Mobility Increases the Capacity of Ad Hoc Wireless Networks

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***Abstract*—The capacity of ad hoc wireless networks is constrained by the mutual interference of concurrent transmissions between nodes. We study a model of an ad hoc network where n nodes communicate in random source–destination pairs. These nodes are assumed to be mobile. We examine the per-session throughput for applications with loose delay constraints, such that the topology changes over the time-scale of packet delivery. Under this assumption, the per-user throughput can increase dramatically when nodes are mobile rather than fixed. This improvement can be achieved by exploiting a form of *multiuser diversity* via packet relaying.**

***Index Terms*—Ad hoc networks, capacity, mobility, multiuser diversity.**

I. Introduction

A FUNDAMENTAL characteristic of mobile wireless networks is the time variation of the channel strength of the underlying communication links. Such time variation occurs at multiple time scales and can be due to multipath fading, path loss via distance attenuation, shadowing by obstacles, and interference from other users. The impact of such time variation on the design of wireless networks permeates throughout the layers, ranging from coding and power control at the physical layer to cellular handoff and coverage planning at the networking layer.

An important means to cope with the time variation of the channel is the use of *diversity*. Diversity can be obtained over time (interleaving of coded bits), frequency (combining of multipaths in CDMA systems), and space (multiple antennas or multiple base stations). The basic idea is to improve performance by creating several independent signal paths between the transmitter and the receiver.

These diversity modes pertain to a point-to-point link. Recent results point to another form of diversity, inherent in a wireless network with multiple users. This *multiuser diversity* is best motivated by an information theoretic result of Knopp and Humblet [8]. They focused on the uplink in the single cell, with multiple users communicating to the base station via time-varying fading channels. To maximize the total information theoretic capacity, they showed that the optimal strategy is to schedule at any one time only the user with the best channel to transmit to the base station. Diversity gain arises from the fact that, in a system with many users whose channels vary *independently*, there is likely to be a user with a very good channel at any one time. Overall system throughput is maximized by allocating at any time the common channel resource to the user that can best exploit it. Similar results can be obtained for the downlink from the base station to the mobile users [11].

Strategies of this type incur additional delay, because packets have to be buffered until the channel becomes strong relative to other users. Therefore, the time scale of channel fluctuations that can be exploited through multiuser diversity is limited by the delay tolerance of the user or application. For example, for applications that can tolerate delays on the order of fractions of seconds to several seconds, short time-scale fading due to constructive and destructive interference of multiple signal paths can be taken advantage of. In this paper, the focus is on applications that are so asynchronous in nature that they can tolerate end-to-end delays of minutes or even hours. On such a long time-scale, even more diversity gain can be obtained because the *network topology* changes significantly over time due to user mobility. Examples of such applications include electronic mail, database synchronization between a mobile terminal and a central database, and certain types of event notification.

We demonstrate in this paper that these ideas have ramifications to the design of wireless networks beyond classical cellular architectures. We will focus on mobile ad hoc networks that have no fixed base stations and with multiple pairs of users wanting to communicate with each other. Gupta and Kumar [6] proposed a model for studying the capacity of *fixed* ad hoc networks, where nodes are randomly located but are immobile. Each source node has a random destination in the network to which it wants to communicate. Every node in the network acts simultaneously as a source, a destination for some other node, as well as relays for others' packets. The main result shows that as the number of nodes per unit area n increases, the throughput per source-to-destination (S–D) pair decreases approximately like $1/\sqrt{n}$. This is the best performance achievable even allowing for optimal scheduling, routing, and relaying of packets in the networks and is a somewhat pessimistic result on the scalability of such networks, as the traffic rate per S–D pair actually goes to zero.

In this paper, we introduce mobility into the model and consider the situation when users move independently around the network. Our main result shows that the average long-term throughput per S–D pair can be kept *constant* even as the number of nodes per unit area n increases. This is in sharp

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contrast to the fixed network scenario and the dramatic performance improvement is obtained through the exploitation of the time variation of the users' channels due to mobility. We observe that our result implies that, at least in terms of growth rate as a function of n, there is no significant loss in throughput per S–D pair when there are many nodes in the network as compared to having just a single S–D pair. A caveat of this result is that the attained long-term throughput is averaged over the time-scale of node mobility and, hence, delays of that order will be incurred.

In the fixed ad hoc network model, the fundamental performance limitation comes from the fact that long-range direct communication between many user pairs is infeasible due to the excessive interference caused. As a result, most communication has to occur between nearest neighbors, at distances of order $1/\sqrt{n}$, with each packet going through many other nodes (serving as relays) before reaching the destination. The number of hops in a typical route is of order $\sqrt{n}$. Because much of the traffic carried by the nodes are relayed traffic, the actual useful throughput per user pair has to be small.

With mobility, a seemingly natural strategy to overcome this performance limitation is to transmit only when the source and destination nodes are close together, at distances of order $1/\sqrt{n}$. This is reminiscent of the Infostation architecture [4], where users connect to the infostations only when they are close by. However, this strategy turns out to be too naive in the present situation. The problem is that the fraction of time two nodes are nearest neighbors is too small, of the order of $1/n$. Instead, our strategy is for each source node to split its packet stream to as many different nodes as possible. These nodes then serve as mobile relays and whenever they get close to the final destination, they hand the packets off to the final destination. The basic idea is that since there are many different relay nodes, the probability that at least one is close to the destination is significant. On the other hand, each packet goes through at most one relay node and, hence, the throughput can be kept high. Although the basic communication problem is point-to-point, this strategy effectively creates multiuser diversity by distributing packets to many different intermediate nodes that have independent time-varying channels to the final destination.

II. Model

The ad hoc network consists of n nodes all lying in the disk of unit area (of radius $1/\sqrt{\pi}$). The location of the ith user at time t is given by $X_i(t)$. Nodes are mobile, and we assume that the process $\{X_i(\cdot)\}$ is stationary and ergodic with stationary distribution uniform on the disk; moreover, the trajectories of different users are independent and identically distributed (i.i.d.).

We now describe the session model. We assume that each of the n nodes is a *source* node for one session and a *destination* node for another session. Let us stipulate that the source node i has data intended for destination node $d(i)$. We assume that each source node has an infinite stream of packets to send to its destination. The S–D sassociation does not change with time, although the nodes themselves move.

We next describe the transmission model. At (slotted) time t, let $P_i(t)$ be the transmit power of node i and $\gamma_{ij}(t)$ be the channel gain from node i to node j, such that the received power at node j is $P_i(t)\gamma_{ij}$. At time t, node i transmits data at rate R packets/s to node j if

$$\frac{P_i(t)\gamma_{ij}(t)}{N_0 + \frac{1}{L}\sum_{k\neq i} P_k(t)\gamma_{kj}(t)} > \beta \tag{1}$$

where β is the signal-to-interference ratio (SIR) requirement for successful communication, N_0 is the background noise power, and L is the *processing gain* of the system. For a narrowband system $L = 1$, while for a spread-spectrum CDMA system L is larger than 1. In this paper, we only consider large-scale path loss characteristics in the fading channel model. The channel gain is given by

$$\gamma_{ij}(t) := \frac{1}{|X_i(t) - X_j(t)|^\alpha}$$

where α is a parameter greater than 2.

Packets can be transmitted directly from a source to its destination or they can go through one or more other nodes serving as relays. We assume each node has an infinite buffer to store relayed packets. At any time t, a *scheduler* chooses which nodes will transmit packets, which packets they will transmit, and the power levels $P_i(t)$ at which the packets are transmitted from node i. Note that the scheduler implicitly specifies a relay policy, as the scheduled transmissions can be from source to destination, source to relay, relay to relay, or relay to destination.

The objective of the scheduler is to ensure a high long-term throughput for each S–D pair. More precisely, consider a scheduling and relay policy π. Let $M_i^\pi(t)$ be the number of source node i packets that destination $d(i)$ receives at time t under policy π. Given the random trajectories of the users, we shall say a long-term throughput of $\lambda(n)$ is feasible if there is a policy π such that *for every* S–D pair i we have

$$\liminf_{T\to\infty} \frac{1}{T}\sum_{t=1}^{T} M_i^\pi(t) \geq \lambda(n). \tag{2}$$

We note that the throughput $\lambda(n)$ is a random quantity as it depends on the random locations of the users. The performance criterion is in terms of a throughput level common to all S–D pairs. The indexing by the system size n emphasizes that we are interested in studying the asymptotic behavior as n becomes large.

Our model basically follows the one used in [6], except that the nodes are mobile as opposed to fixed.

III. Results

A. Fixed Nodes

First, we review results of Gupta and Kumar [6]. The node positions $\{X_i\}$ are i.i.d. and uniformly distributed in the disk of unit area, but fixed over time. The destination for each source node is a randomly chosen node in the network and the destinations are all chosen independently. The following results yield upper and lower bounds on the asymptotically feasible throughput.

Theorem III-1 (Main Result 4 in [6]): There exists constants c and c' such that

$$\lim_{n\to\infty} \Pr\left\{\lambda(n) = \frac{cR}{\sqrt{n\log n}} \quad \text{is feasible}\right\} = 1$$

and

$$\lim_{n\to\infty} \Pr\left\{\lambda(n) = \frac{c'R}{\sqrt{n}} \quad \text{is feasible}\right\} = 0.$$

Thus, within a factor of $\sqrt{\log n}$, the throughput per S–D pair goes to zero like $R/\sqrt{n}$ in the case when the nodes are fixed.

This result can be intuitively understood as follows. Every bit has to travel at least the distance that separates its source from its destination. It may travel this distance either through a single direct transmission or through multiple transmissions via relay nodes.

Assume for simplicity that all transmitting nodes transmit at the same power P. Let us focus on the transmission from a node i to a node j. From (1), it is can be seen that transmission from i to j will be unsuccessful whenever there is another transmitting interferer k with distance $|X_k - X_j| \le (\beta/L)^{1/\alpha}|X_i - X_j|$. In other words, there cannot be another sender in a disk of radius proportional to the transmission distance $|X_i - X_j|$. Hence, a (successful) transmission over a distance d incurs a cost proportional to d^2 by excluding other transmissions in the vicinity of the sender i. In order to maximize the *transport capacity* of the network, i.e., the total number of meters traveled by all the bits per time unit, it is therefore beneficial to schedule a large number of short transmissions. The best we can do is to restrict transmissions to neighbors, which are at a typical distance of $1/\sqrt{n}$. The transport capacity is then at most $\sqrt{n}$ bit·m/s. As there are n sessions, each with an expected distance of $\Theta(1)$, it follows that the throughput per session can at best be $O(1/\sqrt{n})$.

B. Mobile Nodes Without Relaying

The reason why the throughput for fixed nodes goes to zero is that the number of relay nodes a packet has to go through scales as $\sqrt{n}$. However, in our model of mobile nodes, any two nodes can be expected to be close to each other from time to time. This suggests that we may be able to improve the capacity by not relaying at all and only letting sources transmit directly to destinations. We now show that without relaying, there is no way to achieve a $\Theta(1)$ throughput per S–D pair.

We first need the following lemma. This fact is already established in the proof of [6, Th. 2.1(ii)], but we include the proof here for completeness.

Lemma III-2: Consider a scheduling policy that schedules direct transmissions only. Fix an arbitrary time t. Let $\mathcal{S}(t)$ be the set of source nodes that are scheduled successful transmission to their respective destinations. Then

$$\sum_{i\in\mathcal{S}(t)} |X_i(t) - X_{j(i)}(t)|^\alpha \le B$$

where

$$B := 2^\alpha \pi^{-\alpha/2} \frac{\beta + L}{\beta}.$$

Proof: Writing down the SIR inequalities, we get for every $i \in \mathcal{S}(t)$

$$\frac{P_i(t)\gamma_{i,j(i)}(t)}{N_0 + \frac{1}{L}\sum\limits_{k\in\mathcal{S}(t),\, k\neq i} P_k(t)\gamma_{k,j(i)}(t)} \ge \beta.$$

This is equivalent to

$$\frac{P_i(t)\gamma_{i,j(i)}(t)}{N_0 + \frac{1}{L}\sum\limits_{k\in\mathcal{S}(t)} P_k(t)\gamma_{k,j(i)}(t)} \ge \frac{\beta L}{\beta + L}.$$

Substituting

$$\gamma_{ij}(t) = \frac{1}{|X_i(t) - X_j(t)|^\alpha}$$

we obtain the bound

$$|X_i(t) - X_{j(i)}(t)|^\alpha \le \frac{\beta+L}{\beta L} \frac{P_i(t)}{N_0 + \frac{1}{L}\sum\limits_{k\in\mathcal{S}(t)} \frac{P_k(t)}{|X_k(t) - X_{j(i)}(t)|^\alpha}} \tag{3}$$

$$\le \frac{\beta+L}{\beta L} \frac{P_i(t)}{N_0 + \frac{1}{L}\left(\frac{\pi}{4}\right)^{\alpha/2} \sum\limits_{k\in\mathcal{S}(t)} P_k(t)} \tag{4}$$

since $|X_k(t) - X_{j(i)}(t)| \le 2/\sqrt{\pi}$. Summing over all active S–D pairs at time t, we obtain

$$\sum_{i\in\mathcal{S}(t)} |X_i(t) - X_{j(i)}(t)|^\alpha \le \frac{\beta+L}{\beta L} \frac{\sum\limits_{i\in\mathcal{S}(t)} P_i(t)}{N_0 + \frac{1}{L}\left(\frac{\pi}{4}\right)^{\alpha/2} \sum\limits_{k\in\mathcal{S}(t)} P_k(t)} \le 2^\alpha \pi^{-\alpha/2} \frac{\beta+L}{\beta}$$

which proves the lemma upon setting

$$B := 2^\alpha \pi^{-\alpha/2} \frac{\beta + L}{\beta}.$$

■

This lemma shows that the number of simultaneous long-range communication is limited by interference. Since the distance between the source and destination is $\Theta(1)$ most of the time, this limitation in turn puts a bound on the performance of any strategy which uses only direct communication.

Theorem III-3: Assume that the policy is only allowed to schedule direct transmission between the source and destination nodes, i.e., that no relaying is permitted. If c is any constant satisfying

$$c > \left[2^\alpha \left(1 + \frac{2}{\alpha}\right) \pi^{-\alpha/2} \frac{\beta + L}{\beta}\right]^{1/(1+\alpha/2)}$$

then

$$\Pr\left\{\lambda(n) = cn^{-(1/(1+\alpha/2))} R \quad \text{is feasible}\right\} = 0$$

for sufficiently large n.

This result says that without relaying, the achievable throughput per S–D pair goes to zero at least as fast as $n^{-(1/(1+\alpha/2))}$.

Proof: We will argue by contradiction. Fix a $c > 0$ and a policy π that schedules direct transmission only, and suppose a throughput of $\lambda(n) = \mathrm{c}n^{-(1/(1+\alpha/2))}R$ is feasible. Focus on a source node i, and let $\mathcal{A}_T(i)$ be the set of time instants up until time T where node i is scheduled successful transmission to the destination $d(i)$. By definition of feasible throughputs,

$$\liminf_{T\to\infty} \frac{|\mathcal{A}_T(i)|}{T} \geq \mathrm{c}n^{-(1/(1+\alpha/2))}. \tag{5}$$

Consider the process

$$D_i(t) := |X_i(t) - X_{j(i)}(t)|^\alpha, \qquad t = 1, 2, \ldots.$$

By stationarity and ergodicity of this process, (5) implies that almost surely

$$\liminf_{T\to\infty} \frac{1}{T} \sum_{t\in\mathcal{A}_T(i)} D_i(t) \geq \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z)$$

where F is the cumulative distribution function (cdf) of the random variable $D_i(t)$. This holds for all source nodes i. Summing over all i, we have

$$\liminf_{T\to\infty} \frac{1}{T} \sum_{i=1}^{n} \sum_{t\in\mathcal{A}_T(i)} D_i(t) \geq n \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z)$$

which is equivalent to

$$\liminf_{T\to\infty} \frac{1}{T} \sum_{t=1}^{T} \sum_{i\in\mathcal{S}(t)} D_i(t) \geq n \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z).$$

Here $\mathcal{S}(t)$ is the set of source nodes which are scheduled successful transmission by the policy at time t. The last inequality in turn implies that there must exist a time τ, such that

$$\sum_{i\in\mathcal{S}(\tau)} D_i(\tau) \geq n \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z). \tag{6}$$

Conditional on $X_{j(i)}(t) = x$ in the open disk D, it holds that for $z^{1/\alpha} < |\pi^{-1/2} - x|$

$$\Pr\left\{D_i(t) < z | X_{j(i)}(t) = x\right\} = \pi z^{2/\alpha}$$

the probability that node i is within a neighborhood of radius z from node $d(i)$. Hence

$$\begin{aligned}
&\lim_{z\to 0} F(z)/z^{2/\alpha} \\
&\quad = \lim_{z\to 0} z^{-2/\alpha} \int_{x\in D} \Pr\left\{D_i(t) < z | X_{j(i)}(t) = x\right\} dx \\
&\quad = \int_{x\in D} \lim_{z\to 0} z^{-2/\alpha} \Pr\left\{D_i(t) < z | X_{j(i)}(t) = x\right\} dx = \pi
\end{aligned}$$

where the interchange of limit and integration follows from the Dominated Convergence Theorem.

Substituting this into the integral in (6), we get

$$\lim_{n\to\infty} n \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z) = \frac{c^{1+\alpha/2}}{\pi^\alpha(1+2/\alpha)}.$$

If

$$c > \left[2^\alpha \left(1 + \frac{2}{\alpha}\right) \pi^{\alpha/2} \frac{\beta + L}{\beta}\right]^{(1/(1+\alpha/2))}$$

then

$$\lim_{n\to\infty} n \int_0^{F^{-1}(\mathrm{c}n^{-(1/(1+\alpha/2))})} z\, dF(z) > B$$

where

$$B := 2^\alpha \pi^{-\alpha/2} \frac{\beta + L}{\beta}.$$

Hence, for sufficiently large n, inequality (6) contradicts Lemma III-2. For sufficiently large n, the probability that $\mathrm{c}n^{-(1/(1+\alpha/2))}R$ is a feasible throughput is zero. ■

The intuition behind this result is that if transmissions over long distances are allowed, then there are many S–D pairs that are within range; however, for the reasons discussed in the fixed-node case, interference limits the number of concurrent transmissions over long distances; the throughput is *interference limited*. On the other hand, if we constrain communication to neighboring nodes, then there is only a small fraction of S–D pairs that are sufficiently close to transmit a packet. Hence, the throughput is *distance limited*. Theorem III-3 gives the optimal throughput given these two constraints.

C. Mobile Nodes With Relaying

In the previous section, we have seen that the throughput per session decreases with n if only direct transmissions between sources and destinations are allowed. If we want to increase throughput beyond this limitation, we have to find a way to communicate only locally (to overcome the interference limitation), while making sure that there are actually enough sender–receiver pairs that have packets to transmit (to overcome the distance limitation). Direct communication does not suffice; we need to do relaying.

Theorem III-4 demonstrates that it is, in fact, possible to schedule $\Theta(n)$ *concurrent successful transmissions* per time slot with local communication. However, the question is how we should forward packets between sources and destinations such that we can make use of these transmissions. We propose to achieve this by spreading the traffic stream between the source and the destination to a large number of intermediate *relay nodes*. Each packet goes through one relay node that temporarily buffers the packet until final delivery to the destination is possible. For a source–destination pair S–D, all the other $n - 2$ nodes can serve as relay nodes. The goal is that in steady-state, the packets of every source node will be distributed across all the nodes in the network, hence ensuring that every node in the network will have packets buffered destined to every other node (except itself). This ensures that a scheduled sender–receiver pair always has a packet to send, in contrast to the case of direct transmission.

The question is how many times a packet has to be relayed in order to spread traffic uniformly to all nodes. In fact, as the node location processes $\{X_i(t)\}$ are independent, stationary, and ergodic, it is actually sufficient to *relay only once*. This is because

the probability for an arbitrary node to be scheduled to receive a packet from a source node S is equal for all nodes and independent of S. Each packet then makes two hops, one from the source to its random relay node and one from that relay node to the destination. As no packet is transmitted more than twice, the achievable total throughput is $\Theta(n)$.[1]

We now make the above argument rigorous. We first exhibit a scheduling policy π to select random sender–receiver pairs in each time slot t, such that all the pairs can successfully transmit in time slot t. We will then use this policy as a building block to achieve $\Theta(1)$ throughput per S–D pair for large n.

The scheduling policy π is as follows. Let us focus on a particular time slot t. To simplify notation, we will drop the time index t in the following discussion. We fix a *sender density* parameter $\theta \in (0, 1)$. We randomly designate $n_S = \theta n$ of the nodes as senders in each time slot and the remaining n_R nodes as *potential* receivers. Specifically, we randomly pick one out of $\binom{n}{n_S}$ equally likely partitions of the n nodes into the set of senders $\mathcal{S}$ and the set of potential receivers $\mathcal{R}$. Each sender node transmits packets to its nearest neighbor *among all nodes in* $\mathcal{R}$, using unit transmit power ($P_i = 1$). Among the n_S sender–receiver pairs, we retain those for which the interference generated by the other senders is sufficiently small that transmission is possible. Let N_t be the number of such pairs. Theorem III-4 shows that the number of feasible sender–receiver pairs N_t is $\Theta(n)$. Note that the set of sender–receiver pairs is random and that it depends only on the node locations $\{X_i\}$.

Theorem III-4: For the scheduling policy π, the expected number $\mathrm{E}[N_t]$ of feasible sender–receiver pairs is $\Theta(n)$, i.e.,

$$\lim_{n\to\infty} \frac{\mathrm{E}[N_t]}{n} = \phi > 0. \tag{7}$$

Furthermore, for two arbitrary nodes i and j, the probability that (i, j) is scheduled as a sender–receiver pair is $\Theta(1/n)$.

We can now apply this scheduling policy π to our basic problem. The overall algorithm is divided into two phases: 1) scheduling of packet transmissions from sources to relays (or the final destination (cf. Fig. 1) and 2) scheduling of packet transmissions from relays (or the source) to final destinations (cf. Fig. 2). These two phases are interleaved: in the even time-slots, phase 1 is run; in the odd time-slots, phase 2 is run.

In phase 1, we can apply the scheduling policy π to transmit packets from sources to relays or destinations. In phase 2, we again apply the policy π, but this time to transmit packets from relays to final destinations (or, as in phase 1, from a source directly to the destination). More specifically, when a receiver is identified for a sender under π, the sender checks if it has any packets for which the receiver is the destination; if so, it will transmit it. It should be noted that every packet goes through at most two hops: it is transmitted once in phase 1 from its source to an intermediate relay and once in phase 2 from a relay to the final destination. We allow for packets to be directly transmitted from their source to their destinations in both phases, if a sender–receiver pair happens to be a source–destination pair as well.

[1]It should be emphasized that packets are *not* copied at a source and sent along different two-hop routes; rather, the overall packet stream is split across the different routes.

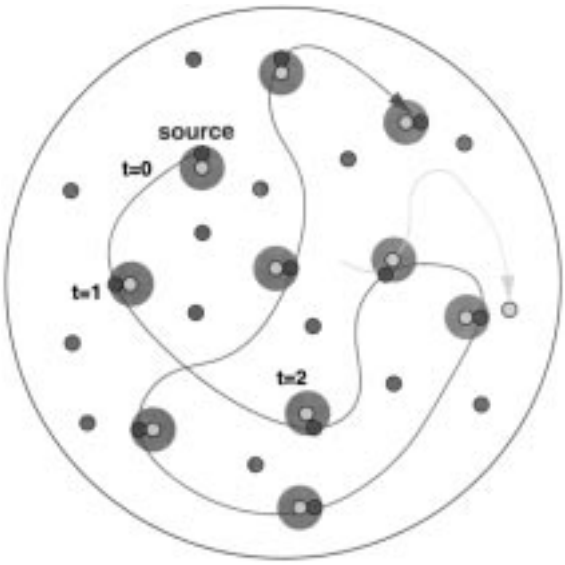

Fig. 1. In phase 1, each packet is transmitted by the source to a close-by relay node.

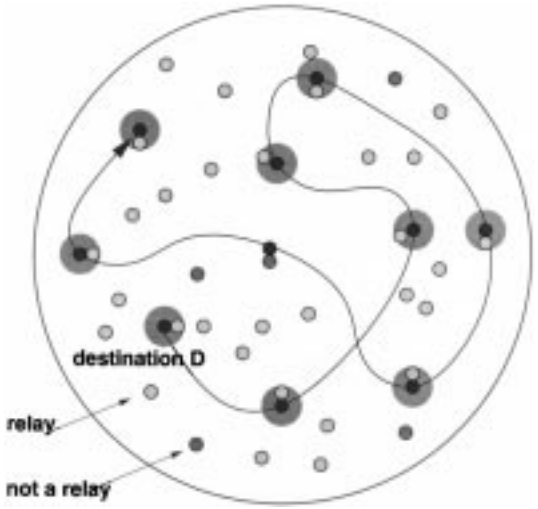

Fig. 2. In phase 2, a packet is handed off to its destination if the relay node is close by.

Let us analyze the throughput per S–D pair under this two-phased scheme. As π only depends on node locations and because the node locations $\{X_i(t)\}$ are i.i.d., stationary, and ergodic, the long-term throughput between any two nodes is equal to the probability that these two nodes are selected by π as a feasible sender–receiver pair. According to Theorem III-4, this probability is $\Theta(1/n)$. Now, for a given S–D pair, there is one direct route and $n - 2$ two-hop routes which go through one relay node R. The throughput over the direct route is $\Theta(1/n)$. For each two-hop route, we can consider the relay node R as a single server queue (cf. Fig. 3). Applying Theorem III-4, we see that both the arrival rate and the service rate of this queue is the same and $\Theta(1/n)$. Summing over the throughputs of all the $n - 1$ routes, it can be seen that the total average throughput per S–D pair is $\Theta(1)$. We have proved the following theorem, which is the main result of this paper.

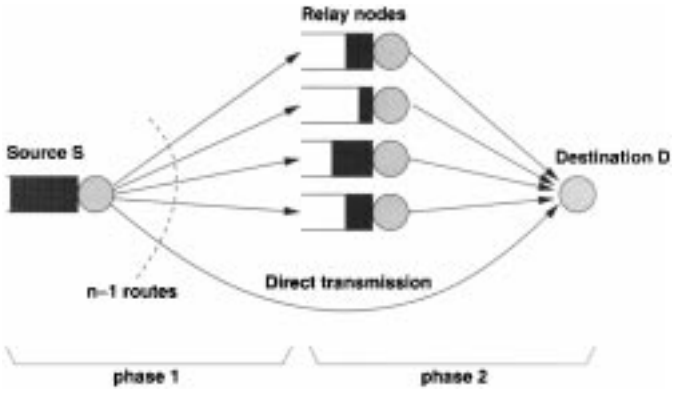

Fig. 3. The two-phase scheduling policy viewed as a queuing system, for a source–destination pair: in phase 1, a packet at S is served by a queue of capacity $\Theta(1)$ and is forwarded either to the destination or to one of $n-2$ relay nodes with equal probability. The service rate at each relay node R is $\Theta(1/n)$, for a total session rate of $\Theta(1)$.

Theorem III-5: The two-phased algorithm achieves a throughput per S–D pair of $\Theta(1)$, i.e., there exists a constant $c > 0$ such that

$$\lim_{n\to\infty} \Pr\{\lambda(n) = cR \quad \text{is feasible}\} = 1.$$

Note that the largest possible throughput is $c = \phi/2$. We now prove Theorem III-4.

Proof: We consider a fixed time t. Let $U_1, \ldots U_{n_S}$ be the random positions of the senders in $\mathcal{S}$. Let $V_1, \ldots, V_{n_R}$ be the positions of nodes in the receiver set $\mathcal{R}$. These random variables are i.i.d. uniformly distributed on the open disk of unit area. For each node $s \in \mathcal{S}$, let its intended receiver $r(s) \in \mathcal{R}$ be the node that is nearest to s among all nodes in $\mathcal{R}$.

We now analyze the probability of successful transmission for each chosen sender–receiver pair. By symmetry, we can just focus on one such pair, say $(1, r(1))$. The event of successful transmission depends on the positions $U_1, \ldots U_{n_S}$ and $V_1, \ldots, V_{n_R}$. Let Q_i be the received power from sender node i at receiver node $r(1)$, and

$$Q_i = |U_i - V_{r(1)}|^{-\alpha}.$$

The node $r(1)$ satisfies

$$r(1) = \arg\min_j |U_1 - V_j|.$$

The total interference at node $r(1)$ is given by $I = \sum_{i\neq 1} Q_i$. The SIR for the transmission from sender 1 at receiver $r(1)$ is given by

$$\text{SIR} = \frac{Q_1}{N_0 + \frac{1}{L} I}.$$

We now analyze the asymptotics of Q_1 and I as $n \to \infty$. Now

$$Q_1 = \max_{j=1,\ldots,n_R} Z_j$$

where $Z_j = |U_1 - V_j|^{-\alpha}$. Let us first condition on $U_1 = u$ for some u in the open disk. A disk centered at u and of radius

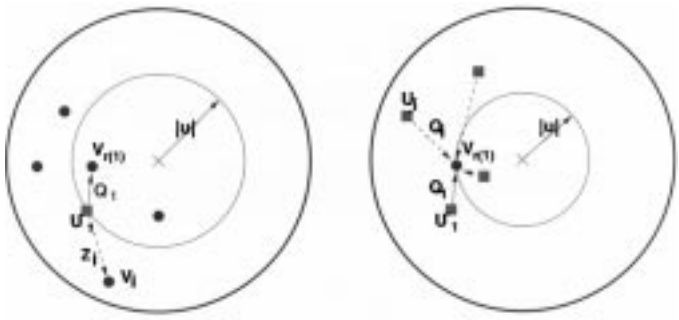

Fig. 4. An illustration of random variables used in the proof: sender location U_1, receiver location $V_{r(1)}$, received signal power Q_1, scaled distance to random receiver Z_i, and scaled interfering sender distance Q_i.

$r < (\pi^{-1/2} - |u|)$ lies entirely inside the unit disk (cf. Fig. 4). Then, for every $z > r^{-\alpha}$ and for all j, we have

$$\Pr\{Z_j > z | U_i = u\} = \Pr\left\{|V_j - u| < z^{-(1/\alpha)}\right\} = \pi z^{-(2/\alpha)}. \tag{8}$$

Conditional on $U_1 = u$, the random variables Z_j's are i.i.d. By a standard result on the asymptotic distribution of extremum of i.i.d. random variables [1, pp. 258–260], the extremum Q_1 of n_R i.i.d. random variables whose cdf satisfies

$$\lim_{n_R\to\infty} \frac{1 - F_Z(x)}{1 - F_Z(kx)} = k^b \tag{9}$$

satisfies

$$\lim_{n_R\to\infty} \Pr\{Q_1 \le a_{n_R} x\} = \exp(-x^{-b}) \tag{10}$$

where a_{n_R} is given by $F_Z^{-1}(1 - 1/n_R) = (\pi n_R)^{\alpha/2} = [(1-\theta)\pi n]^{\alpha/2}$. Thus, the asymptotic distribution of Q_1 conditional on $U_1 = u$ depends only on the tail of the distribution of the Z_j's and is given by

$$\lim_{n\to\infty} \Pr\{Q_1 < a_{n_R} x | U_1 = u\} = F_{Q_\alpha^*}(x) \tag{11}$$

where Q_α^* has a cdf

$$F_{Q_\alpha^*}(x) = \begin{cases} \exp(-x^{-2/\alpha}), & x \ge 0 \\ 0, & x < 0. \end{cases}$$

Hence, for every $x > 0$, we have

$$\begin{aligned} &\lim_{n\to\infty} \Pr\{Q_1 < a_{n_R} x\} \\ &\quad = \lim_{n\to\infty} \int_{u\in D} \Pr\{Q_1 < a_{n_R} x | U_1 = u\}\, du \\ &\quad = \int_{u\in D} \lim_{n\to\infty} \Pr\{Q_1 < a_{n_R} x | U_1 = u\}\, du = F_{Q^*{}_\alpha}(x). \end{aligned}$$

The interchange of limit and integration follows from the Dominated Convergence Theorem. We conclude that

$$[(1-\theta)\pi n]^{-\alpha/2} Q_1 \xrightarrow{\mathcal{D}} Q_\alpha^*. \tag{12}$$

We now turn to the interference $I = \sum_{i=2}^{n_S} Q_i$. Conditional on $V_{r(1)} = u$, we observe that for $i \neq 1$, $Q_i's$ are i.i.d. and have the same distribution as the Z_is conditional on $U_1 = u$. Hence, the distribution of Q_i conditional on $V_{r(1)} = u$ has the same tail as given in (8). From the theory of stable random variables [3, pp. 448, Th. 2], it follows that, conditional on $V_{r(1)} = u$,

$$\left[\pi\Gamma\left(1-\frac{2}{\alpha}\right)n_S\right]^{-\alpha/2} I = \left[\pi\Gamma\left(1-\frac{2}{\alpha}\right)\theta n\right]^{-\alpha/2} I \xrightarrow{\mathcal{D}} I_\alpha^* \tag{13}$$

where I_α^* has the stable distribution with characteristic exponent $2/\alpha$ and does not depend on u.

Again, the asymptotic limit above depends only on the tail of the conditional distributions of the individual Z_is, which does not depend on u. Using a similar argument as above for Q_1, we conclude that (13), in fact, holds unconditionally.

Finally, we claim that the signal power Q_1 and the total interference I are asymptotically independent (although they are in general not independent for finite n.). The argument is as follows. Equation (13) implies that the total interference I is asymptotically independent of $V_{r(1)}$, since the limiting distribution of I conditional on $V_{r(1)} = u$ does not depend on u. Note also that, conditional on $V_{r(1)}$, U_1 and I are independent. Hence, in fact, I is asymptotically independent of the pair $(U_1, V_{r(1)})$. But the signal power Q_1 is a continuous function of U_1 and $V_{r(1)}$ and, hence, by the Continuous Mapping Theorem, I and Q_1 are asymptotically independent.

Combining this last fact with (12) and (13), we get the result on the probability of successful transmission from node 1 to node $r(1)$

$$\begin{aligned}\Pr\{\mathrm{SIR} > \beta\} &= \Pr\left\{\frac{Q_1}{N_0 + \frac{1}{L}I} > \beta\right\} \\ &\to \Pr\left\{\frac{Q_\alpha^*}{I_\alpha^*} > \beta^*\right\} > 0 \end{aligned} \tag{14}$$

where

$$\beta^* = \frac{\beta}{L}\left[\frac{\theta}{1-\theta}\Gamma\left(1-\frac{2}{\alpha}\right)\right]^{\alpha/2} \tag{15}$$

where $\Gamma(s) = \int_0^\infty x^{s-1}e^{-x}\,dx$ is the standard gamma function. The last inequality follows from the fact that Q_α^* and I_α^* can be chosen to be independent and Q_α^* has infinite support.

Therefore, as there are $n_S = \theta n$ senders attempting to transmit, the expected number of feasible sender–receiver pairs is $\mathrm{E}[N_t] = \theta n \cdot \Pr\{\mathrm{SIR} > \beta\}$, i.e., $\phi = \theta \cdot \Pr\{\mathrm{SIR} > \beta\}$. Furthermore, as π only depends on node locations, and as the node locations $\{X_i\}$ are i.i.d., the probability of success of any specific sender–receiver pair is equal and, thus, $\Theta(1/n)$. This completes the proof. ■

The essence of the proof of Theorem III-4, and the fundamental reason why we can have $\Theta(n)$ concurrent nearest neighbor transmission, is the fact that the received power at the nearest neighbor is of the same order as the total interference from $\Theta(n)$ number of interferers. A similar phenomenon has been observed by Hajek *et al.* [7] in the cellular setting, where they have shown that, provided $\alpha > 2$, the capture probability of the nearest transmitter to the base station does not go to zero as the number of interferers become large. A similar result has also been obtained by Shepard [10]. Although these results may seem surprising on first sight, they are all based on the following property: if $W_1, \ldots, W_n$ are i.i.d. random variables such that the cdf $F(w)$ decays slower than w^{-1} as $w \to \infty$, then the largest of them is of the same order as the sum. In the context of our problem, W_is are the received powers from the transmitting nodes.

The technical complication in the proof of Theorem III-4 is due to the fact that both the distribution of the received power from the sender and the distribution of the interference depends on the location of the receiver. This is primarily due to the edge effects of the disk, and this dependency would not be present if, for example, the nodes are randomly located on the surface of a sphere. Fortunately, in the regime we are interested in, the asymptotic distributions depends only on what happens in the local neighborhood around the receiver, and this is independent of where the receiver is in the open disk.

Our channel model considers only large-scale path-loss characteristics (power decay with distance), but does not include multipath fading or shadowing effects. Hajek *et al.* [7] showed that the limiting probability of capture in their problem depends only on the roll-off exponent α, but not on these other channel effects even when they are included. While their results are not directly applicable to our setting, we nevertheless believe that this robustness property to other channel effects carries over.

D. Distributed Implementation

Although in our problem formulation we allow for central coordinated scheduling, relaying, and routing, it should be noted that the algorithm obtained in the constructive proof above can be implemented in a completely distributed manner. At each time instant, each node can randomly and independently decide whether it wants to be a sender or a potential receiver. Each sender then seeks out a potential receiver nearest to it, and attempts to send data to it. In an even phase, senders only forward packets from sources to relays, and in an odd phase, they only forward packets from relays to destinations. The access is uncoordinated; in fact, multiple senders may attempt to transmit to the same receiver. Whether a sender is "captured" is a random event, much like standard MAC random access protocols. What our analysis showed is that the probability of success is reasonable even in a network with many users.

Note that the two-phased algorithm used in the proof was chosen for mathematical convenience. As the capacity in both phases is identical, the expected delay experienced by a packet from source to destination would actually be infinite even for a finite number of nodes n if the capacity of the first phase is used fully. It is straightforward to fix this problem, e.g., by allowing both source-to-relay (S–R) and relay-to-destination (R–D) transmissions to occur concurrently, but giving absolute priority to R–D (phase 2) transmission in all scheduled sender–receiver pairs. A detailed study of local scheduling strategies and their impact on end-to-end delay is the subject of future work.

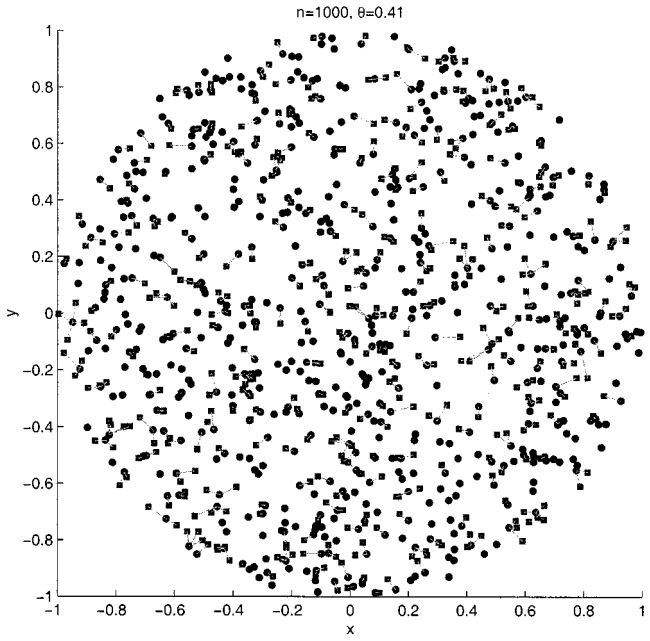

Fig. 5. Example of a random topology with $n = 1000$ nodes for sender density $\theta = 0.41$. Senders are depicted as squares, receivers as circles. A line connects each sender to its closest receiver.

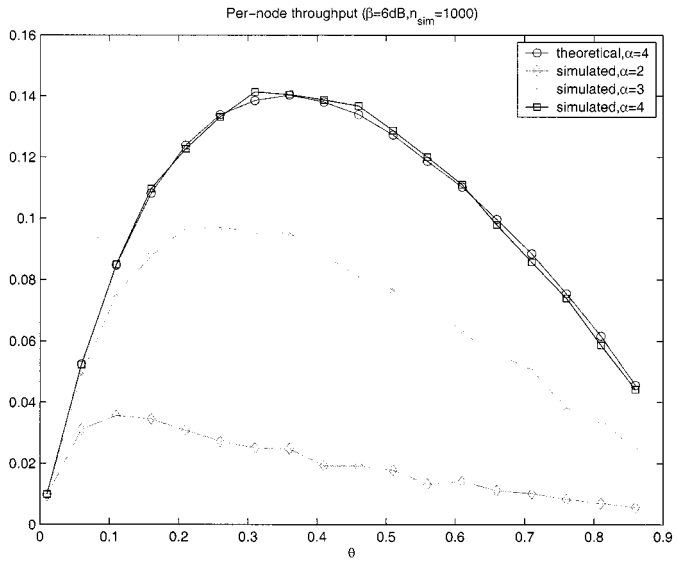

Fig. 6. The normalized per-node throughput, as a function of the sender density θ, for different values of α. For $\alpha = 4$, the throughput predicted by the model is also shown.

E. Numerical Results

We have examined the throughput capacity both through numerical evaluation of the asymptotic probability of capture developed in Section III-C, and through simulation of random network topologies.

We have evaluated the asymptotic fraction of feasible pairs ϕ for the special case $\alpha = 4$, because for this case, the normalized interference I_α^* has a Lévy distribution[2] [9], with cdf

$$F_{I_\alpha^*}(x) = 2\left[1 - Q\left(\sqrt{\frac{\sigma}{x}}\right)\right] \tag{16}$$

where $Q(.)$ is the standard Gaussian cdf, with $\sigma = 1/2$.[3] It is therefore straightforward to numerically evaluate (14) through Monte-Carlo simulation.

We have compared the fraction of feasible pairs ϕ for $\beta = 6$ dB and $L = 1$ predicted by our model with simulations based on $n = 1000$ nodes (cf. Fig. 5). The simulation results are averaged over 20 random topologies. Fig. 6 shows the simulated normalized throughput for $\alpha = 2$, 3, and 4, and the throughput predicted by the asymptotic model for $\alpha = 4$. There is very good agreement between the analytical model and simulation results.

It is evident from the figure that, given α, there exists an optimal sender density θ that maximizes the throughput. If θ is too small, then we do not exploit the potential for spatial channel reuse. If θ is too large, then the interference power becomes too dominant. The optimal θ obviously depends on α. For small α, interference limits the spatial channel reuse. Hence, the sender density has to be small. For large α, interference is more localized, and the optimal θ and the maximum throughput are larger.

F. Sender-Centric Versus Receiver-Centric Approach

In the proof of Theorem III-4, we have used a *sender-centric* approach, in that it is the senders that select the closest receiver to send to. We could also have considered a *receiver-centric* approach, where each receiver selects the closest sender from which to receive. It might seem that the situation is symmetric, and that a similar proof would carry through to arrive at the same result. However, this is not the case.

In the sender-centric approach, several senders may select the same receiver. This is not problematic from a technical point of view. By analogy, in the receiver-centric approach, it is possible that several receivers select the same sender. We can either assume that the sender has to select only one receiver to which to send to, or we can assume that a sender is indeed able to generate signals for several receivers. Both assumptions lead to difficulties in an analogous proof. Under the former assumption, we have to account for the elimination of sender–receiver pairs because the sender has to be unique; simple worst-case bounds can be found, but turn out to be too crude to improve upon the sender-centric capacity. Under the latter assumption, we have to account for the fact that a single sender can generate several unit-power interference signals (or analogously, the fact that the desired signal is only a fraction of unit power). We have not found an elegant way to integrate these complications into the above proof.

However, note that the receiver-centric approach is preferable in terms of the SIR for a *single* receiver. The reason is that in the receiver-centric approach, the signal from the selected sender is always the strongest. If $\{Q_i\}$ are the received powers from the n_S senders, then the received signal power is $\max(Q_i)$, while the remaining $n_S - 1$ signals are interference. On the other hand, in the sender-centric approach used in our proof, the designated receiver is selected as the maximum of an *independent* set $\{Z_i\}$

[2]There is no closed form for the distribution or density function of I_α^* for general α; only the Laplace transform of its density is known explicitly [3], [9] and is given by $\psi_{I_\alpha^*}(s) = \exp(-s^{2/\alpha})$.

[3]This can be seen by comparing the Laplace transform of the density of nonnegative strictly stable random variables in [3, p. 448] with the expression for the characteristic function of general stable random variables in [9, p. 5].

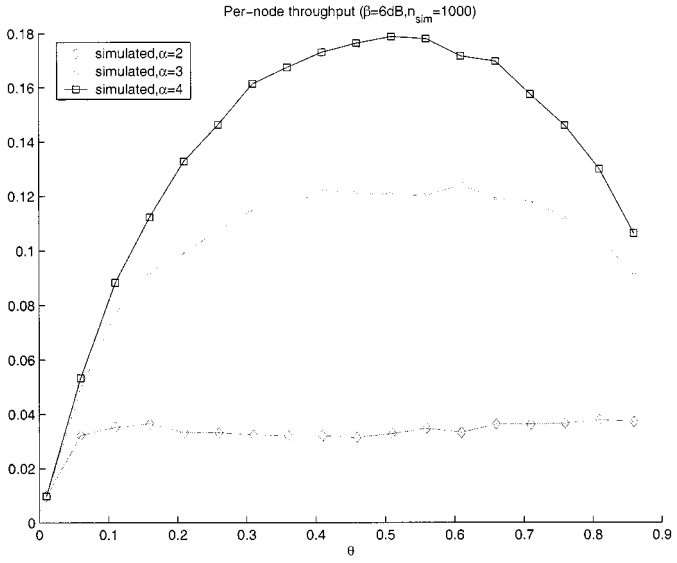

Fig. 7. The normalized per-node throughput for the *receiver-centric* case, as a function of the sender density θ for different values of α.

of n_R random variables, where Z_i has identical distribution as Q_i.[4] The received signal power is $\max(Z_i)$, and the interference power is $\sum Q_i$ (where the sum is over $n_S - 1$ terms).

Let us assume first that $\theta = 1/2$, i.e., $n_S = n_R = n/2$. The power of the received signal is the maximum of n_S i.i.d. random variables in both cases; hence, they are distributed equally. However, the interference in the receiver-centric case is stochastically smaller than in the sender-centric case: in the former, the interference is the sum of $n_S - 1$ random signal powers, whereas in the latter, it is the sum of n_S random signal powers *minus the strongest of these signals*. Therefore, the SIR for the receiver-centric approach is larger on average than in the sender-centric approach. We have simulated the normalized per-node throughput for the receiver-centric approach as shown in Fig. 7. As expected, the throughput is slightly higher than in the sender-centric approach.

IV. Discussion

The central philosophy behind this work is that the delay tolerance of applications can be usefully exploited in a mobile wireless network. This philosophy has been embodied in earlier work on the *Infostation* [4], designed for delay-tolerant data applications. An Infostation is a high-speed wireless base station that does not provide ubiquitous coverage but only allows a mobile user to communicate when the user is nearby. The motivation is that if delay is unimportant, then capacity for an user is maximized by using the entire transmit power budget when the user is close to the base station, and no power when the user is far away. This strategy is motivated by an information theoretic result on point-to-point fading channels [5].

The work on Infostation focuses on point-to-point links in isolation and aims to maximize link throughput for a given power budget. In contrast, the work presented here shifts the emphasis to the network view of *interference management* between many concurrent point-to-point links (S–D pairs).

[4]Ignoring edge effects.

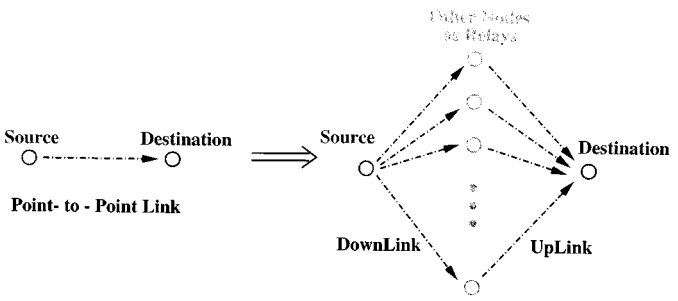

Fig. 8. How relaying can create multiuser diversity.

According to Theorem III-3, it is impossible to support a high throughput per S–D pair by direct communication even if transmission is scheduled only when sources and destinations are close by each other. Instead, this basic idea has to be combined with a two-hop relaying strategy to achieve high throughput.

Our solution exploits a form of *multiuser diversity*, and is best visualized in Fig. 8. Focusing on a specific S–D pair, the direct *point-to-point* link is a statistically poor channel, since it is only strong a small fraction of the time (when the source and destination are close by). By using all the other nodes in the network as relays, however, communication between the source and destination is now performed through two "multiuser" links: a "downlink" from the source to all the relays, and an "uplink" from the relays to the final destination. Due to a multiuser diversity effect, the throughput of the "downlink" is high: at any one time, there is likely to be a relay node close to the source, to whom the source can transmit information. Similarly the throughput of the "uplink" is also high: at any one time, there is likely to be a relay node close to the destination, from whom it can receive information. Hence, the overall throughput is much higher than that of the direct point-to-point link. This is in essence a *statistical multiplexing* effect due to the fact that there is a large number of users in the network.

It should be noted that the view of diversity here is very different from the more traditional technique of *path diversity*. In path diversity routing, copies of the same packets are forwarded along different routes to provide redundancy against uncertain channel conditions and network connectivity. In multiuser diversity routing, each packet is sent along only one route to take advantage of the closeness of the relay node.

V. Conclusion

In this paper, we have examined the asymptotic throughput capacity of large mobile wireless ad hoc networks. Our results show that direct communication between sources and destinations alone cannot achieve high throughput, because they are too far apart most of the time. We propose to spread the traffic to intermediate relay nodes to exploit the multiuser diversity benefits of having additional "routes" between a source and a destination. Two-hop routes are sufficient to achieve the maximum throughput capacity of the network within the limits imposed by the interference model. This explains the dramatic performance improvement over a fixed ad hoc network, where $\Theta(\sqrt{n})$ intermediate relay nodes are necessary.

The improvement in throughput is dramatic, but we would like to emphasize that this result is obtained under several idealistic assumptions. In particular, we assume the complete mixing of the trajectories of the nodes in the network. It would be interesting to study how much throughput can be achieved when nodes have less random mobility patterns. Recent results suggest that high throughput per S–D pair is still achievable even when the nodes' mobility is much more constrained [2]. Specifically, it was shown that if each node is restricted to move along a randomly placed line segment, the per-node throughput capacity is still $\Theta(1)$. Thus, the two-dimensional mobility pattern assumed in the present paper is not a necessary condition for the result to hold.

This paper focuses on the performance metric of *throughput* without taking into consideration *delay*. The delay experienced by the packets under the strategy proposed in this paper is large, increasing with the size of the system. As such, the result should be viewed as a theoretical one. What the theory does suggest is that for delay tolerant applications, there is ample opportunity to trade off delay and throughput by exploiting mobility. The result of this paper can be considered as an extreme point in the tradeoff, without any constraint on the delay. With a tighter delay constraint, the maximum achievable throughput must decrease. It would be interesting to characterize the optimal tradeoff between throughput and delay and to determine the kind of strategies that achieves this tradeoff.

The ideas in this paper are not very relevant to real-time applications such as voice communications. However, wireless data services are expected to grow quickly over the next few years. A subset of these services, such as email and database synchronization, do indeed possess very loose delay constraints (on the order of hours). Also, wireless devices are bound to become smaller and more pervasive in the future; they will not only be carried by humans, but integrated into physical objects (such as cars, electrical appliances, etc.) It is unlikely that the density of base-stations will keep pace, due to regulatory and environmental hurdles in deploying them. Thus, there is a clear opportunity for wireless ad hoc networks to extend the reach of wireless communication. Our results suggest that delay-tolerant applications can take advantage of node mobility to significantly increase the throughput capacity of such networks.

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