DCF modeling and optimization of long-distance IEEE802.11n point-to-point links

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Abstract

This work describes extensions to the well-known Distributed Coordination Function (DCF) model to account for IEEE802.11n point-to-point links. The developed extensions cover adaptions to the throughput and delay estimation for this type of link as well peculiarities of hardware and implementations within the Linux Kernel. Instead of using simulations, the approach was extensively verified on real-world deployments at various link distances. Additionally, trials were conducted to optimize the CW_{min} values and the number of retries to maximize throughput and minimize delay. The results of this work can be used to estimate the properties of long-distance 802.11 links beforehand, allowing the network to be planned more accurately.

1 Introduction

Rural areas often lack affordable broadband Internet connectivity. The resulting digital divide limits the access to knowledge, health care and other services for billions of people. Compared to the high expenses for traditional operator equipment (e.g. CAPEX and especially OPEX), Wireless LAN hardware offers an alternative, cost-efficient back-haul connectivity [1]. Several publications described how rural areas can be connected to the Internet by using WiFi based Long-Distance networks (WiLDs) [1] [2] [3]. All these approaches use IEEE802.11 [4] off-the-shelf hardware.

We have shown in [5] that the physical layer data rate of an 802.11 radio can be increased up to 270 Mbps over a distance of 10 km through exploiting the so-called next generation wireless standard 802.11n in combination with cross polarized (MIMO) high gain antennas. Using uni-directional traffic pattern[1], we found a real throughput of 200 Mbps. However, especially this uni-directional traffic assumption does not hold when users really access the Internet.

To control the medium access in 802.11, networks the IEEE standardized the Distributed Coordination Function (DCF). This function was specifically designed to account for contiguous stations in a cell and spatial restrictions of a few hundred meters at most. This scenario is significantly different to that of WiLDs. We therefore evaluate the throughput and the delay under different link properties and thus provide a complete view on the capabilities of the default 802.11 MAC layer for long-distance links in bi-directional and saturated traffic. This evaluation is carried out using a modeling approach carefully validated with real-world deployments.

In Section 2, a short summary of previous work and a motivation for this novel approach are provided. Section 3 introduces the reader to the basic background of the work. Section 4 introduces the conducted modeling extensions accounting for high throughput long-distance point-to-point links. Sections 5 and 6 present how the validation, estimation and optimization were conducted. The paper ends with a conclusion in Section 7.

2 Previous work and motivation

There are two main reasons why this work is important: the lack of previous work on point-to-point links and researchers' previous reliance on simulations rather than real world measurements. After the first IEEE802.11 standard was published in 1997, several researchers [6] [7] [8] focused on developing models for the DCF. These models evolved over time by including extensions to refine initial assumptions and to account for traffic class separation introduced by the Enhanced Distributed Channel Access (EDCA). However, researchers [8] have only partially addressed the use-case of long-distance point-to-point links, and in this work they focused on large deployments of Wireless Mesh Networks (WMNs) and the outdated 802.11b standard rather than on high throughput point-to-point links. To validate their modeling approach, nearly all researchers use simulations instead of measurements on real hardware [9]. However, the outcome of a possible capacity estimation is only useful when it provides sufficient accuracy on real deployments as well. In fact, researchers focus on the discrepancies between simulations and testbeds and find huge gaps in the results of throughput and delay [10]. Possible reasons for these discrepancies are the peculiarities of the WiFi cards, the implementation of the MAC and driver, and the packet processing in the Kernel.

As a basis for this work, a revised version of Bianchis original DCF modeling approach is used [6]. In their work, the DCF modeling is based on conditional probabilities, and the delay is estimated using Little's law. Several researchers already extended this model, for example to account for a BER on the physical layer [9], and increased the accuracy by including two effects: backoff-freezing [11] and anomalous slots [7]. These extension will be used in this work.

The model extensions and novel scenario developed in this work not only permit the capacities for WiLD links in bi-directional traffic to be estimated but also allow the MAC parameters to be optimized.

[1]Uni-directional, UDP, large payload and high aggregation factors.

3 Background

This section introduces the basics of the DCF and long-distance 802.11 links.

3.1 DCF

In general there are two different techniques associated with DCF. The basic access technique is a two-way hand-shake characterized by transmitting a packet, immediately followed by the receiver sending an acknowledgement (ACK). In addition to this basic access method, the standard specifies a four way hand-shake (often referred as request-to-send/clear-to-send (RTS/CTS)) [4]. When a frame arrives at the head of the transmission queue, a station monitors the channel for a DIFS idle time. To detect an idle channel, 802.11 uses a carrier sense function on the physical layer and a virtual carrier sense function on the MAC layer. To prevent all stations transmitting at the same time as soon as the medium is sensed as idle, the back-off process starts. At the beginning of this collision avoidance process, a random number, uniformly distributed between zero and the current Contention Window (CW)-size, is generated.

The back-off process uses a slotted and discrete time scale. The slot time duration depends on the underlying physical layer and should be equal to the time needed for every station to detect a transmission on the medium. The back-off time counter is represented by the randomly generated number, which is subsequently decremented as long as the carrier sense functions report an idle medium. If a transmission starts and the channel becomes busy during a back-off decrease, the value is frozen and the process pauses. When the channel becomes idle again for the time of a DIFS, the process is resumed with the frozen value [9]. When the counter reaches zero, the station is allowed to start the transmission in the current slot. If the ACK is not received within the so-called ACK-timeout, the frame is considered as lost and needs to be retransmitted after another random back-off. A transmission can be aborted because another station randomly picked the same transmission slot. Initially assigned to CW_{min}, the back-off scheme doubles the value of CW in this case until it reaches CW_{max}:

$$\boldsymbol{CW_{i+1} = 2(CW_i + 1) - 1} \qquad (1)$$

This doubling of CW decreases the probability of two stations picking the same back-off slot, but increases the back-off time.

3.1.1 Modeling

Since the following extensions directly build up on previous work done by Bianchi [6], only the main aspects will be summarized. [6] and [9] contain further details.

Being able to mathematically describe the DCF relies on two important observations. First, the presence of a discrete time scale (slot time). Second, the fact that only three different basic events (P) can occur all linked with a fixed time (T) for this event:

- Transmission successful: P_s and T_s
- Collision: P_c and T_c
- Idle: P_i and slot time (σ)

For example, the time for a successful transmission (T_s) consists of the time to transmit a MPDU (depended on the used MCS and the payload size), a SIFS, and afterwards the transmission of an ACK followed by a DIFS.

In [6] the authors have proven that it is possible to express all three events using two probabilities: constant transmission- (τ) and a constant collision (p) probability. These probabilities can be obtained by the numerical solution of two non-linear equations and strongly depend on the back-off scheme (β), the number of stations (N) in the ad-hoc cell, CW_{min} and the number of retransmissions (R).

Together with these three basic events, an addition to account for erroneous frames (ζ) and the expected payload size, the following equation can be used to obtain the maximum saturation throughput for the ad-hoc cell [9]:

$$\boldsymbol{S = \frac{(1-\zeta)P_s E[Payload]}{P_i\sigma + (1-\zeta)P_s T_s + \zeta P_s T_e + (1 - P_i - P_s)T_c}} \qquad (2)$$

To obtain the average access delay (D), [9] suggest using Little's law with an addition to account for ultimately dropped frames after the retry limit (R) has been reached:

$$\boldsymbol{D_{Access} = \frac{N}{S/E[P]} - \frac{E[slot]p^{(R+1)}}{1 - p^{(R+1)}} \sum_{i=0}^{R} 1 + \beta_i} \qquad (3)$$

3.2 Long-distance constraints

The applicability of the 802.11 standard to long-distance links needs to deal with major constraints on the phy and MAC layer.

The major constraint limiting the range of WiLD is the SNR needed for higher modulations (MCS). The SNR is mainly[2] reduced by two major physical conditions [12]:

- Free-space path loss (FSPL)
- Fresnel zone.

Both conditions attenuate the radio signal and are influenced by the selected frequency and the distance between receiver and transmitter. Using high gain antennas (up to 34 dBi) and high-power wireless cards (up to 600 mW) makes it possible to stretch these constraints and provide high throughput over long-distance [5]. However, common WiFi-cards operate in the ISM-band, which is regulated by different organizations and governments.

The constraints of WiLD links on the phy are evident; the situation on the MAC layer is more complex. The main problem is the increased propagation time resulting in unwanted MAC-layer timeouts. Basically the increased range needs to be taken into account by increasing the timings for two parameters:

- Slot time (σ)
- ACK timeout.

The slot time needs to be increased to give all stations the chance to detect any ongoing transmissions on the medium during the back-off interval. The ACK timeout represents the time the transmitter waits for an ACK from the receiver before marking the frame as lost.

[2]Compare [12] for additional aspects.

4 Model extensions

To account for 802.11n long-distance links, the modeling approach needs several adaptations. These adaptions need to account for

- MAC adaptions needed for WiLD links
- The 802.11n standard
- Peculiarities of hardware and implementations.

To account for the enlarged timings of WiLD transmission, the slot time is no longer considered as a fixed value defined by the standard; instead it changes dynamically according to the distance of link[3].

$$\sigma(d) = \sigma + 3 * \mathrm{cc} = \sigma + 3 * \left\lceil \frac{2 * d\,[m]}{300\,[m/\mu s] * 3} \right\rceil \quad (4)$$

This equation uses the coverage class (cc), which is defined in the standard [4] as an adaption of the slot time according to the propagation delay in a non-continuous way. One additional step of the cc correlates to an extra distance of 450 m. This redefinition of the physical slot time implies a change of all other Interframe Spaces (IFS). Additionally, the propagation time needs to be added to the three basic timings *twice* since the delay will occur for the transmission of the PPDU and the subsequent ACK.

Taking the different MCS into account makes it possible to integrate the physical layer extensions into the modeling approach. A bandwidth of 40 MHz and the MIMO technique increase the number of databits per OFDM symbol and therefore decreases the time of the basic event. If MIMO is used and assuming the HT-Greenfield-Frame format for ad-hoc cells, the physical layer preamble is enlarged by several fields [4].

Several changes are needed to include frame aggregation as the main 802.11n MAC layer extension into the approach. As described in [5], only one aggregation technique is currently implemented in recent mac80211 versions of the Kernel in ad-hoc mode: A-MPDU. This technique aggregates completely formatted MAC frames and uses the block ACK protocol to efficiently acknowledge sub-frames through a bitmap. The number of aggregated Bytes is controllable:

$$AMPDU_{max} = 2^{13+i}\ Bytes, \forall i \in \{-3, -2, \ldots, 3\} \quad (5)$$

However, it is also limited due to fairness considerations among stations. The maximum transmission time on the medium for a PPDU may not exceed a certain level. In the kernel, this maximum time is set to 4 ms. To calculate the number of aggregated blocks for an A-MPDU frame the following equations holds:

$$N_B = \min\left(N_{AMPDU}, \left\lfloor \frac{4[ms] * MCS[Bps]}{Payload + L_{MAC-HDR}} \right\rfloor\right) \quad (6)$$

In combination with a 4 Byte delimiter and a MAC Header for each of the sub-frames, the A-MPDU technique increases the basic timings and the expected payload size. An additional adaption for the inclusion of the A-MPDU technique is the correct usage of the probability for a corrupted frame. The probability that an error in all sub-frames occurred can be expressed as [13]:

$$\zeta = \prod_{i=1}^{N_B} \left(1 - (1 - BER)^{L_i}\right) \quad (7)$$

However, this case should not regularly occur unless there is constant interference on the link. To account for selective retransmission, the average number of transmitted payload bits needed for the throughput estimation can be redefined in the following way:

$$E[P] = \sum_{i=1}^{N_B} (L_i - 32)(1 - BER)^{L_i} \quad (8)$$

To calculate delay, an additional effect needs to be considered. The so-called block ACK reordering takes place at the receiver if one or more sub frames of an aggregated transmission are damaged. In this case, the receiver buffers the complete block of received frames for a defined time and waits for the retransmitted damaged packets to arrive. If the frames arrive during this time, the packets get reordered to their original arrangement and are afterwards handed to higher layers. The following equation accounts for this:

$$D = D + (N_B * P_{BER} * T_{Reorder}) \quad (9)$$

Additional aspects arose while validating this approach. Most of these aspects were related to using real hardware instead of simulations.

For the basic timing of a collision, the ACK timeout [4] and not an EIFS must be used. In current implementations, the MCS used for the transmission of the ACK is not bounded to the lowest possible MCS for all participants in the cell. Instead, the MCS of the ACK rises with the MCS of the data frame. However, only the MCSs with a coding rate of ½ are used to ensure a reliable transmission of the ACK.

An additional aspect to consider for an accurate modeling of 802.11n WiLD links is the buffer size of the transmission queue since this size influences the throughput and the delay of bi-directional traffic saturation, as shown in Figure 1.

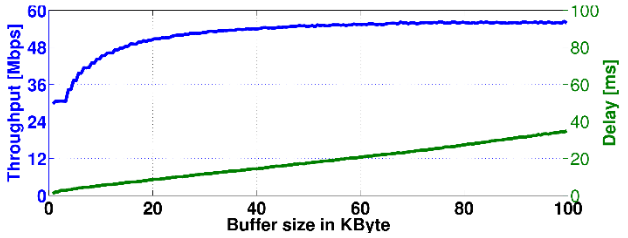

Fig 1: Influence of the buffer size, MCS 7, agg 3

To model the delay including the buffer, the following approximation holds:

$$D_{Sys} = D_{Access} + \frac{D_{Access} N_{Buffer}}{2} + D_{Process} \quad (10)$$

The results indicate that the delay suffers from a rather constant processing time in saturated traffic conditions (approximately 0.2 ms). The assumption is, however, that this time is highly dependent on the hardware used.

For the reduction of the throughput occurring for small buffer sizes, the following approximation (11) is used. Assuming a finite packet arrival rate at the MAC layer, after some transmissions there are no more packets

[3] In [14] we introduced a link calibration algorithm that is capable of determining the distance.

queued in the buffer to send in an aggregated packet. The empty buffer reduces the expected payload size.

$$E[P] = E[P] - E[P]\left(\sum_{k=0}^{\infty} P_S^k - \sum_{k=0}^{(1+(N_{Bu}/N_B))} P_S^k\right) \quad (11)$$

5 Validation

Three links at different distances (2 km, 5 km, 7.8 km) were set up at the Fraunhofer Campus in Sankt Augustin. Additionally, a laboratory environment using high frequency cables instead of directional antennas (no measurable distance) was used to verify the measurements. All links use the same hardware for genuine comparison of the results.

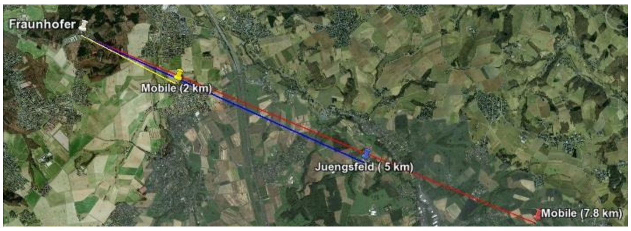

Fig 2: Long-Distance Links for model Validation

The measurements were conducted using outdoor capable routers (Intel Atom N2800 cpu) equipped with four Atheros AR9220 based high power wireless cards (MikroTik RH52n). As OS, Debian 6.0 and a modified Linux kernel based on the revision 3.7.10 were chosen. Our modifications in the 802.11 MAC of this kernel enabled us to adjust the timings to match the propagation delay. Additionally DCF parameters such as CW_{min} and the maximum number of retries of a packet could be modified. To provide sufficient SNR even for higher MCS, directional antennas with a gain of 24 dBi were used for all long-distance links.

The model and the measurement system to compare the calculated results and the measurements were implemented. The throughput and the delay measurements were carried out using a self-developed tool called 80211Analyzer and the well-known packet generator mgen. The Analyzer evaluates packets at the receiver using a virtual 802.11 device in monitor mode, permitting access to relevant statistics.

To accurately measure the delay across the two distributed systems, a hardware clock sync using the Precision Time Protocol (PTP) was executed before each measurement. Additionally, a method was developed to compensate for the linear clock drift by quantifying the time variation before and after each measurement. This method results in nearly constant and known time deviation. Since the measurements used bi-directional traffic and the interesting variable is the delay of packets independent of their origin on the link, the mean value for the combined measured delay from each station provided sufficient accuracy to evaluate the delay of the model.

On the network layer, IP and UDP were chosen to avoid the known behavior of TCP [12] on erroneous wireless links. Nearly 5000 different measurements were conducted (with a duration of 30 s) each for:

- different distances (0 km ,2.5 km,5 km,7.5km)
- all available MCS (0-7 SISO and 8-15 MIMO)
- two different payload sizes (100 and 1450 Byte)
- all possible values for CW_{min}(1,3,..,255)
- and retries (0-7).

To evaluate the accuracy of the model, a single value for the delay and the throughput was used:

$$T|D_{Diff}(d,R,CW,MCS,agg) = \left|\frac{Measure - Model}{Measure}\right| \quad (12)$$

To provide an aggraded view on different dependencies, the numerous calculated diff values for all link properties were combined using the mean or the median function.

The first point of interest is the accuracy of the developed modeling approach depending on the link distances as shown in Figure 3.

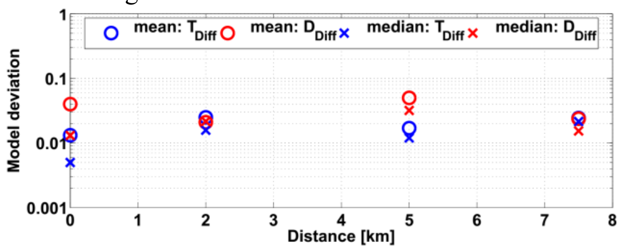

Fig 3: Model deviation at different distances, showing mean and median differences between model and measurements in throughput (T) and delay (D)

As Figure 3 shows, the deviation between model and measurement is independent of the link distance. Additionally, the difference at all distances is around 0.02 for the throughput and slightly higher for the delay. Since the delay calculation directly relies on the throughput calculation, the higher deviation is expected. This independency of the model accuracy was evaluated for different MCS, CW_{min} values and the number of retries.

However, for higher A-MPDU aggregation factors, the accuracy of the delay decreases to 0.1. This decrease is also responsible for the difference between the mean and the median. This higher deviation derives from the delay calculation since higher aggregation factors require a higher buffer, which multiplies with a calculated error for the access delay (cf. eq 10). Nevertheless, the introduced model extensions provide sufficient accuracy especially under the premise that they are verified against real measurements instead of simulations, which are usually used in this field of study [6] [9].

6 Estimation and Optimization

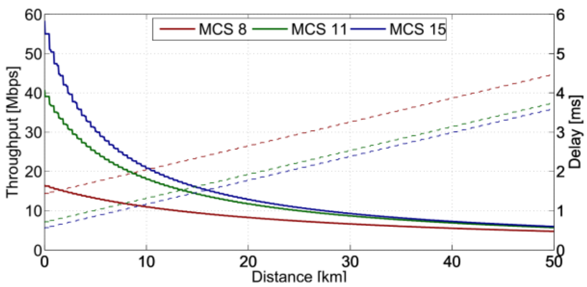

Fig 4: Estimated throughput and delay of two stations. 20 MHz, 1450 Byte Payload, no aggregation

This successfully verified model can be used in many ways. Figure 4 shows the estimated throughput and delay of two stations with bi-directional traffic in contention using just the physical enhancements of 802.11n for distances up to 50 km. While the delay increases nearly linearly and moderately, the throughput decreases significantly to a nearly constant value for all contemplated MCS. This substantial decrease is attributed to the increased back-off in comparison to the actual transmission time of the frame and ACK.

The key factor in enabling high throughput over large distances is the MAC layer aggregation of 802.11n as shown in Figure 5.

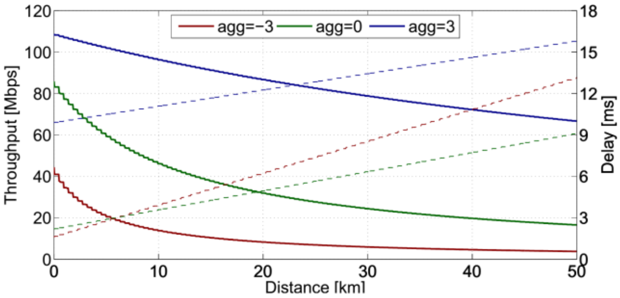

Fig 5: Estimated throughput and delay of two stations. 20 MHz, 1450 Byte Payload, MCS 15

However, using high aggregation factors introduces additional delay, which provides a trade-off in terms of QoS. A WiLDs' designer needs to be aware of this trade-off especially in multi-hop scenarios.

An interesting dependency especially for long-distance links is the influence of the BER on the throughput and the delay as exemplarily shown in Figure 6.

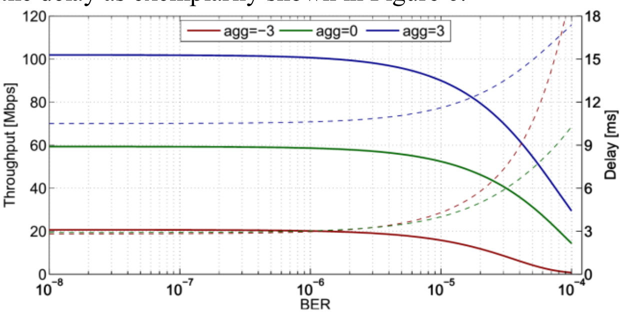

Fig 6: Estimated throughput and delay for different BER. 20 MHz, 1450 Byte Payload, 5 km, MCS 15

Starting at a BER of 10^{-6}, the throughput decreases significantly. This effect is even stronger on long-distance links since, besides the needed retransmission, the exponential back-off scheme doubles CW_{min}, leading to more idle time on the medium. The delay suffers also from the reordering time at the receiver, leading to the significant delay increase for erroneous links.

Since the IEEE does not define the standard [4] under the premise of WiLDs, potential optimization of the MAC parameters can be calculated using the developed modeling approach. Instead of the default (15) for CW_{min}, a decreased value can reduce the back-off. This decrease leads to an increased collision probability, which should be avoided especially for high aggregation factors. The default (7) for retrying a frame has a positive influence on the throughput but leads to additional delay.

The goal of the following optimization is to find the optimum combination of these parameters to minimize the delay and maximize the throughput alike. Since this is a multi-objective problem, additional effort is needed to find the Pareto optimal solutions. A single function is used to scalarize the multi-objective problem:

$$\textbf{maximize}\left(\sqrt{\left(\frac{F}{D_i/D_{min}}\right)^2+\left(\frac{S_i}{S_{max}}\right)^2}\right) \qquad (13)$$

Instead of minimizing the delay, the normalized and reciprocal is used. In combination with the normalized throughput, this scalarization leads to a single value, which can be maximized using common techniques. To normalize the values of delay and throughput, both the maximum throughput (S_{max}) and the minimum delay (D_{min}) are calculated for each set of input variables beforehand.

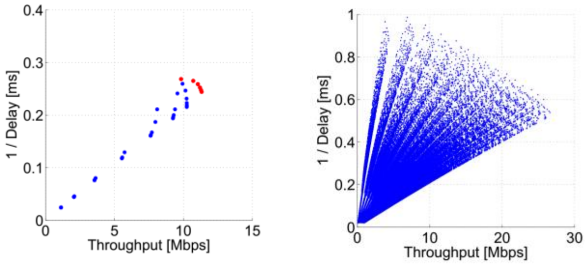

Fig 7:a) single subset b) single MCS

Figure 7a provides a view on a single input set before normalization. Each point is associated with a pair of CW_{min} and R. The marked points in Figure 7a represent the Pareto optimal solution, which can be seen as a graphical explanation of equation 13. A factor (F) is included in the scalarization function to weight the influence of the delay to the solution. In this work, this factor was chosen equal to one, and additional research is needed to study its influence.

The input variables for the optimization process consist of the MCS (0-15), the payload size, the aggregation factors, and the distance up to 20 km. The result of this process are the optimum values for CW_{min} and R maximizing equation 13.

This optimization's first unexpected result is that the optimal number of retries (R) is found at a fixed value of one (a single chance to retry a frame) for all possible combinations of input parameters. While the possible throughput increase for additional chances of a retransmission is low, the increase in delay reduces this parameter to this optimum value.

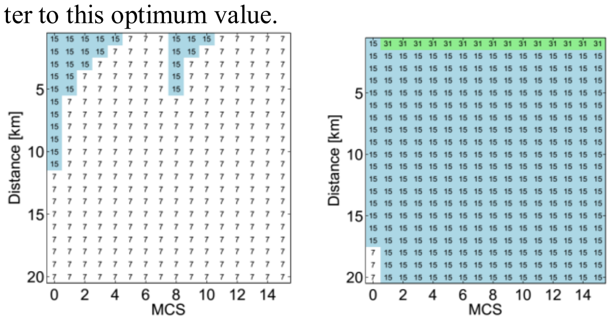

Fig 8a/b: Optimal CW_{min} value for agg=-2 and agg=3 CW_{min}; green=31, blue=15, white=7

For CW_{min} the situation is more complex. Figure 8 provides a view on the optimum CW_{min} value for an aggregation factor of minus two (Fig. 8a) and three (Fig. 8b) dependent on the link distance and the MCS with 20 MHz bandwidth and 750 Byte payload.
For the aggregation factor of minus two, the optimum CW_{min} value is primarily found at 7. For longer distances, higher MCS or both, a CW_{min} value of 7 is the optimal value. For higher MCS, the transmission time of the packets is less (compared to lower MCS) and the influence of the back-off time rises, increasing the idle time on the medium and decreasing the throughput. A collision of a packet is more acceptable in comparison to lower MCS. The same justification applies to the influence of the distance. At lower distances, the idle time on the medium during the back-off is less, and the throughput reduction due to possible collisions increases. This effect can be avoided by using a CW_{min} of 15. For a high aggregation factor of three (Fig 8b) a value of 15 defines the optimum for nearly all combinations, due to the increased A-MPDU packet length.

7 Conclusion

This work provides extensions to the well-known DCF modeling approach by Bianchi [6] to estimate 802.11n long-distance point-to-point to links. An additional contribution to the state-of-the-art is the extensive verification of the model on real deployments instead of on simulations. Designers of WiLD networks can use this model to estimate the properties of the network beforehand, thus enabling a more accurate network planning.
The conducted optimization process shows that an optimal value for CW_{min} and the number of retries exists depended on the properties of the link.

7.1 Future work

The optimization process can be enriched by the buffer-sizes and potential loss. Especially when using TCP as transport protocol, the loss should be included to obtain the optimum MAC parameters. The authors have already started with the inclusion of different traffic classes in the approach. This addition is especially important for prioritizing different services in the network by using the EDCA MAC layer function. With knowledge about traffic patterns occurring on the network, a shift in up- and downlink capacity can be considered by using varying MAC parameters on each station. The results of this work can be used to compare the DCF to different MAC layers approaches such as a FDD or a token-based system for long-distance links.

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8 References

[1] D. Henkel, S. Englander, M. Kretschmer und C. Niephaus, "Connecting the unconnected; Economic constraints and technical requirements towards a back-haul network for rural areas," in GLOBECOM Workshops, 2011.

[2] B. Raman und K. Chebrolu, "Experiences in using WiFi for rural internet in India," IEEE Communications Magazine, Bd. 45, No. 1, pp. 104-110, 2007.

[3] Á. Rendón, A. Martínez und M. Dulcey, "Rural Telemedicine Infrastructure and Services in the Department of Cauca, Colombia," in Telemedicine and e-Healt, 2005.

[4] "IEEE Standard for Information technology-Telecommunications and information exchange between systems Local and metropolitan area networks-Specific requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications,“ IEEE Std 802.11-2012, pp. 1-2793, 2012.

[5] M. Rademacher, M. Kretschmer und K. Jonas, "Exploiting IEEE802.11n MIMO Technology for Cost-Effective Broadband Back-Hauling,“ Fifth International IEEE EAI Conference on e-Infrastructure and eServices for Developing Countries, 2013.

[6] G. Bianchi und I. Tinnirello, "Remarks on IEEE 802.11 DCF performance analysis,“ IEEE Communications Letters, No. 8, pp. 765-767, 2005.

[7] C. Foh und J. Tantra, "Comments on IEEE802.11 saturation throughput analysis with freezing backoff counters,“ IEEE Communication Letters, No. 2, p. 2005.

[8] S. Reigadas, A. Martinez-Fernandez, J. Ramos-Lopez und J. Seoane-Pascual, "Modeling and Optimizing {IEEE} 802.11 DCF for Long-Distance Links,“ IEEE Transactions on Mobile Computing, No. 6, pp. 881-896, 2010.

[9] B. Bing, Emerging Technologies in Wireless LANs - Theory, Design, and Deployment, Cambridge: Cambridge University Press, 2008.

[10] K. Tan, D. Wu, A. Chan und P. Mohapatra, "Comparing simulation tools and experimental testbeds for wireless mesh networks," IEEE International Sumposium on a World of Wireless Mobile and Multimedia Networks, pp. 1-9, 2010.

[11] E. Ziouva und T. Antonakopoulos, "Reprint CSMA/CA Performance under High Traffic Conditions: Throughput and Delay analysis," 2012.

[12] C. Lüders, Lokale Funknetze - Wireless LANs (IEEE 802.11), Bluetooth, DECT, Würzburg: Vogel Verlag Und Druck, 2007.

[13] Y. Lin und V. Wong, "Frame aggregation and optimal frame size adaption for IEEE802.11n WLANs," Global Telecommunications Conference, pp. 1-6, 2006.

[14] M. Kretschmer, T. Horstmann, P. Batroff, M. Rademacher und G. Ghinea, "Link calibration and property estimation in self-managed wireless back-haul networks,“ Asia-Pacific Conference on Communications, pp. 232-237, 2012.