Modelling Channel Error in IEEE 802.11 Networks

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Abstract – Modeling the performance of IEEE 802.11 networks has been done since the standard first appeared. Most of the models assume saturated traffic conditions and do not distinguish between traffic classes and above all, assume the wireless link to be error-free. Without making any of these assumptions, in this paper we illustrate the effect of channel errors in particular on the Quality-of-Service (QoS) in IEEE 802.11 networks. For this purpose we developed our own deep simulator and workload generator. Our results show that, in the case where we had access to hardware test bed results, the results from our model approximate reality reasonably well.

Index Terms—IEEE 802.11, DCF, EDCF, QoS, simulation, channel error, workload models

I. Introduction

The original version of the standard IEEE 802.11 was released in 1997, before it became obvious that such networks would have to carry multimedia traffic such as video and real-time voice traffic as well as data. In 2005, IEEE 802.11e-2005 or 802.11e appeared as an approved amendment to the IEEE 802.11 standard that defines a set of QoS enhancements for wireless Local Area Network (LAN) applications through modifications to the Media Access Control layer (MAC).

IEEE 802.11 in its earlier versions has been modeled often many times [2] – [9], as discussed in Sec. II below. Usually the network being modeled is restricted in that it is assumed to be saturated with fixed-packet traffic, does not distinguish between traffic classes and all channels are assumed error free. While useful, such models are not very realistic. In this paper we remove all three these restrictions and offer a discrete event simulator we developed ourselves to illustrate the QoS results we obtained.

The reader is referred to any standard textbook such as that by Rappaport [29] for a discussion of the Distributed Coordination Function (DCF) and Enhanced Distributed Channel Access (EDCA) which was introduced with the IEEE 802.11e version of the standard. EDCA caters for multimedia traffic and thus makes QoS possible for different traffic classes. The work discussed here caters for both DCF and EDCA Medium Access Control (MAC) methods. In particular the traffic classes considered are those defined in the standard, namely

1. *Voice* traffic (VO) that caters for real-time voice traffic.
2. *Video* that caters for real-time video data, such as MPEG video packets that vary in size and are of variable bit rate.
3. *Best-effort* (BE) can be classified as all other kinds of non-detrimental traffic. This type of traffic is insensitive to QoS metrics, such as jitter, packet loss, and latency. A good example is Peer-To-Peer (P2P) traffic.
4. *Background*, this is designed for traffic that has no delay requirements. An example of background traffic is web browsing data.

Wireless transmission is error prone, and introduces relatively more transmission failures than wired mediums. Factors such as multi-path fading and noise and interference affect the transmitted signal.

The Orthogonal Frequency Division Multiplexing (OFDM) scheme is the choice of modulation in 802.11g/e networks. We will discuss this in more detail in the following sections.

The contributions of this paper are to investigate the effect of channel modelling in 802.11g and 802.11e networks giving particular attention to the QoS in these networks. Our aim is to accurately represent channel conditions that are present in real environments. This work removes assumptions that are normally included in modelling of such networks, such as ideal channel conditions and saturated fixed-packet workload models. We developed an Orthogonal Frequency Division Multiplexing (OFDM) modulation channel model, with Rayleigh multi-path fading and Additive White Gaussian Noise (AWGN). We also developed a multimedia workload model, which is representative of real network traffic. We then show how networks perform in different channel conditions, and which 802.11 protocol is better suited for QoS-sensitive applications.

In the next section we discuss previous work related to ours and in Sect. III we discuss communications channels with a view to developing the channel model described in Sect. V. Sect. IV briefly motivates why we developed a discrete event simulator from scratch and Sect. VI discusses the workload model that we developed to generate traffic for the simulator.

II. Earlier Work

All the analytic models [2] – [9], other than [3] and [7], assume ideal channel conditions, meaning that the channel does not suffer from nearby interfering networks operating at the same frequency, noise, channel fading and path loss. If there is no collision, a packet is transmitted successfully. This is not always the case in real networks since, even without collisions, a packet can be transmitted with error if the channel fades. Pham [3] modeled realistic channel conditions by using a slow Rayleigh fading radio channel link between moving nodes. Szczypiorski *et al.* [7] implement both error-free and error-prone transmissions. For error-prone transmissions, errors in the transmission medium are randomly distributed and in Challa [8] all stations have the same bit error rate.

Lyakhov *et al.* [9] presented an analytical model for estimating saturation throughput for 802.11 wireless LANs in the presence of noise, which distorts transmitted frames. The assumption of the absence of noise may overestimate the throughput because of the inevitability of the presence of electromagnetic noise in large cities. Other sources of noise may be neighboring LAN devices operating on the same license-free 2.4 GHz frequency band, multi-path fading, and adjacent channel interference.

Aguiar *et al.* [21] present a path loss model, called the two-ray model for channels. The two-ray model, also known as “plane earth”, is a model based on the theory of optics, which takes the reflection of waves on the earth surface into account. It also assumes Line-Of-Sight (LOS) and no influence on propagation besides the earth surface. It is often used to describe propagation over water or over open fields. Three waves are considered for its derivation:

• Direct waves,
• Waves reflected on the earth,
• Surface waves.

Surface waves become insignificant a few wavelengths above the earth surface and are therefore not significant for mobile communications. The two-ray model assumes LOS, as well as no other objects surrounding the path between the transmitter and receiver. These assumptions are not valid in many realistic environments, like urban, suburban and indoor environments, where non-LOS (NLOS) is as common as LOS.

III. COMMUNICATION CHANNELS

A wireless communication channel can be considered to comprise of a propagation channel, a radio channel, a modulation channel or digital channel.

The propagation channel lies between the transmitter and receiver antennas and it is only influenced by the phenomena that influence the propagation of electromagnetic waves. The transmitted signal is only affected by attenuation; therefore, this type of channel has a multiplicative effect on the transmitted signal. The radio channel consists of a propagation channel, and both the transmitter and receiver antennas. The transmitted signal is only affected by attenuation but, unlike propagation channels where the attenuation might be modified by the used antennas, the antenna influence is strictly linear. The signal is the same as with propagation channels, but it might also be scaled by the use of antennas. The modulation channel consists of a radio channel plus all system components involved from the output of the modulator on the transmitter side to the input of the demodulator on the receiver side, such as amplifiers. Due to the process of amplifying the received signals, some additive effects that damage the signal come into play, such as noise and interference. The digital channel consists of the modulation channel plus the modulator and demodulator. The input bits are grouped into analogue representations, called symbols, which are then transmitted. The bits are almost always grouped along with check bits, which are used to calculate whether the signal has been transmitted accurately.

Next, we focus on OFDM modulation, which is used by the 802.11g and 802.11e standards at 54Mbps data rate [1], as well as channel fading, interference and noise.

(1) OFDM Transmission

OFDM is a digital multi-carrier modulation method by which a signal is split into several narrowband channels at different frequencies [31]. Data are divided among a large number of closely spaced and orthogonal sub-carriers, which are used to carry data. Each sub-carrier is modulated with a conventional modulation scheme, such as quadrature amplitude modulation or phase shift keying, at a low symbol or bit rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth. The entire bandwidth of the system is occupied by a single source of data, and the data is transferred in parallel. Since a smaller amount of the data is carried on each sub-carrier, and there are lower bit rates per carrier, the influence of inter-symbol interference is significantly reduced.

In 802.11g/e at 54Mbps, OFDM with 52 sinusoidal sub-carriers, a symbol period T, and a frequency separation of 1/T is used.

(2) Fading

Multi-path versions of the same signal will interfere with one another, with the result that there will be fluctuations of amplitudes and phases in the received signal. This phenomenon, and its effect on the received signal, is known as small-scale, multi-path fading [31].

There are several factors that influence fading in radio channels:
• Multi-path propagation,
• Speed of the mobile device,
• Speed of the transmitting object in the environment.

Multi-path propagation is the phenomenon that results in radio signals reaching the receiving antenna by at least two or more different paths, mainly due to refraction of the signals, or reflection from water bodies and terrestrial objects, such as mountains or buildings.

The effects of fading can be characterized by a multi-path time delay spread or a Doppler spread. Multi-path time delay spread determines whether a channel experiences flat or frequency selective fading, and Doppler spread determines whether a channel experiences slow or fast fading.

Flat fading occurs when all of the spectral components of the signal are equally affected by fading. The coherence bandwidth of a channel defines the bandwidth over which a channel will experience flat fading. If the bandwidth of a transmission is less than the coherence bandwidth, the channel experiences flat fading otherwise, it experiences frequency-selective fading. Frequency-selective fading is more difficult to model, as it does not affect all parts of the transmission equally. Flat fading channels are often referred to as narrowband channels, and frequency-selective fading channels are known as wideband channels.

The rate at which fading fluctuates determines whether or not the channel experience fast or slow fading. If the bit period is greater than the average period of a fading signal, the channel experiences fast fading; otherwise, it is slow fading. In OFDM, the total channel is a frequency selective channel. However, the channel used by each sub-carrier in an OFDM system is a flat fading channel.

If there is no dominant propagation along a LOS between transmitter and receiver, and all reflected signals are received with similar signal strengths, then the channel experiences Rayleigh fading. In this case, the envelope of a single multi-path component can be characterized by a Rayleigh distribution. If the dominant propagation results in a signal much stronger that the others along an LOS, then the fading is known as Ricean fading, and the amplitude gain is characterized by a Ricean distribution. Rayleigh fading is characteristic of heavily built-up urban environments which we decided to be typical for what we would like to consider.

(3) Interference and Noise

Interference is anything which alters, modifies, or disrupts a signal as it travels along a channel from transmitter to receiver. In general, it refers to the addition of unwanted signals to a useful signal. Noise, on the other hand, is defined as an unwanted perturbation to a wanted signal. If at least two interference sources are active or exist, interference may be modeled as a white Gaussian process, adopting many of the characteristics of noise. In this paper, we focus on Additive White Gaussian Noise (AWGN). AWGN is one of the most common noise sources in communication systems [30]. The term “white” refers to the frequency spectrum, which is deemed continuous and uniform within the frequency band of interest. The thermal noise at the receiver amplifier shows uniform noise power per unit bandwidth at all frequencies.

IV. DEEP SIMULATION

For a simulation one can either apply existing platforms, such as the OPNET Modeler [20], ns2 [11], QualNet [26] or OMNeT++

[22], or develop one's own simulator by implementing the simulation engine and components in a language such as Java. The OPNET Modeler Wireless Suite, as one instance, provides high fidelity modeling, simulation, and analysis of a broad range of wireless networks. But these simulators are complex, general-purpose software suites and it is seldom clear which details of the network stack are being modeled or where all the associated parameters may be found. More controversially, some of these systems are commercial products and, for proprietary reasons or otherwise, do not make clear [14] whether one is assured that the simulation has stabilized before sampling, what the sample sizes are, or whether the sampling is done to ensure identically and independently distributed (i.i.d) variables.

In a controversial paper, Cavin [25] illustrated these points by showing the deviation in results among widely-adopted simulators, such as OMNeT++ and ns2, for a sample mobile ad-hoc network experiment. Statements such as *". . . users are responsible for verifying for themselves that their simulations are not invalidated because the model implemented in the simulator is not the model that they were expecting. . . "* [23] are not very encouraging when one is conducting a scientific study. Nevertheless, these general simulation platforms and libraries are obviously useful as witnessed by the many published studies that use them.

Some authors, such as Bianchi *et al*. [2] and the authors [24] prefer to know exactly what is being simulated and therefore prefer the second option. They

• sacrifice the convenience of these software platforms and, at the same time,

• spend much time delving down into the minute system detail to ensure that these are represented by the simulation model. It is then possible to measure and record exactly those performance values relevant to the study.

In contrast to the use of general purpose simulation platforms, we call this *"deep simulation"*. Deep simulators make no claim to be general or to replace the existing general simulation platforms.

We used the Java programming, whose flexibility, extensive libraries and algorithmic capabilities supersedes the higher level languages provided by platforms such as OPNET Modeler, ns2, QualNet or OMNeT++. Space does not allow us however to describe our simulator in detail and the reader is referred to the paper by Pileggi *et al.* [16] for details. The simulator is event-driven and modular. It comprises three sub-models; called the Machine Model (MM), Channel Model (CM), and Workload Model (WLM). The MM and CM interact during transmission, and there is an interface between the MM and the WLM that facilitates packet arrivals into the network.

V. CHANNEL MODEL

The channel model described in this section is derived from an OFDM modulation scheme proposed by [30] and [32] and a Rayleigh multi-path fading scheme by [31].

In the simulator, the CM determines whether a packet or frame has been transmitted successfully or not over the wireless medium. Since channel modeling is the focus of this paper, a detailed discussion of its implementation is given in this section. The CM is consulted each time a packet (or frame) is transmitted. At the end of the data transmission, the size of the data unit transmitted is given as input to the CM. The CM, in turn, makes use of this data size and the bit energy-to-noise ratio (Eb/NO) to determine whether the frame was successfully transmitted or not. The Eb/NO is an independent integer variable supplied by the user of the model.

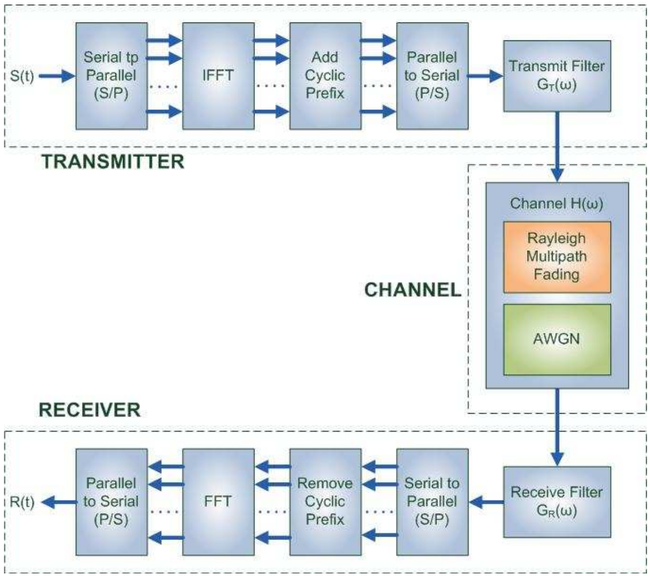

Figure 1: OFDM Channel with Rayleigh multi-path fading and AWGN

Figure 1 illustrates the CM, and how data flow from transmitter to receiver. The Rayleigh Fading and AWGN components are not part of the OFDM modulation scheme, but are external conditions that affect the efficiency of the CM. We decided to model the OFDM modulation scheme because it is the scheme employed by the 802.11g and 802.11e networks, and we wanted to represent these wireless networks as accurately as possible. We also modeled multi-path fading and noise, since these factors significantly influence the efficiency of the medium. The model passes the size of the packet as a parameter. The Signal-To-Noise Ratio (SNR) is then calculated using the assumed Eb/NO ratio using:

$$SNR = Eb/NO + 10 * \log_{10}(nDSC / nFFT) +$$
$$10 * \log_{10}(Data\ Symbol\ Duration / Total\ Symbol\ Duration)$$

where

nDSC is the number of sub-carriers, and *nFFT* is the FFT size.

In order to emulate a transmitted packet, a random generator is used to generate bits stored in an array of the same size as the input frame. If the random number has a value less than 0.5, it is stored as a 0; otherwise it is stored as a 1. Next, Binary Phase Shift Keying (BPSK) modulation is applied, and it changes all 0's to -1 and all 1's remain as 1's in the array, using the following calculation:

$$(2 * array[x]) - 1$$

where

array[x] denotes a bit in the frame at position x

There are 52 sub-carriers specified for OFDM at 54Mbps, and the FFT sample size is set to 64. All the BPSK modulated bits are assigned to each of the sub-carriers. A *reshaping* function is used to assign these symbols. Each sub-carrier will contain a number of symbols calculated by dividing the number of bits by the number of bits per symbol. This data is then stored in a 2-dimensional array with the number of sub-carriers and number of symbols as the dimensions. As illustrated in Fig. 1, the next step is to take the Inverse Fast Fourier Transform (IFFT) of the modulated symbols in the sub-carriers, which renders the frequencies in the sub-carriers orthogonal.

In multi-path fading channel models, many time-delayed versions of a transmitted waveform might be found at the receiver. Without a Guard Interval (GI), the waveforms would interfere with each other. The cyclic prefix (CP) is the most commonly used GI technique [32]. In the CP scheme, the GI is a copy of the partial waveform, meaning that part of the waveform is taken and appended to the waveform. The CP is appended to the beginning of the data set. This is done by taking the last 16 columns in the array and appending them to the beginning of the array. This ensures that the waveform is continuous [30]. The multi-path channel is executed in this step. A set of complex values corresponding to the number of symbols is created. Complex numbers are used because they are periodic, and they can be used to represent phases, since they have a real part and an imaginary part. The fading on each symbol is independent. The frequency response of the fading channel on each symbol is computed and stored. This is done by applying the Fast Fourier Transform (FFT) algorithm. The frequency response is the measure of the models output spectrum in response to the input signal [30]. This will be used on the received signal. The next step is the convolution of each symbol with the random channel. This is a mathematical operation on two functions, *a* and *b*, that produces a third function that is viewed as a modified version of one of the original functions [30].

The multiple symbols are then concatenated, one after the other, to form a long sequence of data that will be transmitted over the medium. At this step AWGN is added to the transmitted signal. The AWGN is modeled as Gaussian complex numbers, and randomly generated values are added to the long transmission sequence. The SNR is used as a function to determine the severity of the noise on the transmission sequence. A higher SNR results in a smaller the effect of noise on the data.

As shown in Fig. 1 the received transmission sequence is turned back into multiple symbols at the receiver and the CP is removed from the data set to leave only the relevant symbols. The received symbols are then converted from a time domain to a frequency domain. This is achieved by using the FFT algorithm. The time domain shows how the signal changes over time, and this is the domain in which the signal is transmitted over the medium. The frequency domain, however, shows how much of the signal lies within each given frequency band over a range of frequencies, which is the case with the OFDM sub-carriers.

Each element in the received symbol is divided by its corresponding element in the frequency response that was calculated previously. This gives a measure of how efficient the channel is when transmitting data. BPSK demodulation occurs, and all -1's are converted to 0's and all 1s remain as 1s. This leaves a binary sequence of the received packet. This resembles the received data packet or frame. The corresponding bits from the transmitted and received packets are compared. A counter is incremented whenever there is a mismatch. If the counter is greater than 0 after the comparison, it means that the packet was received with an error, otherwise a count of zeros signals successful transmission.

VI. WORKLOAD MODEL

The performance of a network largely depends on the traffic load. Hence a performance model of a network is incomplete without a model of representative workloads in order to produce dependable results. In our simulation we used a traffic model which generates four different traffic classes; *Voice Over IP* (*VoIP)*, *Video*, *Best-Effort*, and *HTTP* Traffic.

(1) Voice

We developed a VoIP traffic model from Heffes et al. [10], who used a two-state Markov Modulated Poisson Process (MMPP) to model aggregate traffic from voice sources. MMPP is a double stochastic Poisson process in which a Markov chain determines the transitions between phases. A Poisson process is used to determine the number of arrivals in a time frame at each phase.

The importance of the MMPP model is that it is capable of catching inter-frame dependency between consecutive frames. This is indicated by the Index of Dispersion for Counts (IDC). IDC is defined as the variance-to-mean ratio for the number of arrivals of increasing size in non-overlapping blocks [10]. For a general MMPP, we use a transition matrix of the modulating Markov chain and another matrix whose diagonal elements contain the arrival intensities that correspond to the different states of the chain. Packet sizes were chosen from [11].

(2) Video

Video frames are long, and they typically span multiple IEEE 802.11 packets. Packet headers may contain information that is critical for keeping the traffic synchronized. Video quality suffers a lot from packet loss. If one packet is lost, it will affect other subsequent packets and therefore the propagation effects of packet loss can be significant.

We developed a Log-Normal traffic generator for video traffic [18]. The model is based on analysis carried out on data from the PPLive P2P streaming video network.

Investigations by [18] showed that the majority of the recorded data are from UDP flows, with TCP flows only accounting for 0.46% of the data [18] PPLive uses UDP as the main bearing protocol for video content. Network traffic was measured by [18] by connecting to a popular financial program at 9 AM, monitoring 2 hours of traffic flow, and then all the UDP packets were divided to analyze the time between consecutive packets as well as the packet size distribution [18]. The Kolmogorov-Smirnov Test (K-S Test) was used to determine the best fitting PDF for modelling video traffic IATs [18]. A K-S test tries to determine if two datasets differ significantly, and it has an advantage of making no assumption about the distribution of data [19].

The Log-Normal PDF was chosen for the video frame inter-arrival time (IAT) distribution.

(3) Best-Effort

Best effort traffic can be classified as all other kinds of traffic that are not QoS-sensitive. This type of traffic is insensitive to QoS metrics, such as jitter, packet loss, and latency. Peer-to-Peer (P2P) traffic and Email are typical examples.

In this work, emphasis is put on P2P traffic due to the vast volumes of traffic generated by this application. P2P file sharing applications are often designed to use any and all available bandwidth, which could impact QoS-sensitive applications, like voice and video unless the network is managed well. IEEE 802.11 wireless LANs have less bandwidth compared to wired technologies (typically 54Mbps for 802.11g and 802.11g), and the use of P2P applications with these networks results in higher network latency for the users. Since EDCA gives lower priority to this traffic, it is interesting to see whether transmission of higher priority, QoS-sensitive traffic is not affected. It is also important to see how this compares to the DCF.

We used a Markov Arrival Process (MAP) to generate packet inter-arrival times (IAT). The model was based on [13], which proposed a Pareto PDF for the IAT distribution. Analysis from P2P sources showed that the traffic has a heavy-tailed distribution. This comes from the analysis of the data flow size against the flow holding time [13]. The assumptions of Pareto distribution were verified by several heavy-tailed tests: De Haan's moment method, Hill estimator, and Q-Q plot [13].

(4) HTTP Traffic

The World Wide Web (WWW) is a major source of traffic in the internet. HTTP traffic is traffic generated when a user is using a WWW browser. We also used a MAP to generate the IATs. Analysis from several WWW traffic sources suggested the Weibull distribution as a fitting distribution for the IATs [14].

(5) Traffic Mapping

The traffic sources are mapped to the corresponding ACs in the EDCA protocol. The DCF protocol randomly selects a traffic source for the next packet

VII. Experimentation And Results

In our experiments, we investigated the effect of channel errors in both DCF and EDCA, using a realistic workload and channel model. The performance metrics of interest are *normalized aggregate throughput*, *access delay*, and *bit error rate*.

Normalized aggregate throughput is the fraction of the number of bits sent in a given time duration to the maximum possible number of bits that could be sent in the given time duration. This is expressed as a decimal, and it is between 0 and 1. Access delay is the time taken from when a packet becomes available for transmission to the time it is received by the receiver without error. The access delay is expressed in microseconds. The bit error rate is the fraction of the number of bits in error per packet to the size of the packet. This metric is expressed as a decimal between 0 and 1.

Two sets of experiments were carried out; one for the DCF, and another one for the EDCA protocol. In each set, 8 experiments were executed for 8 Eb/NO values; 15, 20, 25, 30, 35, 40, 45, 50, in a network of 10 stations. Tables 1 and 2 show the model parameters that were used for the DCF and EDCA protocols, respectively. The EDCA protocol specifies 4 access categories (ACs), and each AC should have its own AIFS value and contention window (CW) parameters. AC 0 in this case denotes the lowest priority AC, with AC3 being the highest priority.

Parameter	**Value**
SIFS	16 µS
Slot Time	9 µS
ACK Frame Size	112 Bits
RTS Frame Size	160 Bits
CTS Frame Size	112 Bits
UDP	64 Bits
Snap	64 Bits
IP	160 Bits
Preamble Length	16 µS
PLCP Header	4 µS
Signal Extension	6 µS
MAC Header	304 Bits
RX Start Delay	25 µS
Propagation Delay	1 µS
Channel Speed	54 Bits / µS
Control Frame Speed	24 Bits / µS
CW Min	15
CW Max	1023

Table 1: Parameter values for the physical layer in DCF

Parameter	**Value**
SIFS	16 µS
Slot Time	9 µS
ACK Frame Size	112 Bits
RTS Frame Size	160 Bits
CTS Frame Size	112 Bits
UDP	64 Bits
Snap	64 Bits
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Signal Extension	6 µS
MAC Header	304 Bits
RX Start Delay	25 µS
Propagation Delay	1 µS
Channel Speed	54 Bits / µS
Control Frame Speed	24 Bits / µS
AIFS (AC0)	7 µS
AIFS (AC1)	3 µS
AIFS (AC2)	2 µS
AIFS (AC3)	2 µS
TXOP Limit (AC0)	0 mS
TXOP Limit (AC1)	0 mS
TXOP Limit (AC2)	(6.016 x 1000) µS
TXOP Limit (AC3)	(3.264 x 1000) µS
CW Min (AC0)	31
CW Max (AC0)	1023
CW Min (AC1)	31
CW Max (AC1)	1023
CW Min (AC2)	15
CW Max (AC2)	31
CW Min (AC3)	7
CW Max (AC3)	15

Table 2: Parameter values for the physical layer in EDCA

In each experiment, the simulator executed 30 runs. The relevance of having 30 simulation runs is in order for the results to achieve statistical significance, resulting in normally distributed data sets, which enables the system calculate the sample mean and confidence intervals for each number of stations [15]. The density of the IATs in the workload model was increased in order to have a lot of packets in the system, but without distorting the IAT distributions. This is because we are interested in the worst case scenarios in network performance, i.e. we want to see how a network performs under a heavy, albeit not saturated, workload.

Data samples were recorded from the start of the simulation, since the system stabilized soon after the start of each run. An experiment was a single-run experiment with duration of 30 seconds for 1 contending station. Throughput was sampled using consecutive observation windows with duration of 100 microseconds. The throughput variation remained constant in time, showing that the system is in a stable state from very early on in the experiment.

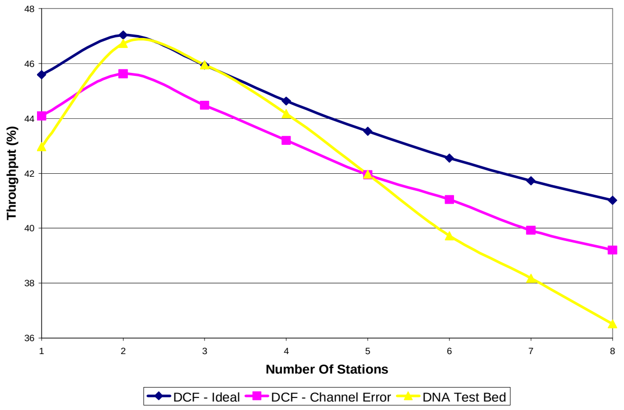

Figure 2: Comparison of the DNA Hardware test bed results to the simulation

Figure 2 compares results from the DCF model with ideal channel conditions, DCF with channel errors, at Eb/NO of 45, and results from the DNA hardware test bed [28]. An Eb/NO of 45 was chosen because the hardware test bed was executed in an isolated environment.

The results show that using a channel model approximates the results relatively better than using ideal channel conditions.

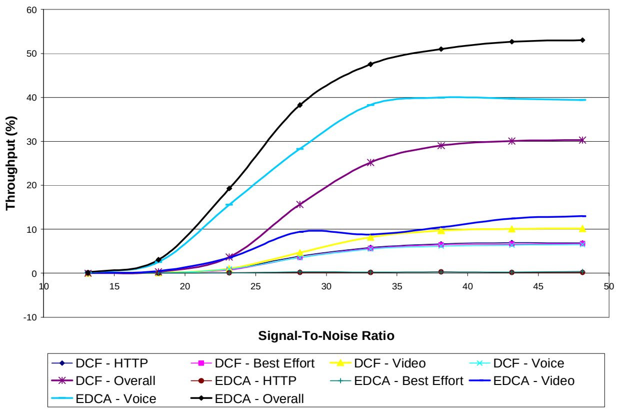

Figure 3: Signal-To-Noise Ratio vs. Normalized Throughput

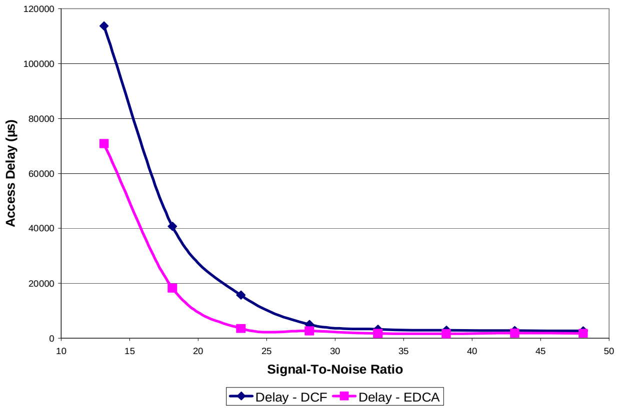

Figure 4: Signal-To-Noise Ratio vs. Access Delay

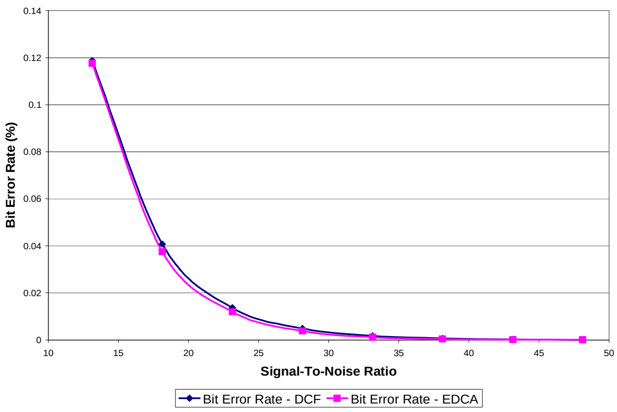

Figure 5: Signal-To-Noise Ratio vs. Bit Error Rate

Figure 3 shows the normalized throughput as the signal-to-noise ratio increases for all ACs in DCF and EDCA. It also shows the aggregate throughput for all ACs for both protocols. The throughput increases rapidly as the ratio increases, but later stabilizes.

EDCA achieves higher throughput than the DCF overall. This is because the EDCA protocol, on average, sends more packets per given time than DCF. Comparing Tables 1 and 2, we see that the minimum and maximum CW values for AC 3 in table 2 are smaller than those of the DCF. This means that, on average, packets sent by the EDCA in AC 3 have smaller back-off values, resulting in packets waiting shorter periods before being transmitted. Voice traffic achieves the highest throughput for EDCA, followed by video traffic. Best-effort and HTTP traffic have very low throughputs. Throughput is evenly distributed, on the other hand, among voice, video, best-effort, and HTTP traffic for DCF.

The above analysis shows that QoS-sensitive applications can are more suited to EDCA. DCF does not prioritize any traffic; therefore, it does not accommodate QoS-sensitive applications that well.

AC 2 and AC 3 also have non-zero Transmission Opportunity (TXOP) limits. This means that if the AC acquires the channel, it can send as many packets as it can within the TXOP duration, only waiting for a SIFS interval before transmitting the next packet. This means that, as expected, the EDCA achieves higher normalised throughput than the DCF. A TXOP limit of zero means that the AC can only transmit 1 packet when it acquires the transmission channel.

The access delay and bit error rate, shown in figures 4 and 5, respectively, decrease with an increasing signal-to-noise ratio, as expected. However, it is clear that, for both MAC protocols, after some point, the effect of channel error quality starts to affect these performance metrics more significantly and exponentially. For the DCF and EDCA, it seems that a SNR >= 33 does not impact performance significantly. The delay and bit error rate decrease because the number of errors decrease as the signal-to-noise ratio increases, resulting in more packets being transmitted.

The overall delay is lower for the EDCA compared to the DCF, and this is because the higher priority queues in EDCA can send multiple packets, one after a SIFS event, and this decreases the average waiting time of a packet. Every DCF packet contends for the channel, which results in longer waiting times.

The bit-error rate should be similar, since the channel model operates the same no matter the protocol in use.

VIII. Conclusion

In this work, we took great care to accurately represent the 802.11g and 802.11e protocols in a simulation environment. These protocols were accompanied by a channel model and a workload model. The channel model was a combination of the OFDM modulation scheme, Rayleigh multi-path fading, and AWGN. The choice of OFDM was motivated by the fact that it is used in both 802.11g and 802.11e. The nature of Rayleigh multi-path fading is that which is experienced in heavily built environments, such as urban metropolitans, which was the focus of our work. AWGN introduces the effect of noise in the wireless medium.

A suitable workload model was developed. The density of the IATs was increased in order to introduce more packets into the system, but without distorting the IAT distributions.

The efficiency of the wireless medium is significantly affected by the SNR, and the higher the SNR, the more efficient the medium. The EDCA protocol, on average, transmits more packets than the DCF, and it showed that it is better suited to QoS-sensitive applications.

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