Frame Aggregation and Optimal Frame Size Adaptation for IEEE 802.11n WLANs

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***Abstract*— The IEEE 802.11a/b/g have been widely accepted as the *de facto* standards for wireless local area networks (WLANs). The recent IEEE 802.11n proposals aim at providing a physical layer transmission rate of up to 600 Mbps. However, to fully utilize this high data rate, the current IEEE 802.11 medium access control (MAC) needs to be enhanced. In this paper, we investigate the performance improvement of the MAC protocol by using the two frame aggregation techniques, namely A-MPDU (MAC Protocol Data Unit Aggregation) and A-MSDU (MAC Service Data Unit Aggregation). We first propose an analytical model to study the performance under uni-directional and bi-directional data transfer. Our proposed model incorporates packet loss either from collisions or channel errors. Comparison with simulation results show that the model is accurate in predicting the network throughput. We also propose an optimal frame size adaptation algorithm with A-MSDU under error-prone channels. Simulation results show that the network throughput performance is significant improved when compared with both randomized and fixed frame aggregation algorithms.**

I. Introduction

With the successful deployment of IEEE 802.11a/b/g wireless local area networks (WLANs) and the increasing demand for real-time applications over wireless, the IEEE 802.11n Working Group is standardizing a new Medium Access Control (MAC) and Physical Layer (PHY) specification [1]–[3]. At the PHY layer, 802.11n will use MIMO (multiple-input multiple-output) and OFDM (orthogonal frequency division multiplexing) to increase the bit rate to be up to 600 Mbps.

The throughput performance at the MAC layer can be improved by aggregating several frames before transmission. Frame aggregation not only reduces the transmission time for preamble and frame headers, but also reduces the waiting time during CSMA/CA (Carrier Sense Multiple Access - Collision Avoidance) random backoff period for successive frame transmissions. The frame aggregation can be performed within different sub-layers. In 802.11n [1]–[3], frame aggregation can be performed either by MAC Protocol Data Unit Aggregation (A-MPDU) or MAC Service Data Unit Aggregation (A-MSDU).

Although frame aggregation can increase the throughput at the MAC layer under ideal channel conditions, a larger aggregated frame will cause each station to wait longer before its next chance for channel access. Thus, there is a tradeoff between throughput and delay for frame aggregation at the MAC layer. Furthermore, under error-prone channels, corrupting a large aggregated frame may waste a long period of channel time and lead to a lower MAC efficiency. As a result, there is a need to study the effects of channel conditions on the chosen frame aggregation schemes.

We now describe some related work on the analytical modeling of the MAC layer performance. Bianchi [4] proposes a two-dimensional Markov chain to determine the saturation throughput of WLAN using the distributed coordination function (DCF). Cali *et al.* [5] approximate the DCF function as a p-consistent CSMA protocol and propose an adaptive mechanism for adjusting the contention window sizes to achieve an optimal throughput. Tay and Chua [6] propose a model based on average value analysis and study the effects of contention window sizes on the throughput performance. Liu and Stephens [7] study the bidirectional frame aggregation in 802.11 DCF by extending Tay and Chua's model. All of the above models assume an ideal wireless channel with no channel errors.

Wireless channels are usually error-prone and the effects of packet errors have an impact on the system performance. Several papers [8]–[12] extend the above system models to study the throughput performance under different channel error conditions. Yin *et al.* [13] study the effects of packet size in an error-prone channel for IEEE 802.11 DCF and conclude that there is an optimal packet size under a certain bit error rate (BER) to achieve the maximum throughput. Ci and Sharif [14] propose an optimal frame size predictor based on Kalman filter to maintain a committed goodput. Most of these studies assume that a single bit error can corrupt the whole frame. This assumption may not be true for the 802.11n MAC layer with frame aggregation.

In this paper, we provide a unified approach to study the saturation throughput and delay performance of the proposed frame aggregation schemes in the new 802.11n proposals under error-prone channels. Both uni-directional and bi-directional transfer are being considered. The analytical model provides an accurate prediction for system performance which is validated by extensive simulation experiments. Based on the analysis, we propose an optimal frame size adaptation algorithm. Simulation results show that there are significant throughput improvements when compared with the randomized and fixed frame aggregation algorithms. These studies serve as the basis for further optimization of the system parameters in the 802.11n WLANs.

The rest of this paper is organized as follows. Section II provides an overview of 802.11n frame aggregation techniques.

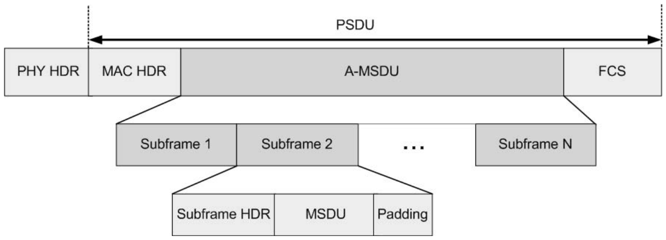

Fig. 1. Frame Format for A-MSDU.

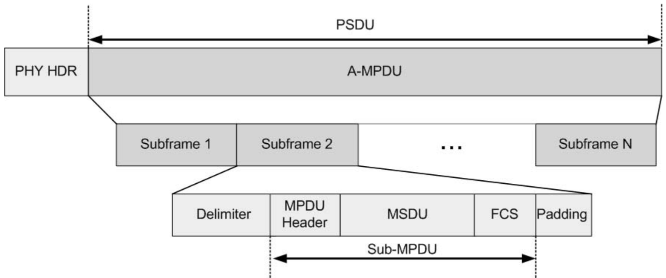

Fig. 2. Frame Format for A-MPDU.

In Section III, we present an analytical model to determine the throughput and delay performance for different frame aggregation schemes under error-prone channel conditions. Simulations are presented in Section IV to validate the accuracy of the analytical model. An optimal frame size adaptation algorithm is proposed in Section V. Conclusions and future work are given in Section VI.

II. BACKGROUND ON IEEE 802.11N

In this section, we provide an overview of several new MAC mechanisms in 802.11n, including the frame aggregation, block acknowledgement, and bi-directional data transmission.

There are two ways to perform frame aggregation at the MAC layer. The first technique is by concatenating several MAC Service Data Units (MSDUs) to form the data payload of a large MAC Protocol Data Unit (MPDU). The PHY header and MAC header, along with the frame check sequence (FCS), are then appended to form the Physical Service Data Unit (PSDU). This technique is known as *MSDU Aggregation* (A-MSDU). Figure 1 shows the frame format for A-MSDU.

The second technique is called *MPDU-aggregation* (A-MPDU). It begins with each MSDU appending with its own MAC header and FCS to form a sub-MPDU. An MPDU delimiter is then inserted before each sub-MPDU. Padding bits are also inserted so that each sub-MPDU is a multiple of 4 bytes in length, which can facilitate subframe delineation at the receiver. Then, all the sub-MPDUs are concatenated to form a large PSDU. Figure 2 shows the frame format for A-MPDU.

The 802.11n also specifies a bi-directional data transfer method. If RTS/CTS is used, the current transmission sequence of RTS (Request To Send) - CTS (Clear To Send) - DATA (Data frame) - ACK (Acknowledgement) only allows the sender to transmit a single data frame. In the bi-directional data transfer method, the receiver may request a *reverse* data transmission in the CTS control frame. The sender can then grant a certain medium time for the receiver on the reverse link. The transmission sequence will then become RTS-CTS-DATAf-DATAr-ACK. This facilitates the transmission of some small feedback packets from the receiver and may also enhance the performance of TCP (Transmission Control Protocol) which requires the transmission of TCP ACK segments.

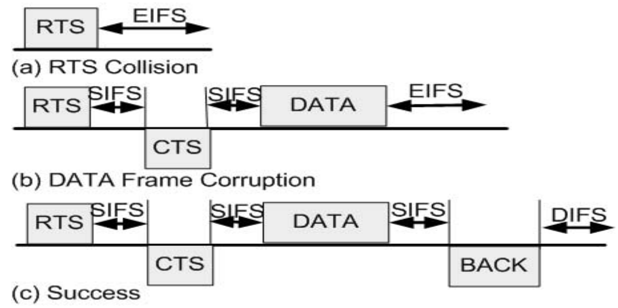

Fig. 3. Uni-directional RTS/CTS Access Scheme.

In all of the above cases, Block Acknowledgement (BACK) can be used to replace the previous ACK frame. The BACK can use a bit map to efficiently acknowledge each individual sub-frame within the aggregated frame. For the bi-directional data transfer, the reverse DATAr frame can contain a BACK to acknowledge the previous DATAf frame.

In the rest of this paper, we will focus on the study of frame aggregation and bi-directional data transfer schemes.

III. THE ANALYTICAL MODEL

We extend Bianchi's model [4] to study the A-MPDU, A-MSDU frame aggregation under error-prone channels. In the analytical model, we assume that there are N mobile stations in the WLAN. Each mobile station has saturated traffic. The wireless channel has a bit-error-rate (BER) of P_b. The minimum contention window size is W and the maximum backoff stage is m. Since the size of an aggregated frame is large, the RTS/CTS access scheme is generally more efficient than the basic access scheme. As a result, this paper only discusses the access scheme with RTS/CTS. In 802.11 WLANs, the control frames (RTS, CTS, BACK) are transmitted at the basic rate which is much lower than the data rate. Thus, the control frames are more robust in combating errors. Since the sizes of these control frames are much smaller than an aggregated data frame, they have a much lower frame error rate. In addition, the PLCP (Physical Layer Convergence Procedure) preamble and header are also transmitted at a lower rate. To simplify the analysis, we do not consider the frame error probabilities for control frames and preambles.

The possible timing sequences for A-MPDU and A-MSDU in the uni-directional transfer case are shown in Figure 3. The timing sequences for bi-directional data transfer are shown in Figure 4. In both figures, the DATA frame represents either an A-MPDU or an A-MSDU frame.

The system time can be broken down into virtual time slots where each slot is the time interval between two consecutive countdown of backoff timers by non-transmitting stations.

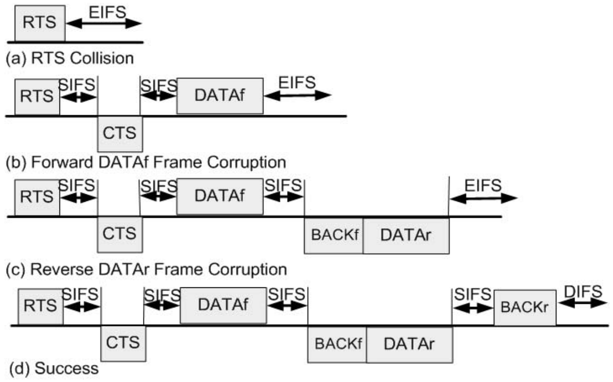

Fig. 4. Bi-directional RTS/CTS Access Scheme.

From [4], the transmission probability τ in a virtual slot is:

$$\tau = \frac{2(1-2p)}{(1-2p)(W+1)+pW(1-(2p)^m)} \tag{1}$$

where p is the unsuccessful transmission probability conditioned on that there is a transmission in a time slot. When considering both collisions and transmission errors, p can be expressed as:

$$p = 1-(1-p_c)(1-p_e) \tag{2}$$

where $p_c = 1-(1-\tau)^{(N-1)}$ is the conditional collision probability and p_e is the error probability on condition that there is a successful RTS/CTS transmission in the time slot. For uni-directional transfer, p_e is the error probability corresponding to the error case in Figure 3(b). For bi-directional transfer, we define p_e as a 2×1 vector $\mathbf{p_e} = [p_{e,1}, p_{e,2}]^T$ (where T is the transpose) corresponding to the two error cases in Figure 3(b) and (c). In the following, we will use the vector form for generality, and equation (2) for the bi-directional case is:

$$p = 1-(1-p_c)(1-p_{e,1}) \tag{3}$$

Note that only $p_{e,1}$ for Figure 3(b) contributes to p. This is because in the case of Figure 3(c), we follow our previous assumption that the $BACK_f$ control frame is error-free. Thus, $BACK_f$ for the forward frame is always successful and the $DATA_f$'s sending station will not double its contention window in this case.

The probability of an idle slot is:

$$P_{idle} = (1-\tau)^N \tag{4}$$

The probability for a transmission in a time slot is:

$$P_{tr} = 1-P_{idle} = 1-(1-\tau)^N \tag{5}$$

The probability for a non-collided transmission is:

$$P_s = \frac{N\tau(1-\tau)^{(N-1)}}{P_{tr}} \tag{6}$$

The transmission failure probability due to error (no collisions but having transmission errors) is:

$$\mathbf{p_{err}} = P_{tr}\ P_s\ \mathbf{p_e} = [p_{err,1}, p_{err,2}]^T \tag{7}$$

where $p_{err,1}$ and $p_{err,2}$ correspond to the two different error timing sequences for the bi-directional transfer in Figure 4. $\mathbf{p_{err}}$ reduces to a scalar for the uni-directional case.

The probability for a successful transmission (without collisions and transmission errors) is:

$$P_{succ} = P_{tr}P_s(1-p_{e,1}-p_{e,2}) \tag{8}$$

The network's saturation throughput can be calculated as:

$$S = \frac{E_p}{E_t} \tag{9}$$

where E_p is the number of payload information bits successfully transmitted in a virtual time slot, and E_t is the expected length of a virtual time slot. We have:

$$\begin{aligned} E_t &= T_{idle}P_{idle} + T_cP_{tr}(1-P_s) \\ &\quad + \mathbf{T_e}^T\mathbf{p_{err}} + T_{succ}P_{succ} \end{aligned} \tag{10}$$

where T_{idle}, T_c and T_{succ} are the idle, collision and successful virtual time slot's length. $\mathbf{T_e}$ is the virtual time slot length for an error transmission sequence. Similar to $\mathbf{p_e}$, it corresponds to a scalar for the uni-directional case, and a 2×1 vector for the bi-directional transfer timing sequence.

Apart from throughput, we study the average access delay experienced by each station in the uni-directional case. The access delay is defined as the delay between the time when an aggregated frame reaches the head of the MAC queue and the time that the frame is successfully received by the receiver's MAC. With the saturation throughput S, each frame takes an average of L_p/S to transmit (L_p is the aggregated frame's payload length). There are N stations competing for transmission. On average, the access delay is:

$$d = N\frac{L_p}{S} \tag{11}$$

To calculate S and d from equations (9) and (11), the parameters of E_p, T_{idle}, T_c, T_{succ}, $\mathbf{T_e}$ and $\mathbf{p_e}$ need to be determined. T_{idle} is equal to the system's empty slot time σ.

$$T_c = RTS + EIFS \tag{12}$$

where RTS is the transmission time for an RTS frame. The other parameters are case-dependent and will be discussed separately in the following subsections.

A. Uni-directional MAC

In the uni-directional case, the equations for T_{succ}, T_e and E_p are as follows:

$$\begin{aligned} T_{succ} &= RTS + CTS + DATA + BACK \\ &\quad + 3SIFS + DIFS \end{aligned} \tag{13}$$

$$\begin{aligned} T_e &= RTS + CTS + DATA + EIFS \\ &\quad + 2SIFS \end{aligned} \tag{14}$$

$$E_p = L_pP_{succ} = L_pP_{tr}P_s(1-p_e) \tag{15}$$

where CTS, BACK and DATA are the transmission time for CTS, BACK and the aggregated data frame, respectively.

For A-MSDU, the equations for p_e and E_p are:

$$p_e = 1-(1-P_b)^L \quad (16)$$
$$E_p = (L-L_{hdr})(1-p_e) \quad (17)$$

where L is the aggregated MAC frame's size, and L_{hdr} is the total length of MAC header and FCS.

For A-MPDU, error occurs when all the sub-frames become corrupted. The variables p_e and E_p can be expressed as:

$$p_e = \prod_i (1-(1-P_b)^{L_i}) \quad (18)$$
$$E_p = \sum_i (L_i - L_{subhdr})(1-P_b)^{L_i} \quad (19)$$

where i is from 1 to the total number of aggregated sub-MPDUs, and L_i is the size for the i^{th} sub-MPDU . L_{subhdr} is the total size of each sub-MPDU's delimiter, header, and FCS.

B. Bi-directional MAC

For the bi-directional MAC transfer function, there are also the two aggregation methods: A-MPDU and A-MSDU. Due to the space limitation, we only present the results for A-MSDU aggregation in this paper. The A-MPDU case can be derived in a similar way based on the following discussions.

For error in the forward frame (see Figure 4(b)), we have:

$$\begin{aligned} T_{e,1} &= RTS + CTS + DATAf + 2SIFS \\ &\quad + EIFS \quad (20) \\ p_{e,1} &= 1-(1-P_b)^{DATAf} \quad (21) \end{aligned}$$

For error in the reverse frame (see Figure 4(c)), we have:

$$\begin{aligned} T_{e,2} &= RTS + CTS + DATAf + BACKf \\ &\quad + DATAr + 3SIFS + EIFS \quad (22) \\ p_{e,2} &= (1-P_b)^{DATAf}[1-(1-P_b)^{DATAr}] \quad (23) \end{aligned}$$

For a successful bi-directional frame transmission:

$$\begin{aligned} T_{succ} &= RTS + CTS + DATAf + BACKf + DATAr \\ &\quad + BACKr + 4SIFS + DIFS \quad (24) \end{aligned}$$

Since we assume that the $BACK$ control frame is transmitted at the basic rate, $DATAf$ will be successfully received in the case of Figure 4(c). Thus, the expected successful payload information transmitted E_p can be expressed as:

$$\begin{aligned} E_p &= (L_f + L_r - 2L_{hdr})P_{succ} + (L_f - L_{hdr})p_{err,2} \\ &= (L_f + L_r - 2L_{hdr})P_{tr}P_s(1-p_{e,1}-p_{e,2}) \\ &\quad + (L_f - L_{hdr})P_{tr}P_s p_{e,2} \quad (25) \end{aligned}$$

IV. Simulation and Model Validation

To verify the accuracy of the analytical model proposed in Section III, simulations are carried out in the ns-2 simulator [15] for throughput and delay performance comparison with the analytical model. The parameters used in the simulation are from [16]. They are also shown in Table I.

Basic Rate	54 Mbps
Data Rate	144.44 Mbps
PLCP Preamble	16 μs
PLCP Header	48 bits
PLCP Rate	6 Mbps
MAC Header	192 bits
FCS(Frame Check Sequence)	32 bits
Time Slot	9 μs
SIFS	16 μs
Sub-frame Header for A-MSDU	14 bytes
Delimiter for A-MPDU	4 bytes

TABLE I
Simulation Parameters

A. Uni-directional Data Transfer

In this simulation, there are 10 wireless nodes and one access point in the network. All the wireless nodes have saturated CBR (constant bit rate) traffic directed to the access point. The BER varies from 0 to 10^{-3}. All the data packets passed down to the MAC layer are 100 Bytes in length. The number of packets aggregated in one MAC frame varies from 1 to 80, which leads to an aggregated payload size from 100 Bytes to 8 KBytes.

Figures 5 and 6 show the saturation throughput and access delay for the A-MSDU aggregation. Figures 7 and 8 show the saturation throughput and access delay for the A-MPDU aggregation. All the lines in the figures are the results obtained from the analytical model. The simulation results are shown in discrete marks. Comparison with the simulation results show that the analytical model is accurate in predicting the network performance.

From these figures, we can observe that the saturation throughput decreases and the delay increases with increasing BER for both aggregation schemes. A-MSDU achieves a higher throughput than A-MPDU under ideal channel conditions (i.e., BER = 0). This is due to the fact that A-MSDU includes a lower overhead in the aggregation process than A-MPDU. However, under error-prone channels, the advantage of A-MSDU quickly diminishes. The curves in Figure 5 show that the throughput under A-MSDU first increases, and then decreases with increasing aggregated frame size in error-prone channels. This is because without the protection of FCS in individual sub-frames, a single bit error may corrupt the whole frame which will waste lots of medium time usage and counteract the efficiency produced by an increased frame size. For A-MPDU, the throughput monotonically increases with increasing aggregated frame size. As a result, it is more beneficial to use A-MSDU under good channel conditions and A-MPDU under bad channel conditions.

Although the throughput increases by increasing the aggregated frame size for A-MPDU, the frame size cannot be increased indefinitely due to the delay constraint by many applications. As a result, we need to choose the proper aggregation scheme and adapt their parameters according to the different channel conditions and application requirements in order to achieve an optimal performance. This paper provides the performance analysis model. An extensive study for a good

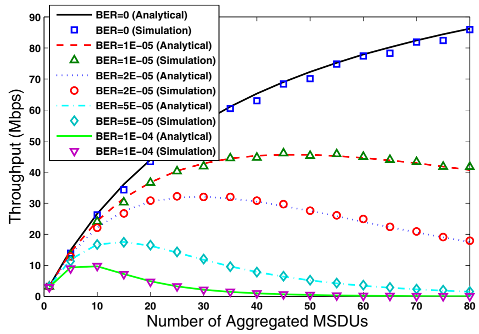

Fig. 5. Saturation Throughput for A-MSDU.

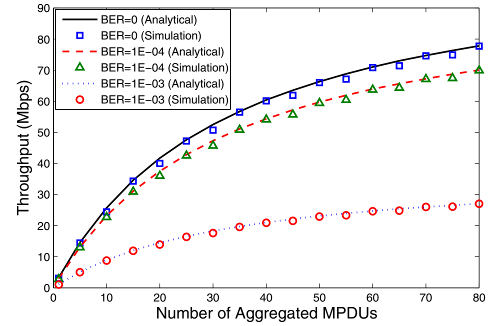

Fig. 7. Saturation Throughput for A-MPDU.

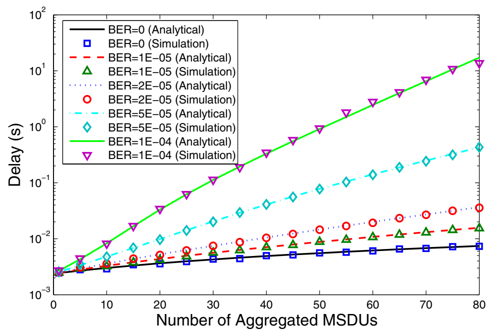

Fig. 6. Access Delay for A-MSDU.

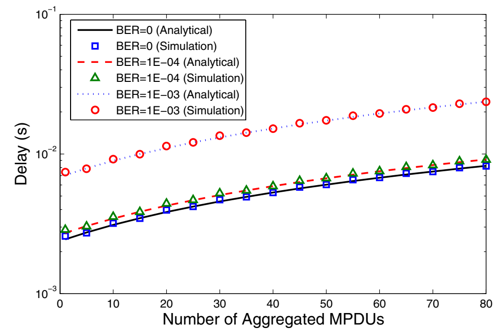

Fig. 8. Access Delay for A-MPDU.

aggregation scheduler is beyond the scope of this paper. In Section V, we will investigate the performance of a simple optimal frame size adaptation algorithm for A-MSDU under error-prone conditions.

B. Bi-directional Data Transfer

The saturation throughput performance for A-MSDU with bi-directional data transfer under different BER is shown in Figure 9. The numbers of aggregated MSDUs in the forward and reverse data aggregation are set to 20 and 1, respectively. The number of stations is varied from 5 to 30. The simulation results validate the accuracy of the analytical model in predicting the network performance. Comparing with Figure 5, the bi-directional transfer provides not much gain from the aspect of saturation throughput performance. Its major contribution to the system improvement is the interaction to the higher layer protocols (e.g., TCP) for the transfer of acknowledgement segments in a timely manner.

V. Optimal Frame Size Adaptation for A-MSDU

From Figure 5, we can observe that A-MSDU may reach a maximum throughput under different BER conditions. The optimal aggregated frame size L^* to achieve this maximum throughput varies with the channel's BER condition. To further determine the relationship between L^* and the number of contending stations, we conduct an experiment in which the number of stations changes from 10 to 30. The other parameters are the same as in Section IV. The analytical and simulation results are shown in Figure 10.

From Figure 10, we can observe that the optimal aggregated frame size L^* is very sensitive to BER, but rather insensitive to the number of contending stations in the network. To this end, we propose a simple and efffective frame aggregation adaptation algorithm to be as follows: First, we determine the L^*-BER curve from the analytical model in Section III by using an average number of stations N in the network. The L^*-BER curve gives the optimal aggregated frame size L^* under different channel bit-error-rate. Before transmitting an aggregated A-MSDU frame, the sending station will obtain an estimation of the channel BER, consult the L^*-BER curve for an optimal L^*, and then construct the aggregated frame with a size that is close to the optimal frame size.

The channel BER is a function of the modulation scheme and the SNR (Signal-to-noise ratio). In general, for a given modulation and coding scheme, the BER can be determined either from a theoretical or an empirical BER-SNR curve. The SNR is measured at the receiver for each received frame. With the help a closed-loop feedback mechanism, this SNR may be efficiently updated to the sender. For example,

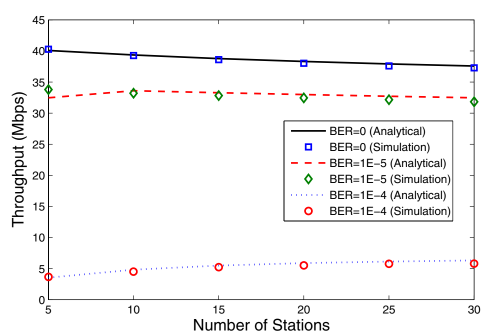

Fig. 9. Saturation Throughput under Bi-directional Data Transfer.

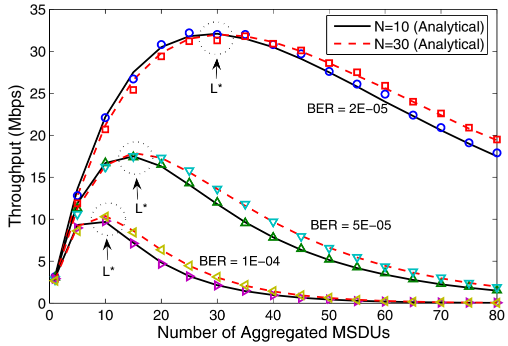

Fig. 10. A-MSDU Throughput under Different Number of Stations.

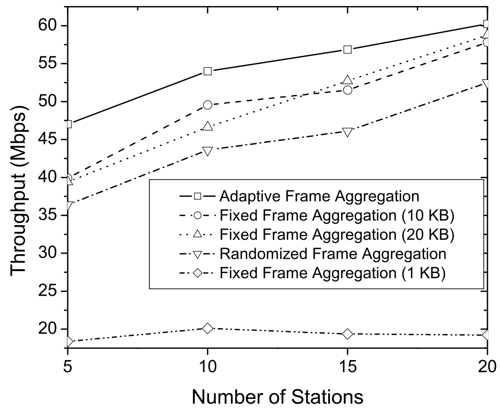

Fig. 11. Throughput under Different Frame Aggregation Schemes.

in the 802.11n proposals, there is a new MAC feature for channel management which is called the *receiver-assisted link adaptation* [1]. Channel conditions are fed back to the sender by control frames in a closed-loop fashion. In this paper, we assume that such a feedback mechanism is available in 802.11n to provide the sending station with the channel SNR information.

To determine the effectiveness of our proposed frame adaptation algorithm, we conduct an ns-2 simulation. We use the Channel B error model from [16], which models a typical large open space and office environments which have non-line-of-sight conditions, and 100 ns rms delay spread. The 144.44 Mbps data rate is used which leads to an effective transmission range of 25 m. The network topology consists of an open space of 50 m $\times$ 50 m. The access point is fixed at the center of the area. There are N wireless nodes in the network, and they move according to a random waypoint mobility model. The maximum speed is 5 m/s and the pause time is 5 s. All the wireless terminals are saturated with CBR traffic. The number of stations N is varied from 5 to 20. The throughput performance of the optimal frame size adaptation algorithm is compared with a fixed frame aggregation model and a randomized frame aggregation model, where the aggregated frame sizes are randomly distributed between the minimum (100 B) and maximum (20 KB) frame size allowed. From the simulation results shown in Figure 11, we can observe that the adaptive frame aggregation algorithm achieves better throughput performance than the other two algorithms.

VI. Conclusions

In this paper, we conducted a thorough study of the newly proposed A-MSDU and A-MPDU frame aggregation schemes in 802.11n WLANs under error-prone channels. The effects on system throughput and delay performance with the uni-directional and bi-directional data transfer methods are analyzed by both analytical and simulation methods. Comparison with the simulation results show that the analytical model is accurate in predicting the network performance. We also proposed a simple and effective optimal frame size adaptation algorithm for A-MSDU under error-prone channels. Simulation results show that it achieves a higher throughput performance than two other heuristics. For future work, we plan to design better frame aggregation schemes by taking into account the different throughput and delay requirements for QoS flows. The interaction between MAC layer's bi-directional frame aggregation and higher layer protocols, such TCP, also needs further investigation.

Acknowledgment

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