Modeling and Analyzing IEEE 802.11n MAC Frame Aggregation Techniques

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ABSTRACT: *To meet the increased expectations and demand for higher data rates in Wireless LANs (WLANs) several new physical layer technologies were developed. However, the actual achievable throughput of WLANs is restricted by the overheads of IEEE 802.11 medium access control (MAC) layer. Several frame aggregation mechanisms have been proposed to improve the MAC layer performance in 802.11n. This paper models and analyzes some of the proposed key aggregation mechanisms under ideal and error-prone channel conditions. For analysis we use well known Bianchi's analytical model and apply it on various aggregation strategies. We also compare the analytical details of various techniques and provide a unified analytical framework for continued research in this direction.*

Keywords: 802.11n, CSMA/CA, Frame Aggregation, MAC, WLANs

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1. Introduction

Wireless LAN has become ubiquitous as it provides mobility and cost-effective solution to present communication needs. The current standard for WLAN, 802.11n [1], focuses on providing high throughput by employing multiple input multiple output (MIMO) antennas with orthogonal frequency division multiplexing (OFDM) technique at PHY layer [2, 3]. However, it has been shown that the overall throughput does not increase with the increase in physical data rate [4, 5]. On the other hand, the MAC efficiency drops when data rate is increased. Figure 1 shows the effect of increased data rate on the MAC efficiency in ideal channel conditions for different frame sizes. The primary reason for this behavior is the overheads related to physical layer headers and contention time. These overheads do not decrease proportionately with the increase in physical data rates and dominate the frame transmission time at higher physical data rates. To cope up with this problem, IEEE 802.11n Task Group (TGn) proposed MAC frame aggregation methods where multiple MAC frames are sent together as one physical-layer service data unit (PSDU) thereby improving the overall throughput and efficiency.

IEEE 802.11n TGn has defined two frame aggregation schemes, namely aggregateMAC Service Data Unit (A-MSDU) and aggregate MAC Protocol Data Unit (A-MPDU). In A-MSDU , multiple MSDUs are concatenated into a single MPDU before transmitting to a receiver. This results in reduced overhead due to the absence of MAC header and FCS in individual MSDUs. It also minimizes channel idle time (fewer SIFS and backoffs). A-MPDU concatenates several MPDU subframes (each with their own MAC header and FCS) that are destined to the same receiver and forms single Physical-layer Service Data Unit (PSDU).

However, transmitting large frames is not encouraged in error-prone channels because single bit-error causes all frames to be retransmitted [9]. Aggregation with Fragment Retransmission (AFR) [4] scheme tries to address this issue by providing mechanisms for partial retransmission of affected frames. In addition to these schemes, two-level aggregation has also been proposed in which A-MSDU and A-MPDU are combined in two stages thus benefitting from the advantages of each aggregation. Few schemes [6, 7] tried to improve MAC efficiency by sending a train (burst) of frames after winning DCF contention window, and hence sharing contention overhead across multiple frames. In BlockAck strategy [8] a train of frames are transmitted without waiting for individual acknowledgments (ACK), and then the whole block is acknowledged (BACK) with single acknowledgment frame, thereby reducing the overhead due to ACKs and SIFS.

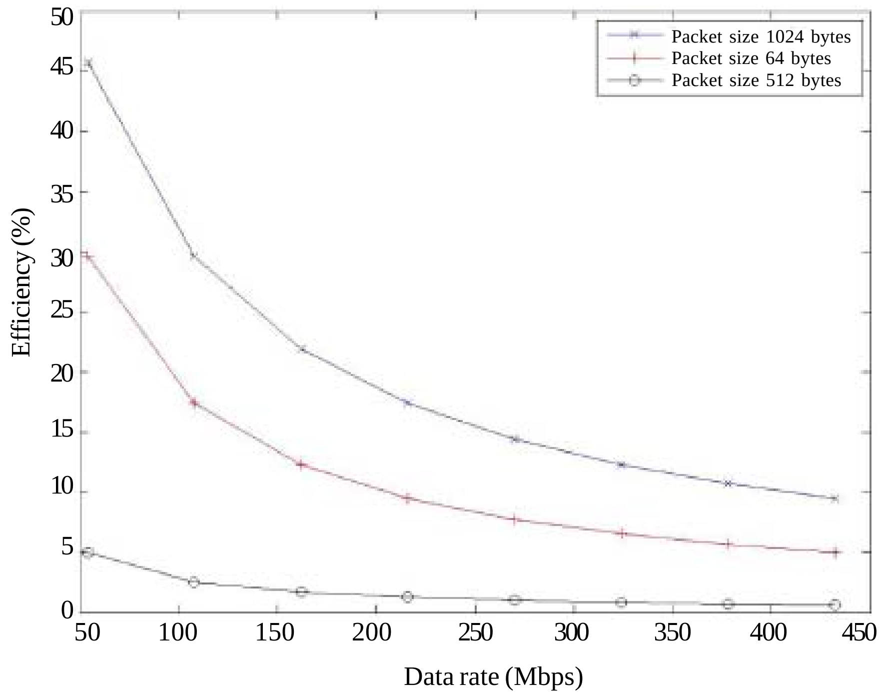


Figure 1. MAC efficiency in ideal channel conditions for various frame sizes

In addition to aggregation, 802.11n standard specifies bidirectional data transfer method over a single transmission opportunity (TXOP)[10, 11]. When a sender is allocated a TXOP, it informs surrounding stations (STAs) the time that the channel will remain busy. However, many times the transmission finishes before the reserved time and channel remains idle. In bidirectional method, the receiver STA is allowed to send packets to the sender STA in the reverse direction for the remaining TXOP time. This feature is useful in sending small feedback packets to the sender during the actual data transmission period.

802.11 has been extensively analyzed [13, 14, 15] and various models have been proposed in order to better understand the performance of 802.11 DCF throughput. DCF is a carrier sense multiple access with collision avoidance (CSMA/CA) scheme with binary slotted exponential backoff. Bianchi's analytical model [12] is one of the widely used schemes that is not only simple but it can also predict accurately system throughput for a number of wireless stations in ideal channel conditions. We have used Bianchi's analytical model to analyze various proposed schemes to enhance the saturation throughput in 802.11n. Our contribution is the comprehensive analytical treatment of 802.11n by exploring several enhancement schemes proposed in the latest protocol. Our analysis encompasses the following important scenarios:

1. DCF two-way handshake

2. DCF four-way handshake

3. Aggregation with fragment retransmission (AFR)

4. Aggregated-MPDU (A-MPDU)

5. Aggregated-MSDU (A-MSDU)

6. A-MPDU and A-MSDU with bidirectional data transfer

7. Two-level frame aggregation

In each of the above scenarios, we investigate the key parameters involved in the equation for saturation throughput and highlight the changes in those parameters for the case of ideal as well as error-prone channels.

2. Bianchi's Model

For the throughput analysis of 802.11n, we use Bianchi's model [12] which treats the backoff window size in the protocol as bi-dimensional Markovian chain. Using this chain, Bianchi attempts to compute the probability that a station transmits in a randomly chosen slot time and ultimately derives normalized system throughput as the fraction of time the channel is used to successfully transmit payload bits.

Figure 2. DCF two-way handshake transmission sequence

If n be the number of contending stations, $b(t)$ be the stochastic process representing each station's backoff time counter, m be the maximum backoff stage such that $CW_{max} = 2^m CW_{min}$, $s(t)$ be the stochastic process representing the backoff stage $(0, \ldots, m)$ of a station at time t such that $W_i = 2^i CW_{min}$, $i \in (0, m)$, then the bi-dimensional process $\{s(t), b(t)\}$ can be modeled with discrete-time Markov chain assuming constant conditional collision probability p. Using this Markov chain, the transition probabilities are given as follows:

$$\begin{aligned} &P\{0, k \mid i, 0\} = (1 - p) / W_0 && k \in (0, W_0 - 1), i \in (0, m) \\ &P\{m, k \mid m, 0\} = p / W_m && k \in (0, W_m - 1) \\ &P\{i, k \mid i - 1, 0\} = p / W_i && k \in (0, W_i - 1), i \in (1, m) \\ &P\{i, k \mid i, k + 1\} = 1 && k \in (0, W_i - 2), i \in (0, m) \end{aligned} \qquad (1)$$

Using the above equations, Bianchi derives the stationary probability (τ) that a station (STA) transmits in a randomly chosen slot time as follows [12]:

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

3. Throughput Analysis

Using Bianchi's model, we analyze the probability of successful transmission and the MAC throughput in the various proposals suggested for 802.11n. In the entire analysis, we will assume that there are fixed number of stations in the WLAN and each transmitting station has saturated traffic, that is, each station is working in full load and has attained stable condition. In other words, each transmitting station has always data to send. Also, we assume that regardless of the number of retransmissions the conditional collision probability for each frame is constant and independent.

The mathematical notation used in this paper is summarized in Table 1. We tried to maintain consistency in the symbols instead of a wide variety of notations used in the literature.

3.1 DCF with Two-way Handshake

First we consider basic Distributed Coordinated Function (DCF) which consists of two-way handshake protocol where a station sends a frame and waits for its acknowledgment in a unidirectional channel. The possible time sequence for basic DCF is shown Figure 2. The normalized throughput (S) for ideal and error-prone channels are calculated as follows.

3.1.1 Ideal Channel

For an ideal channel, the successful transmission means only one STA transmits out of n STAs at any given time. The probability that a STA does not transmit in randomly chosen slot time is $1 - \tau$. The probability that $n - 1$ STAs do not transmit will be $(1 - \tau)^{n-1}$. For an ideal channel, the unsuccessful transmission is because of collision and hence the probability that a transmitted packet encounters a collision (p) is given by

$$p = p_c = 1 - (1 - \tau)^{n-1} \quad (3)$$

Since the probability that no STA transmits is $(1 - \tau)^n$, the probability P_{tr} that at least one station transmits will be

$$P_{tr} = 1 - (1 - \tau)^n \quad (4)$$

The probability of exactly one transmission is $\binom{n}{1} \tau (1 - \tau)^{n-1}$. Therefore, the probability P_s that a transmission is successful is given by the probability that exactly one station transmits, conditioned on at least one station transmits [12].

$$p_s = \frac{n\tau (1 - \tau)^{n-1}}{1 - (1 - \tau)^{n-1}} \quad (5)$$

Symbol	Meaning
τ	Stationary probability that a station transmits in randomly chosen time
W	Minimum congestion window size
m	Maximum backoff stage, $i \in (0, m)$ where $W_i = 2^i W$
n	Total number of stations
f	Number of fragments or sub frames in an aggregated frame
$E[P]$	Expected payload
T_e	Virtual time slot length for error transmission sequence
T_s	Average time the channel is sensed busy because of successful transmission
T_c	Average time the channel is sensed busy by noncolliding stations because of a collision
T_{sifs}	Time duration for transmitting a SIFS
T_{difs}	Time duration for transmitting a DIFS
T_{eifs}	Time duration for transmitting an EIFS
T_{ack}	Time duration for transmitting an ACK
T_{hdr}^{phy}	Time duration for transmitting a physical header
T_{hdr}^{mac}	Time duration for transmitting a MAC header
T_f	Time duration for transmitting one AFR frame payload
L_{frag}	Fragment length in bytes
L_{fcs}	FCS length in bytes
L	Total MAC frame length in bytes
L_{hdr}	Total length of MAC header in bytes
$L_{hdr+fcs}$	Total length of MAC header and FCS in bytes
L_{data}	Full payload of MAC frame in bytes
p	Probability of unsuccessful transmission
p_b	Probability of single bit error
p_e	Error probability for non-ideal channel
p_c	Collision probability

Table 1. Mathematical Notation

The normalized throughput S is the ratio of expected payload transmitted in a slot time to the expected length of a slot time. If $E[P]$ is average packet payload size, then the normalized throughput for ideal channel S_{ic} is given by:

$$S_{ic} = \frac{P_{tr} P_s E\,[\mathrm{P}]}{P_1 \sigma + P_2 T_s + P_3 T_c} \tag{6}$$

Where P_1, P_2 and P_3 are the probabilities that a slot is empty, slot contains a successful transmission, and slot contains a collision respectively and is the duration of an empty slot time. Thus, $P_1 = (1 - P_{tr})$, $P_2 = P_{tr} P_s$, $P_3 = P_{tr} (1 - P_s)$.

Ignoring the transmission delay, T_s and T_c can be written as (see Figure 2):

$$T_s = T_{data} + T_{sifs} + T_{ack} + T_{difs} \tag{7}$$

$$Tc = T_{data} + T_{eifs} \tag{8}$$

The expected payload for DCF two-way handshake under ideal channel conditions is given by:

$$E\,[P] = (L - L_{hdr\,+\,fcs}) \tag{9}$$

3.1.2 Channel with Error

Since the channel is error prone, the unsuccessful transmission could be due to frame collision or channel error. Hence, the unsuccessful transmission probability (p) in Equation 2 needs to be adjusted for both collisions and transmission errors. Thus, p can be expressed as

$$p = 1 - (1 - p_c)\,(1 - p_e) \tag{10}$$

where $p_c = 1 - (1 - \tau)^{n-1}$ is the collision probability and p_e is the error probability on the condition that there is a successful transmission in the time slot. The throughput Equation 6 can now be written for error prone channel as

$$S_{ce} = \frac{P_{tr} P_s E\,[\mathrm{P}]}{P_1 \sigma + P_2 T_s + P_3 T_c + P_e T_e} \tag{11}$$

where T_e is the virtual time slot length for error transmission sequence. It can be given as

$$T_e = T_{data} + T_{eifs} \tag{12}$$

T_c and T_s remain the same. Also note that in Equation 11, P_2 is now equal to $P_s P_{tr} (1 - p_e)$.

The only unknowns left out in Equation 11 are p_e and $E\,[P]$. As we are aware that any single bit error would corrupt the entire frame, thus

$$p_e = 1 - (1 - p_b)^L \tag{13}$$

$$E\,[P] = (L - L_{hdr\,+\,fcs})\,(1 - p_e) \tag{14}$$

3.2 DCF with Four-way Handshake

DCF with four-way handshake uses RTS/CTS (Ready to Send/Clear to Send) control packets. A station that wishes to send a data frame exchanges RTS/CTS frames with the intended recipient before sending the actual data frame. This mode fixes the "*hidden node*" problem. The possible time sequence for DCF with four-way handshake in unidirectional data transfer is shown in Figure 3. DCF four-way handshake throughput analysis is similar to DCF two-way handshake analysis of Section 3.1. The normalized throughput is given by Equation 6 and Equation 11 for ideal and error-prone channel conditions respectively. The only difference is in the time durations T_s, T_c and T_e, which are given by:

$$T_s = T_{rts} + 3T_{sifs} + T_{cts} + T_{data} + T_{ack} + T_{difs}$$

$$T_c = T_{rts} + T_{eifs} \tag{15}$$

$$T_e = T_{rts} + T_{cts} + T_{data} + T_{eifs} + 2T_{sifs} \quad (16)$$

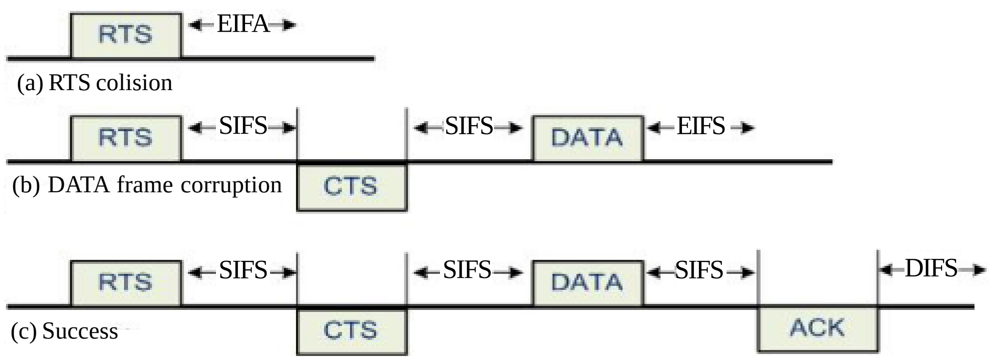


Figure 3. RTS/CTS unidirectional access

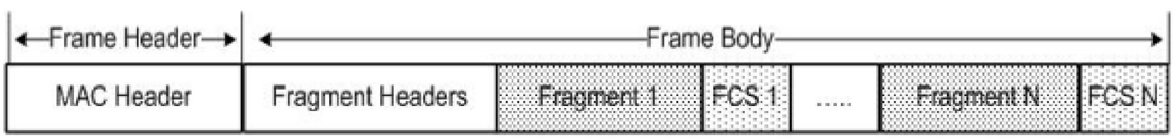


Figure 4. Aggregation with fragment retransmission

3.3 Aggregation with Fragment Retransmission (AFR)

AFR [4] scheme works by combining several packets into one large frame and dividing this frame into multiple fragments. Fragmentation is done to avoid retransmission of the whole frame in case of any packet corruption during the transmission. Only the fragment containing the corrupted packet is retransmitted, thus increasing the overall performance. AFR also employs zero-waiting policy in which frames don't wait for all the fragments from upper layer to arrive. Instead, frames are transmitted as soon as a station wins transmission opportunity. Thus, AFR offers improved throughput through aggregation, fragmentation for selective retransmission and zero-waiting scheme.

AFR can also be modeled using Bianchi's model and the throughput Equation 6 can be directly applied for ideal as well as error-prone channels as follows:

$$S_{a\,fr} = \frac{P_2 E[\mathrm{L}]}{P_1\sigma + P_2 T_s + P_3 T_c} \quad (17)$$

It should be noted that AFR considers partially corrupted frames due to channel noise as successful transmission. Since E[L] is the expected number of successfully transmitted bits instead of frame payload size, it can be calculated as [4]:

$$E[L] = \sum_{i=0}^{f} \binom{f}{i} . \left(p_e^{frag}\right)^i . \left(1 - p_e^{frag}\right)^{f-i} . \left(L - i.L_{frag}\right)$$

where fragment error rate p_e^{frag} is given as

$$p_e^{frag} = 1 - (1 - p_b)^{L_{frag} + L_{fcs}} \quad (18)$$

L_{fcs} is added in Equation 18 as each fragment in AFR data frame has FCS (see Figure 4). Substituting the value of p_e^{frag} in Equation 17 and simplifying to get

$$S_{a\,fr} = \frac{P_2 . (1 - p_e^{frag})}{P_1\sigma + P_2 T_s + P_3 T_c} \quad (19)$$

The other unknown values T_s and T_c in Equation 17 are different for DCF two-way handshake and DCF four-way handshake.

In case of DCF two-way handshake, T_s and T_c are given as:

$$T_s = T_{data}^{afr} + T_{sifs} + T_{ack} + T_{difs} \tag{20}$$

$$T_c = T_{data}^{afr} + T_{eifs} \tag{21}$$

where $T_{data}^{afr} = T_{hdr}^{phy} + Tf$.

For DCF four-way handshake, T_s and T_c are given as:

$$T_s = T_{rts} + T_{cts} + T_{data}^{afr} + 3T_{sifs} + T_{ack} + T_{difs} \tag{22}$$

$$T_c = T_{rts} + T_{eifs} \tag{23}$$

3.4 A-MPDU/A-MSDU Frame Aggregation

In this section we model A-MPDU and A-MSDU schemes using Bianchi's model [12]. Aggregated A-MPDU and A-MSDU frames can be transmitted using either DCF two-way handshake or DCF four-way handshake, and hence the analysis of A-MPDU and A-MSDU for DCF two-way hand shake and four-way hand shake will be similar to the standard analysis of Sections 3.1 and 3.2 respectively. The normalized throughput for A-MPDU and A-MSDU is given by Equations 6 and 11 respectively for ideal and error-prone channel conditions. However, the expected payload values (E[P]) in these equations depend upon the aggregation method and channel condition, and can be estimated as follows.

3.4.1 Ideal Channel

A-MSDU and A-MPDU frame structure is shown in Figure 5. Assume that there are nf subframes in the each aggregated A-MSDU and A-MPDU frame.

In A-MSDU, for each subframe there is an additional overhead of subframe header (14 bytes) and padding (0-3 bytes). Hence, the expected payload size for A-MSDU is

$$E[P] = L - L_{a-msdu}^{oh} \tag{24}$$

where $L_{a-msdu}^{oh} = L_{hdr+fcs} + \sum_{i=1}^{nf}(L_{subhdr} + L_{pad})$.

On the other hand, each subframe in A-MPDU has a separate MAC header, a delimiter (4 bytes), variable size padding (0-3 bytes) and FCS. Hence the expected payload for A-MPDU is given by:

$$E[P] = L - \sum_{i=1}^{nf}(L_{hdr+fcs} + L_{dlim} + L_{pad})$$

The above equation can be rewritten as

$$E[P] = \sum_{i=1}^{nf}(L_i - L_{a-mpdu}^{oh}) \tag{25}$$

where L_i is i^{th} subframe of the aggregated A-MPDU frame and $L_{a-msdu}^{oh} = L_{hdr+fcs} + L_{dlim} + L_{pad}$.

It should be noted that T_s is slightly different for A-MPDU/A-MSDU as single block ACK will be sent instead of individual ACKs.

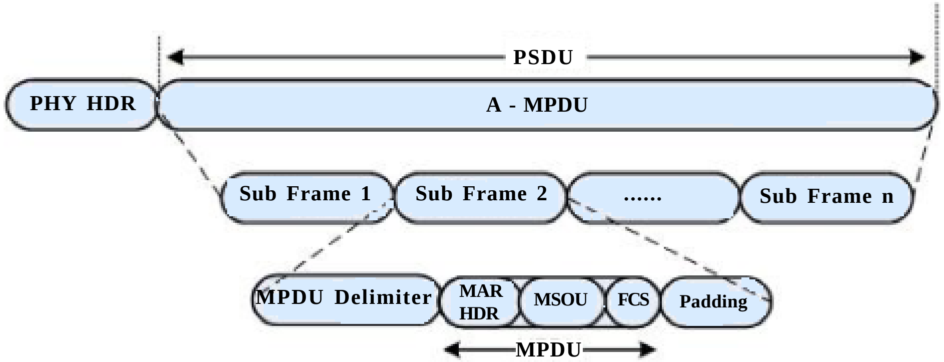
$$T_s = T_{rts} + T_{cts} + T_{data} + T_{back} + 3T_{sifs} + T_{difs}$$

3.4.2 Error Prone Channel

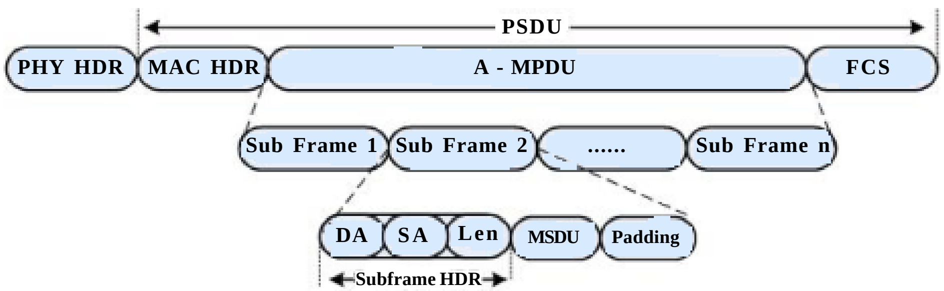
In order to calculate the values for pe and $E[P]$ in Equation 11, we first consider the case for A-MSDU where any single bit error would corrupt the entire frame, thus

$$p_e = 1 - (1 - p_b)^L \tag{26}$$

$$E[P] = (L - L^{oh}_{a-msdu})(1 - p_e) \quad (27)$$



(a) A-MPDU frame structure



(b) A-MSDU frame structure

Figure 5. One-level frame aggregation

In the case of A-MPDU, error occurs when all the sub-frames are corrupted. Therefore, we can write

$$p_e = \prod_{i=1}^{nf} \left(1 - (1 - p_b)^{L_i}\right) \quad (28)$$

$$E[P] = \sum_{i=1}^{nf} \left(L_i - L^{oh}_{a-mpdu}\right)(1 - p_b)^{L_i} \quad (29)$$

where i ranges from 1 to the total number of aggregated sub-MPDUs frames (nf).

3.5 Bidirectional Data Transfer

A key enhancement in 802.11n specifications is bidirectional data transfer. In this section we extend the A-MSDU/AMPDU analysis of Section 3.4 for bidirectional data transfer. The possible time timing sequence for DCF four-way handshake with bidirectional data transfer is shown in Figure 6. The DATA frames in this figure represent A-MPDU/AMSDU aggregated frames.

The normalized throughput equations for bidirectional data transfer remain the same and are given by Equations 6 and 11 respectively for ideal and error-prone channel conditions. However, the expected payload value ($E[P]$) should accommodate both forward and reverse payloads. In the ideal channel case, $E[P]$ is the sum of forward and reverse payloads. If we assume that the data in the reverse direction is also aggregated and it contains nr subframes then $E[P]$ for A-MSDU and A-MPDU is given by the following equations respectively.

$$E[P] = L_f + L_r \;\; 2\, L^{oh}_{a-msdu} \quad (30)$$

$$E[P] = \sum_{i=1}^{nf} (L_i - L^{oh}_{a-msdu}) + \sum_{i=1}^{nr} (L_i - L^{oh}_{a-msdu}) \quad (31)$$

Also T_s should be modified to accommodate the reverse payloads, and it is given by:

$$T_s = T_{rts} + 4T_{sifs} + T_{cts} + T^f_{data} + T^f_{back} + T^r_{data} + T^r_{back} + T_{difs} \tag{32}$$

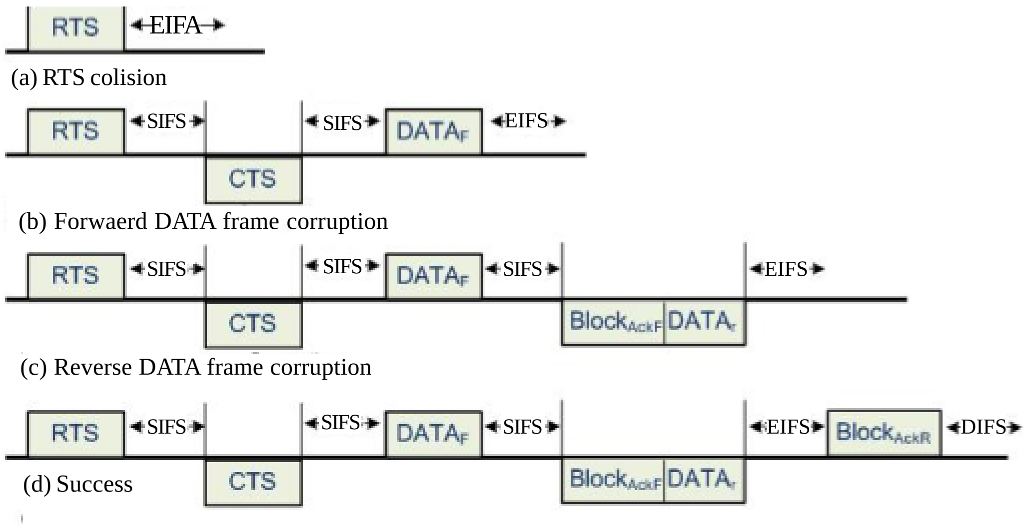


Figure 6. RTS/CTS bidirectional data transfer

3.5.1 Error Prone Channel

In this case, an error can occur during the forward or reverse transmission. Hence the error probability has two components (p_e, f, p_e, r) corresponding to figure 6 (b) & 6 (c). The virtual time slots corresponding to error transmission in the forward and reverse directions are given by:

$$T_{e,f} = T_{rts} + 2T_{sifs} + T_{cts} + T^f_{data} + T_{eifs} \tag{33}$$

$$T_{e,b} = T_{rts} + 3T_{sifs} + T_{cts} + T^f_{data} + T^f_{back} + T^r_{data} + + T_{difs} \tag{34}$$

The error probabilities in the forward and reverse directions are dependent on aggregation method. For A-MSDU, the error probabilities in the forward sequence and reverse sequence are given by

$$p_{e,f} = 1 - (1 - p_b)^{L f} \tag{35}$$

$$p_{e,r} = 1 - (1 - p_b)^{L f} (1 - (1 - p_b)^{L r}) \tag{36}$$

The expected payload for A-MSDU in the bidirectional mode is:

$$E[P] = (E[P]_f + E[P]_r)(1 - p_{e,f}, p_{e,r}) + E^f_p p_{e,r} \tag{37}$$

where $E[P]^f$ and $E[P]^r$ are the expected payloads in the forward and reverse directions respectively and are given by Equation 24.

In case of A-MPDU, the error probability in the forward direction can be calculated as:

$$p_{e,f} = \prod_i (1 - (1 - p_b)^{L_{i,f}}) \tag{38}$$

$$p_{e,r} = 1 - p_{e,f} [\prod_i (1 - (1 - p_b)^{L_{i,r}})] \tag{39}$$

The expected payload for A-MPDU in bidirectional transmission is given by Equation 37, where $E[P]^f$ and $E[P]^r$ are the expected payloads in the forward and reverse directions. These can be calculated by using Equation 25.

3.6 Two-level Frame Aggregation

In this scheme the aggregation is done at two different stages, A-MSDU stage and A-MPDU stage. In the first stage, aggregation is performed by combining all collected MSDUs that satisfy A-MSDU aggregation requirements into a AMSDU frame. In the second stage all A-MSDUs and MSDUs coming from the first stage are combined into A-MPDU frames. Figure 7 illustrates the frame structure of two-level frame aggregation.

3.6.1 Ideal Channel

The expected payload in the ideal channel will be sum of all MPDU payloads after deducting aggregation overhead. If there are n_{mpdu} subframes in the aggregated packet then,

$$E[P] = \sum_{i=1}^{n_{mpdu}} \left(L_i^{mpdu} - L_{a-mpdu}^{oh} \right) \tag{40}$$

where L_i^{mpdu} is the length of i^{th} MPDU, which can contain one or more MSDUs. Presuming MPDUs contain AMSDU frames with n_j^{msdu} MSDUs, then the length of i^{th} MPDU can be approximated as

$$L_i^{mpdu} = (L_i^{a-msdu} - \sum_{j=1}^{n_j^{msdu}} (L_{subhdr} + L_{pad})) \tag{41}$$

where L_i^{a-msdu} is the length of i^{th} A-MSDU frame.

The normalized throughput for this case can be calculated by substituting $E[P]$ in Equation 6.

3.6.2 Error Prone Channel

The expected payload and normalized throughput are affected by the channel errors. Similarly to A-MPDU, the whole packet is discarded when each MPDU has error. Thus, the probability of error in the whole aggregated frame is The frame error in the individual MPDU frames is

$$p_e = \prod_{i}^{n_{mpdu}} \left(p_e^{mpdu_i} \right) \tag{42}$$

where $p_e^{mpdu_i}$ is the probability of error in the i^{th} MPDU, which can be calculated as

$$p_e^{mpdu_i} = \left(1 - (1 - p_b) L_i^{mpdu}\right) \tag{43}$$

The expected payload after accommodating the packet error is given as

$$E[P] = \sum_{i=1}^{n_{mpdu}} \left(L_{mpdu}^{i} - L_{a-mpdu}^{oh} \right)\left(1 - p_e^{mpdu_i}\right) \tag{44}$$

The normalized throughput for this case can be calculated by substituting Equation 44 in Equation 11.

4. Conclusion

In this paper we analyzed and compared the normalized MAC throughput for various aggregation schemes in 802.11n using Bianchi's analytical model. The following are some conclusions that can be drawn from the analysis.

In ideal channel conditions A-MSDU performs very well since there is no overhead of MAC headers & FCSs. Conversely, when BER increases A-MPDU outperforms A-MSDU where single bit error corrupts the whole A-MSDU aggregated frame. A-MPDU obviates the need to resend the entire aggregated frame since the receiver can delineate a received A-MPDU frame and sends a BlockAck allowing individual data frames to be acknowledged or retransmitted. The analytical model of A-MSDU and A-MPDU does not put any restriction on the number of aggregated MSDUs (n_{msdu}) and MPDUs (n_{mpdu}). However, due to practical restrictions on the size of each MPDU, A-MSDU can take benefit from aggregation only when individual MSDUs are smaller. On the other hand, A-MPDU does not suffer from this restriction as it has large PSDU size and it can aggregate relatively higher number of MPDUs. Two-level aggregation benefits from the advantages of A-MPDU as well as A-MSDU thus achieving higher throughput. When the MSDU size is small, the aggregation at A-MSDU level mainly contributes towards the increased throughput. Whereas, when the MSDU size is large, A-MPDU contribute more towards the increase in the overall throughput by combining many MPDUs in one large PSDU. It should be noted that, however, due to inherent complexity of two-level aggregation, the overall processing time may increase.

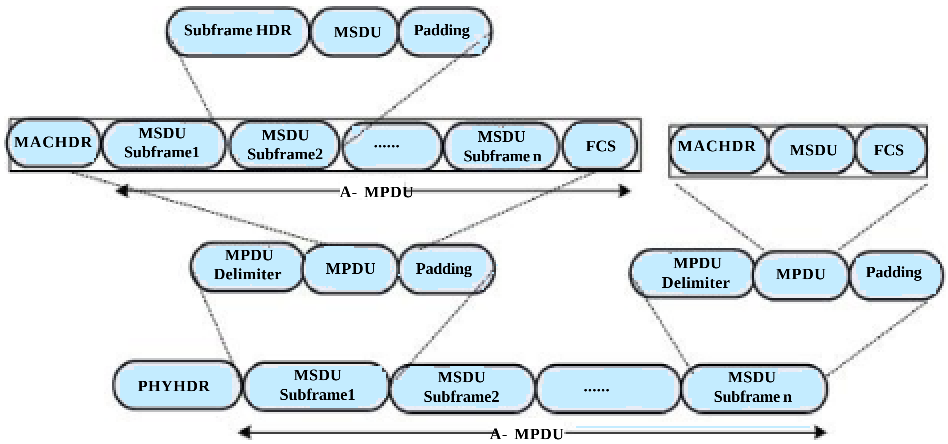


Figure 7. Two-level frame aggregation

Bidirectional data transfer provide significant improvement over unidirectional data transfer when receiver has always data to send in the reverse direction (for example, applications like voice chatting). On the other hand, it won't add any advantage over unidirectional data transfer in terms of MAC throughput if the data is predominantly unidirectional, except that higher layer protocol can benefit from reverse data in terms of timely acknowledgments. Larger frame size increases the probability of collision (P_3) thereby decreasing the throughput. Since aggregation schemes employ larger frame size, they can benefit from four way handshake(RTS/CTS) to reduce the probability of collision because of smaller RTS/CTS control frames. AFR scheme reduces the probability of error (P_e^{frag}) by fragmenting the frame and selectively retransmitting the erroneous fragment thus improving the overall throughput.

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