Analyzing the Effective Throughput in Multi-Hop IEEE 802.11n Networks

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Abstract—**In this paper we characterize the effective throughput for multi-hop paths in IEEE 802.11n based wireless mesh networks. We derive an analytical model capturing the effects of frame aggregation and block acknowledgements, features found in the new IEEE 802.11n standard. We describe the throughput at MAC layer as a function of bit error rate, aggregation level and path length. Using the results of our model as an upper bound, we show that current TCP implementations do not harness the bandwidth provided by the IEEE 802.11n MAC layer. Subsequently we introduce collision induced rate control which uses cross layer feedback to effectively estimate the available bandwidth based on our analytical model. We conduct a comprehensive performance study, showing that results from our analytical model are closely matched by simulation results. Furthermore we show that the proposed protocol achieves up to 50% more goodput and up to 70% less frame collisions compared to TCP-SACK and TCP-NewReno.**

Keywords: IEEE 802.11 wireless networks, Analysis and design of transport protocols, Cross layer design

I. INTRODUCTION

With the emergence of the IEEE 802.11 standard wireless mesh networks have experienced a rapid growth and have drawn increased interest from academia and industry. Mesh networks can be built with relatively low infrastructure expenditure compared to wired broadband. Therefore, they are particularly attractive for providing cost-efficient Internet access for suburban areas with little or no broadband availability. Further areas of application include ubiquitous Internet access for mobile users in metropolitan areas. Despite recent advances in wireless mesh networking, carrier grade service quality in wireless mesh networks, still is an active research area.

The IEEE 802.11n standard [1] is the first IEEE 802.11 standard [2] to introduce a MIMO-based physical layer, providing data rates up to 600Mbit/s and increased tolerance to interference. These features make IEEE 802.11n a promising technology for building carrier grade wireless mesh networks. The high data rates provided by the IEEE 802.11n physical layer can only be harnessed at upper layers if medium access is efficient. Therefore, IEEE 802.11n introduces frame aggregation. Using frame aggregation multiple subframes can be transmitted in sequence, with the overhead for medium access arising only once. Each subframe transmitted in an aggregated frame possesses an own CRC checksum, i.e. the IEEE 802.11n MAC at the receiver can extract individual subframes even if parts of the aggregated frame are erroneous due to lossy channel conditions. Upon reception of an aggregated frame, the receiver can send a BlockAck control frame to acknowledge all correctly received subframes.

For the design of data transport protocols, an understanding of the effective throughput of IEEE 802.11n is required. An appropriate analytical model has to consider the aggregation capabilities as well as partial retransmits occurring in IEEE 802.11n. Furthermore the characteristics of the wireless channel as a scarce resource shared among all mesh nodes within their radio transmission range has to be considered.

In this paper we characterize the effective throughput for multi-hop paths in IEEE 802.11n based wireless mesh networks. We derive an analytical model capturing the effects of frame aggregation and block acknowledgements, features found in the new IEEE 802.11n standard. We describe the throughput at MAC layer as a function of bit error rate, aggregation level and path length. Using the results of our model as an upper bound, we show that current TCP implementations do not harness the bandwidth provided by the IEEE 802.11n MAC layer. Subsequently we introduce collision induced rate control which uses cross layer feedback to effectively estimate the available bandwidth based on our analytical model. We conduct a comprehensive performance study, showing that results from our analytical model are closely matched by simulation results. Furthermore we show that the proposed protocol achieves up to 50% more goodput and up to 70% less frame collisions compared to TCP-SACK and TCP-NewReno.

The remainder of this paper is organized as follows. Section II summarizes related work on analytical models for IEEE 802.11n and data transport optimization for wireless mesh networks. In Section III, we introduce our analytical model, in Section IV the protocol design proposal is described. The analytical model and the protocol design proposal are evaluated in section V. Finally, concluding remarks are given.

II. RELATED WORK

Banchs et al. [3] presented an analytical model for IEEE 802.11e EDCA function. Bianchi et al. [4] presented a model for the throughput of IEEE 802.11 single-hop links under ideal channel conditions and saturated traffic, i.e. each node always has a frame available for transmission. Li et al. [5] proposed an analytical model assuming saturated traffic. They derived the effective throughput, optimal frame and fragment sizes for single-hop links. Opposed to [3], [4] and [5], we consider the effective throughput over multi-hop paths. We do not rely on the assumption of saturated traffic. Such an assumption is too optimistic in case of multi-hop paths, where packets are forwarded by intermediate nodes subject to the spatial reuse constraint of the wireless medium.

Chen et al. [6] characterized the bandwidth delay product for wireless multi-hop paths. They obtained a qualitative upper bound for the throughput considering spatial reuse constraint of the medium. Fu et al. [7] analyzed the effect of pacing for multi-hop wireless networks and proposed a pacing scheme at MAC layer. Opposed to [6] and [7], we obtain quantitative results considering the frame aggregation and BlockAck feature of IEEE 802.11n MAC layer in presence of lossy channel conditions.

Shrivastava et al. [8] analyzed the performance of IEEE 802.11n over single-hop links in a testbed. Karlsson et al. [9] reported on the performance of TCP in the presence of packet aggregation. Opposed to [9] and [8], we provide an analytical model considering the frame aggregation capabilities of IEEE 802.11n.

In [10], we introduced TCP-AP for multi-hop wireless networks. TCP-AP uses adaptive pacing under the assumption of a four-hop interference range. Opposed to [10], we do not rely on the assumption of a static interference range. We provide an estimation scheme which uses cross layer information to estimate the interference range.

Sundaresan et al. [11] proposed ATP, a reliable transport protocol for ad-hoc networks. ATP uses a rate-based transmission scheme based on network feedback similar to ATM. Opposed to [11], we propose an estimation scheme which differentiates between losses due to random bit errors and losses due to frame collision, i.e. congestion. Further, we consider the MAC layer features of IEEE 802.11n.

Bruno et al. [12] introduced an optimized MAC scheme to increase TCP throughput over single-hop links. Kim et al. [13] proposed a modification of the IEEE 802.11 MAC to allow aggregation of unicast and broadcast frames. Vacirca et al. [14] addressed the problem how to design a MAC scheme optimized for TCP. Opposed to [12], [13] and [14], our focus is on the design of a transport layer protocol, not on MAC layer design. Bhandarkar et al. [15] proposed TCP-DCR, a modification at the TCP receiver to cope with lossy wireless channels. Opposed to [15], we propose a sender-side pacing scheme with cross layer rate estimation. Acharya et al. [16] proposed WOOF, an optimized rate adaption scheme for WLAN. Razafindralambo et al. [17] introduced PAS, a dynamic packet aggregation scheme. Opposed to [16] and [17], we consider rate adaption for IEEE 802.11n multi-hop wireless networks.

III. ANALYTICAL MODEL

A. Model Assumptions

To characterize the effective throughput for multi-hop chains, we consider an arbitrary unicast routing protocol and a single traffic flow from one source to one sink. We consider a physical layer bandwidth of B_{DATA} and a single-channel, single-radio configuration. Consistent with the IEEE 802.11n standard [1] (A-MPDU and HT Immediate BlockAck Extension, subclause 9.10.7), we assume that up to N subframes can be transmitted in an aggregated frame, with the overhead (i.e. backoff interval, physical header) for medium access arising only once. Upon reception of an aggregated frame, the receiver sends a BlockAck response, where each correctly received subframe is acknowledged. Upon reception of a BlockAck the sender retransmits all subframes which have not been acknowledged. This procedure is repeated until r_{max} transmission attempts have been made or until all subframes have been acknowledged.

To capture the characteristics of the wireless channel as a shared resource, we assume that two nodes can transmit simultaneously (i.e. without collisions at the receiving nodes due to the hidden terminal problem) if their distance is at least d_{coll} hops. Therefore, the traffic at intermediate nodes is not saturated and the sender has no new subframes available when subframes have to be retransmitted.

TABLE I. MODEL PARAMETERS

Symbol	Description
s	Size of a subframe (in bits)
b_{err}	Bit error rate
p_f	Subframe loss rate
N	Number of subframes per aggregated frame
r_{max}	Max. number of transmission attempts
T_{slot}	Slot time
cw_{min}, cw_{max}	Min. and max. size of contention window
T_{const}	$T_{DIFS} + T_{PHY} + T_{SIFS} + T_{ACK}$
T_{PHY}	Time to transmit physical header
B_{DATA}	Physical layer bandwidth

B. Model Description

We assume that a received frame is discarded by the MAC layer if it contains at least one bit error. Note that this assumption is valid for random distributed bit errors and Cylic Redundancy Check performed by the IEEE 802.11 MAC layer. Assuming a bit error rate b_{err} that accounts for random bit errors of the wireless channel and a subframe size s the probability p_f that a subframe is lost is given by

$$p_f = 1-(1-b_{err})^s \tag{1}$$

Random variable S_N denotes the number of lost subframes in an aggregated frame of size N. Assuming independent losses, S_N follows a binomial distribution $b(N, p_f)$. The probability that k out of N subframes are lost is given by

$$P[S_N = k] = \binom{N}{k} p_f^{\,k}(1-p_f)^{N-k} \tag{2}$$

Therefore, the expected number of lost subframes is

$$E[S_N] = Np_f \tag{3}$$

Next, we derive the probability that all subframes of an aggregated frame are transmitted with l transmission attempts. The number of remaining subframes after each transmission attempt can be modeled by the acyclic discrete time Markov chain (DTMC) depicted in Fig. 1. A state (i_1,i_2) of this DTMC defines that i_1 transmission attempts have been made and i_2 subframes are left to be transmitted. The DTMC resides at time 0 in its initial state $(0,N)$ with probability 1, since no transmission attempt has been made and N subframes are left to be transmitted. Note that the states $(1,0),\dots,(r_{max},0)$ are absorbing states, because all subframes have been submitted. Furthermore $(r_{max},1),\dots,(r_{max},N)$ are absorbing states of this DTMC because the number of maximum retransmission attempts has been reached. Given state (i_1,i_2) there are i_2+1 possible transitions, due to the fact, that all i_2 subframes could be transmitted successfully or $1,\dots,i_2$ subframes could be lost. A transition from state (i_1,i_2) to state (j_1,j_2) can occur if $j_1=i_1+1$ and $j_2 \le i_2$. The transition probability is the probability that j_2 out of i_2 subframes are lost and can be expressed in the probability matrix

$$\mathbf{P} = [p_{ij}] = P\{X_{n+1} = j \mid X_n = i\} \tag{4}$$

where $i=(i_1,i_2)$ and $j=(j_1,j_2)$. The state transition probability of the matrix is given by

$$p_{ij} = \begin{cases} \binom{i_2}{j_2} p_f^{\,j_2} (1-p_f)^{i_2-j_2} & j_1 = i_1+1 \wedge 0 \le j_2 \le i_2 \\ 0 & \text{else} \end{cases} \tag{5}$$

For this DTMC, the initial distribution π^0 is given by

$$\pi_i^0 = \begin{cases} 1, \text{ if i} = (0,N) \\ 0, \text{ else} \end{cases} \tag{6}$$

With this initial distribution and probability matrix $\mathbf{P}$ we get the time-dependent distribution π^n after n steps by $\pi^n=\pi^0\mathbf{P}^n$.

To calculate the probability that exactly l transmission attempts are needed, we define

$$P_{att}(l) = \begin{cases} \pi_i^l, \; i=(l,0) \text{ for } 0<l<r_{\max} \\ \sum_{q=0}^{N} \pi_{i_q}^l, \; i_q = (r_{\max},q) \text{ for } l = r_{\max} \end{cases} \tag{7}$$

The probability of being in $(l,0)$ after l transition attempts is π_i^l where $i=(l,0)$. If $l=r_{max}$ we sum up all probabilities for being in (r_{max},q), because for the time needed it is irrelevant, if this transmission attempt is successful or not.

Let $C(k)$ denote the amount of time required to transmit one aggregated frame in exactly k rounds. $C(k)$ splits into a term, needed in every transmission and a term, dependent on the actual retransmit attempt

$$C(k) = \begin{cases} 0 & \text{, k=0} \\ \dfrac{\min(2^{k-1} cw_{min}, cw_{max})}{2} T_{slot} + \dfrac{sNp_f^{\,k-1}}{B_{DATA}} & \text{, else} \\ +T_{const} + C(k-1) & \end{cases} \tag{8}$$

The expected amount of time T_{onehop} to transmit one aggregated frame (including time for retransmissions) over one hop is given by

$$E[T_{onehop}] = \sum_{k=1}^{r_{max}} P_{att}(k) C(k) \tag{9}$$

Let $size(N,s)$ denote the size of an aggregated frame in bit with N subframes each of size s. Given a multi-hop path of length h hops, two nodes can transmit simultaneously (i.e. without collisions due to interfering radio signals) with a rate w_{max} when their distance is at least d_{coll} hops

$$w_{\max} = \frac{size(N,s)}{\min(d_{coll},h) * E[T_{onehop}]} \tag{10}$$

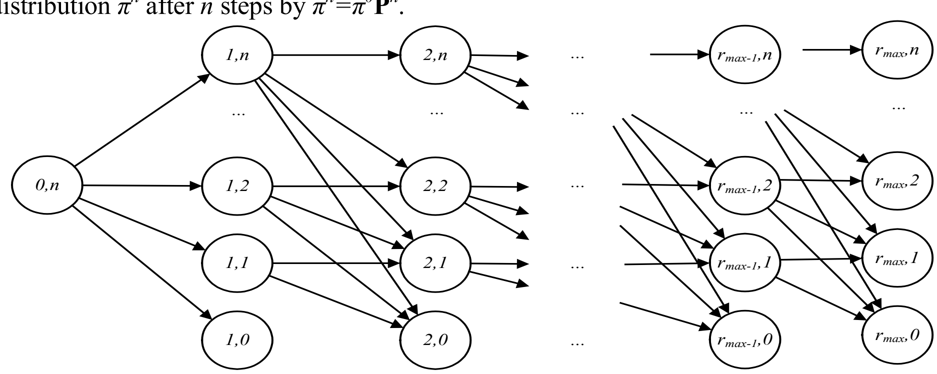

Figure 1. State transition diagram of the acyclic DTMC defining the pmf for $P_{att}(l)$.

IV. Cross Layer Design with Collision Induced Rate Control

Based on the estimation scheme for $P_{att}(k)$ from our analytical model, we propose a collision induced rate control (CIRC) reducing frame collisions and exploiting the frame aggregation capabilities of the IEEE 802.11n MAC layer.

Because the wireless channel is a shared resource, simultaneous transmissions may result in collisions at receiving nodes. Given an interference range of d_{coll} hops, node i can only transmit a packet successfully as soon as node $(i+d_{coll})$ has finished its transmission. For our analytical model we assumed d_{coll} to be known at the source node, as we wanted to obtain an upper bound. However d_{coll} is not known in reality.

To address this problem we use cross layer feedback to estimate the available bandwidth based on our analytical model. The basic idea of CIRC is to identify collisions by comparing the expected number of retransmissions to the actual number of retransmissions as reported by the MAC layer. If the reported number is significantly higher, this is an indication of collisions, as can be observed in Fig. 5. The expected number of retransmissions can be derived using P_{att} as defined in (7) of our analytical model. The rate control is divided into two phases. In exponential phase the algorithm tries to quickly increase the rate according to a factor γ to find an initial estimation of the sending rate. Then a linear phase is entered, to approximate the optimal rate.

To harness the bandwidth provided by the IEEE 802.11n MAC layer the frame aggregation capabilities must be taken into account by upper layers. Hence, upper layer protocols are required to send their data in a way, that it can be aggregated at MAC layer. The proposed protocol maximizes aggregation by monitoring the MAC queue and filling it in bursts to the maximum frame size allowed by IEEE 802.11n. To estimate the rate of these burst transmissions, the sender uses the function *rate_control*, as shown in Fig. 2. To provide reliable data transport the selective ACK scheme of TCP-SACK is adopted.

To enable MAC layer feedback, we propose a minor modification. Therefore, after each final aggregated frame transmission *update_stats* is called. To estimate the subframe error rate p_f and number of retransmitted subframes $R_{current}$, the average percentage of lost subframes is reported based on the received BlockACKs. The maximum of the nodes estimation of these values and the received values from the nodes neighbors is propagated through MAC ACKs. The sender makes use of these two indicators as follows. It first computes P_{att} using p_{avg}. The expected number of retransmitted subframes $R_{expected}$ can then be computed by weighting $P_{att}(k)$ with the expected number of transmitted subframes in the k-th transmission attempt. The expected and actual numbers are compared to a threshold β to determine the level of contention and to adjust the rate accordingly. To mitigate random short term fluctuation of the measured loss rate and retransmits an exponential weighted moving average with smoothing factor α is used.

Sender:

proc rate_control
$R_{current}$ = #retransmitted subframes from MAC
$R_{avg} = \alpha * R_{avg} + (1-\alpha) * R_{current}$
p_f = estimated loss rate from MAC
$p_{avg} = \alpha * p_{avg} + (1-\alpha) * p_f$
$R_{expected}$ = expected # of retransmitted subframes
IF (linear phase)
  IF ($R_{avg} / R_{expected} > \beta$)
    Decrease rate
  ELSE
    Increase rate
ELSE
  IF ($R_{avg} / R_{expected} > \beta$)
    Enter linear phase
    $rate = rate_{laststable}$
  ELSE
    $rate = rate * \gamma$
    $rate_{laststable} = rate$

Forwarding nodes:

proc update_stats
R_{count}=#of retransmitted subframes
$R_{node} = \alpha * R_{node} + (1-\alpha) * R_{count}$
p_{count}=percentage of erroneous subframes in BlockACKs
$p_{node} = \alpha * p_{node} + (1-\alpha) * p_{count}$
$R_{current} = \max(R_{neighbors}, R_{node})$
$p_f = \max(p_{neighbors}, p_{node})$

Figure 2. Algorithmic description of CIRC.

V. Quantitative Evaluation

A. Simulation Setup

We implemented the IEEE 802.11n frame aggregation and block acknowledgement scheme as extension of the normal 802.11 MAC layer for the ns-2 [18] network simulator. As routing protocol AODV is used. All simulation parameters are set according to Table II. In ns-2, all link-layer parameters of IEEE 802.11 are configured to provide a transmission range of 250m and a carrier sensing range as well as an interference range of 550m. The RTS/CTS handshake is disabled and we consider a channel bandwidth of 300 Mbit/s according to IEEE 802.11n while setting the size of transport data packets to 1,460 bytes.

Consistent with the analytical model we consider an equally spaced multi-hop chain comprising $h+1$ nodes (h hops) with a single FTP flow and a 200m inter-node distance. TCP packets traverse along the chain from the leftmost node (i.e., the source) to the rightmost node (i.e., the destination). Nodes in the chain are positioned such that only direct neighbors can communicate with each other over one hop.

We conduct steady-state simulations starting with an initially idle system. In each run, we utilize FTP connections until 55,000 packets are successfully transmitted, and split the output of the experiment in 11 batches, each 5,000 packets in size. The first batch is discarded as initial transient. The considered performance measures are derived from the remaining 10 batches with 95% confidence intervals by the batch means method.

B. *Model Validation*

To validate the proposed analytical model we compare the results from the model to the maximum throughput achievable by optimized paced UDP (meaning that the sender always transmits a fully aggregated frame). As we can observe in Fig. 3, the analytical model can serve as a good upper bound for the maximum achievable throughput on a multi-hop path without as well as with bit errors. Note that with the existence of bit errors the maximum achievable goodput is slightly lower in ns-2 due to the lack of perfect synchronization as assumed by our model.

In a second experiment we examined the mean number of retransmissions needed to successfully transfer a frame. We compared the simulation results with those obtained from our Markov model. It can be seen in Fig. 4 that the results from our Markov model are indeed very close to simulation results.

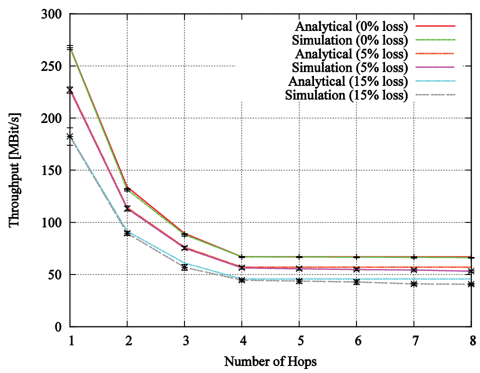

Figure 3. Validation of the analytical model for different subframe loss rates with respect to throughput.

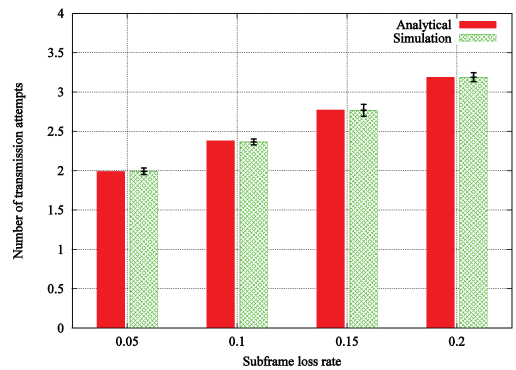

Figure 4. Validation of the analytical model with respect to number of transmissions.

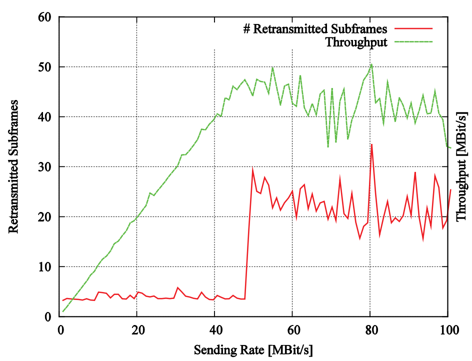

Figure 5. Retransmitted subframes as collision indicator, subframe loss rate 5% and path length 6 hops.

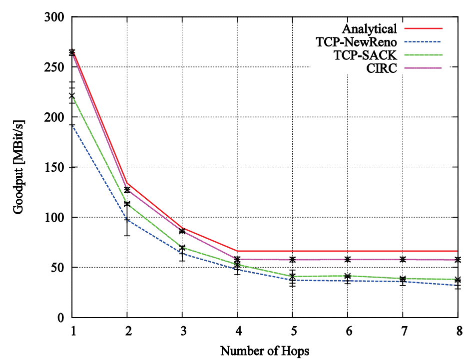

Figure 6. Goodput for chain topology, subframe loss rate 0%.

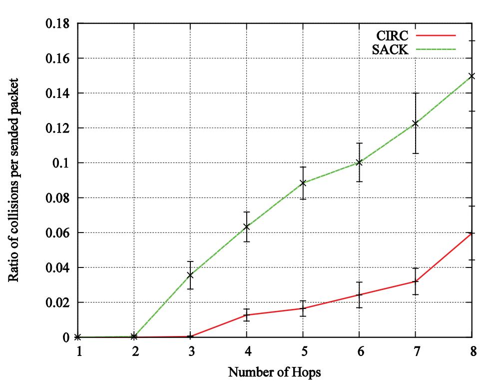

Figure 7. Number of Collisions for chain topology.

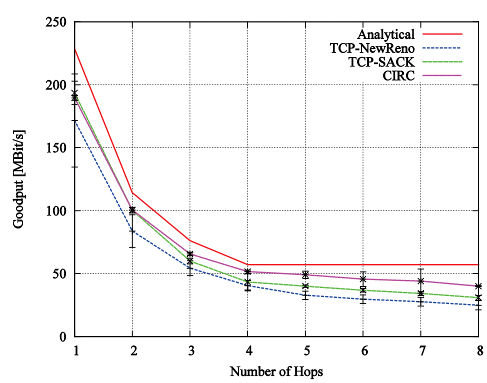

Figure 8. Goodput for chain topology, subframe loss rate 5%.

TABLE II. SIMULATION PARAMETERS

Symbol	*Value*
s	12272 bit (1534 Byte=1460 Byte payload + headers)
p_f	variable
N	42 (max. number of subframes of size 1500 Byte)
r_{max}	7
T_{slot}	9µs
cw_{min}	16
cw_{max}	1024
T_{SIFS}	16µs
T_{DIFS}	34µs
T_{ACK}	20.75µs
T_{PHY}	20µs
B_{DATA}	300 Mbit/s
α	0.75
β	2
γ	1.5

C. Evaluation of Protocol Design Proposal

To evaluate the performance of the proposed protocol we compare the goodput obtained by our protocol to the results from the analytical model, TCP-NewReno and TCP-SACK (TCP with selective acknowledgements) as recommended by [19]. To validate our assumptions we analyzed the average number of retransmits dependent on the sending rate. As can be seen in Fig. 5 this number stays constant up to a certain sending rate and then suddenly rises due to frame collisions. In Fig. 6 we observe that in the absence of bit errors our protocol achieves up to 80% more goodput than TCP-NewReno and up to 50% more than TCP-SACK. According to Fig. 8 we achieve in the presence of random bit errors up to 60% more goodput than TCP-NewReno and up to 30% more goodput than TCP-SACK. Furthermore we examined the amount of collisions on MAC layer. In Fig. 7 we observe that TCP-SACK leads to up to 300% more collisions per transmitted packet than CIRC.

VI. CONCLUSION

In this paper we characterized the effective throughput for multi-hop paths in IEEE 802.11n based wireless mesh networks. We derived an analytical model capturing the effects of frame aggregation and block acknowledgements and gave a protocol design proposal. We showed that our protocol proposal achieves up to 50% more goodput and up to 70% less frame collisions compared to TCP-SACK and TCP-NewReno.

Currently we are implementing a Linux user space prototype of the proposed protocol in our IEEE 802.11n mesh testbed. Further areas of research include the extension of our analytical model for multiple flows. We also plan to validate the model with testbed measurements.

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