-to-end packet delay in 2HR-f-based ad hoc mobile networks. Based on the Theorems 1 and 2 and any setting of $f = n^\delta$, $0 \leq \delta \leq \log_n f_0$, one can easily derive the corresponding order sense results of throughput capacity and packet delay. For example, for an 2HR-f MANET, by setting $m = n^\gamma$, our theoretical models return a $\Theta(n^{\max\{-1,\ \delta+2\gamma-2\}})$ throughput and $O(n^{\min\{1,\ 2-\delta-2\gamma\}})$ delay when $0 < \gamma \leq \frac{1}{2}$, and a $\Theta(n^{\delta-2\gamma})$ throughput and $O(n^{2\gamma-\delta})$ delay when $\gamma > \frac{1}{2}$.

Remark 10: One may also notice that when setting $f = 1$, the Theorem 1 results in a $\Theta(1/n)$ throughput, which is lower than the throughput result $\Theta(1)$ reported in [4]. This is due to the rule of "reception in order" employed in 2HR-f. The restriction of receiving packets according to *request number* ensures that all packets arrive at the destination *in order*, but it wastes the opportunities of receiving "out of order but fresh" packets (i.e., packets with *send number* larger than the current *request number* of destination node). Thus, the benefit of receiving all packets in order comes at the price of a reduced per node throughput.

V. Numerical Results

In this section, we first provide simulation results to verify the theoretical models for the per node throughput capacity and expected end-to-end packet delay, then proceed to explore the maximum per node throughput and corresponding setting of f.

[3] As to be validated in the Section V, the theoretical packet delay bound derived under this assumption is safe and can nicely upper bound the simulated end-to-end packet delay when the network is stable, i.e., $\lambda < \mu$.

A. Simulation Settings

A simulator in C++ was developed to simulate the packet delivery process in a 2HR-f MANET, which is now available at [35]. Similar to the settings adopted in [36], [37], the guard factor here is fixed as $\Delta = 1$, and hence the transmission-group is defined with $\alpha = \min\{8, m\}$. Besides the bi-dimensional i.i.d. mobility model, we also implemented the simulator for the random walk model and random waypoint model, which are defined as follows:

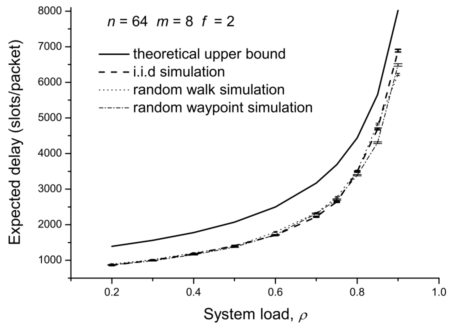
- **Random Walk Model:** At the beginning of each time slot, each node independently and uniformly selects a destination cell among the nine one-hop cells, i.e., the current cell and the eight adjacent cells, and then stays in it for the whole time slot. Each one-hop cell has the probability $1/9$ to become the destination cell of the node.
- **Random Waypoint Model [38]:** At the beginning of each time slot, each node independently and randomly generates a two-dimensional vector $v = [v_x, v_y]$, where the values of v_x and v_y are uniformly drawn from $[1/m, 3/m]$. The node then moves a distance of v_x along the horizontal direction and a distance of v_y along the vertical direction.

B. Model Validation

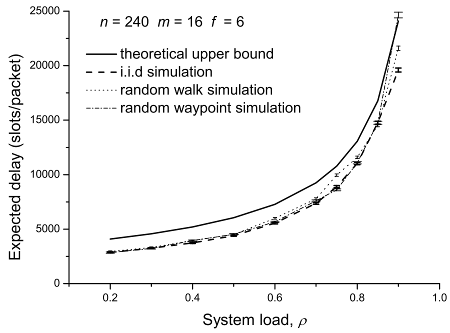
Extensive simulations have been conducted to verify the developed theoretical models. Here, the results of two network scenarios $(n = 64, m = 8)$ and $(n = 240, m = 16)$ are included (the other scenarios can be easily simulated by our simulator as well [35]). For the settings of $(n = 64, m = 8)$ and $(n = 240, m = 16)$, the f_0 is established as 2 and 9, respectively. We fix $f = 2$ for the setting $(n = 64, m = 8)$ and fix $f = 6$ for the setting $(n = 240, m = 16)$, and summarize the corresponding simulation and theoretical results in Fig. 5. Notice that all the simulation results of the expected end-to-end packet delay are reported with the 95% confidence intervals.

The Fig. 5 indicates clearly that for the bi-dimensional i.i.d. mobility model, our theoretical delay results can tightly upper bound the simulated ones when the network is stable, i.e., $\lambda < \mu$ ($\rho < 1$). For example, regarding the network scenario in the Fig. 5a, as the system load $\rho = \lambda/\mu$ (resp. λ) gradually increases from 0.2 up to 0.9 (resp. from 2.96×10^{-4} up to 1.33×10^{-3}), the simulated expected delay increases from 861.86 up to 6888.12, and our theoretical results also increase up and locate rightly above the simulated ones. It can also be observed from the Fig. 5b that for the network scenario of $(n = 240, m = 16)$ there, our theoretical delay results serve as a safe upper bound. A further careful observation of the Fig. 5a and Fig. 5b indicates that when the system load ρ approaches 1 (beyond 0.8), the packet delay rises up sharply and becomes extremely sensitive to the variation of the ρ. The skyrocketing behavior of packet delay when ρ approaches 1 can also serve as an intuitive validation for the throughput capacity derived by our theoretical framework ($\mu = 1.48 \times 10^{-3}$ in Fig. 5a and $\mu = 4.64 \times 10^{-4}$ in Fig. 5b).

In order to further verify the theoretical per node throughput capacity, we examined the $P_a(f+1)$ (i.e., the probability that a packet is received by its destination after all its f copies have been distributed out) for the two network scenarios in the Fig. 5 and summarized the corresponding statistical results in the Fig. 6a and 6b, respectively. The Fig. 6a and 6b indicate clearly that for both the network scenarios, as the system load ρ increases up, the probability $P_a(f+1)$ gradually approaches 1 and thus the destination node receives nearly every packet after the source has distributed out f copies of the packet. The behavior that the $P_a(f+1)$ approaches 1 as the ρ approaches 1 proves the per node throughput capacity established in the (28) of Theorem 1.



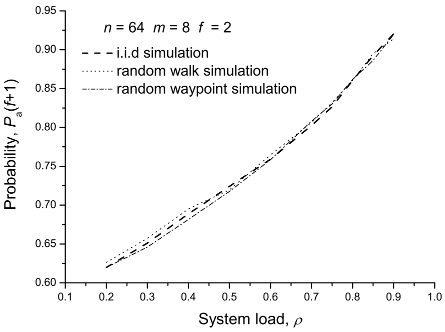
(a) Network setting $(n = 64, m = 8, f = 2)$ with per node throughput capacity $\mu = 1.48 \times 10^{-3}$ (packets/slot).



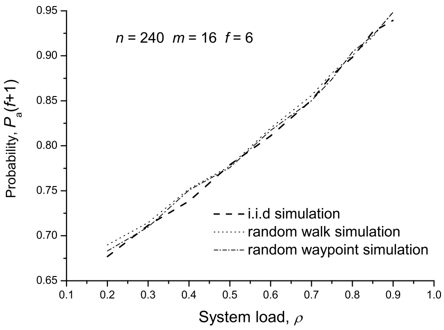
(b) Network setting $(n = 240, m = 16, f = 6)$ with per node throughput capacity $\mu = 4.64 \times 10^{-4}$ (packets/slot).

Fig. 5. Comparisons between the simulation results and the theoretical ones, where the simulation results are provided with 95% confidence intervals.

It is interesting to observe from the Fig. 5 and 6 that, under both the network settings there, regarding the expected packet delay and $P_a(f+1)$, the simulation results of random walk model and random waypoint model have very similar varying tendencies as that of the bi-dimensional i.i.d. model. A further careful observation of the Fig. 5, however, indicates that the packet delays of these three models have totally different behaviors as the ρ approaches 1. For example, when $\rho = 0.9$, the simulation result of i.i.d. model (resp. the simulation result of random walk model) has the highest (resp. lowest) packet delay in the Fig. 5a; while in the Fig. 5b, the simulation result of random waypoint model (resp. the simulation result of i.i.d. model) has the highest (resp. lowest) packet delay. More importantly, in the Fig. 5b, the simulation result of random waypoint model at $\rho = 0.9$ even rises over our theoretical

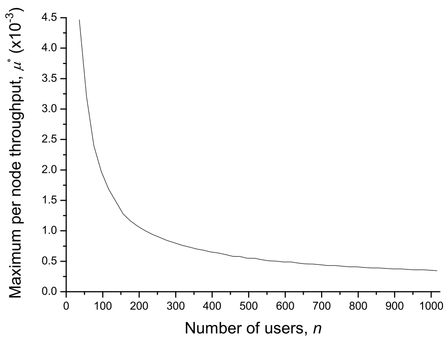


(a) Network setting $(n = 64, m = 8)$.

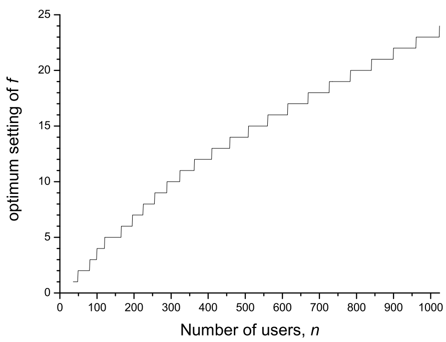


(b) Network setting $(n = 240, m = 16)$.

Fig. 6. Probability $P_a(f + 1)$ vs. system load ρ.



(a) The maximum per node throughput μ^* with $36 \le n \le 1024$.



(b) The optimum setting of f (i.e., the f_0) with $36 \le n \le 1024$, under which the maximum per node throughput μ^* is obtained.

Fig. 7. The maximum per node throughput and corresponding setting of f.

upper bound. Thus, our theoretical model developed under the i.i.d. model can not be directly applied to the random walk model and random waypoint model, for which dedicate new theoretical models are needed to characterize the throughput capacity and end-to-end packet delay there.

C. Performance Analysis

We now explore the maximum per node throughput and corresponding setting of f for a given network scenario (n, m). For the general setting of $\Delta = 1, m = \lfloor\sqrt{n}\rfloor$, we summarize in Fig. 7a how the maximum per node throughput μ^* varies as the number of users n increases from 36 to 1024. The Fig. 7a shows clearly that the μ^* vanishes quickly as n increases. Aside from the maximum per node throughput, we also report in Fig. 7b the corresponding optimal setting of f, i.e., the value of f_0. One can observe from the Fig. 7b that there does not exist a particular optimal value of f which applies to all the cases of n. Actually, an optimal setting of f only applies to a small range of n, and it is a piecewise function of n as shown in the Fig. 7b.

VI. Related Works

A. Two-hop Relay without Redundancy

Since the seminal work in [4], the performance of two-hop relay without redundancy has been extensively explored in the regime of ad hoc mobile networks. Grossglauser and Tse (2001) [4] showed that by employing the two-hop relay scheme, it is possible to achieve a $\Theta(1)$ throughput per node under the i.i.d. mobility model. Later, Gamal *et al.* [8] showed that the $\Theta(1)$ throughput is also achievable under the random walk model, but which comes at the price of a $\Theta(n \log n)$ delay. Mammen *et al.* [10] proved that the same throughput and delay scaling are also achievable even with a variant of the Grossglauser-Tse two-hop relay and a restricted mobility model.

The delay and throughput trade-off of two-hop relay in ad hoc mobile networks has also been widely studied under different mobility models. Perevalov and Blum [5] reported that under the i.i.d. mobility model, the achievable throughput increases as $d^{2/3}$ for moderate values of delay d, and increases as $\Theta(n^{-1/3})$ for a fixed delay value. Later, Gamal *et al.* [6] showed that under the 2-dimensional Brownian motion on a torus of size $\sqrt{n} \times \sqrt{n}$, the delay scales as $\Theta(n^{1/2}/v(n))$, where $v(n)$ is the velocity of mobile nodes. Lin *et al.* [7] also considered the Brownian mobility model, and showed that the $\Theta(1)$ per node throughput is achieved with an expected delay of $\Omega(\log n/\sigma_n^2)$, where σ_n^2 is the variance parameter of the Brownian motion model. Sharma *et al.* [39] showed that when the network is divided into $n^\beta \times n^\beta$ cells, the two-hop delay is $\Theta(n)$ for $0 \le \beta < 1/2$ and $\Theta(n \log n)$ for $\beta = 1/2$ under a family of discrete random direction models, while the delay becomes $\Theta(n)$ for $\beta < 1/2$ and $\Theta(n \log n)$ for $\beta = 1/2$

when a family of hybrid random walk models are considered. More recently, the delay and throughput trade-off has been examined under a correlated mobility model [11], where nodes are partitioned into different groups and all nodes belonging to the same group have to reside concurrently within a circular region around the group center.

B. Two-hop Relay with Redundancy

In the case of allowing packet redundancy, Neely and Modiano [18] considered a modified version of the Grossglauser-Tse two-hop relay algorithm for ad hoc mobile networks, and proved that under the i.i.d. mobility model it achieves $O(1/\sqrt{n})$ throughput and $O(\sqrt{n})$ delay with exact $\sqrt{n}$ redundancy for each packet. Sharma and Mazumdar explored the order sense delay and capacity trade-off in ad hoc mobile networks with multiple redundancy for each packet, and proved that it achieves $\Theta(T_p(n)\sqrt{n})$ delay under the random waypoint mobility model [19] and achieves $O(T_p(n)\sqrt{n\log n})$ delay under the Brownian mobility model [20], where $T_p(n)$ is the packet transmission time. Moraes *et al.* [21] considered an extension of the two-hop relay, where a source node broadcasts each packet once and all users within its transmission range are regarded as the relays, and showed that it can also achieve the $\Theta(1)$ throughput.

VII. Conclusion

We considered in this paper a general 2HR-f relay algorithm, where each packet is delivered to at most f distinct relay nodes and should be received in order at its destination. Theoretical models were further developed for such a network to derive the achievable per node throughput and a tight upper bound for the expected end-to-end packet delay in closed-forms. Extensive simulations by a network simulator were conducted, which verify that our theoretical model can accurately characterize the network throughput and delay performance under the 2HR-f relay. With our closed-form results and any setting of $f = n^{\delta}$, $0 \le \delta \le \log_n f_0$, one can easily derive the corresponding order sense results of throughput capacity and packet delay. For example, by setting $m = n^{\gamma}$, our theoretical models return a $\Theta(n^{\max\{-1,\ \delta+2\gamma-2\}})$ throughput and $O(n^{\min\{1,\ 2-\delta-2\gamma\}})$ delay when $0 < \gamma \le \frac{1}{2}$, and a $\Theta(n^{\delta-2\gamma})$ throughput and $O(n^{2\gamma-\delta})$ delay when $\gamma > \frac{1}{2}$ for the 2HR-f MANETs. We also explored the maximum per node throughput and determined the corresponding optimal setting of f. We found that in general the optimal setting of f varies with n, and an optimal setting of f only applies to a small range of n.

The theoretical models and closed-form results in this paper were developed mainly based on the key observation that in a real-world 2HR-f MANET, as each mobile node is not only buffer storage-limited but also energy-limited, the f should be limited to a small value ($f \le f_0$ here). Therefore, one of our future research directions is to extend the theoretical models in this paper to analyze the throughput and delay performance of a general 2HR-f MANET where $f > f_0$ or to examine the impact of node buffer space on the network throughput and delay performance. Notice that the throughput capacity and delay upper bound derived in this paper hold only for the bi-dimensional i.i.d. mobility model and the transmission-group based scheduling scheme, so our another future research direction is to develop theoretical models for other more commonly used mobility models and MAC schemes, like the random walk model, the random waypoint model, and the 802.11 DCF.

Appendix A
Proof of the Lemmas 1, 2 and 3

Proof of Lemma 1: For the concerned active cell, the contention for transmitting opportunity happens if and only if there are at least two nodes in it. Thus, such contention probability is given by

$$\begin{aligned} &1-\Big(1-\frac{1}{n}\Big)^n-\binom{n}{1}\frac{1}{n}\Big(1-\frac{1}{n}\Big)^{n-1}\\ &=1-\Big(1-\frac{1}{n}\Big)^{n-1}\Big(2-\frac{1}{n}\Big)\to 1-2\mathrm{e}^{-1} \end{aligned}$$

Regarding the contention probability for receiving opportunity, we can see that for the concerned active cell, it will not have contention for receiving opportunity only under the following cases: 1) it has no node inside; 2) it has one or two nodes inside, but its eight adjacent cells contain no node, 3) it has one node inside, and its eight adjacent cells also contain only one node. Notice that these three cases are mutually exclusive, thus the contention probability for receiving opportunity can be determined as

$$\begin{aligned} &1-\Big(1-\frac{1}{n}\Big)^n-\sum_{k=1}^{2}\binom{n}{k}\Big(\frac{1}{n}\Big)^k\Big(1-\frac{9}{n}\Big)^{n-k}\\ &\quad-\binom{n}{2}\binom{2}{1}\frac{1}{n}\cdot\frac{8}{n}\Big(1-\frac{9}{n}\Big)^{n-2}\\ &=1-\Big(1-\frac{1}{n}\Big)^n-\Big(1-\frac{9}{n}\Big)^{n-2}\Big(\frac{19}{2}-\frac{35}{2n}\Big)\\ &\to 1-\mathrm{e}^{-1}-\frac{19}{2}\mathrm{e}^{-9} \end{aligned}$$

Proof of Lemma 2: For a given time slot and a tagged flow, its source node S conducts a packet transmission iff the following three events happen simultaneously: S is in some active cell, S is selected as the transmitter, and there is at least one other node in the same cell of S or its eight adjacent cells. Consider a tagged active cell, and the probability that S is inside is $\frac{1}{\alpha^2}$. Then we can see that inside this cell, the S can be selected as the transmitter only under the following two mutually exclusive cases: the cell contains only node S; or the cell contains at least one other node aside from node S. Further notice that given there are k other nodes inside this cell (resp. the eight adjacent cells of this cell), the other $n-1-k$ nodes can be in any cell of the other m^2-1 (resp. m^2-9) cells. Summing up the probabilities under these two cases, then we have

$$\begin{aligned} p_1=\frac{1}{\alpha^2}\Big\{&\sum_{k=1}^{n-1}\binom{n-1}{k}\Big(\frac{1}{m^2}\Big)^k\Big(\frac{m^2-1}{m^2}\Big)^{n-1-k}\frac{1}{k+1}\\ &+\sum_{k=1}^{n-1}\binom{n-1}{k}\Big(\frac{8}{m^2}\Big)^k\Big(\frac{m^2-9}{m^2}\Big)^{n-1-k}\Big\} \end{aligned} \tag{37}$$

The source node S conducts a source-to-destination transmission iff the following three events happen simultaneously: S is in an active cell, S is selected as the transmitter, and the node D is either in the same cell with S or in one adjacent cell of S. Consider a tagged active cell, the S can conduct a source-to-destination transmission with the D only under the following two mutually exclusive cases: both the S and D are inside this cell; or the S is inside this cell while the D is inside the eight adjacent cells of this cell. If we further assume that aside from the node S and D, there are k other nodes inside this cell, $k \in [0, n-2]$, the probability that the node S is selected as the transmitter is $\frac{1}{k+2}$ (resp. $\frac{1}{k+1}$) under the former case (resp. under the latter case). Summing up the probabilities under these two cases, then we have

$$\begin{aligned} p_2 &= \frac{1}{\alpha^2}\Big\{\sum_{k=0}^{n-2}\binom{n-2}{k}\Big(\frac{1}{m^2}\Big)^k\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{1}{m^2(k+2)} \\ &\quad + \sum_{k=0}^{n-2}\binom{n-2}{k}\Big(\frac{1}{m^2}\Big)^k\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{8}{m^2(k+1)}\Big\} \\ &= \frac{1}{\alpha^2}\Big\{\sum_{k=0}^{n-2}\binom{n-1}{k+1}\Big(\frac{1}{m^2}\Big)^{k+1}\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{1}{k+2} \\ &\quad - \sum_{k=0}^{n-2}\binom{n-2}{k+1}\Big(\frac{1}{m^2}\Big)^{k+1}\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{1}{k+2} \\ &\quad + \sum_{k=0}^{n-2}\binom{n-2}{k}\Big(\frac{1}{m^2}\Big)^{k+1}\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{8}{k+1}\Big\} \\ &= \frac{1}{\alpha^2}\Big\{\frac{9-m^2}{n-1}+\frac{m^2}{n}-\frac{8}{n-1}\Big(\frac{m^2-1}{m^2}\Big)^{n-1} \\ &\quad + \Big(\frac{m^2}{n-1}-\frac{m^2}{n}\Big)\Big(\frac{m^2-1}{m^2}\Big)^{n}\Big\} \end{aligned} \tag{38}$$

Similarly, the S conducts a source-to-relay or relay-to-destination transmission iff the following four events happen simultaneously: S is in an active cell, S is selected as the transmitter, there is at least one other node (except S and D) in the same cell of S or its eight adjacent cells, and the node D is in one of the other m^2-9 cells. Consider a tagged active cell, the probability that the D is in one of the other m^2-9 cells (excluding this cell and its eight adjacent cells) is $\frac{m^2-9}{m^2}$. Further notice that the node S can conduct a source-to-relay or relay-to-destination transmission with some other node only under the following two mutually exclusive cases: this cell contains only node S; or this cell contains at least one other node aside from node S. Further notice that if we assume that there are k ($k \in [1, n-2]$) other nodes inside this cell (resp. the eight adjacent cells of this cell), the other $n-2-k$ nodes can be in any cell of the other m^2-1 (resp. m^2-9) cells. Summing up the probabilities under these two cases, then we have

$$\begin{aligned} p_3 &= \frac{m^2-9}{m^2\alpha^2}\Big\{\sum_{k=1}^{n-2}\binom{n-2}{k}\Big(\frac{1}{m^2}\Big)^k\Big(\frac{m^2-1}{m^2}\Big)^{n-2-k}\frac{1}{k+1} \\ &\quad + \sum_{k=1}^{n-2}\binom{n-2}{k}\Big(\frac{8}{m^2}\Big)^k\Big(\frac{m^2-9}{m^2}\Big)^{n-2-k}\Big\} \end{aligned} \tag{39}$$

After some basic algebraic operations, the (2), (3) and (4) can be easily derived from (37), (38) and (39), respectively.

Proof of Lemma 3: Given that there are already j copies of packet P_h inside the network, we know that the source node S has distributed $j-1$ replicas of the packet to $j-1$ distinct relay nodes. Suppose that in the next time slot, the destination node D will directly receive P_h from S with probability $p_{s\to t}(j)$, and receive P_h from some relay, say node R, with probability $p_{r\to t}(j)$. Then we have

$$p_{s\to t}(j) = p_2 \tag{40}$$

Notice that the D will receive P_h from relay R iff the following six events happen simultaneously: R is in an active cell, R is selected as the transmitter, the destination node of the flow originated from R is not in the one-hop neighbor of R, D is in the one-hop neighbor of R, D is selected as the receiver, and the R chooses to conduct a relay-to-destination transmission. Thus, the $p_{r\to t}(j)$ can be determined as

$$\begin{aligned} & p_{r\to t}(j) \\ &= \phi_1\sum_{t=0}^{n-3}\binom{n-3}{t}\sum_{k=0}^{t}\binom{t}{k}\Big(\frac{1}{m^2}\Big)^{k+1}\Big(\frac{8}{m^2}\Big)^{t-k} \\ &\qquad \cdot\Big(1-\frac{9}{m^2}\Big)^{n-3-t}\frac{1}{t+1}\Big(\frac{1}{k+2}+\frac{8}{k+1}\Big) \qquad (41) \\ &= \phi_1\frac{1}{n-2}\Big(\frac{m^2}{n-1}-\frac{m^2}{n-1}\Big(1-\frac{1}{m^2}\Big)^{n-1} \\ &\qquad -\Big(1-\frac{9}{m^2}\Big)^{n-2}\Big) \\ &= \frac{p_3}{2(n-2)} \end{aligned} \tag{42}$$

where $\phi_1 = \frac{1}{2\alpha^2}(1-\frac{9}{m^2})$.

Notice that in the next time slot, the D may receive packet P_h either from S or from one of its $j-1$ relays, and these events are mutually exclusive, so we have

$$P_r(j) = p_{s\to t}(j) + \sum_{i=1}^{j-1} p_{r\to t}(j) \tag{43}$$

After substituting (40) and (42) into (43), the (5) follows.

According to the source-to-relay transmission in the Procedure 2, a relay node is randomly selected from the one-hop neighbors of node S, so the S can successfully deliver out a new copy of packet P_h iff a node other than these $j-1$ relay nodes that have already received copies of P_h is selected as receiver in the source-to-relay transmission. Thus, we have

$$\begin{aligned} P_d(j) &= \phi_1\sum_{k=1}^{n-2}\binom{n-2}{k}\sum_{i=0}^{k}\binom{k}{i}\Big(\frac{1}{m^2}\Big)^i\Big(\frac{8}{m^2}\Big)^{k-i} \\ &\qquad \cdot\Big(1-\frac{9}{m^2}\Big)^{n-2-k}\frac{1}{i+1}\frac{n-j-1}{n-2} \\ &= \frac{n-j-1}{2(n-2)}\cdot p_3 \end{aligned} \tag{44}$$

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