Performance Evaluation of Multihop Ad Hoc WLANs

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ABSTRACT

Ongoing technological advances in portable devices, coupled with the need for continuous connectivity while mobile, have made ad hoc networks a compelling research and development topic, particularly in a challenging multimedia multihop scenario. The ability of IEEE 802.11's ad hoc mode of operation, as a dominating wireless local area network (WLAN) protocol, to serve multihop networks requires thorough investigation. In this article, through considering crucial real-life physical phenomena and avoiding as many confining assumptions as possible, system performance measures such as delay and packet failure rate are evaluated. As a result, the importance of adequate selection of the system parameters toward performance improvement is underscored. Moreover, the simulation results imply that by complementing through priority provisions, coordination, route reservation, clustering, and optimum channel coding considerations, the IEEE 802.11 medium access control (MAC) protocol can survive in a multihop scenario. The custom simulation environment developed features modularity, comprising traffic generator, mobility, wireless channel, and IEEE 802.11 protocol modules, and is capable of accommodating many more of the physical phenomena involved.

INTRODUCTION

Mobile computing continues to enjoy rapid growth thanks to ongoing technological advances in portable devices, which have led to lowering power consumption and lengthening the battery life span. Simultaneously, the need for continuous connectivity while roaming has put ad hoc networks at the center of attention of researchers and the industry over the past few years. Portability and mobility are greatly consistent with the infrastructureless nature of ad hoc networks. This has made mobile ad hoc networks (MANETs) a compelling research topic. On the other hand, viewed as either a standalone network or a last-mile wireless connection to the Internet, ad hoc networks have the capability of expansion through multihopping. Multihop MANETs, in situations where they are supposed to support multimedia, present a challenge in many aspects. This is attributed to the fact that applications of different natures demand different treatments. Multihop MANETs [1, 2] have features that distinguish them from wired and infrastructured wireless networks. These features can be enumerated as: unpredictable link properties, node mobility, limited battery life, route maintenance, security, hidden terminal, exposed terminal, and capture effect.

In a multihop ad hoc network, nodes communicate with each other using several wireless links, and there is no fixed infrastructure such as a base station. Each node in the network also acts as a router, forwarding data packets to other nodes. One of the important challenges in the design of multihop ad hoc networks (not our concern here) is the development of routing protocols that can efficiently find routes between two communication nodes [3]. In multihop ad hoc networks, due to the numerous variables involved, system multidimensionality grows significantly, thus making analytical modeling a harder task. Turning to simulation enables us to investigate a greater number of phenomena and possibly more involved models regarding traffic, channel, and so on. In such a complex system, adequate selection of the system parameters can lead to considerable improvement in performance, especially for time-sensitive applications. Among similar attempts in the literature, [4] studies the performance of the multihop ad hoc IEEE 802.11 wireless LAN (WLAN) in various mobility and traffic volumes. It proposes a piggyback reservation protocol in which neighbors that hear the data packet are blocked and avoid colliding by returning acknowledgment (ACK). Furthermore, they learn about the next packet transmission time. Likewise, neighbors of the receiver that hear the ACK will avoid transmitting at the time when the receiver is scheduled to receive the next data packet. Another related work [5] compares two different feedback mechanisms, explicit (per hop) vs. end-to-end ACK, in multihop scenarios. The concept of adaptive clustering application to multihop ad hoc networks was introduced in [6]. In the proposed architecture, nodes are organized into nonoverlapping clusters. The clusters are independently controlled and dynamically reconfigured as nodes move. Adaptive clustering uses code separation among different clusters. However, the

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Since our main emphasis in this article is the interaction of physical and MAC layers, we try to make the routing algorithm as simple and straightforward as possible. As such, we assume that all nodes are aware of each other's geographical location.

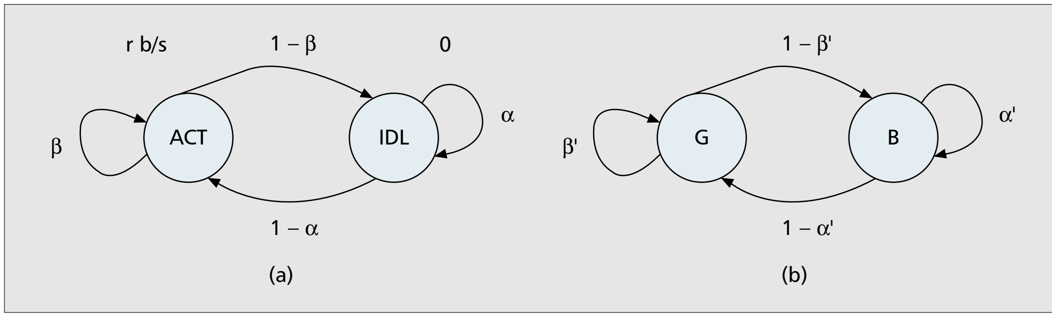

Figure 1. *a) On-off traffic generation model; b) Gilbert model for time variations of the channel.*

work is not particularly about the IEEE 802.11 protocol. A very recent related work [7] discusses priority considerations that are implemented through queuing and contention window manipulations. The motivation for our work herein is to compensate for the fact that all the past simulation efforts have adopted too many restrictive assumptions, and to find out if 802.11's ad hoc mode can serve as the core building block of a multihop MANET. We have tried to develop a thorough simulation environment spanning both medium access control (MAC) and physical (PHY) layers that considers mobility and varying wireless channel as well as forward error correction (FEC) using turbo coding.

The simulation is arranged in a module-based manner where each independently implemented module copes with one unique process, while exchanging data with other modules through some defined function values. This makes module replacement/augmentation task quite simple. Aiming at time-sensitive applications, our goal is to evaluate the performance measures of the system such as end-to-end delay and packet failure rate in different situations (e.g., traffic, mobility) and against different choices of system parameters (e.g., fragmentation factor, buffer length, retransmission limit). The article is organized as follows. A topological description of the system is presented. Traffic generation, mobility, channel, and 802.11 protocol modules are introduced. Key assumptions made throughout the simulation are intermittently mentioned when discussing the modules. Computer simulation results are presented afterward. Concluding remarks then appear.

System Description

A number of n_{tot} nodes with limited buffer size are uniform-randomly located inside a rectangular area, laid diagonally between (x_{min},y_{min}) and (x_{max},y_{max}). These upper and lower limits, together with the transmitter's wireless range, determine whether a single-hop or multihop scenario is dealt with for the source-destination communicating pair of each call. All nodes broadcast their transmissions omnidirectionally. As mentioned earlier, routing is the first concern that distinguishes single-hop from multihop. Since our main emphasis in this article is the interaction of PHY and MAC layers, we try to make the routing algorithm as simple and straightforward as possible. As such, we assume that all nodes are aware of each other's geographical location. This can be achieved through location information (probably acquired by global positioning system, GPS) exchange among the nodes and certainly results in some bandwidth cost. The routing strategy is then defined as follows. Each node with a packet ready to be sent picks another node as its intermediate destination, provided that it is inside the source node's coverage area and has the least Euclidean distance to the packet's final destination. While the latter may not be the best routing strategy (in terms of cost and reliability), in most cases it leads to the least number of hops traversed and is of course easy to implement. However, any other routing algorithm [8] can easily be incorporated in our simulation environment. The most important point that will not be further discussed is the bandwidth cost of whatever routing strategy is used. This degrades system performance below the presented results. In the initialization phase, some nodes are uniform-randomly selected as mobile and the rest are left as fixed nodes. The mobility status will prevail for the whole simulation time. The parameter

$$P_{mobile} = \frac{n_{mobile}}{n_{tot}}$$

along with v, mobile node's speed, represent the mobility volume of the system. In a similar way, uniform-randomly selected nodes are responsible for generating traffic, while all nodes might serve as forwarding and final destination nodes at any time during the simulation. A parameter representing the population of traffic generating (active) nodes is defined as

$$P_{mobile} = \frac{n_{active}}{n_{tot}}.$$

This parameter may serve as the system's traffic volume indicator as well. Each active node is pre-assigned a node as its final destination that remains unchanged throughout the course of simulation. This is not considered a simplifying or restrictive assumption since all the nodes are randomly located over the network area.

Simulation Modules

Traffic Generation

Considering voice connection as a representative of real-time applications, the packet generation model employed is an on-off traffic model. This model has been used extensively in the literature to model voice communications (e.g., [9]), mostly because it can efficiently imitate silence/talk spurt periods of human conversation and can easily be simulated. Figure 1a illustrates the traffic generation model. While in the active state (ACT in the figure), the model acts as a constant bit rate (CBR) source generating r b/s, and while in the idle state (IDL in the figure), no packet is generated. Parameters α and β are self-loop transition probabilities in IDL and ACT states, respectively. The slot time of the model, t_{slot}, is considered long enough to accommodate the longest packet size generated above the node's link layer. The average idle and active states dwell times are calculated as $T_{IDL} = t_{slot}(1 - \alpha)^{-1}$ and $T_{ACT} = t_{slot}(1 - \beta)^{-1}$, respectively. Upon packet generation at the source node, vector function pkt(SEQ_NO,CP_VER,HOP_VER) is responsible for carrying over all the information pertinent to the generated packet, such as generating node identity, final destination node identity (specified at the transport or higher layers), intermediate node identity (specified at the network layer as a result of routing strategy application), packet location in the queue, and temporal information. Each packet is uniquely identified by its 3-tuple identifier (SEQ_NO,CP_VER, HOP_VER) as explained shortly.

Mobility

A special case of a random walk mobility model (with constant speed and limited movement directions) was adopted to predict mobile nodes' movements throughout the network. The model features completely random and memoryless mobility; as such, it might not be suitable for all kinds of applications. A complete treatment of mobility models in ad hoc networks and the importance of model choice regarding performance results can be found in [10]. In our model, all nodes designated as mobile periodically (each Δ/v s where the input parameter Δ denotes displacement step length in meters) update their locations. This moves the mobile node as much as Δ along either the x or y axis. Large-scale fading and shadowing phenomena are directly related to the macro movements of the nodes, so they are handled in the mobility module. Using a two-slope curve model [11] and log-normal distribution for path loss and shadowing, respectively, we have

$$\begin{aligned} P_L(d) &= P_L(d_0) + 10n\log_{10}\left(\frac{d}{d_0}\right) + X(0,\sigma)\ ; \ d \geq d_0 \\ P_L(d) &= 20\log_{10}\left(\frac{4\pi d}{\lambda}\right) \qquad\qquad ; \ d < d_0. \end{aligned} \tag{1}$$

In Eq. 1, n_{PL} and λ represent terrain-dependent path loss exponent and wavelength, respectively, and X is a normal random variable in dB. Parameter d_0 denotes the breakpoint distance below which the environment has free space characteristics. For simplicity we do not consider correlation in shadowing from one location to another. The displacement, distance, and large-scale fading between all pairs of nodes are calculated and used for channel bit error rate (BER) evaluation as follows.

The Channel Model

In this module the small-scale behavior of the channel, in particular multipath fading as its major contributor, is simulated. The two important parameters that quantify channel behavior with regard to fading classes are coherence

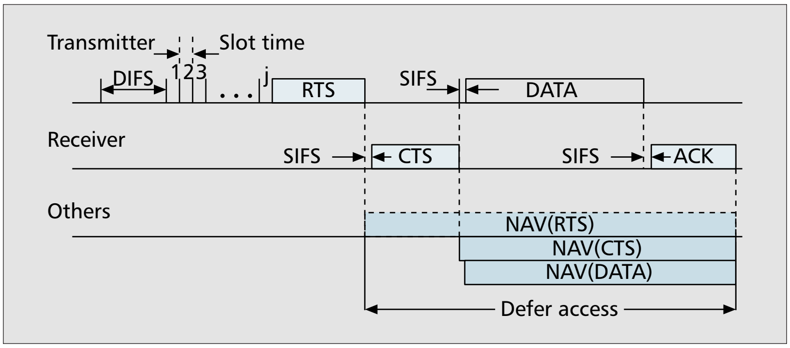

Figure 2. *Random access and data transmission in IEEE 802.11 ad hoc mode of operation using the optional RTS/CTS channel reservation scheme.*

bandwidth (B_c) and coherence time (T_c). Less conservative expressions for the coherence bandwidth and coherence time are [12]

$$B_c \approx \frac{1}{5\sigma_\tau}, T_c \approx \frac{0.423C}{v_m f_c}. \tag{2}$$

In Eq. 2 σ_τ denotes the delay spread of the wireless channel and f_m, v_m, f_c, and C denote maximum Doppler frequency, maximum mobile speed, carrier frequency, and light speed respectively, which are related through $f_m = v_m f_c / C$. The power spectral density of the direct sequence spread spectrum (DSSS) signal in 802.11 is limited to around 20–22 MHz using a transmit spectrum mask [13]. This bandwidth translates to a 9–10 ns delay spread through applying Eq. 2. While this value of delay spread may fall into the rural area range, suburban and urban areas show higher delay spreads [14] and subsequently lower coherence bandwidths. To avoid frequency band variations of the signal strength, at this point we adopt the flat fading channel assumption, which may not be valid for all terrain situations. Regarding the coherence bandwidth and for the system under study, values of $C = 10^8$ m/s, $f_c = 2.4$ GHz, and $v = 10$ m/s give a coherence time of $T_c = 5.29$ µs. Compared to the symbol period of 1 µs, the wireless channel can conventionally be considered slow fading. However, since we are concerned about packet rather than symbol (bit herein), depending on the packet length, a packet may experience several channel conditions. So at any time we consider the channel as a time-invariant stationary process with different statistics during different coherence time units. In order to model the time variations of the channel, a two-state Markov model (Gilbert model) is adopted. Markovian modeling of fading channels is extensively found in the literature (e.g., [15]). Most of these references agree on the suitability of the first order Markov chain in modeling slow to medium fading channels, while higher-order Markov chains are needed for faster fading channels. Moreover, limiting Markov states to two as good and bad channel states has been common due to its efficient trade-off between accuracy and simplicity, and of course its ease of computer simulation. However, good and bad states have been defined in a variety of ways such as having fixed low and high BERs, experiencing no and deep fade, or non- and erroneous

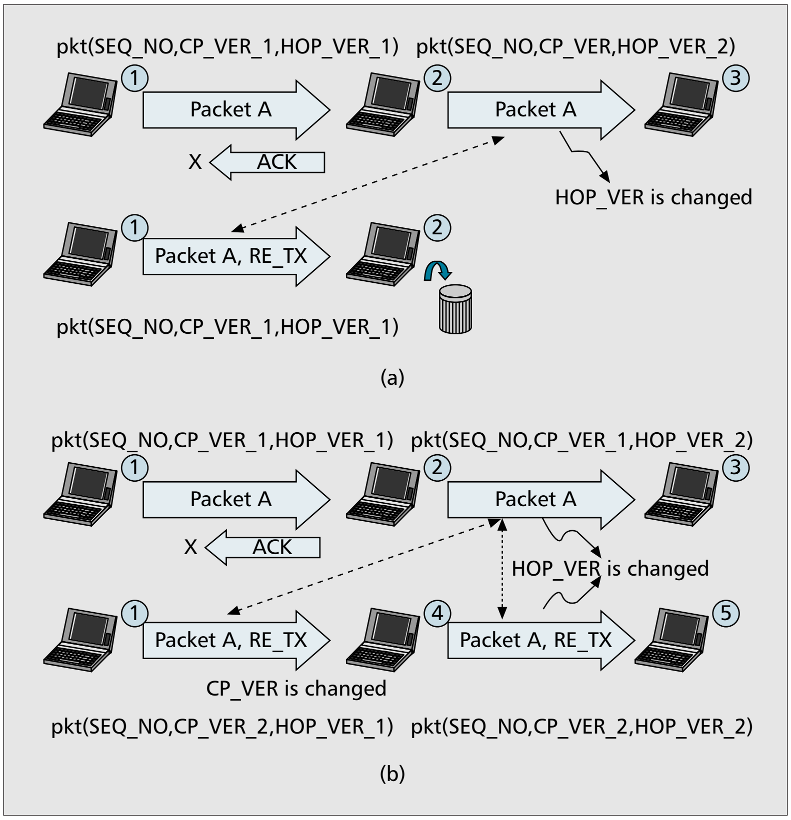

Figure 3. *Upon an ACK failure, retransmission is forwarded to: a) the same previous target node; b) a new target node due to mobility and topology change.*

packet delivery. We assume that a mobile node along its movement alternatively experiences either a line of sight (LOS) or Rayleigh fading channel along its target node. Figure 1b shows the embedded channel model in our simulation with α' and β' denoting the self-loop probabilities of bad (B) and good (G) states. By equating T_c in Eq. 2 to the model's slot time, the average dwell times of the bad and good states are $T_B = T_c(1 - \alpha')^{-1}$ and $T_G = T_c(1 - \beta')^{-1}$, respectively. In the good state the received power (P_{RG}) is calculated from the transmitted power (P_T) as $P_{R_G} = P_T - P_L(d)$ where $P_L(d)$ is defined in Eq. 1 and with all powers in dB. In the bad state, P_{R_G} serves as the average power of the received signal with a Rayleigh distributed envelope from which its power is calculated.

802.11 Protocol Implementation

IEEE 802.11 protocol implementation in ad hoc mode is discussed here. More emphasis is put on the MAC layer while the PHY layer implementation is realized through adopting parameters related to the specific choices of modulation, spreading, and FEC coding.

The MAC Layer — The random backoff algorithm is the contention resolution strategy in the IEEE 802.11 WLAN protocol operating in ad hoc or distributed coordination function (DCF) mode. Figure 2 illustrates the channel access mechanism through the optional request-to-send/clear-to-send (RTS/CTS) channel reservation scheme in ad hoc mode of operation. For a more detailed discussion of the protocol, the reader is referred to [9, 13]. Real-time traffic, depending on the application, requires a compromise between delay and loss. Late packets are not worth more than lost ones. To take care of this, and as an amendment to the protocol toward real-time adaptability, unacknowledged packets are retransmitted a limited number of times (R_{max}) before releasing the channel. Moreover, R_{max} unsuccessful retransmissions are followed by a packet drop to avoid too much latency of late packets. Other mechanisms that result in packet elimination are blocking due to the generating/target node's full buffer, dropping due to reaching some constraints such as time-to-live, maximum hopping, retransmission limit, duplication (explained below), route unavailability, and successful reception. We also assume that fragmentation is done at layers above MAC so we only need to deal with single-fragment MAC protocol data units (MPDUs). The MPDU is then prefixed with PHY layer overhead, coded, and transmitted as the DATA packet (Fig. 2). In terms of performance evaluation, the latter does not seem to produce considerable discrepancy with respect to the MAC fragmentation scheme [16]. Due to the ad hoc nature of the network, the MAC protocol is implemented in a distributed manner (concurrently in all nodes); thus, there are some peculiarities that will be highlighted here. The only verification mechanism regarding a healthy DATA transmission is through ACK exchange (Fig. 2), which itself may or may not fail. Thus, situations may arise in which several copies of the same packet are hopping to the same destination. Figure 3 illustrates such situations. In Fig. 3a node #1 sends packet A to node #3 through intermediate node #2. Although the packet is received correctly at node #2, its corresponding ACK back to node #1 fails, which makes node #1 initiate a retransmission. At the same time, node #2 will forward already queued packet A along its route to the final destination. In Fig. 3b the same event is illustrated, except that due to topology changes (as a result of mobility) the retransmission of packet A is headed toward a different intermediate node, node #4, which in turn will forward it along its route to the final destination. To better simulate the real-life situation and account for the above possibilities, the following measures are put in place:

- Only the originating node can manipulate the function pkt(.),
- Upon correct reception at an intermediate node, a new version of the packet with a different hop identification (HOP_VER) is produced (with this intermediate node as its originator) and queued for further forwarding,
- At the time of retransmission, if the next intermediate node is different from the previous transmission attempt, a new version of the packet with a different copy identification (CP_VER) is produced (at the originating node), and the old one is discarded.

With the above provisions, the multiversion packet phenomenon is well simulated. Since all versions of a packet have the same unique sequence number assigned at generation time, flooding can be partially prevented by discarding the packet whose sequence number (SEQ_NO) has already been met (duplication) at each

Network parameters	
n_{tot}	30
Node's coverage radius	150 m
Network area	400 m × 400 m
Displacement step (Δ)	5 m
Simulation time	10.0 s
Source model parameters	
Node's traffic generation rate	64 kb/s
Avg. talk-spurt/silence period	1.0/1.35 s
Max. packet length (uncoded)	8000 bits
Channel parameters	
Path loss exponent (n_{PL})	3.2
Breakpoint distance (d_0)	10 m
Standard deviation of shadowing (σ)	8 dB
β′/α′	0.967/0.9
IEEE 802.11 standard parameters for DSSS WLANs	
$L_{RTS/CTS/ACK}$	20/14/14 octets
$OH_{MAC/PHY}$	34/24 octets
DIFS/SIFS/slot — time	50/10/20 µs
Channel rate	1 Mb/s
Miscellaneous parameters	
Transmitter power (P_T)	17 dBm
Noise floor (*NF*)	–80 dBm
Simulation slot time	10 µs
Maximum hopping	4 hops

Table 1. *Parameters used in the simulation.*

receiving node. The vector function node (ID_NO), throughout the course of simulation, encompasses the static and dynamic information for node #ID_NO.

PHY Layer — Spreading, modulation, and coding schemes are considered here. According to FCC regulations, a DSSS system shall provide a processing gain of at least 10 dB. This is accomplished by chipping the baseband signal at 1 MHz with an 11-chip PN code. The DSSS system uses baseband modulations of binary phase shift keying (BPSK) or differential binary phase shift keying (DBPSK) to provide 1 Mb/s data rate. The presence of fading in wireless environments motivates the use of channel codes. Channel codes mitigate the adverse effect of fading in wireless channels by adding redundancy and memory to the transmission. Turbo codes are a class of error correction codes that enable reliable communications with power efficiencies close to the theoretical limit predicted by Claude Shannon. The good error performance of turbo codes may be compromised by their bandwidth and computational (hence energy consumption) costs in certain ad hoc applications. For instance, they cannot be afforded in most sensor networks where nodes feature very limited battery resources. As an illustrative example of turbo code application to WLANs, our turbo coding scheme features two identical recursive systematic convolutional (RSC) encoders with parameters $n = 2$, $k = 1$, $K = 3$, $G_0 = 7$, and $G_1 = 5$, with half of the parity bits punctured. The decoder is Log-MAP with 8 iterations. Regarding the interleaver, we consider two types: one, which shuffles fixed short size control packets (RTS,CTS, and ACK), is a block interleaver; the other, which deals with data packets, is a random separated interleaver. While this coding scheme consumes half the bandwidth in the first place, it is hoped it will retrieve much more through BER improvement. However, there is no claim about the optimality of this coding scheme, and no effort is made to evaluate its efficiency since it is not in the main theme of this work. Now the focus is turned toward how transmission health is assessed in our simulation. Data packets, depending on their length and the channel coherence time (T_c in Eq. 2) they face, may experience several channel qualities. The latter information represented by BER_1, BER_2, …, BER_x are carried by pkt(.) function. On the other hand, short control packets always go through a single-quality channel (represented by a single BER) except at rarely occurring boundary situations. The corresponding BERs are extracted from the BER vs. SNR curve in the additive white Gaussian noise (AWGN)/BPSK case with the above employed coding scheme [17, Fig. 5.26]. Accordingly, the packet error rate (PER) for all packet sizes is

$$PER = 1-(1-BER_x)^{\mathrm{mod}(L_i, L_{Tc})}\prod_{m=1}^{x-1}(1-BER_m)^{L_{Tc}}; \quad i = RTS, CTS, ACK, DATA, \tag{3}$$

where L_{Tc} denotes the equivalent bit length of T_c, $mod(a,b)$ accounts for the remainder of dividing a by b, and x is the number of coherence times fitted within one packet.

Since we are aiming at real-time application support, a larger area means more hops, which in turn contributes to unfavorable delay accumulation. The area is limited such that the majority of the source-to-destination transmissions can take place with the number of hoppings less than HOP_LMT = 4.

Simulation Results

In this section the effects of different choices of MAC layer parameters as well as other external network parameters (e.g. mobility, traffic load, etc.) on the system performance are simulated. In particular, packet length, maximum lifetime of a packet (time-to-live, *TTL*), maximum number of times a packet is retransmitted ($ReTX_{max}$), and a node's maximum buffer size (Q_{max}) are examined. This is done in different network conditions by varying the percentage of active nodes (P_{active}), percentage of mobile nodes (P_{mobile}), and mobile node speed (v). However, due to space limitations, only some of the produced results are presented herein. Regarding other parameters, the total number of nodes present in the network (n_{tot}), node traffic generating rate, and network geographical dimensions remain fixed at 30, 64 kb/s, and 400 m × 400 m, respectively, throughout the simulation. Also, throughout the simulation the following values

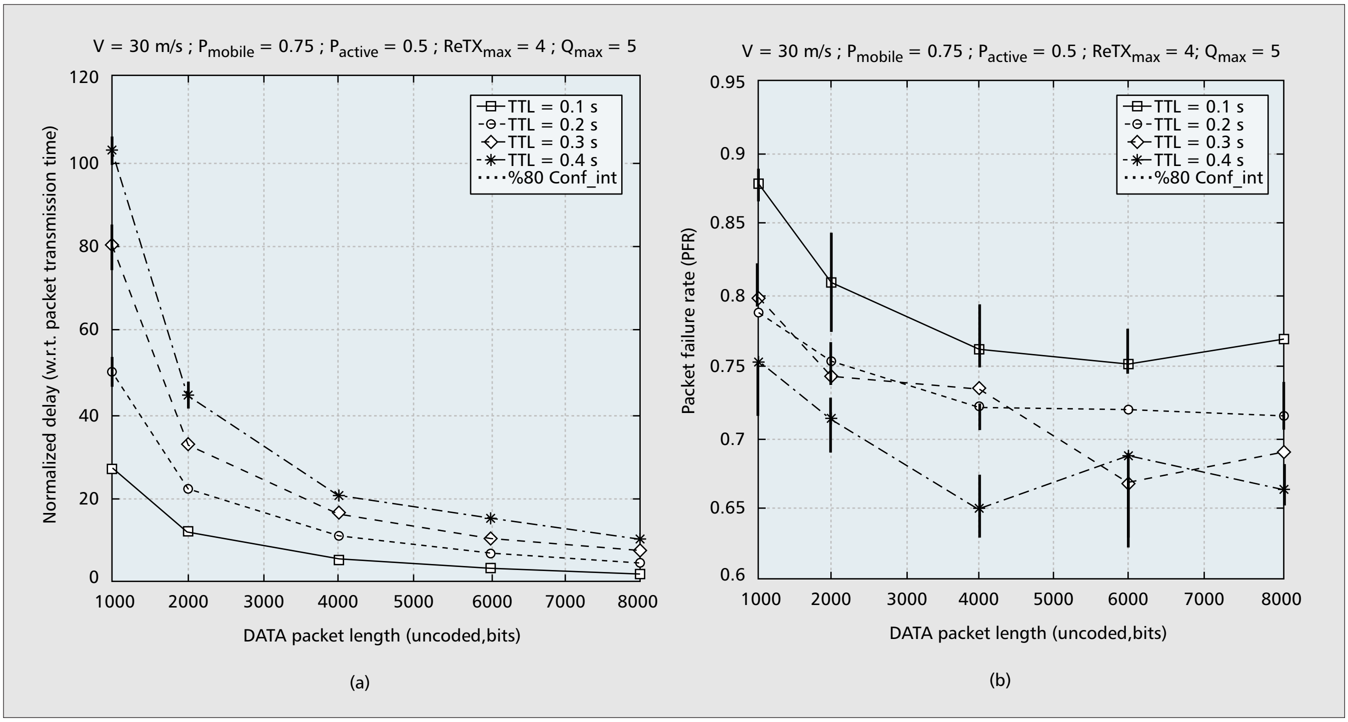

Figure 4. *Performance measures of the system versus DATA packet length with time-to-live (TTL) as a parameter: a) normalized delay; b) packet failure rate.*

were assigned to the above parameters unless they are the parameter subject to variation; *TTL* = 0.4 s, v = 30 m/s, P_{active} = 0.5, P_{mobile} = 0.75, $ReTX_{max}$ = 4, and Q_{max} = 5. These values reflect high traffic load and relatively fast mobility. To give a sense of traffic volume, P_{active} = 0.5 means, when counting in MAC/PHY/FEC overheads and idle/active periods, 15 nodes each generating on average 60 kb/s continuously, and this should be compared against 1 Mb/s channel capacity. Moreover, in multihop situations, generated traffic is multiplied by a factor of number of hops to final destination, which is lower than the effective channel reuse factor. A per user traffic rate of 64 kb/s can be representative of a wide variety of real-time applications. A 400 m × 400 m area together with a node's 150 m coverage radius (which is a typical coverage radius for wireless interfaces in the market) represent our intended multihop scenario. Since we are aiming at real-time application support, a larger area means more hops, which in turn contributes to unfavorable delay accumulation. The area is limited such that the majority of source-to-destination transmissions can take place with the number of hoppings less than HOP_LMT = 4. However, the simulation environment developed has no limitation in handling areas of larger sizes. Regarding the number of nodes, 30 users in the area above would be a normal user population (of course, in certain application scenarios) and ensures reasonable availability of intermediate nodes toward the final destination as well. However, as mentioned earlier, in most of the simulations P_{active} was set to 0.5 to yield a high traffic load scenario. Parameters used in the simulation are tabulated in Table 1. The parameters of the source and channel models are chosen to be consistent with the corresponding parameters in [9]. The other parameters are partially acquired from [11] and Cisco WLAN specification guides. The simulation time is fixed at 10.0 s to accommodate a good number of generated packets and TTL periods, and for the network to reach its steady state. In all the simulation results, each point is acquired by applying the batch means method [18] over 24 (6 × 4) simulation runs to obtain a good level of confidence. Two major measures considered herein as performance indicatives are end-to-end (e.t.e.) delay and packet failure rate (PFR), wherein such a delay has been presented as normalized with respect to the subject DATA packet transmission time. The delay includes the time duration from the moment of packet generation until its healthy reception. Not included are packetization delay, interleaving delay in turbo encoding, propagation delay, delays related to turbo decoder iterations, and other packet processing times at intermediate and destination nodes. However, these delays are both fixed and negligible compared to random transmission and queuing delays. In each simulation run, e.t.e. delay is calculated by averaging over all healthy received packets. Packet failure rate is the ratio of the number of packets dropped to the total number of generated packets throughout the duration of simulation . Different limit violations in the aggregate contribute to the total number of dropped packets. Figures 4–6 illustrate how the above mentioned performance measures are affected by varying DATA packet length (L_{DATA}). In Fig. 4, a group of curves corresponding to different values of TTL, 0.1–0.4 s, are drawn. In the left plot, the normalized delay is decreased by adopting longer packet lengths and lower TTL values. However, this decrease becomes less significant upon passing L_{DATA} =

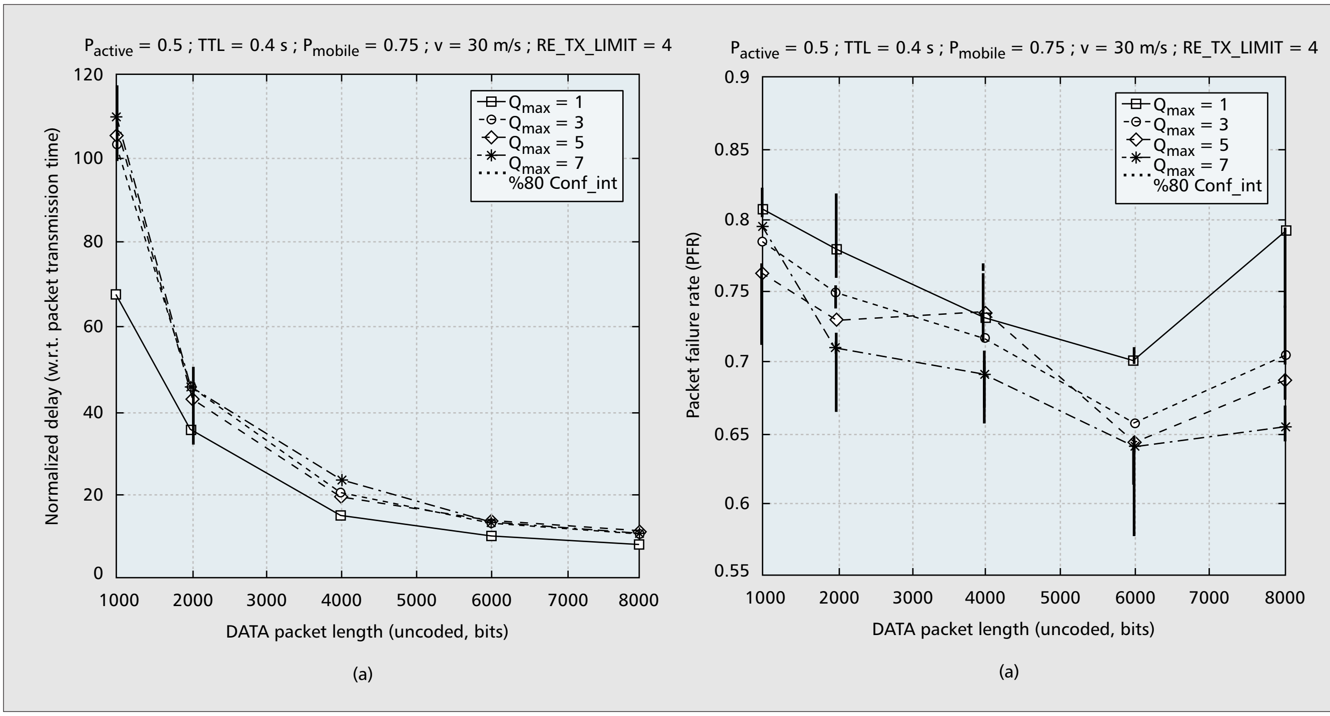

Figure 5. *Performance measures of the system vs. DATA packet length with maximum queue length* (Q_{max}) *as a parameter: a) normalized delay; b) packet failure rate.*

4000 bits. The fact that the channel favors longer packets all the way points to the effectiveness of the FEC scheme employed, and that probably a higher rate turbo coding might be also sufficient. In the right plot, packet failure rate mostly decreases with higher TTLs as may be expected. However, TTL ≈ 0.3 s shows turning points in 4000- and 6000-bit packet lengths. By trading delay for loss or vice versa, we may end up with different choices in the ranges of 0.2 ≤ TTL ≤ 0.4 s and $4000 \leq L_{DATA} \leq 6000$ bits. Figure 5 shows the effect of varying queue size and DATA packet length on system performance measures. Looking at the delay plot, all the curves similarly descend by a climbing packet length. The descents become less significant and particularly indistinguishable beginning at the 4000-bit packet length. The closeness of the curves for $Q_{max} \leq 3$ suggests that beyond $Q_{max} = 3$, this parameter is not a decisive factor in delay performance. On the PFR side, it seems that 6000-bit packet length is the best choice for all queue lengths. Similar to the delay performance, not much is gained by enlarging the queue beyond $Q_{max} = 3$. This means that the parameter pair ($Q_{max} = 3$, $L_{DATA} = 6000$ bits) gives the best combined delay-PFR performance. In the last two figures, the two parameters subject to variation control the dwelling time of the packet in the system (in fact $ReTX_{max}$ has a similar effect). A longer stay of the packet in the system (due to a higher TTL, queue size, or retransmission limit) on one hand may increase its chances of successful delivery, while on the other hand it decreases other packets' chances by making them violate the system parameters limits. The multiple crossings of PFR plots at some points stem from these conflicting effects.

The impact of mobility on system performance can be evaluated by varying either the number of mobile users in the system (P_{mobile}) or the speed of mobile users (v). A higher P_{mobile} (at fixed speed) is translated to deeper topology changes since it involves more nodes. However, since the topology changes take place in a completely random manner, considering the fixed speed, in an average sense one should not expect any wide fluctuations of performance measures for different mobility volumes. On the other hand, a higher node's speed means faster and more frequent topology changes of the network. It also means that a particular long enough packet may experience more channel qualities due to shrinking channel coherence time. Because of the powerful FEC scheme used, many packets that would otherwise be completely faded can be recovered. The delay plot of Fig. 6 shows slightly higher delays for higher speeds while maintaining the same descending pattern with respect to packet length. Since the delay measure only deals with successfully received packets, it cannot clearly reflect the phenomenon described above. The PFR plot illustrates how longer packets may benefit from faster mobility. It also shows that $L_{DATA} = 6000$ bits can be a good choice for different node speeds.

To better visualize the impact of adequate selection of system parameters and traffic load on performance, Fig. 7 illustrates performance measures corresponding to another set of choices: $TTL = 0.25$ s, $v = 30$ m/s, $P_{mobile} = 0.75$, $ReTX_{max} = 2$, $Q_{max} = 4$, and different load conditions. The selected values are intended to improve the combined delay-PFR performance. All the earlier graphs should be compared against the diamonds ($P_{active} = 0.5$) in Fig. 7. To evaluate the performance under different traffic loads, the ratio of active users (generating traffic) is varied over P_{active} = 0.1–0.5. In Fig. 7, as expected, all the plots in general show more degradation with higher loads. Again, delay plot favors longer DATA packets; interestingly, 6000-bit packet could be the best choice for all traffic volumes regarding the PFR performance.

Finally, in interpreting the quantities of the performance measures, it should be noted that the results pertain to pure basic 802.11 protocol, and no extra efforts such as revising the protocol, optimizing the FEC scheme employed, or clustering have been made.

CONCLUSION

In order to investigate the survivability and performance of IEEE 802.11's DCF and the significance of adequate selection of the system parameters in a multihop scenario, an inclusive simulation was performed. The simulation, com-

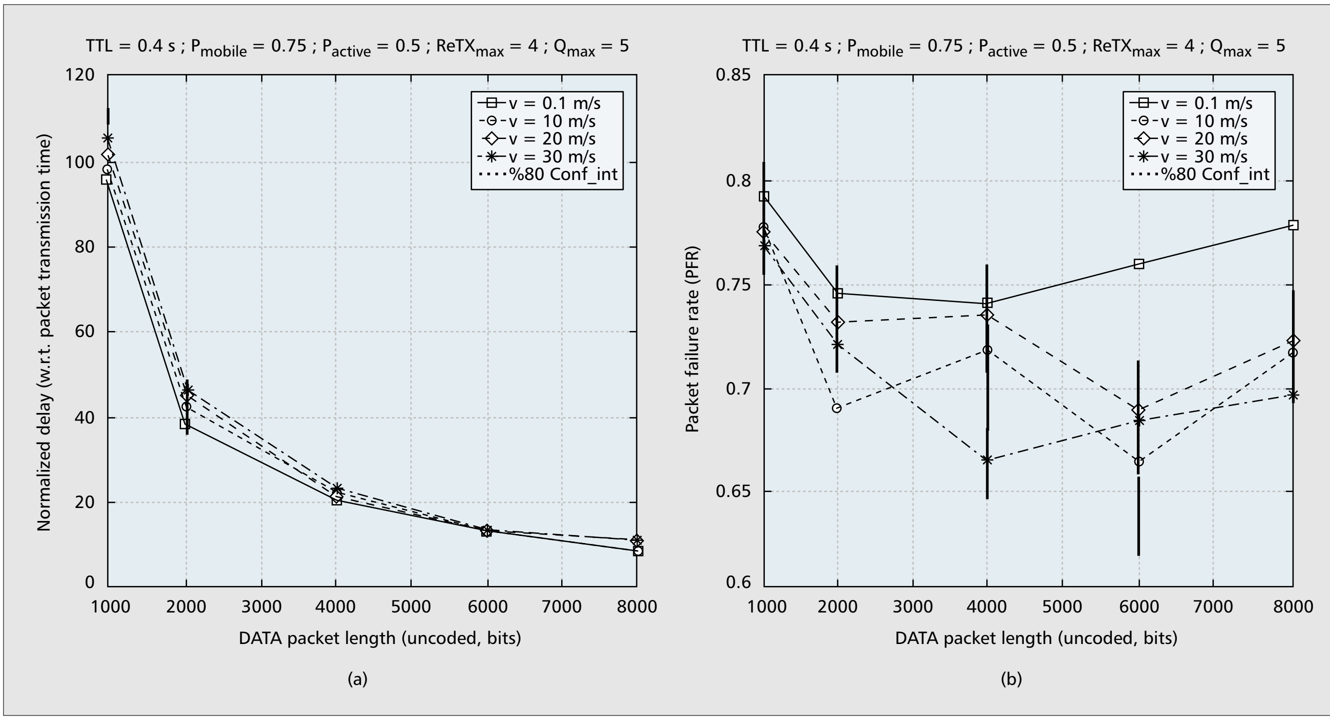

Figure 6. *Performance measures of the system vs. DATA packet length with the mobile nodes' speed (v) as a parameter: a) normalized delay; b) packet failure rate.*

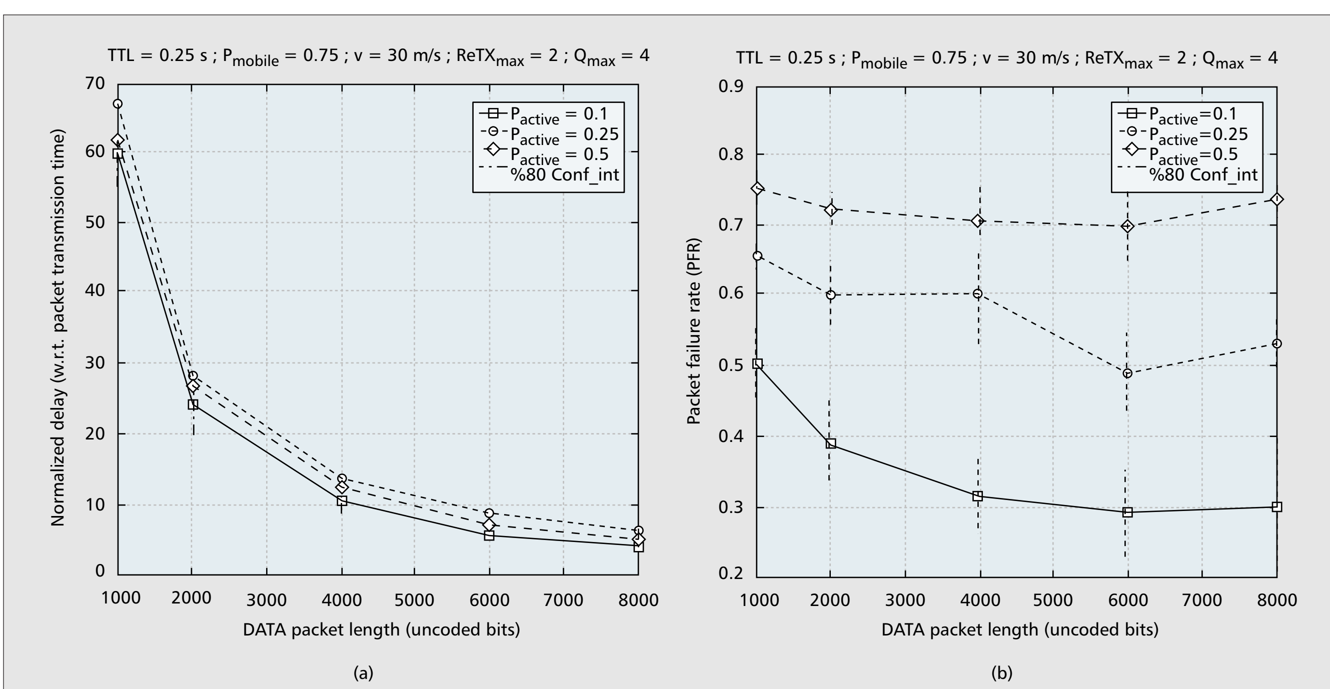

Figure 7. *Performance measures of the system vs. DATA packet length with the percentage of active nodes (P_{active}) as a parameter and a different choice of other parameters: a) normalized delay; b) packet failure rate.*

prising traffic generation, mobility, wireless channel, and IEEE 802.11 MAC protocol, was intended to evaluate system performance measures, such as delay and packet failure rate, of special interest to real-time applications. Here we do not consider throughput, since the maximum bandwidth available cannot be clearly defined due to the multichannel nature of multihop ad hoc WLANs. Most of the results corresponding to the performance measures above do not demonstrate a monotonic trend against the parameters considered. While a higher TTL, queue size, and retransmission limit allow a particular packet to stay longer in the system and have more chances of delivery, it contributes to a more crowded system and in turn more chances of collision, queue overflow (for later coming packets), TTL violation (for earlier arriving packets), and thus final failure. These conflicting issues signify the importance and necessity of adequate selection of the parameters involved. The only monotony seen is the decrease in normalized delay vs. packet length. The nonlinear declining behavior, generally more profound for short packet lengths although dependent on the other parameters, is basically due to the effectiveness (or maybe oversufficiency) of the FEC scheme employed.

An important observation is that apart from the maximum delay, which was constrained by the TTL (i.e., chosen to fulfill the delay requirement of a variety of time-sensitive applications), the results of other performance measurements do not seem promising for real-time applications. This is not surprising, given the worst case deployment scenario of this article. Multihop ad hoc networking, high traffic load, lack of coordination among nodes, no facility for route reservation or clustering, and other issues all contribute to such a worst case. Nevertheless, future advances in the aforementioned areas will make the IEEE 802.11 ad hoc mode of operation a viable approach to handling real-time traffic in wireless environments.

An important observation is that apart from the maximum delay, which was constrained by the TTL, the results of other performance measure do not seem promising in view of real-time applications. This is not surprising, given the worst case deployment scenario of this article.

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