Simulcast Packet Transmission in Ad Hoc Networks

Kiung Jung and John M. Shea, *Member, IEEE*

***Abstract*—In previous work, unequal error-protection techniques have been applied to improve the throughput of a wireless communication system in which a transmission is received by several radios with different capabilities. For instance, these capabilities may correspond to differences in path loss, fading, or interference. By taking advantage of the broadcast nature of the channel, additional messages for the more-capable receivers can be included on transmissions to the less-capable receivers at very little cost (in terms of required energy at the transmitter or error probabilities at the receivers). This technique has been termed *simulcasting* or *multicast signaling*. In this paper, we consider the use of these techniques in an ad hoc network. These techniques impact the link throughput, end-to-end throughput, and network connectivity. We investigate how the choice of parameters for the simulcasting technique affects these network performance metrics. The results indicate that a properly chosen simulcasting technique can improve the link and end-to-end throughput in wireless ad hoc networks with only a slight degradation in other metrics, such as network connectivity.**

***Index Terms*—Ad hoc networks, multicast signaling, nonuniform modulation, simulcasting, unequal-error protection.**

I. Introduction

In MOST wireless ad hoc networks, a radio's ability to communicate with its neighbors often varies considerably because of differences in channel conditions, such as propagation loss and interference levels. In unicast transmissions, in which a single transmitter communicates to a single receiver, the transmitter can compensate for these variations in capability by using adaptive signaling if the channel conditions are accurately known. However, the shared channel is not necessarily used effectively because a signal that is intended for one radio may also be received by other radios in the system that have much better link conditions than the original intended receiver. We refer to such radios as *more-capable* radios. In this scenario, additional messages could be included for the more-capable radios at little expense to the original destination of the unicast transmission. Similarly, broadcast transmissions, which are intended for all of a radio's neighbors, are often required for network maintenance in ad hoc networks. Broadcast transmissions are generally ineffective in their use of the shared communication medium because the transmissions must be designed to allow reception by the least capable of a radio's neighbors. Thus, for any broadcast transmission there are often many more-capable receivers that could successfully receive additional messages that are simultaneously transmitted with the broadcast message.

The concepts behind the simultaneous transmissions schemes were originally explored in the context of broadcast channels by Cover and Bergmans [1], [2]. Pursley and Shea have previously shown that modulation and coding schemes can be modified to allow the inclusion of additional messages for more-capable receivers at very little cost to the performance at the less-capable receiver [3]–[5]. In these papers, the term *multicast signaling* is used to refer to such techniques. However, in ad hoc networks, the term *multicasting* refers to a process that is primarily associated with the network layer in which a single message is delivered to multiple destinations, not all of which are necessarily neighbors of the source radio. In this paper, we refer to our techniques as *simulcasting* to distinguish them from multicasting and to convey their ability to simultaneously transmit multiple messages to different neighboring radios of a transmitter. We have previously shown that nonuniform phase-shift-key (PSK) constellations provide a simple and effective way to convey multiple messages from a single transmitter to two receivers of different capabilities [3]–[5].

In this paper, we investigate the performance of simulcast transmissions in a wireless ad hoc network. We consider a system that uses slotted ALOHA [6]–[10] for channel access. The routing algorithm used is a minimum-hop routing algorithm that is modified to incorporate the simulcast capability. The packet selection mechanism is also modified to provide efficient use of the simulcast capability. We use nonuniform PSK to illustrate how the design of the simulcasting technique affects link throughput, end-to-end throughput, and network connectivity. We present analytical and simulation results for random topologies with a simple good/bad channel model and no mobility. The results indicate that simulcast transmission can improve the link and end-to-end throughput in wireless ad hoc networks at a small cost to network connectivity.

II. Network Model

Before developing the application of simulcasting in ad hoc networks, we first provide an overview of the network model used in this research. The network model was chosen to be as fundamentally simple as possible, while still providing insight into the effects of using simulcasting. The system is a slotted transmission system, where we assume that all radios are perfectly synchronized. The packet arrival process is modeled by a Bernoulli random process. We assume that the radios have large packet buffers. Multiple access is provided by slotted-ALOHA [8].

Our physical-layer models are also selected to avoid obscuring the effects of simulcasting among other physical-layer

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The authors are with the Wireless Information Networking Group, Department of Electrical and Computer Engineering, University of Florida, Gainesville, FL 32611-6130 USA (e-mail: jshea@ece.ufl.edu).

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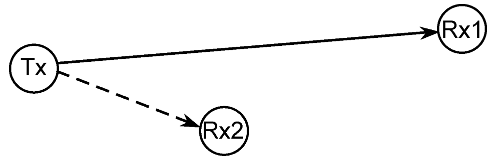

Fig. 1. Simple scenario illustrating simulcasting at the link level.

phenomena. We begin by specifying some maximum transmission range at which a basic message can be received with a target error probability. Radios are considered to be neighbors if they are within that maximum transmission range. A *packet collision* occurs whenever a radio transmits a packet during a time slot when there is also a transmission by any of the neighbors of the packet's designated recipient. We assume that signals from radios that are not neighbors can neither be received nor cause a packet collision by interfering with transmissions from a radio's neighbors. Furthermore, we assume that all collisions result in packet errors and that there is immediate and perfect feedback on packets that collided or were, otherwise, received in error. Retransmissions occur after a back-off period that is chosen according to a geometric random variable, as discussed in Section II-B.

A. Simulcast Transmission

Simulcasting has the potential to increase the average link throughput by allowing some radios to simultaneously transmit multiple packets to several different receivers, while using approximately the same network resources required to transmit a single packet with unicasting. The easiest way to visualize this is in terms of propagation distance, which generally results in lower average received energy at the more-distant receivers. This is illustrated in Fig. 1 for the case of two receivers. Suppose that the power spectral density of the noise is the same at the two receivers and the only difference in received power is due to the difference in propagation distances. Then, receiver 2 will be more-capable than receiver 1 in the sense that the higher signal-to-noise ratio at receiver 2 will allow it to successfully recover a message transmitted with a higher code rate or higher order modulation than can be successfully recovered at receiver 1. Thus, in the terminology of [3]–[5], receiver 1 is a *less-capable receiver*, and receiver 2 is a *more-capable* receiver. By using unequal error-protection modulation or coding, each time that the transmitter sends a message to receiver 1, it can include extra messages that can be recovered by receiver 2 because of its higher signal-to-noise ratio. In this case, the message intended for receiver 1 is called a *basic message*, and the messages intended for receiver 2 are called *additional messages*. We refer to these as the *class* of the message.

Simulcast transmission can be achieved in many ways but depends on the ability to achieve a different level of error protection for the basic message than for the additional messages. One simple way that this unequal error protection can be achieved is through nonuniform modulation [3], [4]. For instance, one of the simplest examples is the nonuniform quadriphase-shift key (QPSK) constellation illustrated in Fig. 2. For this constellation, the nonuniform spacing makes it much easier for a receiver to correctly recover the first bit than the second bit. Thus, the first

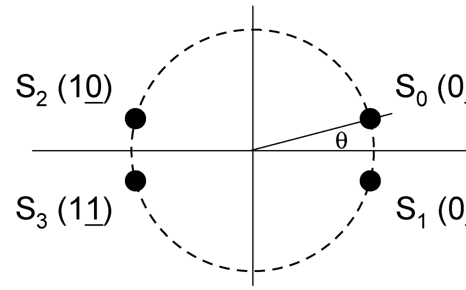

Fig. 2. Nonuniform 4-PSK that achieves different levels of error protection for each bit.

bit can be used to send a basic message that is intended for a less-capable receiver or for all of a radio's neighbors, while the second bit is used to convey an additional message that can only be recovered by more-capable receivers. Thus, this technique can be used to simultaneously send two packets in a single slot, effectively doubling the link throughput. However, the use of this or any other simulcasting technique will result in some degradation in performance at the less-capable receiver if the transmit power is unchanged. In this paper, we assume that the transmit power is fixed, and the effects of this degradation on network performance are investigated. Furthermore, we assume that the simulcasting scheme is also not adapted to the network topology; in other words, the *offset angle* θ shown in Fig. 2 is the same at all nodes in the network. For most simulcasting techniques, the performance degradation to the less-capable receivers can be made very small, while still achieving a significant gain from transmissions to more-capable receivers. Further discussion and examples are given in Section III-A.

Simulcast transmission can be achieved through a variety of other unequal error-protection techniques. These include other types of nonuniform modulation [11]–[14], unequal error-protection coding [15]–[20], combined modulation and coding schemes [5], [21]–[24], and space–time coding [25]. Any of these techniques can be used for simulcasting in ad hoc networks. However, in order to be able to demonstrate the advantages and disadvantages of simulcasting in the context of ad hoc networks, we employ the simple example of nonuniform QPSK described above for the remainder of this paper. With this scheme, each transmission can include at most two classes of message: a basic message packet and an additional message packet. All packets are assumed to be of the same length.

In [3] and [4], two important parameters are introduced that provide a simple physical-layer characterization of simulcast transmission schemes that carry only two classes of messages. The parameters are the *degradation* and the *capability disparity*. Both of these parameters are typically specified in decibels. In general, these parameters must be specified in terms of the target error probabilities for the basic and additional messages. In this work, the target error probabilities for these messages are assumed to be equal. By using a simulcast signaling scheme instead of a traditional signaling scheme that only conveys one basic message, the performance of the basic message is degraded. The degradation measures the additional amount of energy that must be received to achieve the same performance for the basic message with a simulcast signaling scheme as is

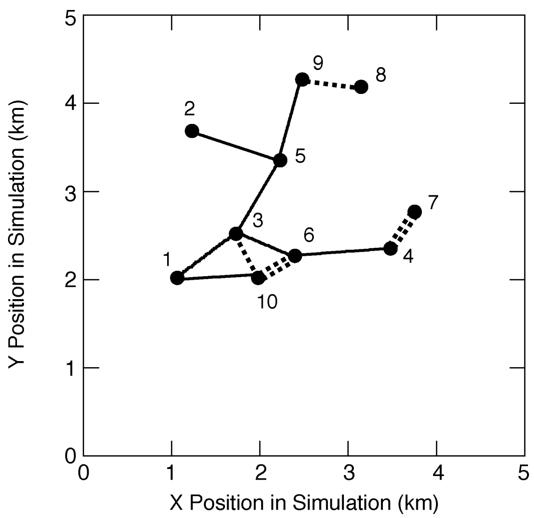

Fig. 3. Link capabilities for a ten-node wireless network. Solid lines indicate less-capable links. For degradation of 0.5 dB and disparity of 9.1 dB, all the links shown with dashed lines are more-capable links. For degradation of 0.3 dB and disparity of 11.4 dB, only the links shown with two dashed lines are more-capable links.

achieved with a traditional signaling scheme. The capability disparity, or simply *disparity*, is a measure of how much more capable a receiver must be in order to recover an additional message in comparison to a receiver that only recovers the basic message. It can be calculated as the amount of additional energy that is required at a more-capable receiver to recover the additional message at the target error probability in comparison to the amount of energy required at a less-capable receiver to recover the basic message at the target error probability. Typical values for the degradation and disparity from [4] are 0.5 and 9.1 dB, respectively.

In the context of an ad hoc network, the concepts of more-capable and less-capable receivers must be extended, as each radio may act as a transmitter or receiver at different times. When a radio is acting as a receiver, its capability level will depend on its link (channel) from the transmitting radio. Therefore, we define the radio links as being more-capable or less-capable *links*. For the results presented in this paper, we assume that the only differences in link qualities are caused by differences in propagation distance. This also implies that links are symmetric, so if the link from radio 1 to radio 2 is a more-capable link, then so is the link from radio 2 to radio 1. Radios are able to discover the capabilities of neighboring radios during network maintenance or during regular packet transmission.

An example of link map from our simulation is illustrated in Fig. 3. The diagram in Fig. 3 actually shows the link capabilities for two different values of degradation and disparity, as explained below. The maps are based on typical degradation and disparity values from [4] and exponential path loss proportional to the fourth power of distance. The figure illustrates the link capabilities for two scenarios: 1) $\theta = 19.25^\circ$, which yields a degradation of 0.5 dB and disparity of 9.1 dB and 2) $\theta = 15^\circ$, which yields a degradation of 0.3 dB and disparity of 11.4 dB. For scenario 1, the solid lines represent the less-capable links, and the dashed lines (including those links shown with two dashed lines) represent the more-capable links. For the more stringent requirement on the degradation and the higher disparity of scenario 2, the links shown with two dashed lines are the more-capable links, and all the other links are less-capable.

The use of simulcasting also causes some performance degradation to the less-capable links. For a fixed transmission power, the degradation results in the transmission range for the basic message being smaller for simulcasting than for unicasting. Thus, some links may break, which will cause two main effects to the network. First, a link may be critical to network connectivity, and when that link breaks, the network will become disconnected. Second, some routes may become longer because a node that is reachable in a single hop with unicasting may no longer be directly reachable. The increase in the length of routes will reduce the end-to-end throughput. The expected number of links that break increases as the degradation increases, while the expected number of more-capable links increases as the degradation increases (and the required disparity decreases). Thus, the simulcast signaling scheme should be designed to ensure that the increase in link throughput from having a greater number of more-capable links translates into an increase in end-to-end throughput and that the impact on network connectivity is minimal. Results on this tradeoff are given in Section IV.

As previously mentioned, we assume that the basic and additional messages require the same error probability. In fact, we consider a packet communication scheme in which any packet may be transmitted as either a basic or additional message, depending on the availability of more-capable links. The fact that a packet has been transmitted as one class of message over a link does not affect the class to which it will be assigned on later links. Thus, a packet may start out as an additional message, be transmitted as a basic message over some intermediate links, and be sent over the final link to its destination as an additional message. The only requirement that we place on the transmissions is that additional messages should be transmitted whenever possible in order to improve the network efficiency. This approach differs from the approaches in [4], [5], [11], [12], and [26], in which nonuniform signaling techniques are used to transmit different classes of multimedia messages that may have different requirements on the packet error probability.

Each simulcast transmission contains two full packets, each of which has full headers. Thus, when a radio detects a packet, it will attempt to demodulate and decode the headers for both the basic and additional message. A receiver does not need to know *a priori* whether a packet contains an additional message; if no additional message is present, the receiver will not recover a valid header for that message (typically, the CRC will fail). If neither of the packets is intended for a radio, then as usual, the radio can turn off its transceiver until the next slot to conserve energy. If either or both of the packets is intended for a radio, then they will be recovered in the usual way. We note that we assume that all nodes will listen to the headers at the beginning of each slot. If a sleep schedule is employed to conserve energy at the radios, the performance of simulcasting may be significantly degraded by a reduction in the number of receivers with more-capable links that are awake during any particular slot.

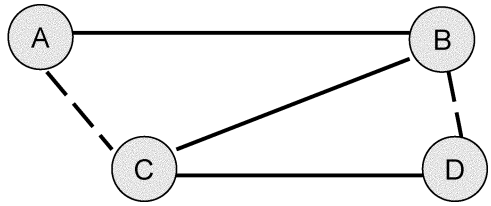

Fig. 4. Example link map for a four-node wireless network.

B. Medium-Access Control Parameters

In the system that we consider, radios contend for the channel via slotted-ALOHA. We assume that the radios have long packet buffers such that every radio will always have a packet to transmit. When a radio suffers a collision, the radio will wait a random backoff time that is selected according to a geometric distribution. When a radio is successful in transmitting, it may immediately transmit in the following slot. For the system parameters that we consider, the performance is dominated by the effects of contention. Thus, the way that the probability of retransmitting in each slot (or equivalently, the average number of slots that the system will back off after a collision) is determined can have a significant effect on the performance of the system. For the results presented in this paper, each radio uses the same retransmission probability in any slot. By adjusting this transmission probability, different average network attempt rates G can be obtained.

C. Routing Algorithm

In this paper, we consider a form of minimum-hop (min-hop) routing [27] in which the routing tables are modified to effectively utilize the capability of simulcasting. Our approach to including simulcasting in the network is designed to allow the transmission of an additional message whenever possible. As previously mentioned, we allow any packet to be sent as an additional message if an appropriate link is available. Whether a packet can be sent as an additional message at any node will depend on the packet's destination and the link capability of the next link on any minimum-hop route to that destination.

The new routing tables are a superset of the standard min-hop routing tables. The standard min-hop routing table is always used for selection of the next-hop radio for the basic message. To this routing table is added a set of *simulcast entries*. For a routing table entry to be a valid simulcast entry, it must have a first hop that is a more-capable link and it must be a minimum-hop route. It is not required that the links after the first link be more-capable links. Thus, as previously mentioned, a packet that is transmitted as an additional message over one link may be transmitted as a basic message over other links, and *vice versa*.

To illustrate the modified routing table, consider the simple four-node network shown in Fig. 4. In this figure, the more-capable links are shown as dashed lines, and the less-capable links are shown as solid lines. In Table I, we present an example routing table for radio A. The routing table is formed as follows. The simulcast entries are specified first. Note that there will be no simulcast entry for destination radio B because there is no minimum-hop route for which the next hop from A is a more-capable link. However, there are simulcast entries for destination radios C and D. Both destinations C and D can be reached in the minimum number of hops by first sending the packet over the more-capable link $A \rightarrow C$. The routing table entries for the basic messages are labeled "Normal Entries" in Table I and are selected from the possible min-hop routes in the usual way. For the results presented in this paper, the normal and simulcast routing-table entries for a particular destination are allowed to be identical, even if other min-hop routes exist.

TABLE I
ROUTING TABLE FOR RADIO A IN FIG. 4

	Destination	Next Hop	No. of Hops
Simulcast Entries:	C	C	1
	D	C	2
Normal Entries:	B	B	1
	C	C	1
	D	B	2

D. Packet-Selection Algorithm

The packet-selection algorithm should also be modified to ensure efficient use of the simulcast capability. At each time that a radio transmits, it will attempt to utilize a more-capable link if one is available. By doing so, the link throughput can be increased because two packets are sent simultaneously by a radio in a single packet transmission interval whenever possible. An important feature of the simulcasting technique is that the basic and additional messages in a transmission are not required to have the same next-hop radio. Thus, for the network illustrated in Fig. 4, radio A can simultaneously send a basic message to radio B at the same time that it sends an additional message to radio C.

The packet-selection algorithm determines which packet(s) in a radio's buffer will be transmitted in any given packet transmission interval. The packet-selection algorithm is a modified first-in–first-out (FIFO) algorithm that ensures that more-capable links are utilized whenever possible. It functions in the following way. Any radio that has any more-capable link will first try to select from its queue the first packet that can be sent as an additional message. This will not necessarily be the first packet in its queue. After the additional message (if available) is selected, then the radio will select the first packet from the remaining set of packets to be sent as a basic message. In the absence of mobility, packets intended for a particular destination will be transmitted in order, thereby minimizing the impact of simulcasting on the out-of-order arrival problem.

A brief example serves to illustrate this packet selection algorithm. Suppose that radio A's packet buffer contains four packets. Each entry in the packet buffer consists of an ordered pair (I, D), where I denotes the packet ID and D denotes the destination node. Then, the packet buffer is an ordered set of such entries. Suppose that radio A's buffer is given by

$$((1, B), (2, B), (3, D), (4, C)).$$

Then, during the first interval in which radio A transmits, it first searches its buffer for the first packet that can be sent as an additional message. To do so, it compares the destination for each packet to the set of destinations in the simulcast entries in the routing table. In this case, the first packet that can be sent as an additional message to its next-hop radio is packet 3, which, based on the simulcast entry for destination D in Table I, will be sent to next-hop radio C. Packet 1 is then selected for transmission as the basic message. So, when radio A transmits, it will simultaneously send messages to radios B and C using simulcast transmission. On radio A's next transmission, packet 4 will be selected as the additional message and packet 2 will be sent as the basic message. Note that this simulcast transmission scheme is significantly different from multicasting that occurs at the network or application layers, in which one message is distributed to a group of different receivers. In simulcasting, multiple messages are simultaneously transmitted to a one or more neighbors of the transmitting radio.

III. Throughput Analysis

In this section, we analyze the link and end-to-end throughputs for simulcasting in an ad hoc network. We consider a fixed network topology with a noise-free channel. Thus, for unicast signaling, link throughput depends on the probability of collision and the time between transmission attempts, and end-to-end throughput depends on the link throughput and the number of hops that the messages must travel. For simulcasting, these throughputs also depend on the simulcasting parameters through the number of messages that can be sent as additional messages and the changes in the number of hops, which are caused by changes in the maximum transmission distance for the basic messages.

The analysis that follows is for two scenarios. In the first, the network topology is fixed, which allows the topological parameters to be easily calculated. In the second, N radios are uniformly distributed over an area A. Except as noted, edge effects are neglected in the calculations.

A. Network Parameters

We begin by considering the effects of simulcasting on the network topology. In particular, we consider two network parameters that have a significant effect on the link and end-to-end throughputs. The first is the network degree N_{deg}, which is the average number of neighbors of a radio. Here, we define a neighbor as any radio that is directly connected to the radio of interest by either a less-capable or more-capable link. The second parameter is R_m, the proportion of radios with more-capable links.

For the noise-free channel, whether two radios are neighbors depends only on the signal-to-noise ratio at one of the radios when the other transmits to it. Since the only factor that affects the signal-to-noise ratio is exponential path loss (no random fading or shadowing is considered), whether two radios share a link depends only on the distance between the radios. Let d_U denote the maximum link distance for unicast signaling, and let $d_\ell(\theta)$ and $d_m(\theta)$ denote the maximum link distances for less- and more-capable links, respectively. These distances can be calculated as

$$d_\ell(\theta) = d_U 10^{-\frac{D(\theta)}{(10n)}}$$

and

$$d_m(\theta) = d_\ell 10^{-\frac{\delta(\theta)}{(10n)}}$$

where $D(\theta)$ and $\delta(\theta)$ are the degradation and disparity (both in decibels), respectively. For nonuniform QPSK, these simplify to

$$d_\ell(\theta) = d_U \left[\cos(\theta)\right]^{\frac{2}{n}}$$

and

$$d_m(\theta) = d_U \left[\sin(\theta)\right]^{\frac{2}{n}}$$

where n is the path-loss exponent. We take $n = 4$ in all that follows.

It is interesting to consider the coverage area of a transmitter, which is defined as the area of the region in which other radios will have a link to that transmitter. Then, the coverage areas for the basic and additional messages are given by πd_ℓ^2 and πd_m^2, respectively. Consider the coverage area as a proportion of the coverage area for unicasting, πd_U^2. For $\theta = 10°$, the proportions of coverage for the basic and additional messages are given by 0.985 and 0.174, respectively. Thus, for a reduction in coverage area of 1.5%, 17.4% of the coverage area supports more-capable links. If $\theta = 20°$, the coverage area for the basic message is 6.0% less than that of unicasting, but the coverage area for the additional message is increased to 34.2% of unicasting. Thus, by appropriately choosing θ, the coverage area for additional message transmission can be made reasonably large without significantly reducing the coverage area for the basic message.

Consider an arbitrary node in a network with nodes uniformly distributed over area A. Let p_ℓ and p_m denote the probabilities that some other node is connected to that node by a less-capable link and more capable link, respectively. Then, $p_\ell(\theta) \approx \pi[d_\ell(\theta)]^2/A$, and $p_m(\theta) \approx \pi[d_m(\theta)]^2/A$, where the approximations come from ignoring the edge effects of the finite area over which the nodes are placed. Then, the network degree is given by

$$N_{\text{deg}}(\theta) = (N-1)p_\ell(\theta) \quad (1)$$

$$\approx (N-1)\frac{\pi\left[d_\ell(\theta)\right]^2}{A^2} \quad (2)$$

$$= N_{\text{deg}}(0)\cos(\theta). \quad (3)$$

For the simulation results in Section IV, the transmission distance is close to the dimension of the simulation area, so the edge effect makes (1) and (2) yield inaccurate estimates if $p_\ell(\theta)$ and $d_\ell(\theta)$ are determined as specified above. However, we find that (3) gives a good approximation if the correct value of $N_{\text{deg}}(0)$ is found via simple topological simulation; therefore, we use this approach for the results in Section IV. The proportion of radios with more-capable neighbors can also be simply calculated by

$$R_m(\theta) = 1 - \left[1 - p_m(\theta)\right]^{N-1}. \quad (4)$$

For the network topology illustrated in Fig. 3, $R_m = 7/10$ for $D = 0.5$ dB ($\theta = 19.25°$), and $R_m = 4/10$ for $D = 0.3$ dB ($\theta = 15°$).

B. Link Throughput

We apply the conventional techniques for link-throughput analysis of slotted ALOHA [8]. Let $E[D_i]$ be the expected value of the delay (in terms of number of slots) required for a packet transmitted by radio i to be successfully received by the designated next-hop radio. Then, the *link throughput* at radio i,

S_i, is defined by $S_i = 1/E[D_i]$. The average link throughput for a network of N nodes is given by

$$S = \frac{1}{N}\sum_{i=1}^{N} S_i. \tag{5}$$

We evaluate (5) for two different scenarios. In *unicast* transmission, simulcasting is not allowed, and each radio sends at most one packet to one next-hop radio during a time slot. For *simulcast transmission*, two packets can be sent simultaneously by a radio during a time slot if that radio has any more-capable links, as described in Section II-A. The link throughputs for unicast and simulcast transmission are denote by S_U and S_S, respectively.

The link throughput will depend on several parameters. Define G_i to be the attempt rate of the ith radio. Let $S_{U,i}$ and $S_{S,i}(\theta)$ be the link throughput at radio i for unicast and simulcast transmission with phase offset θ, respectively. The throughput at radio i depends on the number of neighbor radios $B_i(\theta)$, the probability of collision $C_i(\theta)$, and the retransmission rate for unsuccessful packets $R_i(\theta)$.

1) Unicast Transmission: For unicast transmission, a radio sends only a single message in a slot, and that message is intended for only one of its neighbors. In this case, the throughput for the ith radio can be determined as follows. The attempt rate must satisfy $G_i = S_{U,i} + R_i$, where $R_i = G_i C_i$. Then, the throughput is given by

$$S_{U,i} = G_i(1 - C_i) \tag{6}$$

where, if radio i has B_i neighbors and G is the average attempt rate over all radios, then

$$C_i \approx \frac{1}{B_i}\sum_{j=1}^{B_i} C_{i,j}. \tag{7}$$

Here, $C_{i,j}$ is the probability of collision at the jth neighbor of radio i, which is given by

$$C_{i,j} \approx 1 - (1 - G)^{B_{i,j}} \tag{8}$$

where $B_{i,j}$ is the number of neighbors of the jth neighbor of radio i. The result in (7) is approximate because it assumes equal probability of transmission to each neighbor, and (8) is approximate because the offered load from the potential interferers is replaced by the average offered load. The average link throughput S_U can be approximated by using (6)–(8) in (5).

2) Simulcast Transmission: We consider the link throughput of simulcasting using nonuniform QPSK with parameter θ. First, consider the throughput for the basic message. Although the number of neighbors that can be reached by direct transmission is reduced, the interference range stays constant. Thus, the link throughput for the basic message will be approximately equal to the link throughput for unicasting $S_{U,i}$. Now, consider the additional message. For the case of long packet buffers, if the packet generation rate is sufficiently high, then a radio that has a more-capable link will always have a packet that can be sent as an additional message. Then, the link throughput for the ith radio with simulcast transmission, $S_{S,i}(\theta)$ can be approximated as $S_{S,i}(\theta) \approx 2S_{U_i}$ if radio i has a more-capable link and $S_{S,i}(\theta) = S_{U,i}$, otherwise. Let N be the number of nodes in the network, and let $M(\theta, i)$ be an indicator function such that $M(\theta, i) = 1$ if radio i has a more-capable link and $M(\theta, i) = 0$, otherwise. Then, the throughput for simulcast transmission can be approximated by

$$\begin{aligned} S_S &\approx \frac{1}{N}\sum_i \left[2S_{U,i}M(\theta,i) + S_{U,i}\left(1 - M(\theta,i)\right)\right] \\ &= S_U\left[1 + R_m(\theta)\right]. \end{aligned}$$

So, in the best scenario (long packet buffers), simulcast transmission has the capability to improve the link throughput by a factor of up to $R_m(\theta)$. However, it is not clear that this increase in link throughput will translate into a corresponding increase in end-to-end throughput. This is the topic of the next subsection.

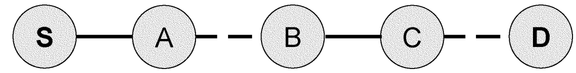

Fig. 5. Example route for estimating end-to-end throughput.

C. End-to-End Throughput

We consider the end-to-end throughput for simulcasting as a function of θ. Then, the end-to-end throughput for unicasting can be found by setting $\theta = 0$. The end-to-end throughput over T time slots is defined by

$$S_{\text{ete}} = \frac{N_D(T)}{T}$$

where $N_D(T)$ is the number of packets that reach their final destination in T time slots. We are interested in steady-state conditions and consider the expected value of S_{ete}, which is not a function of T.

Consider a generic route, as shown in Fig. 5. We analyze the throughput by considering the delay required to transmit two packets (corresponding to the two types of messages) over such a route. Under the best-case scenario, one of the packets can be sent as an additional message over each of the more-capable links (shown as dashed lines). Then, for each less-capable link (shown as solid lines), the expected delay is $2E[D_i]$ for the two packets. For each more-capable link, the expected delay is only $E[D_i]$ for the two packets. Thus, for the example in Fig. 5, the expected delay for both packets to reach the destination (not counting queueing delays) is $6E[D]$, where $E[D]$ is the expected delay at an arbitrary node. Then, the average end-to-end throughput for each packet is

$$S_{\text{ete}} \le \frac{2}{6E[D]} \approx \frac{S_U}{3}.$$

Let $H(\theta)$ be a random variable representing the number of hops in an arbitrary route. Note that as θ changes, the distribution of

TABLE II
POSITIONS OF NODES IN TEN-NODE NETWORK OF FIG. 3

Node ID	X position (km)	Y position (km)
1	1.15	2.00
2	1.53	3.59
3	1.95	2.55
4	3.50	2.25
5	2.40	3.21
6	2.55	2.12
7	3.71	2.57
8	3.10	4.01
9	2.58	4.18
10	2.06	2.00

H changes, as discussed in Section II-A. Then, in general, the end-to-end throughput can be approximated by

$$S_{\mathrm{ete}}(\theta) \approx \sum_{i=1}^{N} \frac{2P(H(\theta)=i)}{iR_m(\theta)E[D] + 2i[1-R_m(\theta)]E[D]} = \frac{S_U}{1-0.5R_m(\theta)} \sum_{i}^{N} \frac{P(H(\theta)=i)}{i}.$$

This expression is approximate because the distribution of the number of hops for packets that take a more-capable link may be different than for packets that do not take any more-capable link. For instance, more-capable links may be used more often than less-capable links to directly transmit a packet to its destination.

Note also that $S_{\mathrm{ete}}(\theta)$ is a nonlinear function of $R_m(\theta)$. Unlike the link throughput, the end-to-end throughput does not increase in direct proportion to $R_m(\theta)$. Note that the summation term will decrease as θ and $R_m(\theta)$ increase, as the number of hops increases. Then, if the distribution of the number of hops is constant, a 50% increase in end-to-end throughput requires at least $R_m(\theta) = 2/3$. For θ, large enough to satisfy this requirement, the expected number of hops may be significantly larger than for unicasting, thereby reducing the gain from the increase in link throughput. The analysis of the distribution of the number of hops is beyond the scope of this paper, so we use the empirical values for our calculations. In Section IV, we present results on the expected gain in S_{ete} from simulcasting and on the optimal choice of θ to maximize S_{ete}.

IV. RESULTS

The performance of simulcasting in ad hoc networks is evaluated using the analytical expressions in Section III and Monte Carlo simulations. We used a custom simulation programmed in MATLAB because this provided us a simple approach to develop a simulation that incorporates the ability to transmit multiple packets to multiple different receivers in a single transmission slot and to adapt the link- and network-layer protocols to take advantage of the simulcasting capability. We begin by considering the link throughput for the fixed network topology of ten nodes illustrated in Fig. 3. The positions of these nodes are given in Table II. The network degree, defined as the average number of links per radio is 2.2.

The results in Fig. 6 show the link throughput performance of the network as a function of the average attempt rate. Solid lines represent the performance predicted by the analysis, and the markers illustrate the performance results from our simulation.

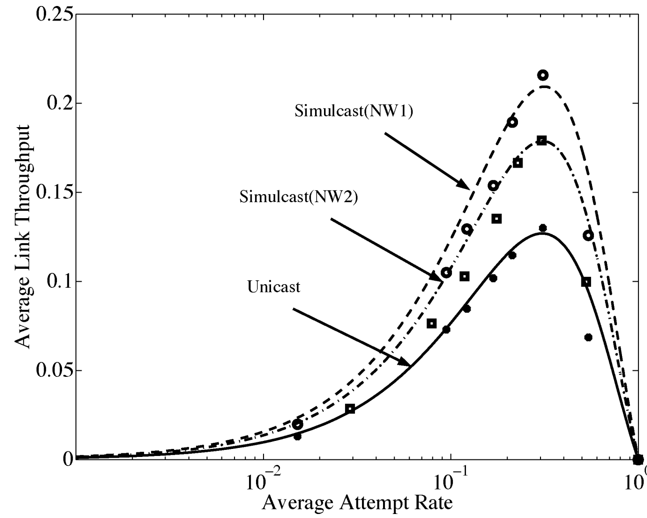

Fig. 6. Throughput in AWGN for the network of ten nodes that is illustrated in Fig. 3. For NW1, the degradation is 0.5 dB, and the disparity is 9.1 dB. For NW2, the degradation is 0.3 dB, and the disparity is 11.4 dB.

The performance is illustrated for three different network configurations. For the results marked "Unicast," the nodes are constrained to not employ the simulcast signaling technique and, thus, each node transmits at most one packet in a time slot. For the results marked "Simulcast(NW1)," simulcast transmission is used, where the more-capable links are determined based on a degradation of 0.5 dB and a disparity of 9.1 dB. The results marked "Simulcast(NW2)" illustrate the performance for a network with fewer more-capable links because the required capability disparity is increased to 11.4 dB, which corresponds to a degradation of 0.3 dB. The results indicate that simulcasting can significantly improve the throughput in the ad hoc network. The simulation incorporates transmission in additive white Gaussian noise (AWGN), where the bit-error probability at the maximum transmission range of 1 km is 10^{-4}. It is assumed that the packet length is 1000 bits and an error-control code is used that can correct up to 10 bit errors. For this case of no mobility, we expect that there will be almost no performance degradation from the noise, as the transmission range for nodes to be considered neighbors is such that the packet error probability is very small. The simulation results match closely with the analytical results.

Next, let us consider a network with $N = 15$ nodes placed at random over a 1 km $\times$ 1 km area. The node positions are chosen from a uniform distribution over the area. We consider the simple physical-layer model of noise-free communication with a maximum link distance for unicasting (or, equivalently, simulcasting with $\theta = 0$) of 381 m. Consider first some basic network parameters as a function of the offset angle θ of the nonuniform QPSK used for simulcasting. The results in Fig. 7 illustrate the network degree (expected number of neighbors) as a function of the offset angle θ. The analytical results are determined from (3). The simulation results are shown for two cases. The results for "all networks" is the average over 100 random topologies. The results for "connected networks only" shows the average network degree for ten of the random topologies that formed a connected network for all $0 \le \theta \le 45°$. The results show the sensitivity of the network degree to the parameter θ. For all networks, unicasting ($\theta = 0$) yields a network degree

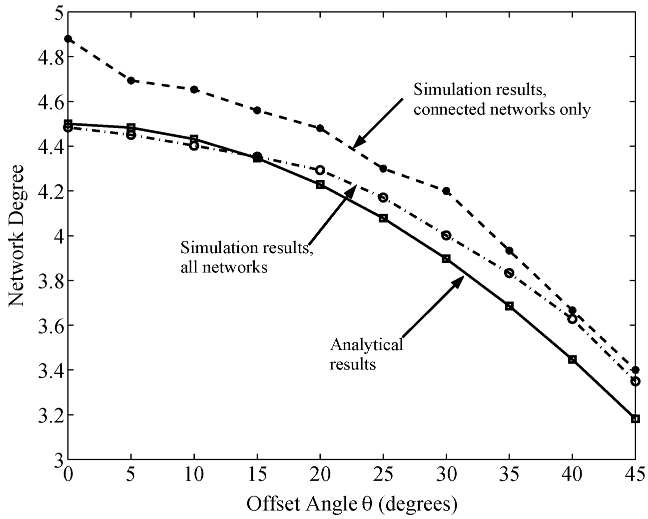

Fig. 7. Network degree as a function of the offset angle θ for simulcasting with nonuniform QPSK in a wireless ad hoc network with random node placement.

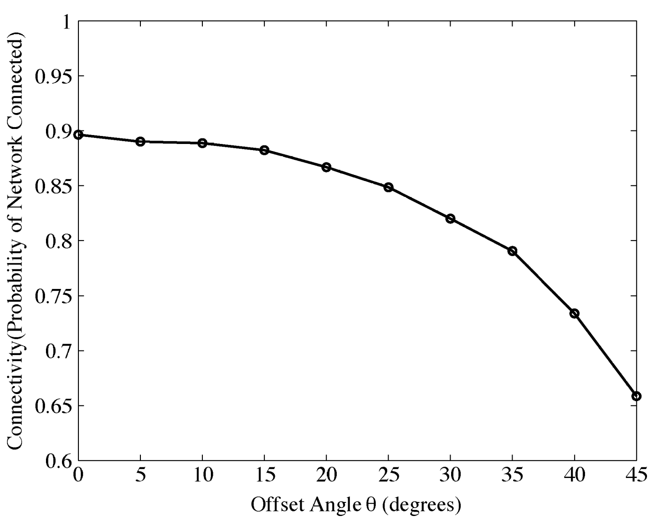

Fig. 8. Network connectivity as a function of the offset angle θ for simulcasting with nonuniform QPSK in a wireless ad hoc network with random node placement.

of approximately 4.5, while for QPSK ($\theta = 45°$), the network degree drops to 3.4. We note that if we only consider connected networks, then the network degree is biased above the value for all networks.

One of the primary effects of changes in the network degree is an impact on the connectivity, which we define as the probability that every node has a route to every other node in a randomly generated network. The connectivity is shown as a function of the offset angle θ in Fig. 8. The unicast link distance of 381 m was chosen because it provides connectivity of approximately 0.9. The network connectivity decreases as θ decreases. However, for $\theta < 25°$, the connectivity remains above 0.85. Thus, if θ is kept small, simulcasting can be used with relatively little impact on network connectivity for a network of this size. As θ approaches its maximum value of 45°, the connectivity rapidly decreases to approximately 0.66. Thus, it is not possible to switch to a uniform QPSK constellation without a significant loss in network connectivity.

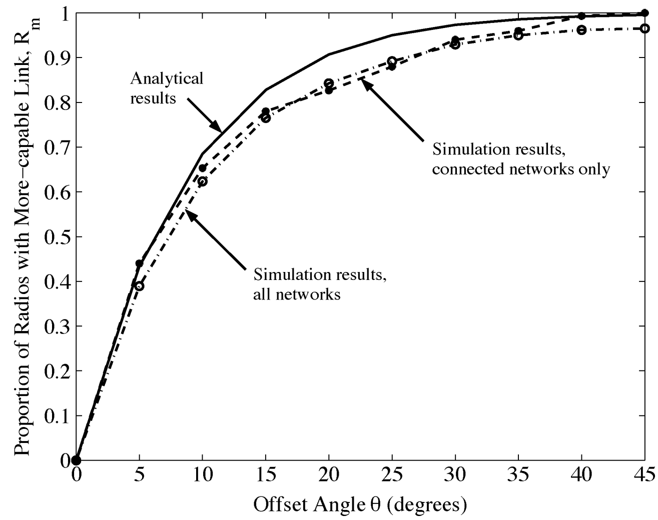

Fig. 9. Proportion of nodes with a more-capable link as a function of θ for simulcasting with nonuniform QPSK in a wireless ad hoc network with random node placement.

The results in Fig. 9 show the expected proportion of radios that have a more-capable link as a function of θ. As shown in Section III, this parameter has an important effect on both the link and end-to-end throughputs. The results show that as θ increases from 0, the proportion of radios with a more-capable link increases rapidly. The analytical expression (4) is shown to closely match the simulation results. There is no significant effect on this parameter of only considering connected networks instead of all randomly generated networks. At the previously mentioned value of $\theta = 25°$, the proportion of radios with a more-capable link exceeds 0.8. Thus, there is little to gain from increasing θ further, and any further increase comes at a significant expense in terms of network connectivity, as shown in Fig. 8.

We next restricted the simulations to ten fixed topologies for which all radios are connected for all $0 \leq \theta \leq 45°$. These ten fixed topologies were randomly selected from the connected topologies among 100 randomly generated topologies. In this way, we can be sure that we can calculate end-to-end throughput for every value of θ for each network. However, the distribution of the nodes will no longer be uniform, which will affect the results. Each topology still consists of 15 nodes distributed over a 1 km $\times$ 1 km area, with maximum link distance of 381 m for unicasting. Each simulation consisted of 1500 time slots after a 100 time slot warm-up period.

The results in Fig. 10 show the average link throughput for unicasting and simulcasting with offset angle $\theta = 25°$ for the networks described above. The simulation and analytical results differ slightly because the analytical results are for randomly generated networks, but the simulation results are for a set of connected networks. The results show that for $\theta = 25°$, the maximum link throughput is almost twice as high with simulcasting as can be achieved with unicasting. From Fig. 8, the network connectivity for $\theta = 25°$ is approximately 0.85 versus 0.9 for unicasting, so the link throughput can be significantly increased with little cost to network connectivity.

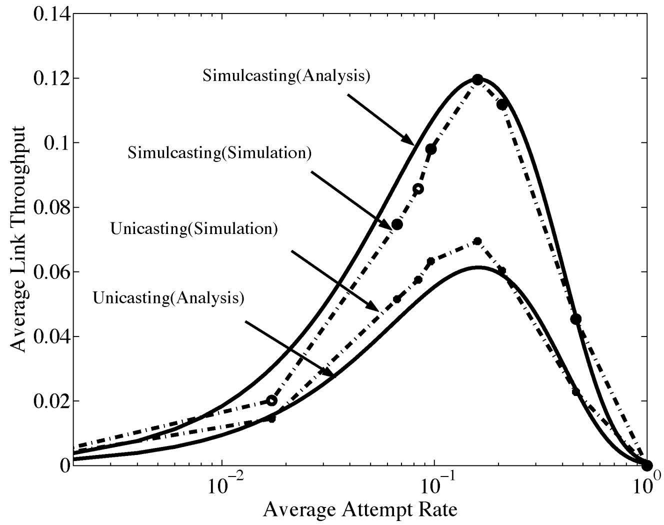

Fig. 10. Link throughput for unicasting and simulcasting with nonuniform QPSK with offset angle $\theta = 25°$ in a wireless ad hoc network with random node placement.

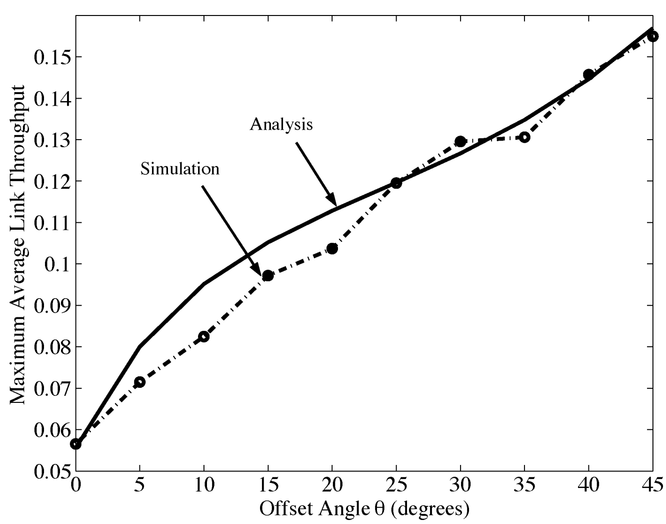

Fig. 11. Maximum (over all attempt rates) link throughput for simulcasting with nonuniform QPSK as a function of offset angle θ.

The results in Fig. 11 show the maximum link throughput achieved as a function of the offset angle θ. Here, the maximum is taken over all possible attempt rates. Note that as θ increases, so does the link throughput that can be achieved. This is reasonable because as long as the network remains connected, each node will have at least one node to which it can transmit. Furthermore, as θ increases, the probability of collision goes down along with the expected number of neighbors, and the number of nodes with more-capable links goes up. The combined effect is that the maximum throughput with $\theta = 45°$ is approximately 2.7 times higher than the maximum throughput with unicasting. However, from Fig. 8, we see that the network connectivity suffers greatly as θ becomes large.

In addition to the impact on network connectivity, increasing θ also affects the length of routes in the network, which may impact the end-to-end throughput. The results in Fig. 12 illustrate the average number of hops in a route as a function of the offset angle θ. As θ increases from 0° to 45°, the average number of

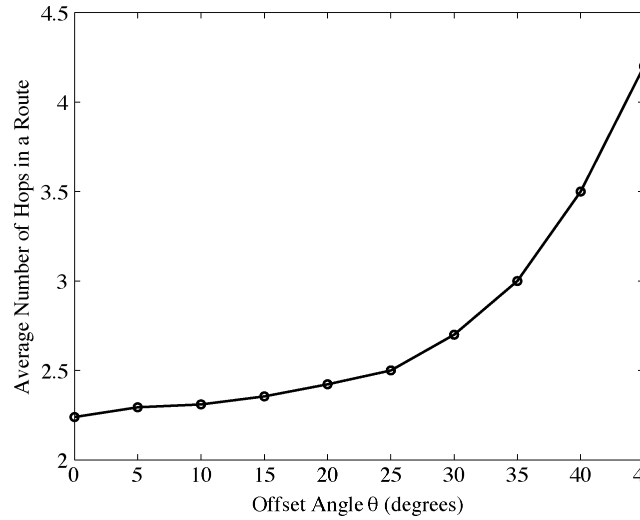

Fig. 12. Average number of hops in a route for simulcasting with nonuniform QPSK as a function of offset angle θ.

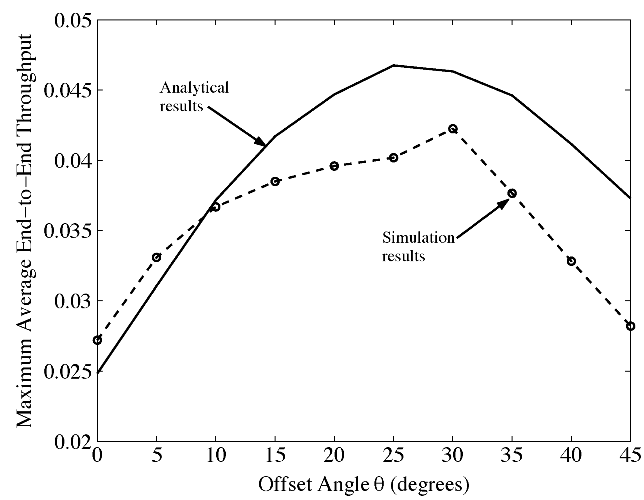

Fig. 13. Maximum (over all attempt rates) end-to-end throughput for simulcasting with nonuniform QPSK as a function of offset angle θ.

hops increase from approximately 2.25 to 4.2. The number of hops increases rapidly as θ increases beyond 20°. The results in Fig. 13 illustrate the maximum average end-to-end throughput as a function of θ. Here, the maximum is over all attempt rates. The solid line illustrates the analytical results (using the empirical values for the expected number of hops), and the dashed line illustrates the simulation results. The results show that the end-to-end throughput is a nonmonotonic function of θ. The analytical results are optimistic for $\theta > 10°$. However, they do show the same trends as the simulation results. We believe that the primary differences in the two curves come from the fact that for the simulation results, the nodes are no longer uniformly distributed because we have enforced that the networks must be connected. The simulation results show that the end-to-end throughput is maximized by $\theta = 30°$. The end-to-end throughput at $\theta = 30°$ is approximately 0.042 versus 0.027 for unicasting. Thus, simulcasting results in an increase in end-to-end throughput of over 55%. If we use a more conservative value of θ in the range $20 \leq \theta \leq 25$, then the end-to-end throughput is still more than 40% higher than unicasting, while having a smaller impact on network connectivity.

V. Conclusion

In this paper, we introduced the use of simulcast transmission techniques in wireless ad hoc networks. We applied a cross-layer approach in which the link- and network-layer protocols were modified to effectively utilize the new capability presented by simulcasting. The performance of simulcast signaling was analyzed and simulated for a wireless ad hoc network that employs slotted ALOHA. We presented detailed results on the effects of varying the offset angle θ when nonuniform QPSK is used for simulcasting. We showed that the signal constellation cannot simply be increased to a larger constellation with uniform spacing without severely affecting the network connectivity. The analytical and simulation results confirm that by choosing the simulcasting parameters appropriately, simulcasting can significantly improve both link and end-to-end throughput for static networks at the expense of a slight decrease in network connectivity.

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Kiung Jung received the B.S. and M.S. degrees in electronic material engineering from Kwangwoon University, Seoul, Korea, in 1988 and 1990, respectively, and the M.S. degree in electrical and computer engineering from University of Florida, Gainesville, FL, in 2001. Currently, he is working towards the Ph.D. degree in electrical and computer engineering at the University of Florida.

From 1990 to 2002, he was with the Electronics and Telecommunications Research Institute (ETRI), Taejon, Korea, where he was mainly involved in the development of CDMA mobile communication systems. His current research is on wireless communications with emphasis on simulcast signaling and wireless ad hoc networks.

John M. Shea (S'92–M'99) received the B.S. degree (with highest honors) in computer engineering, and the M.S. and Ph.D. degrees in electrical engineering from Clemson University, Clemson, SC, in 1993, 1995, and 1998, respectively.

He is currently an Assistant Professor of Electrical and Computer Engineering at the University of Florida, Gainesville. Prior to that, he was a Postdoctoral Research Fellow at Clemson University from January 1999 to August 1999. From 1993 to 1998, he was a Research Assistant in the Wireless Communications Program, Clemson University. He is currently engaged in research on wireless communications with emphasis on error-control coding, cross-layer protocol design, cooperative diversity techniques, and hybrid ARQ.

Dr. Shea was a National Science Foundation Fellow from 1994 to 1998. He received the Ellersick Award from the IEEE Communications Society in 1996. He is an Associate Editor of the IEEE Transactions on Vehicular Technology.