IEEE 802.11 Ad Hoc Networks: Protocols, Performance and Open Issues

Giuseppe Anastasi
Dept. of Information Engineering
University of Pisa
Via Diotisalvi 2 - 56122 Pisa - Italy
Email: g.anastasi@iet.unipi.it

Marco Conti, Enrico Gregori
Istituto IIT
National Research Council (CNR)
Via G. Moruzzi 1 - 56124 Pisa - Italy
Email: {marco.conti, enrico.gregori}@iit.cnr.it

1. Introduction

The previous chapter has presented the activities of the different task groups within the IEEE 802.11 project [IEE], and has highlighted that the IEEE 802.11 is currently the most mature technology for infrastructure-based Wireless LANs (WLANs). The IEEE 802.11 standard defines two operational modes for WLANs: *infrastructure-based* and *infrastructure-less* or *ad hoc*. Network interface cards can be set to work in either of these modes but not in both simultaneously. The infrastructure-based is the mode commonly used to construct the so called Wi-Fi hotspots, i.e., to provide wireless access to the Internet. The drawbacks of an infrastructure-based WLAN are the costs associated with purchasing and installing the infrastructure. These costs may not be acceptable for dynamic environments where people and/or vehicles need to be temporarily interconnected in areas without a pre-existing communication infrastructure (e.g., inter-vehicular and disaster networks), or where the infrastructure cost is not justified (e.g., in-building networks, specific residential communities networks, etc.). In these cases, a more efficient solution can be provided by the infrastructure-less or ad hoc mode.

When operating in this mode stations are said to form an Independent Basic Service Set (IBSS) or, more simply, an ad hoc network. Any station that is within the transmission range of any other, after a synchronization phase, can start communicating. No Access Point (AP) is required, but if one of the stations operating in the ad hoc mode also has a connection to the wired network, stations forming the ad hoc network have a wireless access to the Internet.

The IEEE 802.11 technology is a good platform to implement single-hop ad hoc networks because of its extreme simplicity. Single-hop means that stations must be within the same transmission radium (say 100-200 meters) to be able to communicate. This limitation can be overcome by multi-hop ad hoc networking. This requires the addition of routing mechanisms at stations so that they can forward packets towards the intended destination, thus extending the range of the ad hoc network beyond the transmission radium of the source station. Routing solutions designed for wired networks (e.g., the Internet) are not suitable for the ad hoc environment, primarily due to the dynamic topology of ad hoc networks.

In a pure ad hoc networking environment, the users' mobile devices *are* the network and they must co-operatively provide the functionality that is usually provided by the network infrastructure (e.g. routers, switches, and servers). This approach requires that the users' density is high enough to guarantee the packets forwarding among the sender and the receiver. When the users' density is low networking may become unfeasible.

Even though large-scale multi-hop ad hoc networks will not be available in the near future, on smaller scales, mobile ad hoc networks are starting to appear thus extending the range of the IEEE 802.11 technology over multiple radio hops. Most of the existing IEEE 802.11-based ad hoc networks have been developed in the academic environment, but recently even commercial solutions have been proposed (see, e.g., MeshNetworks[1] and SPANworks[2]).

Other than being a solution for pure ad hoc networking, the IEEE 802.11 ad hoc technology may also constitute an important and promising building block for solving the first mile problem in hot spots. This aspect is related to the understanding of some basic Radio

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1 http://www.meshnetworks.com

2 http://www.spanworks.com

Frequency (RF) transmission principles. Specifically, the transmission range is limited since the RF energy disperses as the distance from the transmitter increases. In addition, even though WLANs operate in the unregulated spectrum (i.e., the users are not required to be licensed), the transmitter power is limited by the regulatory bodies (e.g., FCC in USA and ETSI in Europe). IEEE 802.11a and IEEE 802.11b can operate at several bit rates but since the transmitter power is limited the transmission range decreases when the data rate is increased.

It is expected that the bandwidth request in hot spots will increase very fast thus requiring higher speed access technologies. As explained in the previous chapter in this book, channel speeds for the IEEE 802.11 family continue to increase: 802.11a operates at 54 Mbps, and enhanced versions operating at speeds up to 108 Mbps are also under investigation. Such high-speed WLAN standards are expected to further increase the popularity of wireless access to the backbone infrastructure. On the other hand, increasing the transmission rate (while maintaining the same transmission power) produces a reduction in the coverage area of an AP. Specifically, at 100 Mbps rate the coverage area will correspond to a radius of few meters around the AP. It seems not a feasible solution to spread in a hot spot a large number of APs uniformly and closely spaced. A more feasible solution may be based on the use of a relative low number of multi-rate APs, and the deployment of multi-hop wireless networks that provides access to the wired backbone via multiple wireless hops. When the population in a hot spot is low, the AP can use low transmission rates thus covering a large area. In this case, the users devices can contact the AP directly (i.e., single-hop). When the hot-spot population increases, the data rate is increased as well, and hence some devices cannot anymore directly contact the AP but they need to be supported by other devices for forwarding their traffic towards the AP. By further increasing the data rate, more users can be accommodated in the hot spot but, at the same time, more hops may be necessary for user traffic to reach the AP.

Currently, the widespread use of IEEE 802.11 cards makes this technology the most interesting off-the-shelf enabler for ad hoc networks. However, the standardization efforts concentrated on solutions for infrastructure-based WLANs, while little or no attention was given to the ad hoc mode. Therefore, the aim of this chapter is triple: (*i*) an in-depth investigation of the ad hoc features of the IEEE 802.11 standard, (*ii*) an analysis of the performance of 802.11-based ad hoc networks; and (*iii*) an investigation of the major problems arising when using the 802.11 technology for ad hoc networks, and possible directions for enhancing this technology for a better support of the ad hoc networking paradigm.

The rest of the chapter is organized as follows. The next section briefly describes the architecture and protocols of IEEE 802.11 WLANs. The aim is to introduce the terminology and present the concepts that are relevant throughout the chapter. The interested reader can found the details on the IEEE 802.11 protocols in the standard documents [IEE99].

The characteristics of the wireless medium and the dynamic nature of ad hoc networks make (IEEE 802.11) multi-hop networks fundamentally different from wired networks. Furthermore, the behavior of an ad hoc network that relies upon a carrier-sensing random access protocol, such as the IEEE 802.11, is further complicated by the presence of hidden stations, exposed stations, "capturing" phenomena [XuS01, XuS02], and so on. The interaction between all these phenomena makes the behavior of IEEE 802.11 ad hoc networks very complex to predict. Recently, this has generated an extensive literature related to the performance analysis of the 802.11 MAC protocol in the ad hoc environment that we surveyed in Section 3. Most of these studies have been done through simulation. To the best of our knowledge, only very few experimental analysis have been conducted. For this reason, in Section 4 we extend the 802.11 performance analysis with an extensive set of measurements that we have performed on a real testbed. These measurements were performed both in indoor and outdoor environments, and in the presence of different traffic types. For the sake of comparison with the previous studies, our analysis is mostly related to the basic IEEE 802.11 MAC protocol (i.e., we consider a data rates of 2 Mbps). However, some results related to IEEE 802.11b are also included in Section 5. In the same section, we present some problems (gray zones) that may occur by using IEEE 802.11b in multi hop ad hoc networks. Finally, in Section 6 we discuss some possible extensions to the IEEE 802.11 MAC protocol to improve its performance in multi-hop ad hoc networks.

2. IEEE 802.11 Architecture and Protocols

In this section we will focus on the IEEE 802.11 architecture and protocols as defined in the original standard [IEE99], with a particular attention to the MAC layer. Later, in Section 5, we will emphasize the differences between the 802.11b standard with respect to the original 802.11 standard.

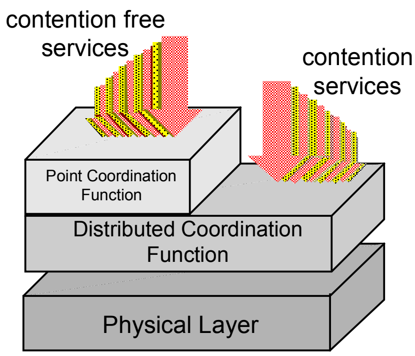


Figure 1. IEEE 802.11 Architecture.

The IEEE 802.11 standard specifies both the MAC layer and the Physical Layer (see Figure 1). The MAC layer offers two different types of service: a contention free service provided by the *Distributed Coordination Function* (DCF), and a contention-free service implemented by the *Point Coordination Function* (PCF). These service types are made available on top of a variety of physical layers. Specifically, three different technologies have been specified in the standard: Infrared (IF), Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS).

The DCF provides the basic access method of the 802.11 MAC protocol and is based on a *Carrier Sense Multiple Access with Collision Avoidance* (CSMA/CA) scheme. The PCF is implemented on top of the DCF and is based on a polling scheme. It uses a *Point Coordinator* that cyclically polls stations, giving them the opportunity to transmit. Since the PCF can not be adopted in ad hoc mode, it will not be considered hereafter.

2.1 Distributed Coordination Function (DCF)

According to the DCF, before transmitting a data frame, a station must sense the channel to determine whether any other station is transmitting. If the medium is found to be idle for an interval longer than the *Distributed InterFrame Space* (*DIFS*), the station continues with its transmission[3] (see Figure 2). On the other hand (i.e., if the medium is busy), the transmission is deferred until the end of the ongoing transmission. A random interval, henceforth referred to as the *backoff time,* is then selected, which is used to initialize the *backoff timer.* The backoff timer is decreased for as long as the channel is sensed as idle, stopped when a transmission is detected on the channel, and reactivated when the channel is sensed as idle again for more than a DIFS (for example, the backoff timer of Station 2 in Figure 2 is disabled while Station 3 is transmitting its frame; the timer is reactivated a DIFS after Station 3 has completed its transmission). The station is enabled to transmit its frame when the backoff timer reaches zero. The backoff time is slotted. Specifically, the backoff time is an integer number of slots uniformly chosen in the interval (0, CW-1). CW is defined as the Backoff Window, also referred to as *Contention Window*. At the first transmission attempt $CW=CW_{min}$, and it is doubled at each retransmission up to CW_{max}. In the standard CW_{min} and

[3] To guarantee fair access to the shared medium, a station that has just transmitted a packet and has another packet ready for transmission must perform the backoff procedure before initiating the second transmission.

CWmax values depend on the physical layer adopted. For example, for the FHSS Phisical Layer *Cwmin* and *Cwmax* values are 16 and 1024, respectively [IEE99].

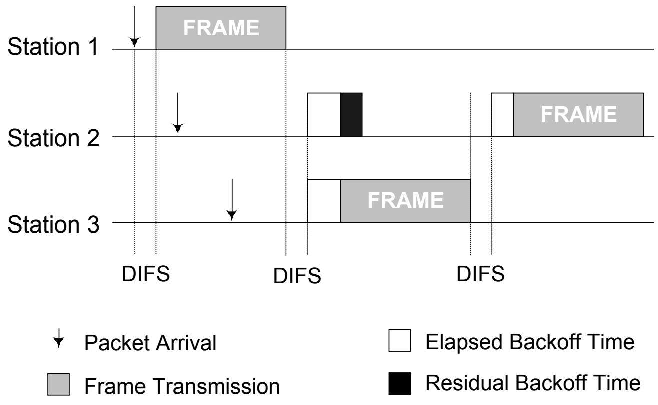


Figure 2. Basic Access Mechanism.

Obviously, it may happen that two or more stations start transmitting simultaneously and a collision occurs. In the CSMA/CA scheme, stations are not able to detect a collision by hearing their own transmissions (as in the CSMA/CD protocol used in wired LANs). Therefore, an immediate positive acknowledgement scheme is employed to ascertain the successful reception of a frame. Specifically, upon reception of a data frame, the destination station initiates the transmission of an acknowledgement frame (ACK) after a time interval called *Short InterFrame Space* (SIFS). The SIFS is shorter than the DIFS (see Figure 3) in order to give priority to the receiving station over other possible stations waiting for transmission. If the ACK is not received by the source station, the data frame is presumed to have been lost, and a retransmission is scheduled. The ACK is not transmitted if the received packet is corrupted. A Cyclic Redundancy Check (*CRC*) algorithm is used for error detection.

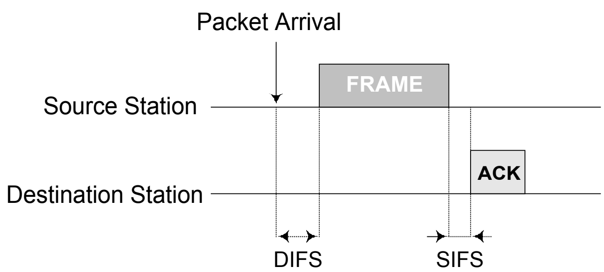


Figure 3. Interaction between the source and destination stations. The SIFS is shorter than the DIFS.

After an erroneous frame is detected (due to collisions or transmission errors), a station must remain idle for at least an *Extended InterFrame Space* (EIFS) interval before it reactivates the backoff algorithm. Specifically, the EIFS shall be used by the DCF whenever the physical layer has indicated to the MAC that a frame transmission was begun that did not result in the correct reception of a complete MAC frame with a correct FCS value. Reception of an error-free frame during the EIFS re-synchronizes the station to the actual busy/idle state of the medium, so the EIFS is terminated and normal medium access (using DIFS and, if necessary, backoff) continues following reception of that frame.

2.2 Common Problems in Wireless Ad Hoc Networks

In this section we discuss some problems that can arise in wireless networks, mainly in the ad hoc mode. The characteristics of the wireless medium make wireless networks fundamentally different from wired networks. Specifically, as indicated in [IEE99]:

- the wireless medium has neither absolute nor readily observable boundaries outside of which stations are known to be unable to receive network frames;
- the channel is unprotected from outside signals;
- the wireless medium is significantly less reliable than wired media;
- the channel has time-varying and asymmetric propagation properties.

In wireless (ad hoc) network that relies upon a carrier-sensing random access protocol, like the IEEE 802.11 DCF protocol, the wireless medium characteristics generate complex phenomena such as the hidden-station and exposed-station problems.

Figure 4 shows a typical "hidden station" scenario. Let us assume that station B is in the transmitting range of both A and C, but A and C cannot hear each other. Let us also assume that A is transmitting to B. If C has a frame to be transmitted to B, according to the DFC protocol, it senses the medium and finds it free because it is not able to hear A's transmissions. Therefore, it starts transmitting the frame but this transmission will results in a collision at the destination Station B.

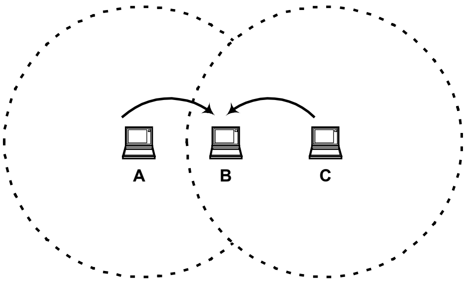


Figure 4. The "hidden station" problem.

The hidden station problem can be alleviated by extending the DCF basic mechanism by a **virtual carrier sensing** mechanism (also referred to as floor acquisition mechanism) that is based on two control frames: *Request To Send* (*RTS*) and *Clear To Send* (*CTS*), respectively. According to this mechanism, before transmitting a data frame, the station sends a short control frame, named RTS, to the receiving station announcing the upcoming frame transmission (see Figure 5). Upon receiving the RTS frame, the destination station replies by a CTS frame to indicate that it is ready to receive the data frame. Both the RTS and CTS frames contain the total duration of the transmission, i.e., the overall time interval needed to transmit the data frame and the related ACK. This information can be read by any listening station that uses this information to set up a timer called *Network Allocation Vector* (*NAV*). While the NAV timer is greater than zero the station must refrain from accessing the wireless medium. By using the RTS/CTS mechanism, stations may become aware of transmissions from hidden station and on how long the channel will be used for these transmissions.

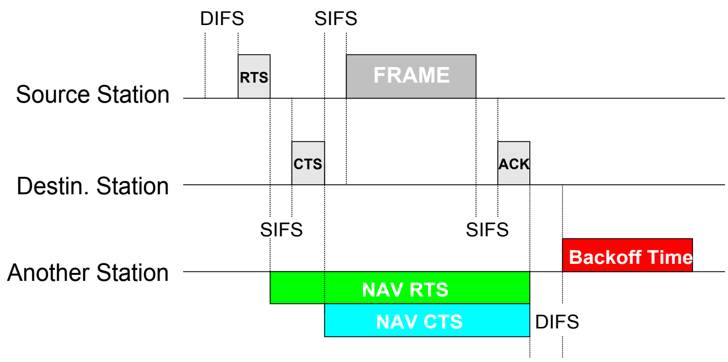


Figure 5. Virtual Career Sensing mechanism.

Figure 6 depicts a typical scenario where the "exposed station" problem may occur. Let us assume that both Station A and Station C can hear transmissions from B, but Station A can not hear transmissions from C. Let us also assume that Station B is transmitting to Station A and Station C receives a frame to be transmitted to D. According to the DCF protocol, C senses the medium and finds it busy because of B's transmission. Therefore, it refrains from transmitting to C although this transmission would not cause a collision at A. The "exposed station" problem may thus result in a throughput reduction.

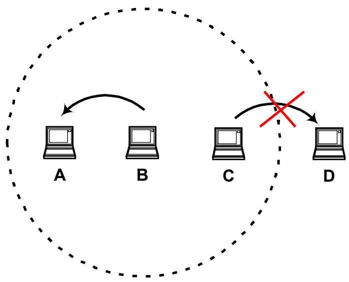


Figure 6. The "exposed station" problem

2.3 Ad Hoc Networking Support

In this section we will describe how two or more 802.11 stations set up an ad hoc network. In the IEEE 802.11 standard, an ad hoc network is named *Independent Basic Service Set* (IBSS). An IBSS enables two or more 802.11 stations to communicate each other without the intervention of either a centralized AP, or an infrastructure network. Hence, the IBSS can be considered as the support provided by the 802.11 standard for mobile ad hoc networking.[4]

Due to the flexibility of the CSMA/CA protocol, to receive and transmit data correctly it is sufficient that all stations within the IBSS are synchronized to a common clock. The standard specifies a Timing Synchronization Function (TSF) to achieve clock synchronization between stations. In an infra-structured network the clock synchronization is provided by the AP, and all stations synchronizes their own clock to the AP's clock. In an IBSS, due to the lack a centralized station, clock synchronization is achieved through a distributed algorithm. In both cases synchronization is obtained by transmitting special frames, called *beacons,* containing timing information.

The TSF requires two fundamental functionalities, namely *synchronization maintenance* and *synchronization acquirement*, that will be sketched below. We only focus on IBSS.

2.3.1 Synchronization maintenance.

Each station has a *TSF timer* (clock) with modulus 2^{64} counting in increments of microseconds. Stations expect to receive beacons at a nominal rate defined by the a BeaconPeriod parameter. This parameter is decided by the station initiating the IBSS, and is then used by any other station joining the IBSS. Stations use their TSF timers to determine the beginning of beacon intervals or periods. At the beginning of a beacon interval each station performs the following procedure:

i. it suspends the decrementing of the backoff timer for any pending (non-beacon) transmission;

[4] To uniquely identify a IBSS it is necessary to associate to it an identification number (IBSSID) that is locally administered and that will be used by any other Station to join the IBSS, i.e., the ad hoc network. When a station starts a new IBSS, it generates a 46-bit random number in a manner that minimizes the probability that the same number is generated by another station.

ii. it generates a random delay interval uniformly distributed in the range between zero and twice the minimum value of the Contention Window.

iii. it waits for the random delay;

iv. if a beacon arrives before the random delay timer has expired, it stops the random delay timer, cancel the pending beacon transmission, and resumes the backoff timer;

v. if the random delay timer has expired and no beacon has been received, it sends a beacon frame.

The sending station sets the beacon timestamp to the value of its TSF timer at the time the beacon is transmitted. Upon reception of a beacon, the receiving station looks at the timestamp. If the beacon timestamp is later than the station's TSF timer, the TSF timer is set to the value of the received timestamp. In other words, all stations within the IBSS synchronize their TSF timer to the quickest TSF timer.

2.3.2 Synchronization acquirement.

This functionality is necessary when a station wants to join an already existing IBSS. The discovery of existing IBSSs is the result of a scanning procedure of the wireless medium during which the station receiver is tuned to different radio frequencies, looking for particular control frames. Only if the scanning procedure does not result in finding any IBSS, the station may start with the creation of a new IBSS. The scanning procedure can be either passive or active.

In a passive scanning the station listens to the channels for hearing a beacon frame. It is worth reminding that a beacon frame contains not only timing information for synchronization, but also the complete set of IBSS parameters. This set includes the IBSS identifier IBSSID, the aBeaconPeriod parameter, the data rates that can be supported, the parameters relevant to IBSS management functions (e.g., power saving management).

Active scanning involves the generation of Probe frames, and the subsequent processing of received Probe Response frames. The station that decides to start an active scanning procedure has a ChannelList of radio frequencies that will be scanned during the procedure. For each channel to be scanned a probe with broadcast destination and is sent by using the DCF access method. At the same time a ProbeTimer is started. If no response to the probe is received before the ProbeTimer reaches the MinChannelTime the next channel of the list is considered. Otherwise, the station continues to scan the same channel until the timer reaches the MaxChannelTime. Then, the station processes all received Probe responses.

Probe responses are sent using normal frame transmission rules as directed frames to the address of the station that generated the Probe request. In an IBSS, only the station that generated the last beacon transmission will respond to a probe request, in order to avoid the waste of bandwidth with repetitive control frames. In each IBSS, at least one station must be awake at any given time to respond to Probe request. Therefore, the station that sent the last beacon remains in the awake state in order to respond to Probe requests, until a new beacon is received. There may be more than one station in a IBSS that responds to a given probe request, particularly in the case where more than one station transmitted a beacon, either due to not receiving successfully a previous beacon, or due to collision between beacon transmissions.

2.4 Power Management

In a mobile environment, portable devices have limited energetic resources since they are powered through batteries. Power management functionalities are thus extremely important both in the infrastructure-based and in the ad hoc modes. Obviously, in the ad hoc mode, i.e., inside an IBSS, Power Saving (PS) strategies need to be completely distributed in order to preserve the self-organizing nature of the IBSS. A station may be in one of two different power states: *awake* (station is fully powered) or *doze* (the station is not able to transmit or receive). Multicast and/or directed frames destined to a power-conserving station are first announced during a period when all stations are awake. An Ad hoc Traffic Indication Map (ATIM) frame does the announcement. A station operating in the PS mode listens to these announcements and, based on them, decides whether it has to remain awake or not.

ATIM frames are transmitted during the ATIM Window, a specific period of time following the beginning of a Beacon period whose length is defined by the aATIMWindow parameter (an IBSS parameter included in the beacon content). During the ATIM Window, only beacon and ATIM frames can be exchanged and all stations must remain awake. Directed ATIM frames are to be acknowledged by the destination station, while multicast ATIMs are not to be acknowledged. Hence a station sends a directed ATIM frame and waits for the acknowledgement. If this acknowledgement does not arrive it executes the backoff procedure for re-transmitting the ATIM frame.

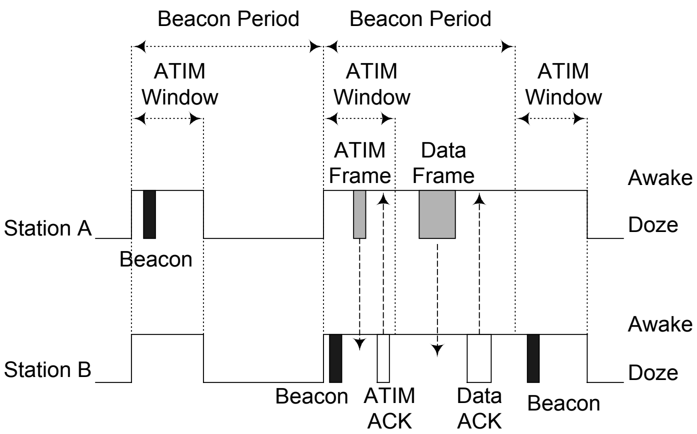


Figure 7. A data exchange between stations operating in PS mode in an ad hoc network.

A station receiving a directed ATIM frame must send the acknowledgement and remain awake for the entire duration of the beacon interval, waiting for the announced data frame. Data frames are transmitted at the end of the ATIM Window according to the DCF access method (see Figure 7). If a station does not receive any ATIM frame during the ATIM Window can enter the doze state at the end of the ATIM window.

3. Simulation Analysis of IEEE 802.11 Ad Hoc Networks

As mentioned above, in this chapter we are primarily interested in the performance provided by the 802.11 MAC protocol in an ad hoc environment. In this framework, almost all previous works are based on simulation and have looked at the performance of TCP applications. Less attention has been devoted to UDP applications (this can be easily justified since, currently, the most popular applications use TCP as the transport protocol).

The previous studies have been pointed out several performance problems. They can be summarized as follows. In a dynamic environment, mobility may have a severe impact on the performance of the TCP protocol [Hol99, Hol02, Cha01, Liu01, FuM02, Ahu00, Dye01]. However, even when stations are static, the performance of an ad hoc network may be quite far from ideal. It is highly influenced by the operating conditions, i.e., TCP parameter values (primarily the congestion window size) and network topology [Fu02, Li01]. In addition, the interaction of the 802.11 MAC protocol (hidden and exposed station problems, exponential backoff scheme, etc.) with TCP mechanisms (congestion control and time-out) may lead to unexpected phenomena in a multi-hop environment. For example, in the case of simultaneous TCP flows, severe unfairness problems and - in extreme cases - capture of the channel by few flows [Tan99, XuS01, XuS02, XuB02, XuG02] may occur. Even in the case of a single TCP connection, the instantaneous throughput may be very unstable [XuS01, XuS02]. Such phenomena do not appear, or appear with less intensity, when the UDP protocol is used [XuG02].

In the next subsections we will briefly survey the findings of the previous studies. To better understand the results presented below, it is useful to provide a model of the relationships existing among stations when they transmit or receive. In particular, it is useful to make a distinction between the transmission range, the interference range and the carrier sensing range. The following definitions can be given.

The *Transmission Range* (*TX_range*) is the range (with respect to the transmitting station) within which a transmitted packet can be successfully received. The transmission range is mainly determined by the transmission power and the radio propagation properties.

The *Physical Carrier Sensing Range* (*PCS_range*) is the range (with respect to the transmitting station) within which the other stations detect a transmission. It mainly depends on the sensitivity of the receiver (the receive threshold) and the radio propagation properties.

The *Interference Range* (*IF_range*) is the range within which stations in receive mode will be "interfered with" by a transmitter, and thus suffer a loss. The interference range is usually larger than the transmission range, and it is a function of the distance between the sender and receiver, and of the path loss model. It is very difficult to predict the interference range as it strongly depends on the ratio between power of the received "correct" signal and the power of the received "interfering" signal. Both these quantities heavily depend on several factors (i.e., distance, path, etc.) and hence to estimate the interference we must have a detailed snapshot of the current transmission and relative station position.

In the simulation studies presented hereafter the following relationship has been generally assumed: $TX\_range \le IF\_range \le PCS\_range$. For example, in the ns-2 simulation tool [Ns-2] the following values are used to model the characteristics of the physical layer: $TX\_range = 250m$, $IF\_range = PCS\_range = 550m$.

3.1 Influence of mobility

Station mobility may severely degrade the performance of the TCP protocol in mobile ad hoc networks (MANETs) [Hol99, Hol02, Cha01, Liu01, FuM02, Ahu00, Dye01]. This is due to the inability of the TCP protocol to manage efficiently the effects of mobility. Station movements may cause route failures and route changes and, hence, packet losses and delayed ACKs. The TCP misinterprets these events as congestion signals and activates the congestion control mechanism. This leads to unnecessary retransmissions and throughput degradation. In addition, mobility may exacerbate the unfairness between competitive TCP sessions [Tan99].

Numerous new mechanisms have been proposed for optimizing the TCP performance in MANETs, including the adaptation of TCP error-detection and recovery mechanisms to the mobile ad hoc environment. [Cha01] proposes to introduce explicit signaling (Route Failure and Route Re-establishment notifications) from intermediate stations to notify the sender TCP of the disruption of the current route, and construction of a new one. Upon receiving a route failure notification the sender TCP does not activate the congestion control mechanism, but simply freezes its status that will be resumed when a Route Establishment notifications is been received.

In [Hol02] an Explicit Link Failure Notification (ELFN) is still used to notify the sender TCP about a route failure. However, no explicit signaling about route reconstruction is provided. [Mon00] presents a simulation study of the ELFN mechanism, both in static and dynamic scenarios. This study points out the limitations of this approach that are intrinsic to TCP properties (e.g., long recovery time after a timeout), and proposes to implement mechanisms below the TCP layer. A similar approach is taken in [Liu01] where the standard TCP is unmodified but new mechanisms are implemented in a thin layer, Ad hoc TCP (ATCP), between TCP and IP. ATCP uses Explicit Congestion Notifications (ECN) and ICMP "destination unreachable" messages to discriminate congestion conditions from link failures, and from packet losses in wireless links. The ATCP takes the appropriate actions according to the type of event recognized.

All previous techniques require an explicit notification from intermediate stations to the sender TCP. To avoid this complexity, a heuristic is used in [Dye01] to distinguish route failures from congestions. When timeouts occur consecutively the sender TCP assumes that a route failure occurred rather than a network congestion. The unacknowledged packet is retransmitted again but the retransmission timeout is not doubled a second time. The retransmission timeout remains fixed until the route is re-established and the packet is acknowledged. An implicit detection approach is also taken in [Wan02] where the authors propose to infer route changes by observing the out-of-order delivery events.

3.2 Influence of the network topology

Even in a static environment, the performances of an ad hoc network are strongly limited by the interaction between neighboring stations [Li01]. Stations' activity is limited by the behavior of neighboring stations (a station must sense the medium before start transmitting) and by stations in its interfering range (interferences may cause collisions at the destination station). For example, it can be shown that in a string (or chain) topology, like the one shown in Figure 8, the expected maximum bandwidth utilization is only 0.25 [Li01]. However, things may be even worse in practice. This discrepancy is due to 802.11 MAC inability to find the optimum schedule of transmissions by itself. In particular, in a chain topology it happens that stations early in the chain starve later stations (similar consideration apply to other network topologies). In general, the 802.11 MAC protocol appears to be more efficient in case of local traffic patterns, i.e., when the destination is close to the sender [Li01].

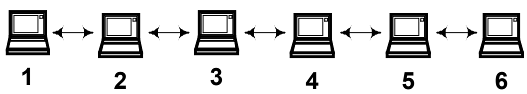


Figure 8. A string network topology.

3.3 Influence of the TCP congestion window size

The TCP congestion window size may have a significant impact on performance. In [Fu03] it is shown that, for a given network topology and traffic pattern, there exist an optimal value of the TCP congestion window size at which the channel utilization is maximized. However, the TCP does not operate around this optimal value and typically grows its average window size much larger, leading to decreased throughput (throughput degradation is in the order of 10-

30% with respect to the optimal case) and increased packet losses. This behavior can be explained by considering the origin of packet losses that in ad hoc networks is completely different than in traditional wired networks. In ad hoc networks, packet losses caused by buffer overflows at intermediate stations are rare events (unless the station buffer is very small), while packet losses due to link-layer contention (i.e., a station that fails to reach its adjacent station, see Section 3.4 and [XuS01]) are largely dominant. Furthermore, the multi-hop wireless network collectively exhibits graceful loss behavior. In general, the link loss probability is insufficient to stabilize the average TCP congestion window around the optimal value. To achieve this objective Fu and others propose two link level mechanisms: Link RED, and adaptive spacing [Fu03]. Similarly to the RED mechanism implemented in Internet routers, the Link RED tunes the packet loss probability at the link layer by marking/discarding packet according to the average number of retries experienced in the transmission of previous packets. The Link RED thus provides TCP with an early sign of overload at link level. Adaptive spacing is introduced to improve spatial channel reuse, thus reducing the risk of stations' starvation. The idea here is the introduction of extra backoff intervals to mitigate the exposed receiver problems. Adaptive spacing is complementary to Link RED: it is activated only when the average number of retries experienced in previous transmission is below a given threshold.

3.4 Effects of the interaction between MAC protocol and TCP mechanisms

The interaction of some features of the 802.11 MAC protocol (hidden/exposed station problem, exponential backoff scheme, etc.) with the TCP protocol mechanisms (mainly, the congestion control mechanism) may lead to several, unexpected, serious problems. S. Xu and Saadawi identified these problems through a simulation analysis of a multi-hop ad hoc network via the *ns* network simulator tool [XuS01] . The same results have been confirmed with a different simulation tool [XuS02]. Recently, similar phenomena have been also observed in other scenarios [XuB02, XuS02].

Specifically, in [XuS01] and [XuS02] it is pointed that the following problems may affect the TCP performance in a multi-hop ad hoc environment.

(i) The instantaneous throughput of a TCP connection may be very unstable (dropping frequently to zero) even when this is the only active connection in the network (*instability problem*).

(ii) In case of two simultaneous TCP connections, it may happen that the two connections can not coexist: when one connection develops the other one is shut down (*incompatibility problem*).

(iii) With two simultaneous TCP connections, if one connection is single-hop and the other one is multiple-hop, it may happen that the instantaneous throughput of the multiple-hop connection is shut down as soon as the other connection becomes active (even if the multiple-hop connection starts first). There is no chance for the multiple-hop connection once the one-hop connection has started (*one-hop unfairness problem*).

The above problems have been revealed in a string network topology like the one shown in Figure 8 where the distance between any two neighboring stations is 200 m and stations are static. According to the 802.11 based Wave-Lan, the nominal transmission radius of each station has been set to 250 m (each station can thus communicate only with its neighboring stations). Furthermore, the sensing and interfering ranges have been set to twice the transmission range, i.e., 500 m [XuS01, XUS02], i.e., the typical setting of the ns-2 simulator.

Below we will provide a brief explanation of how the one-hop unfairness problem arises. Similar explanations can be provided for the instability and incompatibility problems, but are omitted for the sake of space. The reader can refer to [XuS02] for a detailed analysis of all cases.

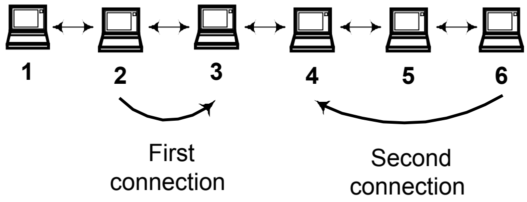


Figure 9. A string topology with two TCP connections. The First connection is one-hop; the second connection is two-hop.

Figure 9 shows two TCP connections. The first connection is from Station 2 to Station 3 (one-hop connection), while the second connection is from Station 6 to Station 4 (two-hop connection). Let us assume, for example, that Station 2 is transmitting a data frame to Station 3 (e.g., a TCP segment), and Station 5 wants to transmit a frame to Station 4. According to the 802.11 MAC protocol, Station 5 tries to send an RTS frame, and, then, wait for the corresponding CTS frame. However, Station 5 never receives this CTS frame.

Most of the RTS transmission attempts tried by Station 5 results in a collision at Station 4 due to the interference of Station 2 (hidden station problem). Station 5 cannot hear the CTS from station 3 because it is out of the transmission range of Station 3 and, thus, it is not aware of Station 2 transmission. However, Station 4 is in the interfering range of Station 2 since the interfering range is larger than the transmission range (twice in ns-2 simulator). Even if Station 4 successfully receives the RTS frame it is not able to reply with the corresponding CTS frame, again due to Station 2. Tough Station 4 is out of the transmission range of Station 2, however, Station 4 can sense the transmission of Station 2 since the sensing range is larger than the transmission range (twice in the ns-2 simulator). This inhibits Station 4 from accessing the wireless medium (exposed station problem).

After failing to receive the CTS frame from Station 4 for seven times, Station 5 reports a link breakage to its upper layer and a route-failure notification is sent to Station 6, i.e., the data packet originator. Upon receiving this notification Station 6 starts a route discovery process. Obviously, while looking for a new route no data packet can flow along the connection and this makes the instantaneous throughput drop to zero.

The above example allows us to understand why the instantaneous throughput of the two-hop connection drops to zero. However, it not yet clear why this throughput remains to zero in *most of the connection lifetime*. To better clarify this point the following additional remarks need to be taken into account.

- Since Station 5 is in the interfering range of Station 3, it has to defer when Station 3 is sending. Therefore, Station 5 can transmit an RTS frame only when Station 3 is not sending.
- In the one-hop connection as soon as Station 2 receives a TCP ACK from Station 3, it immediately prepares itself to send another TCP segment. This means that Station 5 has very few opportunity to find the channel free.
- Data frames (i.e., TCP segments) transmitted by Station 2 are usually much larger in size than RTS frames that Station 5 tries to transmit.

In conclusion, the time available for Station 5 for successfully accessing the channel is very small. In addition, the exponential backoff scheme used by the 802.11 MAC protocol always favors the last succeeding station.

From the above description, it emerges that several features of the multi-hop ad hoc environment contribute to the "capture" of the channel by the one-hop connection. The most important and direct causes are the hidden station and the exposed station problems. These problems, in their turn, are caused by the larger size of the interfering and sensing ranges with respect to the transmission range. However, the random backoff scheme of the 802.11 MAC protocol also contributes by favoring the last succeeding station.

The "capture" effect revealed in [XuS01, XuS02] is not peculiar of the string network topology. Gerla and al. observed the same phenomenon even in other scenarios [XuG02, XuB02]. In [XuG02] they also propose two possible solutions to remove the capture effect: (i) replacement of the binary backoff scheme in the 802.11 MAC protocol by an adaptive retransmission timeout based on the number of active neighboring stations; and (ii) the use of special antennas that reduce interference during packet reception.

4. Experimental Analysis of IEEE 802.11 Ad Hoc Networks

In the previous section we have seen that there exists an extensive literature that has investigated TCP performance in ad hoc networks, especially over the IEEE 802.11 MAC protocol. Most papers report the same type of unfairness problems. The hidden and exposed station problem, the large interference range, and the backoff scheme of IEEE 802.11 MAC protocol, have been recognized as the major responsible for these unfairness problems. All these previous analysis were carried by simulation and, hence, the results observed are highly dependent on the physical layer model implemented in the simulation tool used in the analysis (e.g., GloMosim [Glo02], ns-2 [Ns02], Qualnet [Qua02]). Hereafter, we validate and extend these results by presenting a similar analysis that has been carried on a real testbed. Since the simulation results presented in Section 3 were obtained by considering IEEE 802.11 network cards operating at the nominal bit rate of 2Mbps, most of the measurement studies presented in this section refer to the IEEE 802.11 standard [IEE99]. However, in Section 5 we will also investigate the performance of the IEEE 802.11b ad hoc networks.

It is worth pointing out that, while in the simulation studies presented above the values of TX_range, PCS_range, and IF_range are known and constant, in the real world the physical channel has time-varying and asymmetric propagation properties. Hence, the values of TX_range, PCS_range, and IF_range may be highly variable even during the same experiment.

4.1 Experimental Testbed

The measurement testbed is based on laptops running the Linux-Mandrake 7.2 operating system. The laptops are equipped with Lucent WaveLAN IEEE 802.11 network cards using the DSSS technique, and operating at the nominal bit rate of 2Mbps. The target of our study is the analysis of the TCP performance over an IEEE 802.11 ad hoc network. Since the aim of the study is to investigate the impact of the CSMA/CA protocol on the TCP performance, static ad hoc networks (i.e., the network stations do not change their position during an experiment) with single-hop TCP connections were considered. This allows to remove other possible causes that may interfere with the TCP behavior, e.g., link breakage, route re-computation, etc.

4.2 Indoor Experiments

The indoor experiments were carried out in the scenario depicted in Figure 10. Stations numbered as S1, S2 and S3 have an active ftp session towards Station S4, i.e., data frames are transmitted to S4 that replies with ACK packets. As ftp data transfers are supported by the TCP protocol, in the following the data flows will be denoted as TCP*i*, where *i* is the index of the transmitting station. As shown in the figure, a reinforced concrete wall (represented by the gray rectangle) is located between stations S1 and S2, and between stations S2 and S3. As a consequence, S1, S2 and S3 are outside the *TX_ range* of each other.[5] Furthermore, each Station S*i* (where *i*= {1,2,3}) is in the transmission range of S4. Therefore, this is a typical hidden-station scenario where it is expected that the RTS/CTS mechanism (by avoiding hidden station collisions) should provide a significant throughput gain with respect to the basic CSMA/CA protocol.

Two sets of experiments were performed in this scenario. In the first set only two ftp sessions are active: TCP1 and TCP2. In the second set all three sections are active.

[5] This was verified by running the Ping program for a sufficiently long time from each station to the other stations. In no case a packet was successfully delivered among each couple of stations.

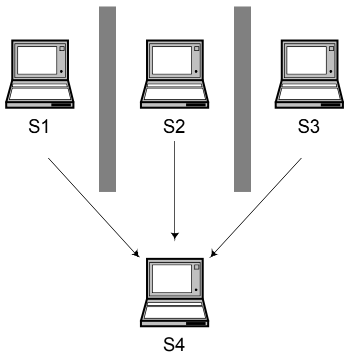
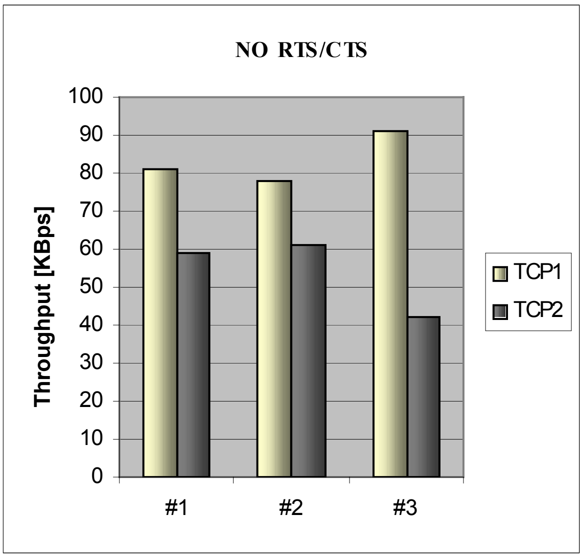


Figure 10. Indoor scenario.

Table 1. Reference Throughputs in Kbyets/sec (KBps)			
	Packet size 1460 Bytes	Packet size 512 Bytes	
	ftp/TCP traffic	ftp/TCP traffic	CBR/UDP traffic
Throughputs Basic Access	145 KBps	125 KBps	165 KBps
Throughputs RTS/CTS	135 KBps	110 KBps	140 KBps



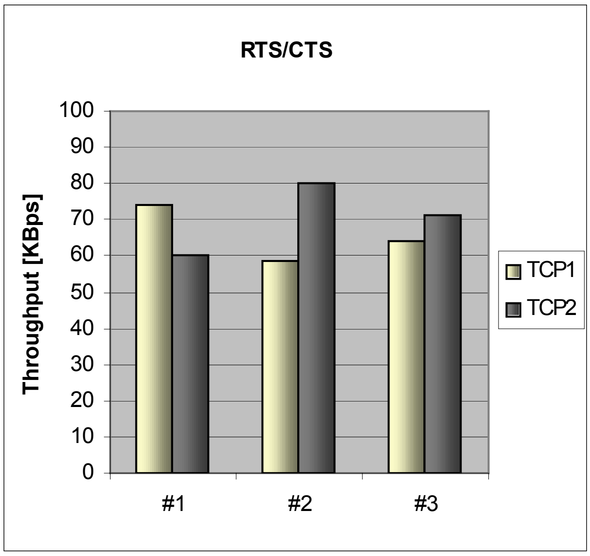


Figure 11. Throughput (in KBps) estimated in the indoor scenario with two ftp sessions with the basic CSMA/CA access (left) and the RTS/CTS mechanism (right), respectively.

To better analyze the results we also performed some reference experiments. Specifically, we measured the maximum throughput (at the application layer) of a single sender-receiver session when the two stations are very close to each other (in the same room), and no other session is active. The estimated throughput represents the upper bound throughput for a sender-receiver session and is reported in Table 1 for different operating conditions.

Let us now start analyzing the results related to the indoor scenario. The results obtained in the scenario with two active sessions (TCP1 and TCP2) are summarized in Figure 11. These results refer to a 60-second ftp transfer that utilizes TCP packets with a 1460-byte payload size. Two types of experiments were done: with and without the RTS/CTS mechanism. For each type, we performed three experiments under the same conditions.

The following remarks can be made based on the above results:

i) the RTS/CTS mechanism does not provide any significant performance improvement with respect to the basic access mechanism;

ii) the RTS/CTS mechanism provides an aggregate throughput slightly lower than the basic access mechanism. This is due to the additional overhead introduced by the RTS and CTS frames,

iii) in each experiment, the aggregate throughput is not very far from the reference throughput reported in Table 1 (i.e., 145 and 135 KBps for the basic access and the RTS/CTS mechanism, respectively).

These observations are confirmed by the results obtained in the scenario with three ftp sessions active and summarized in Table 2. For each set of experiments, Table 2 reports the throughput averaged on all the experiments performed under the same conditions.

Table 2. Throughput (in KBps) estimated in the indoor scenario when all three ftp sessions are active.

	TCP1	TCP2	TCP3	Aggregate
Basic Access	42	29.5	57	128.5
RTS/CTS	34	27	48	109

These results indicate that the carrier sensing mechanism is still effective even if the transmitting stations are “apparently” hidden to each other. This can be explained by remembering that the carrier sensing range is about twice the transmission range. Hence, if two stations (outside the transmission range of each other) are in the transmission range of a third station there is a very high probability that they can sense each other. In these cases, the physical carrier sensing is effective, and hence adding a virtual carrier sensing (i.e., the RTS/CTS mechanism) is useless.

4.3 Outdoor Experiments

To better investigate the phenomena observed in the indoor environment, the testbed was moved to an outdoor space. Each station was located in an open environment (a field without buildings) in order to analyze the TCP behavior when hidden and/or exposed stations may be present. In all experiments the WLAN was set to 2Mbps.

The network scenario for the outdoor experiments is shown in Figure 12. In this scenario, we may have two contemporary active sessions. Specifically, Station S1 communicates with Station S2 (Session 1), while Station S3 is in communication with Station S4 (Session 2). In the figure, the arrows represent the direction of the data flow (e.g., S1 is delivering data to S2), and $d(i,j)$ is the distance between stations Si and Sj. Data to be delivered are generated by either an ftp application, or a Continuous Bit Rate (CBR) application. In the former case the TCP protocol is used at the transport layer, while in the latter case UDP is the transport protocol.

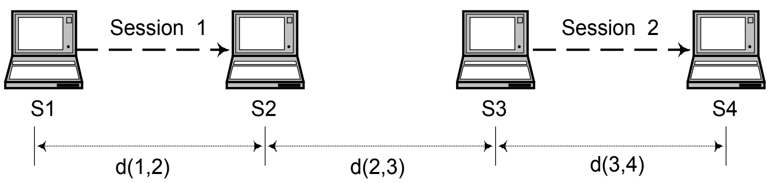


Figure 12. Reference network scenario for the outdoor experiments.

We performed a preliminary set of experiments aimed at estimating the *Tx_range* in the outdoor environment where the experiments were done. We used the following procedure.

We considered a single couple of stations, let's say S1 and S2. Then, starting from zero, we progressively increased the distance d(1,2) between these two stations until they were no longer able to exchange data. For each value of d(1,2) the ping application was used to test the connectivity between the stations. By applying this procedure several times, we obtained that the transmission range is in the order of 40 m. It is worth pointing out that, in a real environment, the value of *TX_range* is not constant. On the other hand, it is highly variable depending on several factors: weather conditions, hour of the day, place and time of the experiment, etc.

Then, we performed several experiments with Session 1 and Session 2 simultaneously active. In all experiments the receiving station is always in the transmission range of its transmitting station - i.e., Station S2 (S4) is in the transmitting range of Station S1 (S3). On the other hand, the distance *d*(2,3) between the two couples of stations[6] is variable. Depending on the actual *d*(2,3) value the following situation can occur.

1. All stations are within the transmission range of each other (Type 1). This means that in our testbed the distance between any two stations must be less than 40 m.
2. Extreme case: the two sessions are far from each other (Type 2). In our testbed this is achieved by setting *d*(2,3)>90 m (i.e., more than twice the minimum transmission range size);.
3. Intermediate case 1: is obtained by setting *d*(2,3)=65 m (Type 3).
4. Intermediate case 2: is obtained by setting *d*(2,3)=15 m (Type 4).

[6] That is, the couple (3,4) with respect to the couple (1,2), and vice-versa.

In all experiments ftp data traffic was transmitted and the TCP protocol was used at the transport layer.[7] For this reason the two sessions will be indicated below as TCP1 and TCP2. The payload size of TCP packets was set to 512 bytes.

Table 3. Throughputs in Kbytes/sec (KBps) measured in Type 1 and Type 2 experiments.

	Type 1		Type 2	
	TCP 1	TCP 2	TCP 1	TCP 2
No RTS/CTS	61	54	122.5	122
RTS/CTS	59.5	49.5	96	100

The results obtained for Type 1 and Type 2 experiments are summarized in Table 3. These experiments produced the expected results. In Type 1 experiments (all stations within the same transmission range) the two ftp sessions fairly share the bandwidth, and the aggregate throughput is close to the *reference throughput* for this configuration (see Table 1). From the above results it also appears that the RTS/CTS mechanism is useless since it only reduces the aggregate throughput (due to the overhead introduced by the RTS and CTS frames).

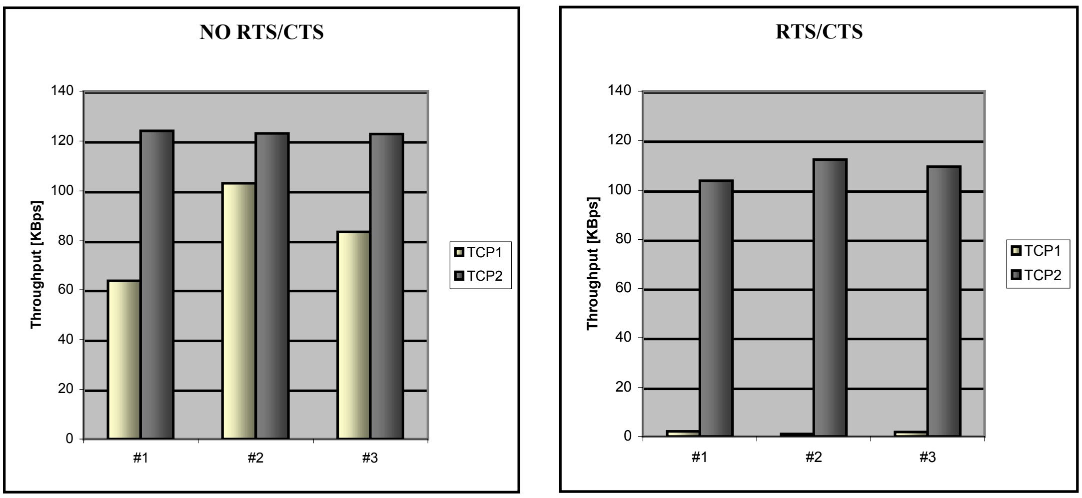


Figure 13. Throughputs (in KBps) measured in the outdoor scenario in Type 3 experiments with (right) and without (left) the RTS/CTS mechanism.

[7] The length of each experiment is 120 seconds.

In Type 2 measurements the two sessions are independent, and they both achieve a throughput very close to the reference throughput. Again, the RTS/CTS mechanism is useless since it only introduces overhead.

Unlike the previous ones, Type 3 and Type 4 experiments exhibited a very strange and unpredictable behavior as shown in Figure 13 and Figure 14, respectively. In Type 3 experiments stations S2 and S3 are 65 m apart from each other. It can be observed that the use of the RTS/CTS mechanism produces a capture of the channel by the second session (i.e., S3-S4). A possible explanation for this behavior is that Station S2 is often blocked by S3 data transmissions to S4. Hence, it may not be able to reply to the RTS frame of S1. On the other hand, session S3-S4 is only marginally affected by session S1-S2 as the only possible impact is due to S3 being blocked by S2's (CTS and ACK) transmissions. When using the basic access mechanism, S1 can start transmitting to S2 without almost any interference from session S3-S4.

It is also worth noting that by using the basic access the second session does not reduce its throughput (actually, the throughput of TCP2 increases as the RTS/CTS overhead is removed). Indeed, with the basic access each session achieve a higher throughput.

To summarize, in this configuration the RTS/CTS mechanism, by adding further correlations between the stations' behavior (S1 cannot start transmitting if S2 does not reply with a CTS frame), produces a block of the first session without providing any advantage to the other one.

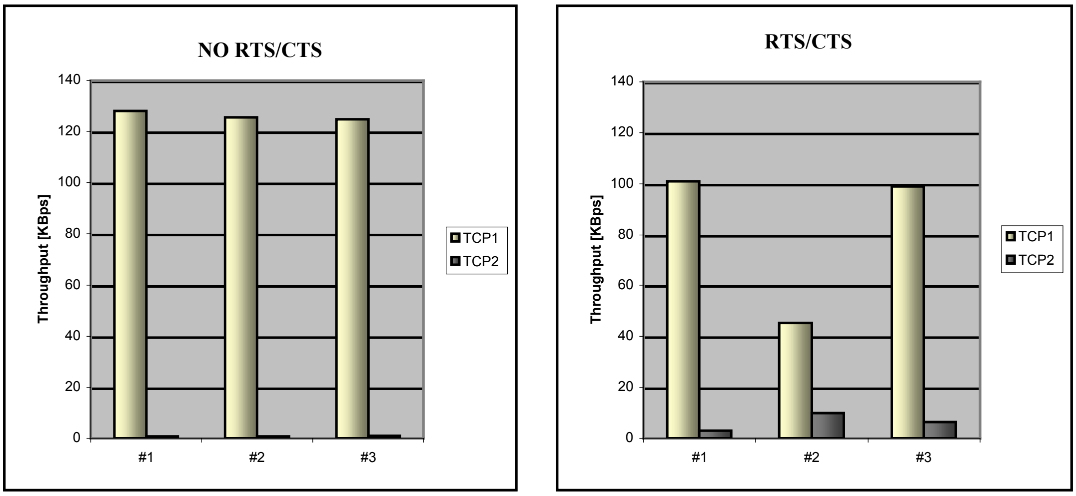


Figure 14.Throughputs (in KBps) measured in the outdoor scenario in Type 4 experiments with (right) and without (left) the RTS/CTS mechanism.

In Type 4 experiments, whose results are shown in Figure 14, we observed the capture of the channel by one of the two TCP connections. In this case the RTS/CTS mechanism provided a little help in solving the problem.

The experimental results presented above confirm the unfairness/capture problems of the TCP protocol in IEEE 802.11 ad hoc networks revealed in previous simulation studies. As briefly discussed in Section 3, the TCP protocol (specifically the flow/congestion control mechanism) by introducing correlations in the transmitted traffic emphasizes these phenomena. This effect is clearly pointed out by the experimental results shown in

Figure 15. This figure still refers to the Type 4 configuration but traffic flows are now generated by CBR sources and the UDP protocol is used instead of TCP. As it clearly appears, the capture effects disappear.

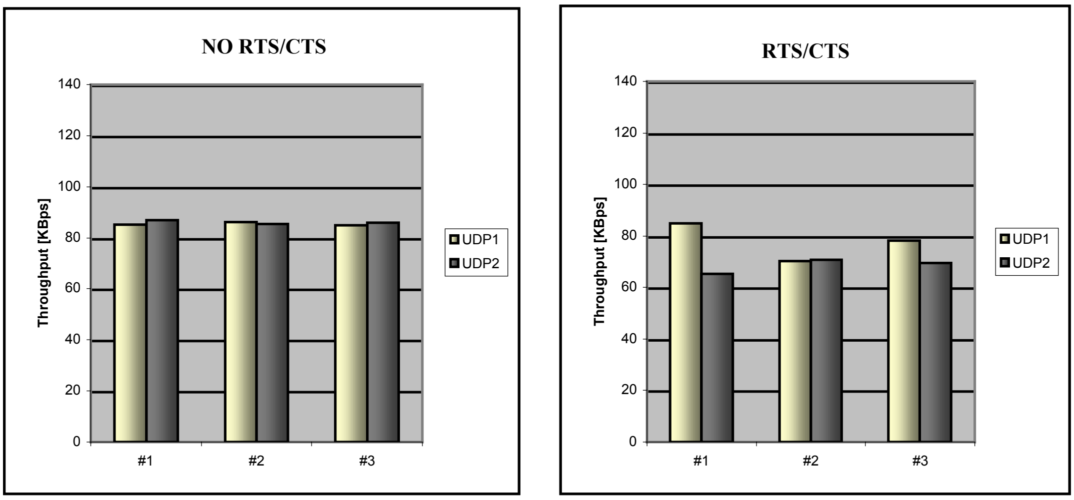


Figure 15. Type 4 experiments with CBR/UDP traffic.

In conclusion, the experimental results have confirmed that, in some scenarios, TCP connections may actually experience significant throughput unfairness, and even capture of the channel by one of the connections, as pointed out in previous simulation studies. Furthermore, it has been clearly shown that the RTS/CTS mechanism might be completely ineffective when there are stations that are outside their respective transmission ranges but within the same carrier sensing range. In these cases the physical carrier sensing is sufficient to regulate the channel access and the virtual carrier sensing (i.e., the RTS/CTS mechanism) is useless.

5. IEEE 802.11b

The results presented in the previous section have been obtained by considering IEEE 802.11-based ad hoc networks. Currently, however, the Wi-Fi network interfaces are becoming more and more popular. Wi-Fi cards implement the IEEE 802.11b standard. It is therefore important to extend the previous experimental analysis to IEEE 802.11b ad hoc networks.

The 802.11b standard extends the 802.11 standard by introducing a higher-speed Physical Layer in the 2.4 GHz frequency band still guaranteeing the interoperability with 802.11 cards. Specifically, 802.11b enables transmissions at 5.5 Mbps and 11 Mbps, in addition to 1 Mbps and 2 Mbps. 802.11b cards may implement a dynamic rate switching with the objective of improving performance. To ensure coexistence and interoperability among multirate-capable stations, and with 802.11 cards, the standard defines a set of rules that must be followed by all stations in a WLAN. Specifically, for each WLAN is defined a *basic rate set* that contains the data transfer rates that all stations within the WLAN must be capable of using to receive and transmit.

To support the proper operation of a WLAN, all stations must be able to detect control frames. Hence, RTS, CTS, and ACK frames must be transmitted at a rate included in the basic rate set. In addition, frames with multicast or broadcast destination addresses must be transmitted at a rate belonging to the basic rate set. These differences in the rates used for transmitting (unicast) data and control frames has a big impact on the system behavior as clearly pointed out in [Eph02].

Actually, since 802.11 cards transmit at a constant power, lowering the transmission rate permits the packaging of more energy per symbol, and this makes the transmission range increasing. In the next subsections we investigate, by means of experimental measurements,

i) the relationship between the transmission rate of the wireless network interface card (NIC) and the maximum bandwidth utilization;

ii) the relationship between the transmission range and the transmission rate.

5.1 Available Bandwidth

In this section we will show that only a fraction of the 11 Mbps nominal bandwidth of IEEE 802.11b cards can be used for data transmission. To this end we need to carefully analyze the overheads associated with the transmission of each packet (see Figure 16). Specifically, each stream of *m* bytes generated by a legacy Internet application is encapsulated by the TCP/UDP and IP protocols that add their own headers before delivering the resulting IP datagram to the

MAC layer for the transmission over the wireless medium. Each MAC data frame is made up of: *i)* a *MAC header*, say MAC_{hdr}, containing MAC addresses and control information,[8] and *ii)* a variable length *data payload*, containing the upper layers data information. Finally, to support the physical procedures of transmission (carrier sense and reception), a *physical layer preamble* (PLCP preamble) and a *physical layer header* (PLCP header) have to be added to both data and control frames. Hereafter, we will refer to the sum of PLCP preamble and PLCP header as PHY_{hdr}.

It is worth noting that these different headers and data fields are transmitted at different data rates to ensure the interoperability between 802.11 and 802.11b cards. Specifically, the standard defines two different formats for the PLCP: Long PLCP and Short PLCP. Hereafter, we assume a Long PLCP that includes a 144-bit preamble and a 48-bit header both transmitted at 1 Mbps, while the MAC_{hdr} and the $MAC_{payload}$ can be transmitted at one of the NIC data rates: 1, 2, 5.5, and 11 Mbps. In particular, control frames (RTS, CTS and ACK) can be transmitted at 1 or 2 Mbps, while data frame can be transmitted at any of the NIC data rates.

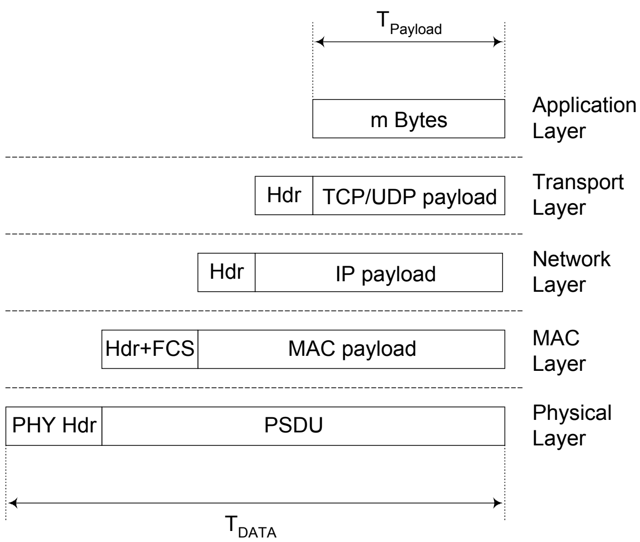


Figure 16. Encapsulation overheads.

[8] Without any loss of generality we have considered the *frame error sequence* (FCS), for error detection, as belonging to the MAC header.

By taking into considerations the above quantities, Equation (1) defines the maximum expected throughput for a single active session (i.e., only a sender-receiver couple active) when the basic access scheme (i.e., DCF without RTS/CTS) is used. Specifically, Equation (1) is the ratio between the time required to transmit the user data and the overall time the channel is busy due to this transmission:

$$Th_{noRTS/CTS} = \frac{m}{DIFS + T_{DATA} + SIFS + T_{ACK} + \frac{CW\min}{2} * Slot\_Time} \qquad (1)$$

where

T_{DATA} is the time required to transmit a MAC data frame; this includes the PHY_{hdr}, MAC_{hdr}, $MAC_{payload}$ and FCS bits for error detection.

T_{ACK} is the time required to transmit a MAC ACK frame; this includes the PHY_{hdr}, and MAC_{hdr}.

$\frac{CW\min}{2} * Slot\_Time$ is the average backoff time.

When the RTS/CTS mechanism is used, the overheads associated with the transmission of the RTS and CTS frames must be added to the denominator of (1). Hence, in this case, the maximum throughput, $Th_{RTS/CTS}$, is defined as

$$Th_{RTS/CTS} = \frac{m}{DIFS + T_{RTS} + T_{CTS} + T_{DATA} + T_{ACK} + 3 * SIFS + \frac{CW\min}{2} * Slot\_Time} \qquad (2)$$

where T_{RTS} and T_{CTS} indicate the time required to transmit the RTS and CTS frames, respectively.

The numerical results presented below depend on the specific setting of the IEEE 802.11b protocol parameters. Table 4 gives the values for the protocol parameters used hereafter.

Table 4. Value of the IEEE 802.11b parameters.

$Slot\_Time$	τ	PHY_{hdr}	MAC_{hdr}	FCS	*Bit Rate(Mbps)*
20 μsec	≤1 μsec	192 bits (2.56 t_{slot})	240 bits (2.4 t_{slot})	32 bits (0.32 t_{slot})	1, 2, 5.5, 11
DIFS	*SIFS*	*ACK*		CW_{MIN}	CW_{MAX}
50 μsec	10 μsec	112 bits + PHY_{hdr}		32 t_{slot}	1024 t_{slot}

In Table 5 we report the expected throughputs (with and without the RTS/CTS mechanism) by assuming that the NIC is transmitting at a constant data rate equal to 1, 2, 5.5. or 11 Mbps, respectively. These results are computed by applying Equations (1) and (2), and assuming a data packet size at the application level equal to m=512 and m=1024 bytes.

Table 5. Maximum throughput at different data rates.

	m= 512 Bytes		m=1024 Bytes	
	No RTS/CTS	RTS/CTS	No RTS/CTS	RTS/CTS
11 Mbps	3.337 Mbps	2.739 Mbps	5.120 Mbps	4.386 Mbps
5,5 Mbps	2.490 Mbps	2.141 Mbps	3.428 Mbps	3.082 Mbps
2 Mbps	1.319 Mbps	1.214 Mbps	1.589 Mbps	1.511 Mbps
1 Mbps	0.758 Mbps	0.738 Mbps	0.862 Mbps	0.839 Mbps

As shown in Table 5, only a small percentage of the 11 Mbps nominal bandwidth can be really used for data transmission. This percentage increases with the payload size. However, even with a large packet size (e.g., m=1024 bytes) the bandwidth utilization is lower than 44%.

The above theoretical analysis has been complemented with the measurements of the actual throughput achieved at the application level. Specifically, we have considered CBR applications that exploits UDP as the transport protocol. Applications operate in asymptotic conditions (i.e., they always have packets ready for transmission) with constant size packets of 512 bytes.

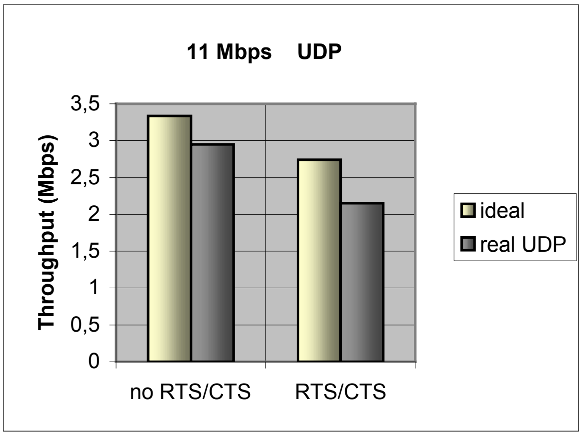


Figure 17. Comparison between the theoretical and the measured throughput.

In Figure 17 the results obtained from this experimental analysis are compared with the maximum expected throughputs calculated according to Equations (1) and (2). The real throughput is very close to the maximum throughput computed analytically. Similar results have been obtained by comparing the maximum throughput according to (1) and (2) when the data rate is 1, 2 or 5.5 Mbps, and the real throughputs measured when the NIC bit rate is set accordingly.

5.2 Transmission Ranges

The dependency between the data rate and the transmission range was investigated by measuring the packet loss rate experienced by two communicating stations whose network interfaces transmit at a constant (preset) data rate. Specifically, four sets of measurements were performed corresponding to the different data rates: 1, 2, 5.5, and 11 Mbps. In each set of experiments the packet loss rate was recorded as a function of the distance between the communicating stations. The resulting curves are presented in Figure 18.

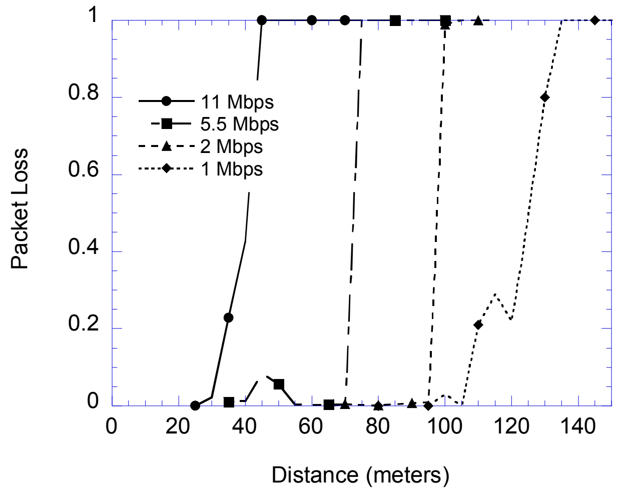


Figure 18. Packet loss rate as a function of the distance between communicating stations for different data rates.

Figure 19 shows the transmission-range curves derived in two different days (the data rate is equal to 1 Mbps). This graph highlights the variability of the transmission range depending on the weather conditions.

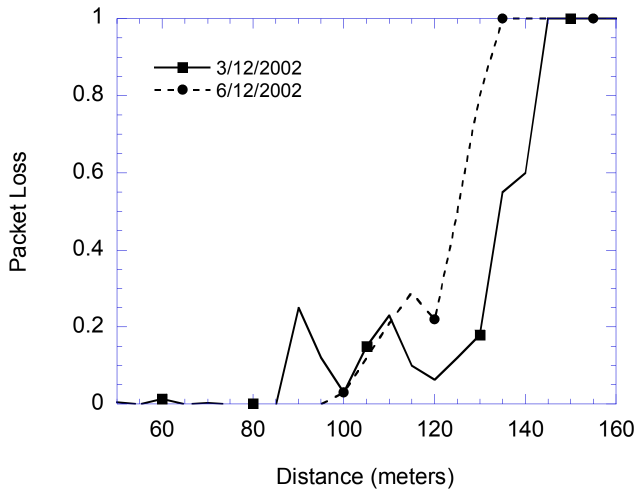


Figure 19. 1-Mbps transmission ranges in different days.

The results presented in Figure 18 are summarized in Table 6 where the estimates of the transmission ranges at different data rates are reported. These estimates point out that, when using the highest bit rate for data transmission, there is a significant difference in the transmission range of control and data frames, respectively. For example, assuming that the RTS/CTS mechanism is active, if a station transmits a frame at 11Mbps to another station within its transmission range (i.e., less then 30m apart) it reserves the channel for a radius of approximately 90 (120) m around itself. The RTS frame is transmitted at 2Mbps (or 1Mbps), and, hence, it is correctly received by all stations within the transmitting station's range, i.e., 90 (120) meters.

Table 6. Estimates of the transmission ranges at different data rates.

	11 Mbps	5.5 Mbps	2 Mbps	1 Mbps
Data *TX_range*	30 meters	70 meters	90-100 meters	110-130 meters
Control *TX_range*			≈ 90 meters	≈ 120 meters

Again, it is interesting to compare the transmission range used in the most popular simulation tools, like ns-2 and Glomosim, with the transmission ranges measured in our experiments. In these simulation tools it is assumed $TX\_range = 250m$. Since the above simulation tools only consider a 2-Mbps bit rate we make reference to the transmission range estimated with a NIC data rate of 2 Mbps. As it clearly appears, the value used in the simulation tools (and, hence, in the simulation studies based on them) is 2-3 times higher that the values measured in practice. This difference is very important for example when studying the behavior of routing protocols: the shorter is the *TX_range*, the higher is the frequency of route re-calculation when the network stations are mobile.

5.2.1 Transmission Ranges and the Mobile Devices' Height

During the experiments we performed to analyze the transmission ranges at various data rates, we observed a dependence of the transmission ranges on the mobile devices' height from the ground. Specifically, in some case we observed that while the devices were not able to communicate when located on the stools, they started to exchange packets by lifting them up. In this section we present the results obtained by a careful investigation of this phenomenon. Specifically, we studied the dependency of the transmission ranges on the devices height from the ground. To this end we measured the throughput between two stations[9] as a function of their height from the ground: four different heights were considered: 0.40 m, 0.80 m, 1.2 m and 1.6 m. The experiments were performed with the Wi-Fi card set at two different transmission rates: 2 and 11 Mbps. In each set of experiments the distance between the communicating devices was set in such away to guarantee that the receiver is always inside the sender's transmission range. Specifically, the sender-receiver distance was equal to 30 and 70 meters when the cards operated at 11 and 2 Mbps, respectively.

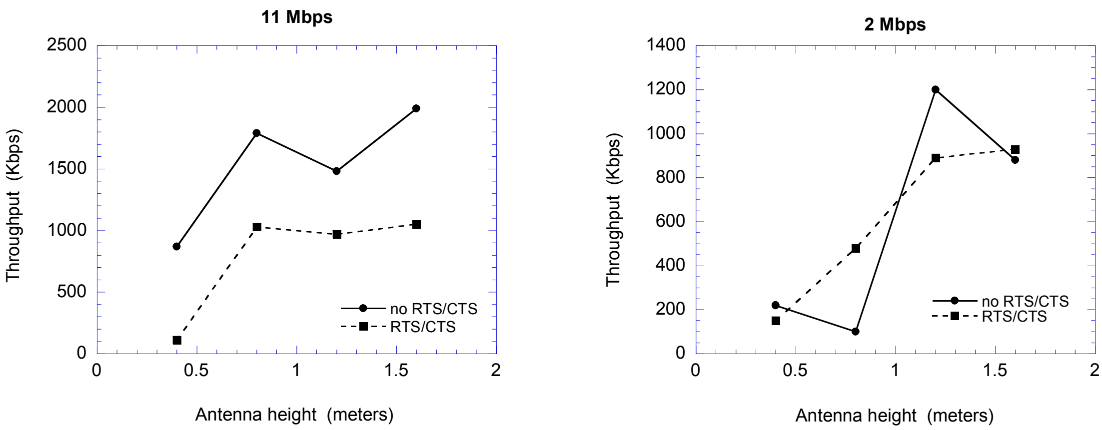


Figure 20: Relationship between throughput and devices' height

As it clearly appears in Figure 20, the height may have a big impact on the quality of the communication between the mobile devices. For example, at 11 Mbps, by lifting up the devices from 0.40 meters to 0.80 meters the throughput doubles, while further increasing the height does not produce significant throughput gains. A similar behavior is observed with a 2 Mbps transmission rate. However, in this case the major throughput gain is obtained lifting up

[9] In these experiments UDP is used as the transport protocol.

the devices from 0.80 meters to 1.20 meters. A possible explanation for this different behavior is related to the distance between the communicating devices that is different in the two cases. This intuition is confirmed by the work presented in [Oba02] that provides a theoretical framework to explain the height impact on IEEE 802.11 channel quality. Specifically, the channel power loss depends on the contact between the Fresnel zone and the ground. The Fresnel zone for a radio beam is an elliptical area with foci located in the sender and the receiver. Objects in the Fresnel zone cause diffraction and, hence, reduce the signal energy. In particular, most of the radio-wave energy is within the First Fresnel Zone, i.e., the inner 60% of the Fresnel zone. Hence, if this inner part contacts the ground (or other objects) the energy loss is significant. Figure 21 shows the Fresnel zone (and its inner 60%) for a sender-receiver couple at a distance D. In the figure, R1 denotes the height of the First Fresnel Zone. As shown in [Oba02] R1 is highly dependent on the stations distance. For example, when the sender and the receiver are at an height of 1 meter from the ground, the First Fresnel Zone has a contact with the ground only if $D > 33$ meters. While at heights of 1.5 and 2 meters the First Fresnel Zone contacts the ground only if D is greater than 73 and 131 meters, respectively. These theoretical computations are aligned with our experimental results.

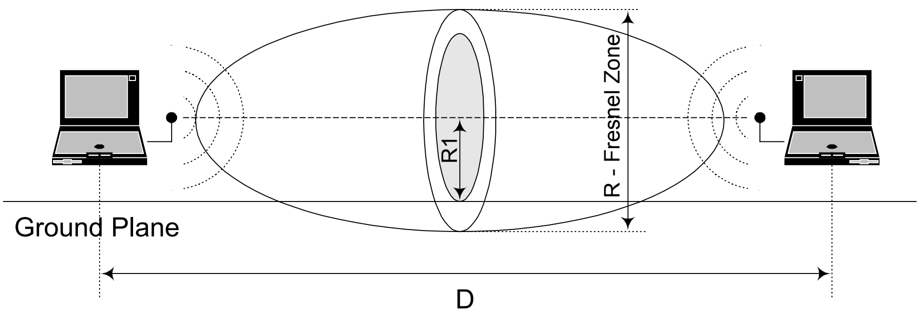


Figure 21: The Fresnel Zone

5.3 Four-Stations Network Configurations

The results presented in the previous sections show that the IEEE 802.11b behavior is more complex than the behavior of the IEEE 802.11 standard. Indeed the availability of different transmission rates may cause the presence of several transmission ranges inside the network. In particular, inside the same data transfer session there may be different transmission ranges for data and control frame (e.g., RTS, CTS, ACK). Hereafter, we show that the superposition of these different phenomena makes very difficult to understand the behavior of IEEE 802.11b ad hoc networks. To reduce this complexity, in the experiments presented below the NIC data rate is set to a constant value for the entire duration of the experiment.[10] Hereafter, we present only the results obtained with the NIC data rate constant, and equal to 11Mbps; more results can be found in [Ana03].

The four-stations configuration presented in Figure 22 was used in the experiments. The results obtained are presented in Figure 23.

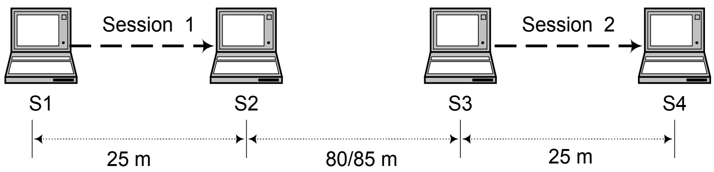


Figure 22. Network configuration at 11 Mbps.

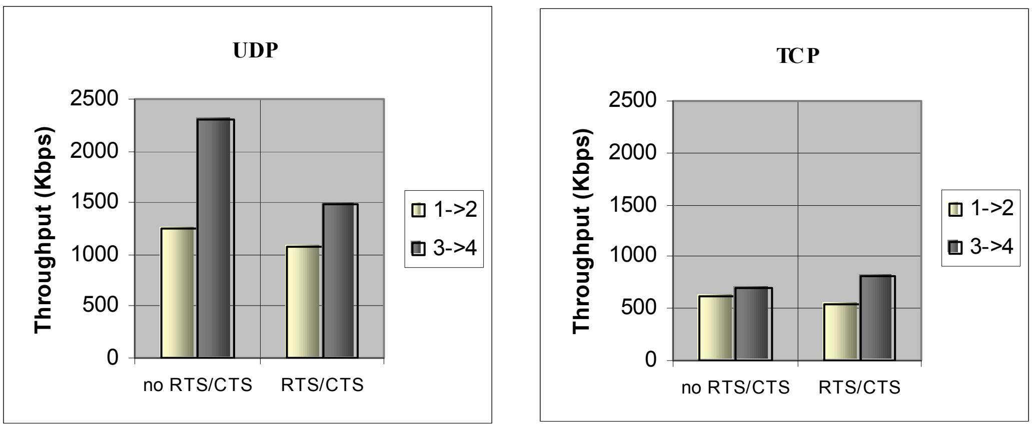


Figure 23. Throughputs at 11 Mbps.

[10] It is worth pointing out that we experienced a high variability in the channel conditions thus making a comparison between the results difficult.

These results were the superposition of several factors. In detail, dependencies were observed between the two connections even though the transmission range was smaller than the distance between stations S1 and S3. Furthermore, the dependency were observed also when the basic mechanism (i.e., no RTS/CTS) is used.[11] To summarize, this set of experiments showed that interdependencies among the stations extends beyond the transmission range. To explain this we hypothesized that all stations were inside the same physical carrier sensing range, and this produced a correlation between active connections whose effect is similar to that achieved with the RTS/CTS mechanism (virtual carrier sensing). The difference in the throughputs achieved by the two sessions when using the UDP protocol (with or without RTS/CTS) can be explained by considering the asymmetric condition that exists on the channel: station S2 was exposed to transmissions of station S3 and, hence, when station S1 sent a frame to S2 this station was not able to send back the MAC ACK. Therefore, S1 reacted as in the collision cases (thus re-scheduling the transmission with a larger backoff). It is worth pointing out that also S3 was exposed to S2 transmissions but the S2's effect on S3 was less marked given the different role of the two stations. When using the basic access mechanism, the S2's effect on S3 was limited to short intervals (i.e., the transmission of ACK frames). When adopting the RTS/CTS mechanism, the S2 CTS forced S3 to defer the transmission of RTS frames (i.e., simply a delay in the transmission), while RTS frames sent by S3 forced S2 to not reply with a CTS frame to S1's RTS. In the latter case, S1 increased the back off and rescheduled the transmission. Finally, when the TCP protocol was used the differences between the throughput achieved by the two connections still existed but were reduced. The analysis of this case is very complex because we must also take into consideration the impact of the TCP mechanisms that: i) reduces the transmission rate of the

[11] A similar behavior is observed (but with different values) by adopting the RTS/CTS mechanism.

first connection, and ii) introduces the transmission of TCP-ACK frames (from S2 and S4) thus contributing to make the system less asymmetric.

5.4 Physical Carrier Sensing Range

Results presented in the previous section seem to indicate that dependencies among the stations extend far beyond the transmission range. For example, taking as a reference the scenario presented in Figure 22, the distance between the two couples of transmitting stations is about three times the transmission range. The hypothesis is that dependencies are due to a large physical carrier sensing that includes all the stations. To validate this hypothesis and to better understand the system behavior we designed some experiments to estimate the physical carrier sensing range. A direct measure of this quantity seems difficult to achieve because the 802.11b cards we utilized do not provide to the higher layers information about the channel carrier sensing. Therefore, we defined an indirect way to perform these measurements. We utilized the scenario shown in Figure 24 with fixed distance between each couple of communicating stations (d(1,2)=d(3,4)=10 meters), and variable distance between the two couples, i.e., d(2,3), is variable.

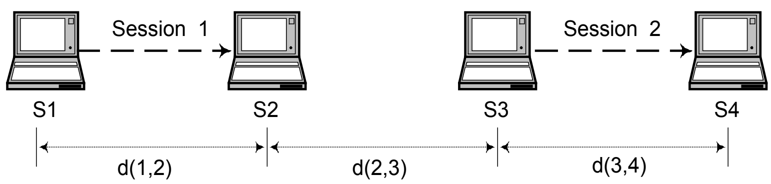


Figure 24. Reference network scenario.

The idea is to investigate the correlation among the two sessions while increasing the distance d(2,3). To measure the correlation degree, just before running each experiment we performed some preliminary measurements. Specifically, we measured the throughput of each session in isolation, i.e., when the other session is not active. Then, we measured the throughput of each session when both sessions are active. Hereafter, $Th_i(x)$ denotes the throughput of session i

(i=1,2) when both sessions are active and d(2,3)=x. Obviously, $Th_i(\infty)$ denotes the throughput of session i (i=1,2), when d(2,3)= ∞, and hence the two sessions are independent. By exploiting these measurements we estimated the correlation existing between the two sessions by the following index:

$$D_1(x) = 1 - \frac{Th_1(x) + Th_2(x)}{Th_1(\infty) + Th_2(\infty)} \quad .$$

The $D_1(x)$ index takes the value 0 if the two sessions are independent. Taken a session as a reference, the presence of the other session may have two possible effects on the performance of the reference session: 1) if the two sessions are within the same physical carrier sensing range, they share the same physical channel; 2) if they are outside the physical carrier sensing range the radiated energy from one session may still affect the quality of the channel observed by the other session. As the radiated energy may travel over unlimited distances, we can expect that $D_1(x)$ may be equal to zero only for very large distances among the sessions [Eph02a].

When the $D_1(x)$ value is greater than zero, the index does not indicate how strong the correlation is. To measure this second aspect we introduce the $D_2(x)$ index:

$$D_2(x) = \frac{Th_1(0) + Th_2(0)}{Th_1(x) + Th_2(x)} \quad .$$

$D_2(x)$ compares the throughput of the two sessions when they are active at the same time and d(2,3)=x, with respect to the two-session throughput when all the stations are inside the same transmission range, i.e., d(2,3)=0. A $D_2(x)$ value equal to 1 indicates the maximum correlation that exists when all stations are in the same transmission range.

By varying the distance d(2,3) we performed several experiments to estimate the above indexes. The results were obtained with the cards transmission rates set to 2 and 11 Mbps, and are summarized in Table 7 and Table 8, respectively. As it clearly appears from the tables, the correlation among session is still marked when d(2,3) is less than or equal to 250 meters, noticeably decreases around 300 meters, and further reduces (but not disappears) when the inter-session distance is about 350 meters.

Table 7: Throughput values (Card rate =11 Mbps, payload size=512 Bytes)

Access Mechanism	Distance	Throughput of Session 1		Throughput of Session 2		$D_1(x)$	$D_2(x)$
		$Th_1(\infty)$ Kbps	$Th_1(x)$ Kbps	$Th_2(\infty)$ Kbps	$Th_2(x)$ Kbps		
No RTS/CTS	x=0	2780	1849	2981	1768	0.37	1.00
	x=150	1950	1500	2950	2250	0.23	0.96
	x=180	2920	2210	3040	1580	0.36	0.95
	x=200	2290	1930	3160	2660	0.16	0.78
	x=250	2820	1700	3170	2760	0.25	0.81
	x=300	2980	2800	3060	2750	0.08	0.65
	x=350	2730	2590	3250	3230	0.03	0.62

From the above results we assume that 250 m is approximately the size of the physical carrier sensing range. After this distance the correlation among the two sessions is due to the mutual impact on the channel quality. A set of measurements is currently ongoing to further verify the exact size of the physical carrier sensing range.

It is worth noting that the physical carrier sensing range is almost the same for the two different transmission rates. Indeed, the physical carrier sensing mainly depends on two parameters: the stations' transmitting power and the distance between transmitting stations. The rate at which data are transmitted have no significant effect on these parameters.

The results obtained confirm the hypotheses we made in the previous section to justify the apparent dependencies existing among the two couples of transmitting stations even if the distance among them is about three times greater than the transmission range.

It is worth noting that the ideal value for $D_1(0)$ is 0.5, i.e., each session gets half of the throughput of the session in isolation. This is not true for CSMA MAC protocol as $Th_1(0)$

($Th_2(0)$) is greater than $Th_1(\infty)/2$ ($Th_2(\infty)/2$). This results is caused by a smaller overhead of the backoff algorithm in the experiments with d(2,3)=0.

Table 8: Throughput values (Card rate =2 Mbps, payload size=512 Bytes)

Access Mechanism	Distance	Throughput Session 1		Throughput Session 2		$D_1(x)$	$D_2(x)$
		$Th_1(\infty)$	$Th_1(x)$	$Th_2(\infty)$	$Th_2(x)$		
No RTS/CTS	x=0	1279	577	1253	561	0.55	1.00
	x=150	1310	880	1310	780	0.37	0.69
	x=180	1310	930	1310	820	0.33	0.65
	x=200	1270	1030	1330	1130	0.17	0.53
	x=250	1300	960	1330	960	0.27	0.59
	x=300	1370	1360	1380	1050	0.12	0.47
	x=350	1360	1110	1400	1390	0.09	0.45

5.5 Channel Model for an IEEE 802.11 Ad Hoc Network

The results presented in this paper indicate that for correctly understanding the behavior of an 802.11 network operating in ad hoc mode, several different ranges must be considered.

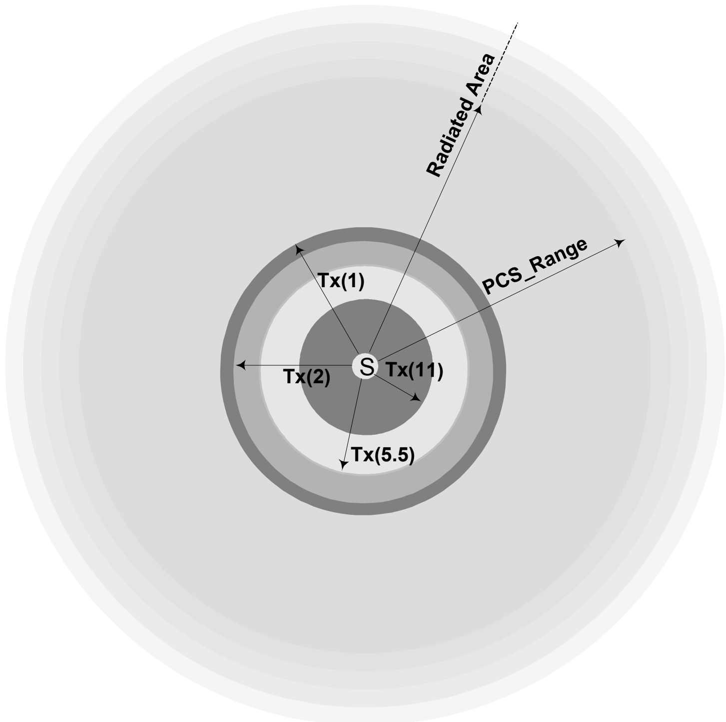


Figure 25: Channel model for an 802.11 ad hoc network.

Specifically, as shown in Figure 25, given a transmitting station S, the stations around will be affected by the station S transmissions in a different way depending on the distance from S and the rate used by S for its transmissions.

Specifically, assuming that S is transmitting with a rate x $\left(x \in \{1,\ 2,\ 5.5,\ 11\}\right)$ stations around it can be partitioned into three classes depending on their distance, *d*, from *S*:

i. Stations at a distance *d* < TX_Range(*x*) are able to correctly receive data from *S,* if *S* is transmitting at a rate lower or equal to *x*;

ii. Stations at a distance *d*, where TX_Range(*x*) < *d* < PCS_Range, are not able to receive data correctly from station *S.* However, as they are in the *S* physical carrier sensing range, when *S* is transmitting they observe the channel busy and thus they defer their transmissions.

iii. Stations at a distance *d* > PCS_Range do not measure any significant energy on the channel when S is transmitting, therefore they can start transmitting contemporarily to S; however, the quality of the channel they observe may be affected by the energy radiated by S. In addition, if *d* < PCS_Range + TX_Range(*x*) some interference phenomena may occur (see below). This interference depends on the IF_Range value. This value is difficult to model and evaluate as it depends on several factors (mainly the power at the receiving site) but as explained before TX_Range(1) < IF_Range < PCS_Range.

Several interesting observations can be derived by taking into consideration points i-iii above. Firstly, the hidden station phenomenon, as it is usually defined in the literature (see Section 2.2), is almost impossible with the ranges measured in our experiments. Indeed, the PCS_Range is more than twice TX_Range(1), i.e., the larger transmission range. Furthermore, two stations, say S1 and S2, that can start transmitting towards the same receiver, R, must be at a distance ≤ 2•TX_Range(1), and thus they are inside the physical carrier sensing range of each other. Hence, if S1 has an ongoing transmission with R, S2 will observe a busy channel and thus will defer its own transmission. This means that, in this scenario, virtual carrier sensing is not necessary and the RTS/CTS mechanism only introduces additional overhead.

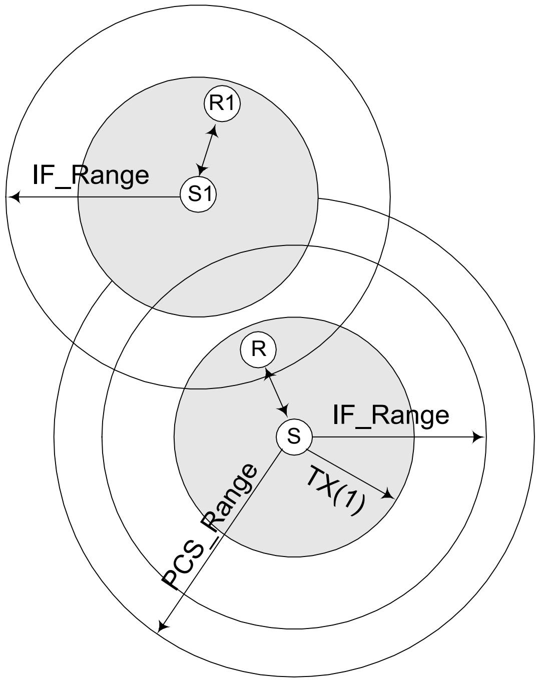


Figure 26 : Interference-based hidden station phenomenon.

While the hidden station phenomenon, as defined in the literature, seems not relevant for this environment point iii above highlights that packets cannot be correctly received due to the interference caused by a station that is “hidden” to the sending station. An example of this type of *hidden station phenomenon* is presented in Figure 26. In this figure we have two transmitting stations, S and S1 that are outside their respectively PCS_Range and hence they are hidden to each other. In addition we assume that the receiver of station S (denoted by R in the figure) is inside the interference range (IF_Range) of station S1. In this scenario S and S1 can be simultaneously transmitting and, if this occurs, station R cannot receive data from S correctly. Also in this case the RTS/CTS mechanism does not provide any help and new coordination mechanisms need to be designed to extend the coordination in the channel access beyond the PCS_Range.

It is worth noting that, in our channel model, the exposed station definition (see Figure 6) must be modified too. In this scenario, exposed stations are those station at a distance PCS_Range- TX_Range(1) < *d* < PCS_Range. Indeed, these stations are exposed to station S transmissions, while they are in the transmission range of stations with *d* > PCS_Range . The following example outline problems that may occur in this case. Let us denote with S1 a station at a distance d from S: PCS_Range < *d* < PCS_Range+TX_Range(*x*). Station S1 can start transmitting, with a rate *x*, towards a station E that is inside the physical carrier sensing of S; station E cannot reply because it observes a busy channel due to the ongoing station S transmissions, i.e., E is exposed to station S. Since station S1 does not receive any reply (802.11 ACK) from E, it assumes an error condition (collision or CRC error condition), hence it backoffs and then tries again. If this situation repeats for several times (up to 7), S1 assumes that E is not anymore in its transmission range, gives up the transmission attempt and (wrongly) signals to the higher layer a link breakage condition, thus forcing higher layers to attempt a recovery action (e.g., new route discovery, etc. – see Section 3).

To summarize, results obtained in the configuration we analyzed indicate that the hidden station and exposed station definitions must be extended. These new hidden-station and exposed-station phenomena may produce undesirable effects that may degrade the performance of an ad hoc network, mainly if the TCP protocol is used. Extending the coordination in the channel access beyond the PCS_Range seems to be the correct direction for solving the above problems.

5.6 The *Communication Gray Zones* Problem

An important problem related to the different transmission ranges of control and data frames is the so-called *communication gray zones* problem [Lun02]. This problem was revealed by a group of researchers at the Uppsala University. While measuring the performance of a their own implementation of the AODV routing protocol [Per99] in an IEEE 802.11b ad hoc network, they observed an unexpected large amount of packets' losses, especially during route changes. They found that the increase in packet loss occurred in some specific geographic areas that they called "*communication gray zones"*. In such zones the packet loss experienced by a station may be extremely high, up to 100%, thus severely affecting the performance of those applications characterized by a continuous packet flow (e.g., file transfers and multimedia streaming). They also found that the ultimate reason for this phenomenon is that a station inside a gray zone is considered as reachable by a neighboring station, based on its routing information, but data communication between the stations is not possible. The same problem was found to affect other routing protocols like OLSR [Cla02] and LUNAR [Tsc02].

To better understand why communication gray zones arise it is worthwhile to briefly recall how the AODV routing protocol works. AODV is a reactive protocol that discovers and maintains routes on demand. When a route to a target station is needed, the AODV protocol broadcasts a route-request message that is then disseminated throughout the network. When the target station (or a valid route to the target station) is found, a route-reply message is sent back to the requesting station by means of a unicast message. While this message travels towards the requesting station, routes are set up inside routing tables of the traversed stations.

In addition to the request-reply mechanism, the AODV protocol uses a sensing mechanism to discover neighboring stations and, based on this, to update, add or remove routes in the routing table. Periodically, each station broadcasts HELLO beacons. Upon reception of an

HELLO message from a neighboring, a station becomes aware that the neighboring station is reachable and can, thus, be used to relay data transmissions. Routing table are, thus, updated accordingly.

Several elements contribute to the occurrence of communication gray zone. In particular, the different properties of HELLO messages with respect to data messages play an important role. These properties, and their effects, are summarized below.

1. **Transmission rate**. Since HELLO beacons are broadcast messages they are transmitted at the basic rate (2 Mbps). On the other hand, data packets (that are unicast) may be transmitted at 11 or 5.5 Mbps. Therefore, HELLO messages have a transmission range larger than data messages.
2. **No Acknowledgement**. In 802.11b broadcast messages are transmitted without acknowledgement. Therefore, a station that receives an HELLO message from a neighboring has no indication whether transmission is possible even in the opposite direction, i.e., there is no indication that the link is bi-directional.
3. **Packet size**. In general, HELLO messages are much smaller in size than data packets. As is well known, small packets have a lower probability to be affected by transmission errors, and minor chances of colliding with other packets. Therefore, it is more likely for an HELLO message to reach a receiver than a data packet, especially when the link quality is poor.

In addition to the above elements, the effects of fluctuating links need to be taken into account, as well. At the border of the transmission range the communication quality tends to be fluctuating. In such conditions it may happen that a station sporadically receives an HELLO message from a neighbor, but this does not imply that consistent communication between the stations is actually possible. Since the AODV protocol updates routing tables based on the neighboring sensing mechanism (i.e., based on the reception of HELLO message) it may occur that stable and longer routes are replaced by shorter but unreliable ones.

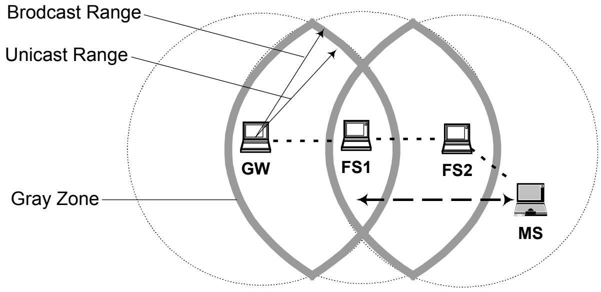


Figure 27. A scenario where communication gray zones can be experienced (taken from [Lun02]).

Figure 27 depicts a scenario pointed out by the researchers at Uppsala University where communication gray zone can be experienced by the mobile station MS (see [Lun02]). In this scenario, stations labeled as GW, FS1 and FS2 are static, while station MS moves forward and back as indicated in the figure. There is an active communication between the gateway Station GW and the mobile station MS. Depending on the physical position of the mobile station, the traffic from MS to GW (and vice-versa) is routed trough one, two or three hops via intermediate stations FS1 and FS2. Theoretically, the MS has always a route towards GW. However, while moving from the initial position to the rightmost position MS will pass through two gray zone. Similarly, two gray zones will be traversed in the reverse path. In [Lun02] it is shown that traversal of the gray zones are associated with time intervals during which MS experience a packet loss of up to 100%. The duration of this time interval, as well as the peak value in the packet loss experienced by the mobile station, depends on the specific routing protocol.

Before proceeding on it is important to highlight that communication gray zone problem can not be revealed by using the current simulation tools (e.g., ns-2). Indeed, in the IEEE 802.11 model implemented by simulation tools both unicast and broadcast transmissions are performed at 2 Mbps and, hence, they have the same transmission range. Furthermore, connectivity is modeled as on/off, i.e., the communication becomes impossible as soon as the distance exceeds the transmission range.

In [Lun02] the authors also propose some possible solutions for alleviating the communication gray zones problem, namely: (i) the exchange of the neighboring set (i.e., stations include their neighboring set in HELLO messages); (ii) the transmission of N-consecutive HELLO messages; and (iii) the introduction of a SNR threshold to discard weak control messages. They have also assessed, by means of an experimental analysis, that the SNR-threshold approach is the most effective and it eliminates the effects of communication gray zones almost completely.

6. Evolution of IEEE 802.11b for Ad Hoc Networks

In ad hoc networks each station logically operates similarly to a router. However, from the physical standpoint, there is a significant difference between a router and a station in an ad hoc network. Typically, a router has multiple network interfaces, and a packet received from one interface is retransmitted through a different interface (see the left side of Figure 28). On the other hand, in a multi-hop ad hoc network a station has a single wireless interface and packets are received from and transmitted through the same interface (see the right side of Figure 28).

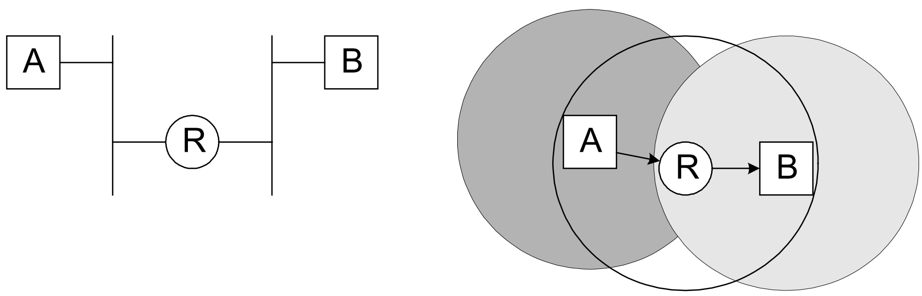


Figure 28. Packet forwarding in a router in a wired network (left) and in ad hoc networks (right).

Current ad hoc network architectures do not take into account this difference, and implement in ad hoc stations the same functionalities of a router. Specifically, packets received from the wireless medium are delivered to the IP layer where a route lookup is performed based on the destination IP address (steps 1-3 in Figure 29-left). If the packet is not destined to the station itself it is passed down to the network interface to be retransmitted (steps 4 and 5 in Figure 29-left).

The difference, from the forwarding standpoint, between an ad hoc station and a router has been recently pointed out by A, Acharya et. alt. who have also proposed an architecture for efficient packet forwarding at stations in multi-hop ad hoc networks [Ach02].

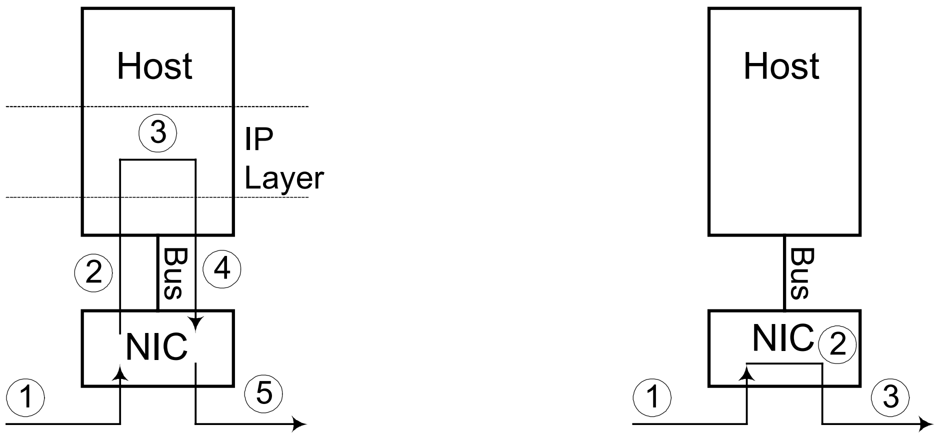


Figure 29. Forwarding in ad hoc stations: legacy approach (left) and NIC based forwarding (right).

The first question addressed in [Ach02] is which is the best architecture for forwarding a packet in ad hoc networks. The answer is highlighted in Figure 29. The left-hand side in the figure depicts the legacy approach for forwarding packets, while the right-hand side shows the new approach proposed by the authors. In the latter case, the forwarding is completely managed by the network interface card (NIC). Upon receiving a packet, the NIC (by exploiting some local information) determines whether or not the packet has to be retransmitted. Only packets destined to the station itself are passed to higher-level protocols. Due to its behavior the proposed architecture has been named Cut-through architecture [Ach02].

The Cut-through architecture provides several advantages that can be classified into two categories. The first category includes advantages that are not related to a specific MAC protocol, while advantages belonging to the second category are strongly related to the random access scheme and the RTS-CTS mechanism used in the IEEE 802.11 MAC protocol. The following advantages belong to the first category:

a) The NIC does not need to interrupt the CPU for packet processing. This could lead to a considerable power saving for example, if the station is used only for packet forwarding purposes. The CPU needs to wake up only for processing route updates.

b) Delays for transferring data from the NIC to the host and vice-versa are avoided.

c) Local traffic does not further delays forwarding traffic. In legacy IEEE802.11 architecture, at the forwarding station, the packet is transferred to the main memory by the NIC. The host CPU is notified (e.g. via interrupts) for further processing of the packet by the IP protocol stack running on the host CPU. The host software (IP protocol stack) would typically queue up the packet in a transmission queue (together with the locally generated packets) and select packets for transmission based on a scheduling algorithm (typically, FIFO). Thus packets generated by applications running at the station can overtake packets to be forwarded. This produces an increase in the end-to-end delay.

As the other advantages are strictly related to the IEEE 802.11 MAC protocol they can be better understood by first considering the operations performed at the MAC layer by stations A, B and R in the scenario depicted in the right side of Figure 28.

6.1 Forwarding Operation: Legacy approach vs. NIC-based approach

Let us consider the case shown in Figure 30, where A is the upstream station, R is the forwarding station, and B the downstream station. In the legacy approach, the data delivery from A to B involves two separate and independent transmissions. For each transmission the IEEE 802.11 MAC protocol (including the RTS/CTS mechanism) is used.

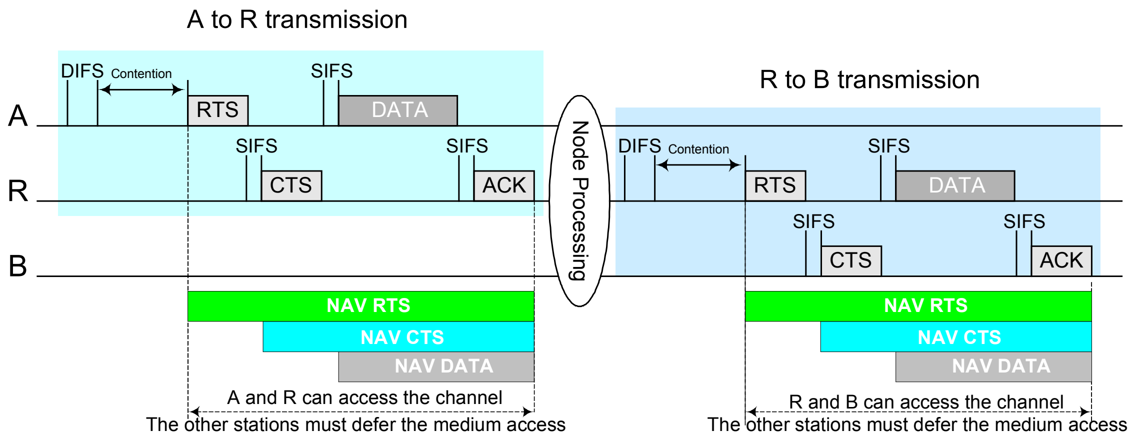


Figure 30. Forwarding operations in the legacy approach.

Specifically, with reference to Figure 30, the transmission from A to R is first performed. At R the packet is passed to the IP protocol, processed by the IP software, and passed down to the NIC for transmission to B. A this time, R has to repeat the same procedure executed by A during the transmission to R. Note that the two transmissions (from A to R, and from R to B) are independent each other from the channel access standpoint.

It is worth noting that, after the first RTS/CTS exchange, stations A and R got the channel control and no other station in the transmission range of A and/or R can access the channel. However, this control is immediately lost after the ACK transmission from R to A. Clearly, from the Station R standpoint, it is not very wise to release the channel control, and immediately after compete again for gaining the channel control. It would be better for R to maintain the exclusive control on the channel. In this case the transmission to B would be done without contention, thus improving the bandwidth utilization (there would be no bandwidth wastage due to collisions and backoff periods), and minimizing the forwarding delay. Obviously, Station R should operate the packet forwarding very quickly so that the transmission from R to B could start immediately after the ACK transmission from R to A. This can be achieved only if the forwarding operation is performed completely inside the NIC (right-hand side of Figure 29).

Figure 31 shows an extension of the IEEE 802.11 MAC protocol to manage packet forwarding in a more efficient way [Ach02]. The basic idea is to give the highest priority to the traffic to be forwarded by extending the channel reservation scheme to allow on-the-fly transmissions. Specifically, upon receiving a frame from A, Station R not only sends back an ACK frame to A but, at the same time, it transmits an RTS frame to further extend the channel reservation.[12] Since the RTS frame transmission occurs while all the other stations within the R's transmission range are still blocked (due to the previous RTS/CTS exchange) station R can immediately get the channel. The extended MAC protocol has been named Data-driven Cut-through Multiple Access (DCMA).

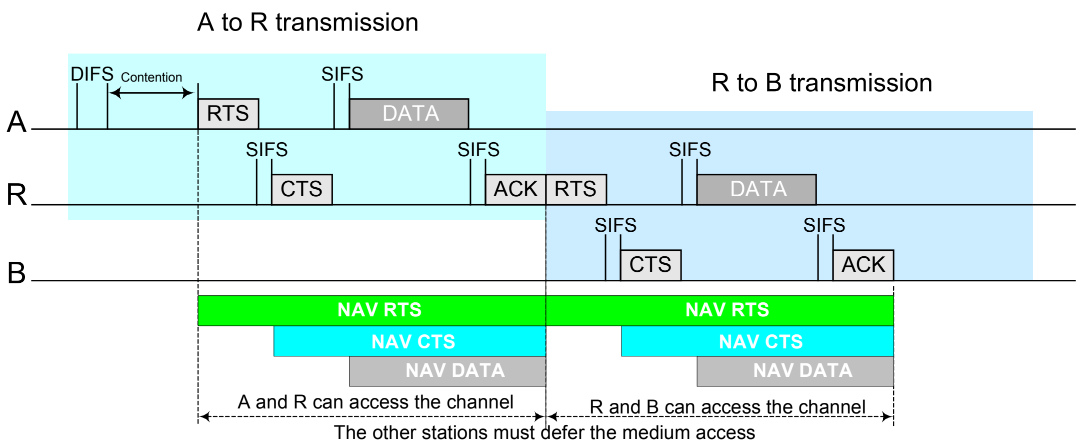


Figure 31. Forwarding operations in the NIC-based approach.

6.2 DCMA MAC protocol

The DCMA MAC protocol is an extension of the IEEE 802.11 DCF and, as such, it follows the associated 4-way handshake involving the exchange of RTS/CTS/DATA/ACK frames. As shown in the previous section, the DCMA attempts to replace the two distinct channel accesses (upstream and downstream) with a combined access. Specifically, DCMA combines the ACK (to the upstream station) with the RTS (to the downstream station) in a single ACK/RTS packet that is sent to the MAC broadcast destination address.

[12] More precisely, the RTS frame is piggybacked to the ACK frame.

The cut-through approach, proposed in DCMA, fails when the downstream station (e.g., B in our example) cannot reply to the ACK/RTS (with a positive CTS). In such a case the forwarding station then simply queues the packet in the NIC queue, and resumes the normal IEEE 802.11 channel access method using the exponential backoff to regulate subsequent access to the shared channel. The channel contention resolution of DCMA is same as that of 802.11, with a station remaining silent as long as any of its one-hop neighbors are either receiving or transmitting a data packet. Accordingly, this protocol does not suffer from any additional penalties, over and above those present in 802.11.

Since DCMA has no notion of future reservations (all access attempts are for immediate transfer of DATA frames), it does not require any modifications or enhancements to the 802.11 NAV—a station simply stays quiet as long as it is aware of (contiguous) activity involving one or more of its neighbours. Any station that overhears an ACK/RTS not addressed to it merely increments the NAV by the time interval included in the ACK/RTS message.

In [Ach02] a simulation analysis of the DCMA scheme is also presented. This analysis was carried out by implementing the DCMA access protocol in the ns-2 simulation tool. Consequently, the ns-2 typical values were used: bit rate of 2 Mbps, transmission range equal to 250 m, and interfering range equal to 550 m. All transmissions, regardless of the frame size, were preceded by an RTS/CTS exchange.

In the simulation study, high-rate sources were used to guarantee a never empty queue at the transmitter. To avoid the interference of TCP mechanisms, UDP protocol was used at the transport layer. The statistics were estimated by considering only the packets correctly received at the receiver. In the forwarding stations routing tables[13] were pre-configured with the shortest path routes to their respective destinations.

Table 9. Throughput comparison (in Kbps).

	Packet size (Bytes)			
	256	512	1024	1536
802.11	159	200	231	258
DCMA	197	242	282	301

Table 10. End-to-end delay comparison (in sec).

	Packet size (Bytes)			
	256	512	1024	1536
802.11	1.00	1.67	2.59	2.83
DCMA	0.50	0.81	1.31	1.73

Several configurations are considered in [Ach02]. For the sake of space only a set of them are discussed here. They refer to a string or chain topology (see Figure 8) where the distance between successive stations is 250 m. A single flow of UDP packets is transmitted from the leftmost to the rightmost station. Several experiments were conducted by varying the size of the payload from 256 to 1536 bytes.

The results obtained by considering a 7-hop chain are summarized in Table 9 and Table 10. It clearly appears that DCMA improves the throughput around 20% with respect to the standard

[13] The contents of these routing tables will be briefly discussed in the following. A complete description can be found in [Ach02].

protocol, while the delay improvement is more significant, ranging from 63% (1536bytes) to 100% (256byte packets).

In [Ach02] the comparison was further extended by considering an increasing number of hops in the chain. The results obtained are consistent with results presented in Table 9 and Table 10: the delay reductions with DCMA are significant (in the order of 50%), while throughput improvements are marginal.

Table 11.Throughput (in Kbps) as a function of the offered load.

	Offered Load (Kbps)										
	250	275	300	325	350	375	400	425	450	475	500
802.11	242	267	292	316	340	364	259	273	256	241	235
DCMA	242	267	292	316	340	364	389	413	376	376	374

Table 12. End-to-end delay (in sec) as a function of the offered load.

	Offered Load (Kbps)										
	250	275	300	325	350	375	400	425	450	475	500
802.11	0.075	0.075	0.075	0.075	0.075	0.075	0.075	0.075	0.75	0.95	1.09
DCMA	0.080	0.080	0.080	0.080	0.080	0.080	1.37	1.53	2.06	2.02	2.08

Finally the influence of the offered load was considered. The results obtained are summarized in Table 11 and Table 12. Specifically, these results are related to a 12-hop chain and have been obtained by increasing the sending rate at the source from 250 Kbps to about 500 Kbps. It clearly appears that there is a different saturation points for the two protocols. The IEEE 802.11 MAC protocol has the maximum throughput at around 0.375 Mbps; after this offered load level the queues start to build up, the end-to-end delay has a significant increase and the throughput decreases. On the other hand, DCMA has the maximum throughput at around 0.425 Mbps. Furthermore, after the saturation point DCMA shows a more stable behavior: the throughputs remains high, and the end-to-end delay is about half that of the standard protocol.

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References

[Ach02] A. Acharya, A. Misra, S. Bensal, "A Label-switching Packet Forwarding Architecture for Multi-hop Wireless LANs", Proceedings of the *ACM Workshop on Mobile Multimedia (WoWMoM 2002)*, Atlanta (GA), September 28, 2002.

[Ahu00] A. Ahuja et al., "Performance of TCP over different routing protocols in mobile ad-hoc networks," Proceedings of the *IEEE Vehicular Technology Conference (VTC 2000),* Tokyo, Japan, May 2000.

[Ana03] G. Anastasi , E. Borgia, M. Conti, E. Gregori, "IEEE 802.11 Ad Hoc Networks: Performance Measurements", Proceedings of the *Workshop on Mobile and Wireless Networks (MWN 2003)*, Providence (Rhode Island), May 19, 2003.

[Cha01] K. Chandran, S. Raghunathan, S. Venkatesan, R. Prakash, "A Feedback Based Scheme for Improving TCP Performance in Ad Hoc Wireless Networks", *IEEE Personal Communication Magazine*, Special Issue on Ad Hoc Networks, Vol. 8, N. 1, pp. 34-39, February 2001.

[Cla02] T. Clausen and al., "Optimized Link State Routing Protocol", Internet Draft, IETF MANET Working Group, November 2002, available at http://menetou.inria.fr/olsr/draft-ietf-manet-olsr-07.txt.

[Dye01] T.D. Dyer, R.V. Boppana "A Comparison of TCP Performance over Three Routing Protocols for Mobile Ad Hoc Networks", Proceedings of the *ACM Symposium on Mobile Ad Hoc Networking & Computing (MobiHoc*), October 2001.

[Eph02] T. Ephremides, "A Wireless Link Perspective in Mobile Networking", ACM Mobicom 2002 keynote speech, available at http://www.acm.org/sigmobile/mobicom/2002/program/

[Eph02a] T. Ephremides, "Energy Concerns in Wireless Networks", IEEE Wireless Communications, August 2002, pp. 48-59.

[FuM02] Z. Fu, X. Meng, S. Lu, "How Bad TCP Can Perform in Mobile Ad Hoc Networks", Proceedings of the *IEEE Symposium on Computers and Communications (ISCC 2002),* Taormina-Giardini Naxos (Italy), July 2002, pp. 298-303.

[Fu03] Z. Fu, P. Zerfos, K. Xu, H. Luo, S. Lu, L. Zhang, M. Gerla, "The Impact of Multihop Wireless Channel on TCP Throughput and Loss", Proceedings of *IEEE INFOCOM 2003,* San Francisco (CA), March 30 – April 3, 2003.

[Glo] GloMoSim, Global Mobile Information Systems Simulation Library, http://pcl.cs.ucla.edu/projects/glomosim/.

[Hol99] G. Holland, N. Vaidya, "Analysis of the TCP Performance over Mobile Ad Hoc Networks", Proceedings of the *ACM International Conference on Mobile Computing and Networking (MobiCom'99)*, Seattle (WA), August 1999, pp. 207-218.

[Hol02] Gavin Holland, Nitin H. Vaidya "Analysis of TCP Performance over Mobile Ad Hoc Networks", *ACM/Kluwer Journal of Wireless Networks* 8(2-3), (2002) pp. 275-288.

[IEE02] Official Homepage of the IEEE 802.11 Working Group, http://grouper.ieee.org/groups/802/11/.

[IEE99] IEEE standard 802.11, "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, August 1999.

[Li01] J. Li, C. Blake, D. De Couto, H. Lee, R. Morris, "Capacity of Wireless Ad Hoc Wireless Networks", Proceedings of the *ACM International Conference on Mobile Computing and Networking (MobiCom'01)*, Rome (I), July 2002, pp. 61-69.

[Liu01] J. Liu, S. Singh, "ATCP: TCP for mobile ad hoc networks", *IEEE Journal on Selected Areas in Communications (J-SAC)*, 19(7):1300-1315, July 2001.

[Lun02] H. Lundgren, E. Nordstron, C. Tschudin, "Coping with Communication Gray Zones in IEEE 802.11 based Ad Hoc Networks", Proceedings of the *ACM Workshop on Mobile Multimedia (WoWMoM 2002)*, Atlanta (GA), September 28, 2002, pp. 49-55.

[Mon00] J.P. Monks, P. Sinha, V. Bharghavan, "Limitations of TCP-ELFN for Ad Hoc Networks", Proceedings of *MoMuc 2000*, Tokyo, Japan, October 2000.

[Ns02] The Network Simulator - ns-2, http://www.isi.edu/nsnam/ns/index.html.

[Oba02] Mohammad S. Obaidat, David G. Green, "An Accurate Line of Sight Propagation Performance Model for Ad Hoc 802.11 Wireless LAN (WLAN) Devices", Proc. ICC 2002, New York City, April 28 – May 2, 2002.

[Per99] C. Perkins, E. Royer, "Ad Hoc On-Demand Distance Vector Routing", Proceedings of the 2nd *IEEE Workshop on Mobile Computing Systems and Applications (WMCSA'99*), February 1999.

[Qua02] Qualnet simulator, http://www.qualnet.com/.

[Tan99] K. Tang, M. Gerla, "Fair Sharing of MAC under TCP in Wireless Ad Hoc Networks", Proceedings of *IEEE MMT'99*, Venice (I), October 1999.

[Tsc02] C. Tschudin, R. Gold, "LUNAR: Lightway Underlay Network Ad Hoc Routing", available at http://www.docs.uu.se/docs/research/projects/selnet/lunar/lunar.pdf.

[XuS01] S. Xu and T. Saadawi, "Does the IEEE 802.11 MAC protocol Work Well in Multihop Wireless Ad Hoc Networks?", *IEEE Communication Magazine*, Volume 39, N. 6, June 2001, pp. 130-137.

[XuS02] S. Xu and T. Saadawi, "Revealing the Problems with 802.11 MAC Protocol in Multi-hop Wireless Networks", *Computer Networks*, Volume 38, N. 4, March 2002.

[XuB02] K. Xu, S. Bae, S. Lee, M. Gerla, "TCP Behavior across Multihop Wireless Networks and the Wired Networks", Proceedings of the *ACM Workshop on Mobile Multimedia (WoWMoM 2002),* Atlanta (GA), September 28, 2002, pp. 41-48.

[XuG02] K. Xu, M. Gerla, "TCP over an IEEE 802.11 Ad Hoc Network: Unfairness Problems and Solutions", *UCLA Computer Science Deptartment Technical Report* – 020019, May 2002.

[Zar03] G. Zaruba, S. Das, "Off-the-Shelf Enablers of Ad Hoc Networks", in Mobile Ad hoc networking, S. Basagni, M. Conti, S. Giordano, I. Stojmenovic (Editors), IEEE Press and John Wiley and Sons, Inc., New York, 2003.