國立交通大學

網路工程研究所

碩士論文

有效率的動態選擇封包合併機制在無線網狀網路中

Efficient Dynamic Frame Aggregation in IEEE 802.11s Mesh Networks

研 究 生：楊宗憲
指導教授：林盈達　教授

中華民國九十七年七月

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研 究 生：楊宗憲	Student: Tsung-Hsien Yang
指導教授：林盈達	Advisor: Dr. Ying-Dar Lin

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碩士論文

A Thesis
Submitted to Institutes of Computer Science and Engineering
College of Computer Science
National Chiao Tung University
in partial Fulfillment of the Requirements
for the Degree of
Master
In

Network Engineering

June 2008

HsinChu, Taiwan, Republic of China

中華民國九十七年七月

國立交通大學

研究所碩士班

論文口試委員會審定書

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有效率的動態選擇封包合併機制在無線網狀網路中

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中華民國九十七年七月一日

Institute of Network and Engineering
College of Computer Science
National Chiao Tung University
Hsinchu, Taiwan, R.O.C.

As members of the Final Examination Committee, we certify that we have read the thesis prepared by Tsung-Hsien Yang entitled Efficient Dynamic Frame Aggregation in IEEE 802.11s Mesh Networks

and recommend that it be accepted as fulfilling the thesis requirement for the Degree of Master of Science.

Committee Members:

Shao Li Tsao

Yuan-Cheng Lai

Ying-Dar Lin

Thesis Advisor: Ying-Dar Lin

Director: Chien-Chao Tseng

Date: 2008-07-01

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中華民國 年 月 日

有效率的動態選擇封包合併機制在無線網狀網路中

學生：楊宗憲　　**指導教授：林盈達**

國立交通大學網路工程研究所

摘要

無線區域網路在真實環境中所達到的實際效能比起理論值是相對地遜色許多。因其MAC協議：CSMA/CA在傳輸過程中，所產生高度額外的負載為主要的因素。再加上現今盛行的多媒體通訊應用和網路控制協議通常使用小型的封包來進行資料的傳輸，如此，若使用新興的技術像802.11n如此高的傳輸速率來傳送資料，那麼在整個傳輸過程中，所耗費在控制協調的比例就相對來的較高。若加上因多點跳躍的傳輸環境，為傳輸單一封包而得耗用更多額外的資源，會更顯著地大幅降低傳輸效能。因此一個解決傳輸效能低落的方法之一是在傳輸封包之前，將小封包聚集成大封包再進行傳送。

故本論文先陳述三種普遍認定的封包合併機制，其使用限制、傳輸特徵及其效益，而後提出一個針對802.11s無線網狀網路傳輸環境下，基於機率上的假設來有效率地動態選擇最適合的封包合併機制的排程演算法。此演算法依據佇列內封包數量的多寡、封包的分布情形和當下的傳輸品質，決定兩件事情:第一是採用何種封包合併機制，第二是何時把合併的封包傳送出去。藉由此排程來提升整體無線網狀網路的頻寬使用效率。透過模擬結果，驗證此演算法能有效地提升整體網路的傳輸吞吐量達將近95%。

關鍵字： 無線網路、封包合併、多點跳躍

Efficient Dynamic Frame Aggregation
in IEEE 802.11s Mesh Networks

Student: Tsung-Hsien Yang **Advisor: Dr. Ying-Dar Lin**

Department of Network Engineering

National Chiao Tung University

Abstract

WLAN achieves poor throughput performance compared to the underlying PHY data rate. This is mainly caused by the overhead of CSMA/CA. Besides, the data of multimedia traffic and control protocols is usually transmitted in small frames. When transmitting a large number of small-size frames with high data rate, such as 802.11n, the ratio occupied for CSMA/CA control overhead is relatively high so that it results in worse efficiency. The degree of throughput degradation is further severe under multi-hop transmissions. Thus, aggregating several small-size frames into one transmission is a way to improve this.

This works first reveal the three common frame aggregation mechanisms about their transmission characters, benefits, and the restriction of usage, and then propose a novel algorithm, which could dynamically adopt the appropriate aggregation mechanism according to hypothesis of probability, to achieve a high-throughput and high-efficiency mesh network. Based on channel conditions, the quantity and the distribution of frames in the transmission queue, two things will be determined, one is what aggregation mechanism to be adopted; the other is when to send the aggregated frames. Through the policy described above, the bandwidth utilization will be maximized as high as possible. Simulation results demonstrated that the algorithm actually increases the channel efficiency of the 802.11 MAC and further improves the overall throughput of wireless mesh networks by approximately 95%.

Keywords: wireless networks, frame aggregation, multi-hop

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Chapter 1 Introduction

With the increasing demand for real-time applications over wireless networks, IEEE 802.11n is proposed to provide a high transmission rate up to 600 Mbps [1], using multiple-input multiple-output (MIMO) and orthogonal frequency division multiplexing (OFDM). However, control protocols, such as ARP and ICMP, and multimedia traffic, such as VoIP, are usually transmitted in small frames. When transmitting small-size frames with such a high data rate, the ratio, which is occupied for CSMA/CA control overhead, including preamble, frame headers, carrier sense waiting time, and a random backoff period, is relatively high so that it results in worse efficiency. Thus, aggregating several frames into one transmission is a way to improve this.

At which sub-layer to aggregate?

Frame aggregation can be performed at different sub-layers. There are three main ways, as shown in Figure 1, to perform frame aggregation, known as (1) MAC Service Data Unit Aggregation (A-MSDU), where multiple MSDUs can be aggregated at the MAC layer and sent to the same receiver via a single MAC Protocol Data Unit (MPDU) with a MAC header, (2) A-MPDU, which consists of a number of MPDU delimiters, each of which is followed by an MPDU to form a PHY Service Data Unit (PSDU), and (3) PHY Protocol Data Unit Aggregation (A-PPDU), which concatenates multiple PSDUs together and adds a PHY header [2][3][4]. The comparison among 3 types of frame aggregation is shown in Table 1. A-PPDU and A-MPDU have the advantage of multiple destination addresses, and are robust to transmission errors, such as collisions, because individual Frame Control Sequence (FCS) is attached. A-PPDU has another advantage of rate change for each PPDU with different modulations. A-MSDU has the highest efficiency because of its small overhead of CSMA/CA, but is restricted to a single destination address and vulnerable to transmission errors.

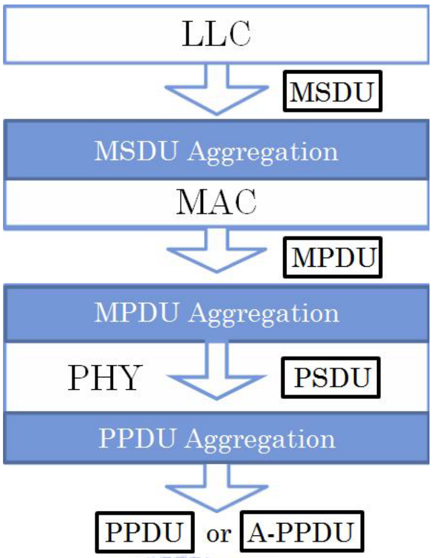
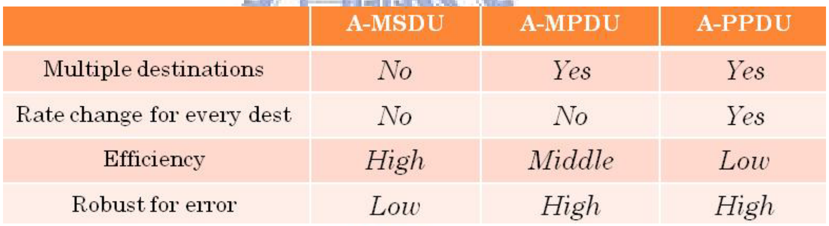


Fig. 1: Layers of WLAN interface



	A-MSDU	A-MPDU	A-PPDU
Multiple destinations	*No*	*Yes*	*Yes*
Rate change for every dest	*No*	*No*	*Yes*
Efficiency	*High*	*Middle*	*Low*
Robust for error	*Low*	*High*	*High*

Table 1: Comparisons of Frame Aggregation Mechanisms

Networks with backhaul links, such as wireless mesh networks, are more suitable for frame aggregation due to frequent frame queuing. A wireless mesh network is composed of gateway nodes, mesh points (MP), mesh access points (MAP), and wireless clients (STA) [5]. Gateways connect the mesh network with the wired Internet. MPs, MAPs, and gateways communicate with one another via wireless medium and form a wireless backbone network. STAs gain network access by associating with a MAP. Each MP or MAP has peer-to-peer neighbors under a mesh topology. But there is only one node permitted to transmit packets at a time under the same collision domain due to CSMA/CA. This situation leads to many frames queued frequently at mesh nodes. Other scenarios are analogous to this situation when an MP or MAP has many peer-to-peer neighbors or the traffic load is large inside a mesh network.

3 communication pairs and 4 transmission types in wireless mesh

Because there are different roles in mesh networks, the peer-to-peer communication among them could be classified into three categories. The three communication pairs are M(A)P-to-M(A)P, MAP-to-STA, and STA-to-MAP. Since an aggregated frame might go through multiple next-hops, i.e. receivers, to reach multiple destinations, there are four transmissions types in this multi-hop environment, namely single destination single receiver (SDSR), multiple destination single receiver (MDSR), multiple destination multiple receiver (MDMR) and single destination multiple receiver (SDMR), which is namely the multi-path issue. Each combination of the communication pairs and the transmission types is suitable for some aggregation mechanisms according to different transmission characteristics. For example, a STA, which has only one link to a MAP, will not choose A-PPDU to aggregate the frames because multi-receivers, MDMR, will not happen to such a transmission. But a MAP may have multiple links to different STAs, it may choose A-PPDU to aggregate the frames because MDMR may happen to the transmission from MAPs to STAs.

In this work, we propose a novel algorithm, called Dynamic Aggregation Selection and Scheduling (DASS), to achieve a high-throughput and high-efficiency mesh. It could dynamically adopt the appropriate aggregation mechanism according to the bit error rate (BER), the communication pair, the transmission type, and the quantity and the distribution of frames in the transmission queue to maximize the bandwidth utilization as high as possible. Besides, traffic load in mesh networks is not balanced. The traffic load near mesh gateways is relatively large so that the mesh nodes close to gateways have more frames to aggregate than others. Through the considerations above and the analysis of past traffic, we could expect how many incoming frames to be aggregated, and then determine an appropriate time to send the aggregated frame. We use Network Simulation 2 (NS-2) to evaluate DASS to compare with a single aggregation mechanism under infinite and steady backlog, and then show the results, including throughput performance and average delay.

Wireless channels are usually error-prone and effects of packet errors have an impact on system performance. Several papers [6] - [9] analyze the throughput performance under different channel error conditions and conclude that there is an optimal packet size under a certain BER to achieve the maximum throughput. Lin and Wong [10] conducts the thorough study of the newly proposed A-MSDU and A-MPDU frame aggregation schemes, and proposes a simple and effective optimal frame size adaptation algorithm for A-MSDU under error-prone channels. All of the studies do not consider how to choose an appropriate aggregation mechanism due to the variations of the quantity and the distribution of frames, the communication pair, and multi-receivers. Moreover, their simulation is under infinite backlog (i.e. all stations have data to transmit at all time), but what is the throughput gain under steady backlog?

The rest of this work is organized as follows. Chapter 2 provides an overview of 802.11n frame aggregation mechanisms, the architecture of 802.11s mesh networks, and the referred analytical model for optimal frame size adaptation. In chapter 3, we present the DASS algorithm and illustrate the detailed operations. Chapter 4 describes the simulation environment and numerical results to observe the behavior of frame aggregation. Finally, chapter 5 concludes this work with future directions.

Chapter 2 Background

2.1 Overview of IEEE 802.11n and 802.11s

2.1.1 Sources of PHY/MAC Overhead

In order to understand throughput inefficiency, first we need to describe MAC's mandatory Distributed Coordination Function (DCF) operation. DCF is a basic medium access mechanism that allows wireless stations (STAs) to access the wireless medium for transmission.

Once a frame arrives at the MAC layer from the upper layers, it enters the transmission queue, which is situated for receiving and buffering incoming data. Then the MAC halts for a certain period of time, named DCF interframe space (DIFS). If the STA senses the channel is busy during that period, it waits till the channel becomes idle. Alternatively, if the medium remains unoccupied, the STA starts a backoff operation with a randomly-selected backoff count value within a contention window. The counter starts to decrement a slot interval as long as the channel remains idle and when it reaches to zero then the frame can be transmitted. When the receiver STA receives the frame successfully, it responds back with an acknowledgement frame (ACK) after a short interframe space (SIFS). If the initiator doesn't receive the ACK, it assumes that the communication was broken or interfered so it commences again the same procedure. An optional mechanism that avoids collisions with a high probability is the Request-to-Send/Clear-to-Send (RTS/CTS) process, where RTS/CTS are two control frames, which are sent from the sender and the receiver respectively to corroborate that the channel is unbound from both sides. Obviously, this functionality can aggravate the channel efficiency as more steps are affixed to the DCF operation.

From the above operation, the overhead needed for each frame, the required additional information that we allow to be transmitted or compulsory operations that are taken in order to guarantee a successful transmission. The derived overhead is the DIFS, Backoff, PHY headers (PCLP Preamble and PLCP Header), MAC header (including FCS), SIFS and ACK. However, we assume that the transmission was successful with the first attempt and no re-transmissions were needed, something that would exponentially accumulate the existing overhead.

2.1.2 802.11n Frame Aggregation Mechanisms

A-MSDU

The purpose of A-MSDU is to allow numerous MSDUs be aggregated and sent to the same receiver via a single MPDU. Thus, channel efficiency rapidly increases, specifically when there are many small MSDUs such as ACKs.

Figure 2 illustrates the architecture of a carrier MPDU which contains an A-MSDU. An A-MSDU concatenates multiple subframes, which consist of a subframe header followed by an MSDU and 0-3 bytes of padding. Since the length of each subframe should be a multiple of 4 bytes, except the last one. Because all MSDUs are compressed into a single MPDU with a single FCS, corruption of one subframe results in the retransmission of the entire A-MSDU. This situation could lead in poor channel utilization in case of transmission errors. There are also some constraints: i) all MSDUs must have the same TID value, ii) lifetime of an A-MSDU should be equal to the maximum lifetime of the MSDUs and iii) the Destinations Address (DA) and Senders Address (SA) parameter values of each subframe header must map to the same Receiver Address (RA) and Transmitter Address (TA) in the MAC header. Thus, broadcasting or multicasting is not allowed.

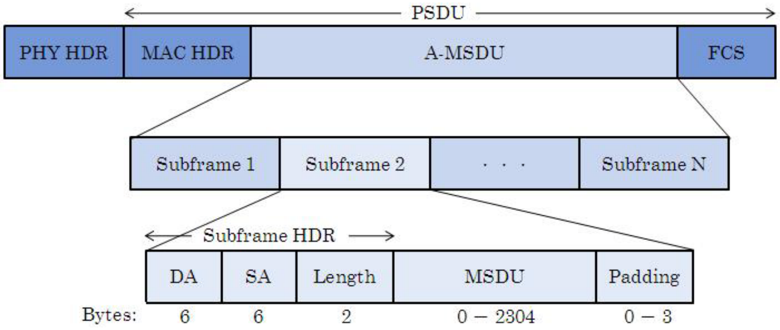


Fig. 2: The frame format of an A-MSDU

A-MPDU

The purpose of A-MPDU is to joint multiple MPDUs to diminish a PHY header. These MPDUs sent to the same receiver could be aggregated into an A-MPDU no matter their TIDs are consistent or not. The number of subframes it could hold is 64 since a Block ACK bitmap field is 128 bytes in length where each frame is mapped in 2 bytes.

The A-MPDU format is shown in Figure 3, where an A-MPDU consists of numerous of MPDU delimiters each followed by an MPDU. The basic operation of a delimiter header is to define the position of the MPDU inside an aggregated frame. Note that the CRC field on a delimiter verifies the authenticity of the 16 preceding bits. The padding bits are added so that each MPDU is a multiple of four bytes in length, which can assist subframe delineation at the receiver’s side.

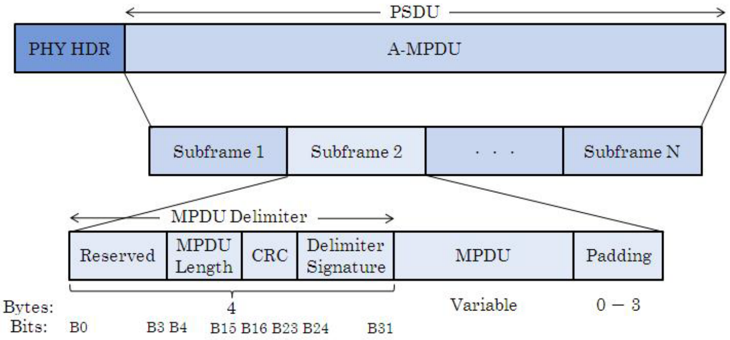


Fig. 3: The frame format of an A-MPDU

A-PPDU

The purpose of A-PPDU is to concatenate multiple PSDUs together to improve efficiency of channel usage. Different PSDUs are separated by a PLCP signal field. An A-PPDU concatenates multiple PSDUs with a common preamble. A-PPDU aggregation is performed in a single medium access, and permits frames to be sent to different destination addresses. Frames could be aggregated into a single PPDU as long as they are being transmitted at the same transmission power level.

Figure 4 shows the format of an A-PPDU. A-PPDU aggregation should be implemented in the PHY layer. A PHY SYNC header is placed before the first SIGNAL field. Subsequent PPDUs without PHY SYNC Headers are continuously transmitted after RIFS (Reduce Inter frame Space) timing that is 0 < RIFS << SIFS. The data rate of each MPDU is independently defined in the SIGNAL field respectively.

Fig. 4: The frame format of an A-PPDU

2.1.3 IEEE 802.11s mesh networks

IEEE 802.11s defines the mesh networking using the IEEE 802.11 MAC/PHY layers that support layer-2 path selection protocols and data forwarding over multi-hop topologies. Figure 5 illustrates the architecture of the mesh networks. Each node which joins a mesh network is called a mesh point (MP). A MP which also plays the role of an AP is called a mesh access point (MAP). A MP which bridges wired networks is called a mesh point portal (MPP). Mostly, a user is a MP or a STA. For the MP case, a user transmits data through its neighbor MPs which forward these data to the destinations. For the STA case, a user transmits data through the MAP and then the MAP forwards these data through the mesh networks. If BSS traffic and mesh forwarding traffic use the same channel, they starve each other because the channel can only be occupied by one side. As a result, they are usually separated into different channels.

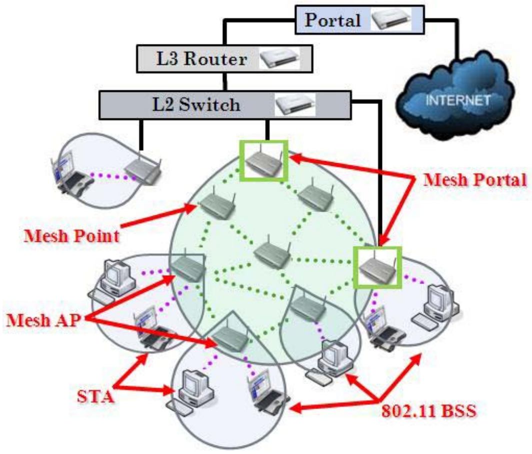


Fig. 5: IEEE 802.11s mesh networks architecture

The usages of frame aggregation mechanisms differ among the different communication pairs, as shown in Table 2.

	MP-to-MP				STA-to-MAP		MAP-to-STA	
	SDSR	MDSR	SDMR	MDMR	SDSR	MDSR	SDSR	MDMR
A-MSDU	◎				◎		◎	
A-MPDU	△	◎			○	◎	○	
A-PPDU	△	△	◎	◎	△	△	△	◎

◎ : Most suitable ○ : Replaceable but less efficient △ : Replaceable but inefficient

Table 2: The adoptive aggregation mechanisms among different transmission pairs

2.2 Related works

Several papers [6] - [9] conclude that an optimal packet size exist under a certain BER to achieve the maximum throughput of frame aggregation. But most of these studies assume that a single bit error can corrupt the whole frame. This assumption might not be true for 802.11n with frame aggregation. Lin and Wong [10] provide a unified approach to study saturated throughput and delay of the proposed frame aggregation schemes, A-MSDU and A-MPDU, under error-prone channels. The analytical model provides an accurate prediction for system performance. Based on the analysis, they propose an optimal frame size adaptation algorithm for A-MSDU aggregation.

The throughput decreases and the delay increases with increasing BER for the A-MSDU and A-MPDU aggregation schemes. A-MSDU achieves a higher throughput than A-MPDU under ideal channel conditions (i.e., BER = 0) due to the fact that A-MSDU includes the lower overhead than A-MPDU. However, under error-prone channels, throughput of A-MSDU decreases quickly often with the aggregated frame size extends a threshold in error-prone channels. This is because no protection of FCS in individual sub-frames, a single bit error might corrupt the whole frame. The above wastes lots of medium time and counteract the enhancement of efficiency contributed by frame aggregation. For A-MPDU, the throughput monotonically increases with increasing the 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.

Chapter 3 Dynamic Aggregation Selection and Scheduling Algorithm (DASS)

This chapter details the concepts and procedures of the proposed Dynamic Aggregation Selection and Scheduling (DASS) algorithm. The DASS algorithm is used to decide which aggregation mechanisms to adopt and when to send frames according to the quantity and distribution of frames in the transmission queue and the predicted frame arrival rate. It is expected to provide high bandwidth utilization to achieve a high-throughput and high-efficiency mesh networks by the dynamic selection of frame aggregation mechanisms.

3.1 Overview of the Algorithm

The goal of frame aggregation is actually to maximize the whole bandwidth utilization. Because of in mesh networks the transmission properties between different roles are not exactly the same, how to base on these characters to adopt frame aggregation mechanisms is an important issue. Based on the principles described above, DASS algorithm is proposed to how to dynamically adopt the appropriate frame aggregation to achieve a high-throughput and high-efficiency mesh network. In the first phase of DASS, each aggregation point filters out the inappropriate aggregation mechanisms before transmission. In latter phases of DASS, the channel quality, the quantity and distribution of frames in the queue, and the predicted frame arrival rate are the most important factors to determine two things : (1) which aggregation to be adopted, (2) when to send the aggregated frame out. The operations of the algorithm are depicted in Fig. 6 and elaborated in the following subsections.

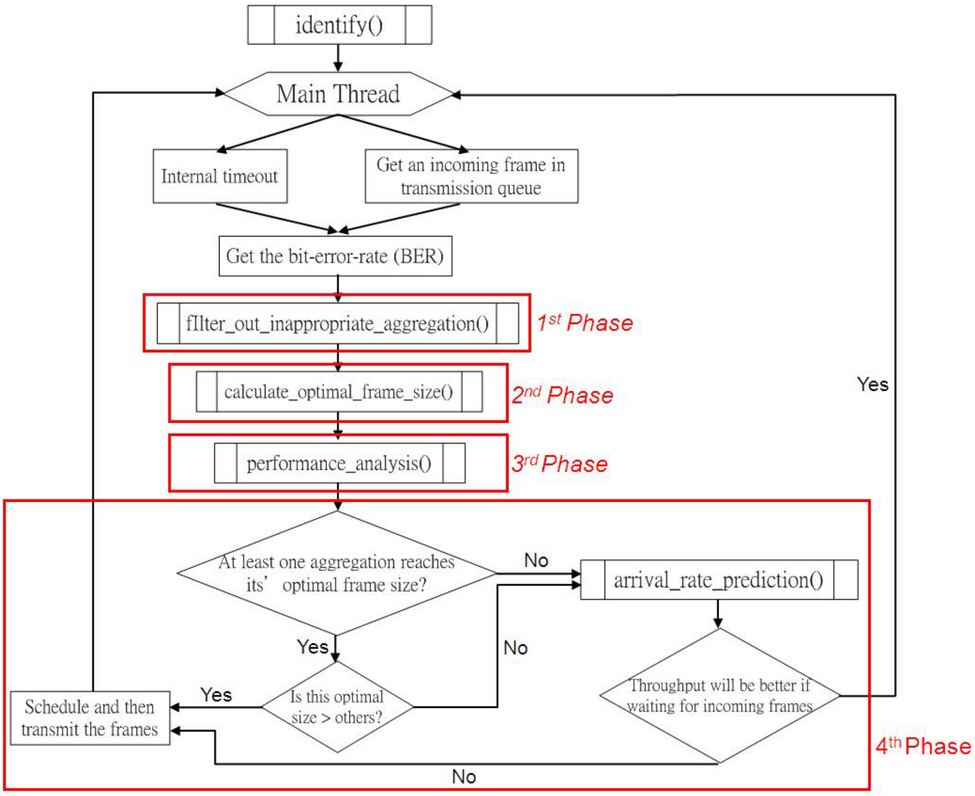


Fig. 6: The flow chart of DASS algorithm

3.2 Detailed Operations of DASS

3.2.1 First Phase: Filtering Out Inappropriate Aggregation Mechanisms

When a mesh node boosts on, it will identify itself as what kind of role it is in mesh. Through the identification, a mesh node can filter out the inappropriate aggregation mechanisms before first transmission. In this paper, we suppose that every STA follows the 802.11 standard to have only one link to its associated MAP. Thus, if a mesh node is a STA, it will not consider A-PPDU to aggregate the frames because multi-receivers, MDMR, will not happen to such a transmission.

3.2.2 Second Phase: Getting the Optimal Frame Size

After properly filtering out inappropriate aggregation mechanisms viewed from a mesh node, we begin to compute the optimal frame size for available aggregation mechanisms, respectively, in the second phase. We adopt and extend Lin and Wong's analytical model to compute the optimal frame size under error-prone channels. In their analytical model, they assume that there are N mobile stations in the WLAN. Since in mesh networks the BSS traffic and the mesh forwarding traffic may be delivered under the same channel, they compete for the transmission opportunities because the channel can only be occupied by one side. Thus, N is redefined as the number of all mesh nodes which can sense each other under the same collision domain. The wireless channel has a bit-error-rate (BER) of P_b, which can be measured through an incoming frame. 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. In 802.11 WLANs, transmitting the control frames at the basic rate, which is much lower than the data rate, makes the control frames more robust in combating errors. To simplify the analysis, they do not consider the frame error probabilities for control frames and preambles.

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

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.

The probability of an idle slot is:

$$P_{idle} = (1-\tau)^N \tag{3}$$

The probability for a transmission in a time slot is:

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

The probability for a non-collided transmission is:

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

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

$$P_{err} = P_{tr} P_s p_e \tag{6}$$

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

$$P_{succ} = P_{tr} P_s (1-p_e) \tag{7}$$

The network's saturation throughput can be calculated as:

$$S = \frac{E_p}{E_t} \tag{8}$$

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:

$$E_t = T_{idle}P_{idle} + T_c P_{tr}(1-P_s) + T_e P_{err} + T_{succ}P_{succ} \tag{9}$$

where T_{idle}, T_c and T_{succ} are the idle, collision and successful virtual time slot's length. T_e is the virtual time slot length for an error transmission sequence.

Apart from throughput, they study the average access delay experienced by each node. 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 nodes competing for transmission. On average, the access delay is:

$$d = N\frac{L_p}{S} \tag{10}$$

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

$$T_c = RTS + EIFS \tag{11}$$

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. The equations for T_{succ}, T_e and E_p are as follows:

$$T_{succ} = RTS + CTS + DATA + BACK + 3SIFS + DIFS \tag{12}$$

$$T_e = RTS + CTS + DATA + EIFS + 2SIFS \tag{13}$$

$$E_p = L_p P_{succ} = L_p P_{tr} P_s (1 - p_e) \tag{14}$$

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:

$$p_e = 1 - (1 - P_b)^L \tag{15}$$

$$E_p = (L - L_{hdr})(1 - p_e) \tag{16}$$

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:

$$p_e = \prod_i (1 - (1 - P_b)^{L_i}) \tag{17}$$

$$E_p = \sum_i (L_i - L_{subhdr})(1 - P_b)^{L_i} \tag{18}$$

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.

3.2.3 Third Phase: Performance Analysis

After getting the optimal frame size of available aggregation mechanisms, we begin to select the adoptive aggregation mechanism with the highest throughput improvement for the mesh node. In second phase, we know that the optimal aggregated frame size is varied under different BER conditions. Since a mesh node may have more than one peer-to-peer neighbors, it is necessary to think about multi-rate issue due to the divergent transmission conditions, which may result in diverse BER between different communication pairs. Thus, the functional analyses have to be considered for different BER between every communication pair. A scenario that a mesh node has these packets destined to some destinations for $Endpoint_{i,j}$ is taken for an example to explain the details of this algorithm. At first, the variables used by this algorithm are defined in the following.

$\forall Endpoint_{i,j}$,

$f(x, BER)$ *is the function of the optimal frame size adaptation algorithm,* x *is the frame size*

$D_{Buffered}(i,j)$ *is the amount of buffered data for* $Endpoint_{i,j}$

Rr_m *is the subset of* $Endpoint_{i,j}$ *through the same destined receiver*

$D_{Rr}(m)$ *is the amount of buffered data for* Rr_m

$T_{Max-MSDU}$: *the current maximum throughput using* $A\text{-}MSDU$

$T_{Max-MPDU}$: *the current maximum throughput using* $A\text{-}MPDU$

$T_{Max-PPDU}$: *the current maximum throughput using* $A\text{-}PPDU$

While transmission queue has incoming frames, DASS will base on available aggregation mechanisms to individually compute the maximum throughput when one of them is adopted. All of the frames in transmission queue are classified according to destination address and TID value.

In A-MSDU, individual frames could only be aggregated when their destination and TID value are the same. BER measured between sending and receiving ends along with the accumulative frame size could then be used as the function input, which in turn gives the corresponding throughput. We repeat this procedure on each set of aggregated frames, and obtain the maximum throughput of the transmission queue under A-MSDU by comparisons. Note that different set of aggregated frames may have the same maximum throughput. For example, the frames, lead to destination A with the TID value equal to 2, and the frames, lead to destination B with the TID value equal to 7, are abundant enough to make the throughput performance reach the greatest benefit.

$$T_{Max-MSDU} = Max(\ f(D_{Buffered}(i,j), BER)) \tag{19}$$

For A-MPDU, the frames with the same receivers can be aggregated. Via routing information, we could know which node the next-hop is if the frame is going to lead to its destination. Thus, each mesh node can classify all the frames in transmission queue according to the next-hop receivers. In A-MPDU, frames can be aggregated as long as having the same receiver. A frame's next-hop is made known via routing information, by which each mesh node might determine the concatenatablility of individual nodes. In a similar way, the maximum throughput of the transmission queue can be obtained. Note that as in A-MSDU, different set of concatenatable frames may have the same maximum throughput.

$$T_{Max-MPDU} = Max(f(D_{Rr}(m), BER)) \tag{20}$$

A-PPDU has no restrictions on concatenation. The maximum throughput is computed in a similar way, except that it's the maximum among all possible frame aggregation.

$$T_{Max-PPDU} = f\,(\sum_{i}\sum_{j=0}^{7} D_{Buffered}\left(i,j\right)) \qquad (21)$$

Through the comparison between the three maximums received after overall calculation, which kind of aggregation mechanisms can be determined to adopt.

3.2.4 Fourth Phase: Scheduling packets

$T_{Waiting}$ *is the duration for waiting for incoming frames for* $Endpoint_{i,j}$ *in state Future*

$D_{Future}\left(i,j\right)$ *is the amount of incoming data for* $Endpoint_{i,j}$ *during state Future*

$R_{\Pr edict}\left(i,j\right)$ *is the predicting frame arrival rate for* $Endpoint_{i,j}$

A_k *is the inter-arrival time between* $frame_k$ *and* $frame_{k+1}$

L_p *is the aggregated frame's payload length*

$Th_{Buffered}$ *is the throughput for transmitting buffered data*

$Th_{\Pr dict}$ *is the throughput for transmitting buffered and incoming data*

MAX_Waiting_Threshold is the maximum duration for waiting for incoming frames

Through the third stage, we can decide which aggregation mechanism to be adopted, and estimate for what the maximal throughput is if transmitting this kind of aggregated frames. During the second stage, under different BER conditions there will be different optimal aggregated frame size for different aggregation mechanisms, called ideal value. And comparing this ideal value with the accumulative frame size has three situations.

The first kind of situation is when the amount of frames is greater than ideal value, and then we must select enough frames from the queue to make the aggregated size approach but smaller than ideal value. For A-MSDU, the selection strategy is First In, First Out (FIFO). However, for A-MPDU and A-PPDU, the selection strategy depends on Quality of Service (QoS) types. The frame with higher QoS type has the higher priority to be sent. If the frames are with the same QoS type, we select the frames with more hop-counts from source node to this aggregation point so that the latency between different end-to-end nodes has smaller variations. The second kind of situation is when the amount of frames is equal to ideal value. Obviously the choice is to aggregate these frames and then send out. The third kind of situation is when the amount of frames is less than ideal value. At this time, DASS will base on past traffic to predict frame arrival rate for this kind of frames. According to the past sixteen frames from now, we could estimate for frame arrival rate by taking the total frame size to divide by the time interval between the past sixteen frames. The equation for $R_{\mathrm{Pr}edict}(i,j)$ is as follow:

$$R_{\mathrm{Pr}edict}(i,j) = \frac{16 * L_p}{\sum_{k=0}^{15} A_k} \tag{22}$$

After computing frame arrival rate, we could make an estimate for whether this kind of frames will come enough to be aggregated and promote the throughput performance in the future. Below we take A-MSDU for an example. If the throughput performance by transmitting the aggregated frame made up of buffered data is defined as $Th_{Buffered}$:

$$Th_{Buffered} = f(D_{Buffered}(i,j), BER) \tag{23}$$

Assume that we will wait $T_{Waiting}$ seconds for oncoming frames in the future, the amount of frame size could be calculated by frame arrival rate:

$$D_{Future}(i,j) = R_{\mathrm{Pr}edict}(i,j) * T_{Waiting} \tag{24}$$

Then we could deduce the equation for the throughput $Th_{predict}$ when waiting $T_{Waiting}$ seconds:

$$Th_{predict} = \frac{D_{Buffered}(i,j) + D_{Future}(i,j)}{\dfrac{D_{Buffered}(i,j) + D_{Future}(i,j)}{f(D_{Buffered}(i,j) + D_{Future}(i,j), BER)} + T_{Waiting}} \tag{25}$$

Through the comparison between $Th_{Buffered}$ and $Th_{predict}$, we could decide whether we will wait for follow-up frames or not.

$$Th_{predict} > Th_{Buffered} \tag{26}$$

If the inequality equation above has the positive solutions, the executing step will go to main thread and hold until the arrival of the follow-up frames or the internal timeout to trigger. If the inequality equation above has no positive solutions, we will immediately aggregate all the frames in the queue and then send it out. Sometimes we determine to wait for the oncoming frames to get higher throughput, but really there are no frames that get in in the future so that makes the throughput drop off. Hence, we have to make a threshold to prevent this situation of indefinite waiting causes the throughput worse and worse. The executing step will automatically go to next step while spending more than the threshold time for waiting, but actually the throughput has decreased since waiting. At this time, the BER value will renew and the algorithm will decide the adopted aggregation mechanism again. The chosen mechanism might be not same as the former one because the quantity and the distribution of frames buffered in the transmission queue might be changed. The maximal waiting threshold is evaluated by Poisson distribution because we assume that the sequence of follow-up frames is shown as Poisson distribution. In probability theory and statistics, spending the threshold time for waiting for follow-up frames to aggregate will cause the throughput to reach the maximal performance under ideal conditions.

$P_{\lambda}(k,T)$ is defined as the Poisson distribution, and the equation is :

$$P_{\lambda}(k,T)=\frac{(\lambda T)^{k}}{k!}e^{-(\lambda T)} \tag{27}$$

λ is set to the number of the received frames per second.

$$\lambda=\frac{R_{\mathrm{Pr}edict}(i,j)}{L_{p}} \tag{28}$$

Thus, for A-MSDU, the equation is expressed as:

$$E(T)=\sum_{k=0}^{\infty}\left[\frac{Min(D_{Buffered}(i,j)+L_{p}*k,8000)}{T_{Buffered}(i,j)+T}*P_{\lambda}(k,T)\right] \tag{29}$$

The computed result is namely the maximal waiting threshold.

Chapter 4 Simulation Results

This chapter verifies the effects of DASS through simulation by the ns-2 simulator in terms of throughput performance under infinite and steady backlog, the accuracy of prediction for frame arrival rate, and the comparisons between different selection strategies. Each scenario considers a set of algorithms supporting certain functionality. The parameters used in the simulation are shown in Table 3.

Parameter	Value
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)
Sub-frame Header in A-MSDU	14 (Bytes)
Delimiter in A-MPDU	4 (Bytes)
Duration of Signal Field in A-PPDU	4 (μs)
RIFS (Reduced Inter Frame Space)	2 (μs)
SIFS (Short Inter Frame Space)	16 (μs)
DIFS (Data Inter Frame Space)	34 (μs)
Size of ACK frame	14 (Bytes)
Size of Block ACK frame	32 (Bytes)

Table 3: Simulation parameters

4.1 Simulation Environment

To test the efficiency of aggregation we assemble a noteworthy scenario that includes 16 MAPs and 10 to 30 STAs in the network. These usage models intend to support the definitions of network simulations that will allow them to evaluate performance of various proposals in terms of, for example network throughput, average latency, packet loss and other metrics. Here, we will study the maximum throughput with the proposed aggregation mechanisms when increasing the offered load with different traffic patterns. From this scenario we also observe the degrading channel efficiency when aggregation is disabled but the system is using in-full its latest PHY layer's capabilities.

For the scenario, we set an infrastructure service area that operates in EDCA mode and includes 8 MPs and 10 to 30 STAs, all operating over a 20 MHz channel and using the same modulation coding scheme. The devices are placed over a distance of 50m and their antennas are on line of sight (LOS). The stations have the same data source that provides varying offered loads (in Mbps) of Constant Bit Rate (CBR) traffic. These CBR sources have no timeout values specified and they may have different TID. And all the data packets passed down to the MAC layer are 100Bytes in length. The BER varies from 0 to 10^{-3}. All simulations are run for 10 seconds.

4.2 Simulation Results

4.2.1 Throughput

Throughput is obviously an important performance metric for discussing the benefit of frame aggregation. In our simulation, we designed different traffic patterns to analyze the numerical results for two topics individually. One of the topics is to discuss the degree of throughput improvement under hybrid or single frame aggregation mechanisms. Thus, for this topic, the simulation was carried out with the saturated traffic and the increase of the number of STAs step by step. Figure 7 shows the throughput under the saturated traffic for frame aggregation. Comparisons with the simulation results show that the degree of the throughput improvement under hybrid adoption is apparently better than the one under single adoption. To contrast with no frame aggregation, DASS could almost promote the overall throughput for 92%. Another phenomenon we observed is that the degree of throughput improvement decreases with the increase of number of STAs. The reason is that with the increase of contentions for bandwidth the time wasted on a CSMA/CA random backoff and the probability of collisions might be raised. The situation would cause the frames to be retransmitted and make the throughput worse. There is one thing worthy to be observed is that why the throughput of the one with waiting mechanism is better than the one without waiting mechanism under saturated traffic. This is because sometimes a STA might adopt A-MSDU to aggregate the frames and then send the aggregated frame to its associated MAP, but the associated MAP might receive the aggregated frame and then consider adopting A-MPDU or A-PPDU to aggregate the received one and the buffered one into a larger size aggregated frame to court the better throughput.

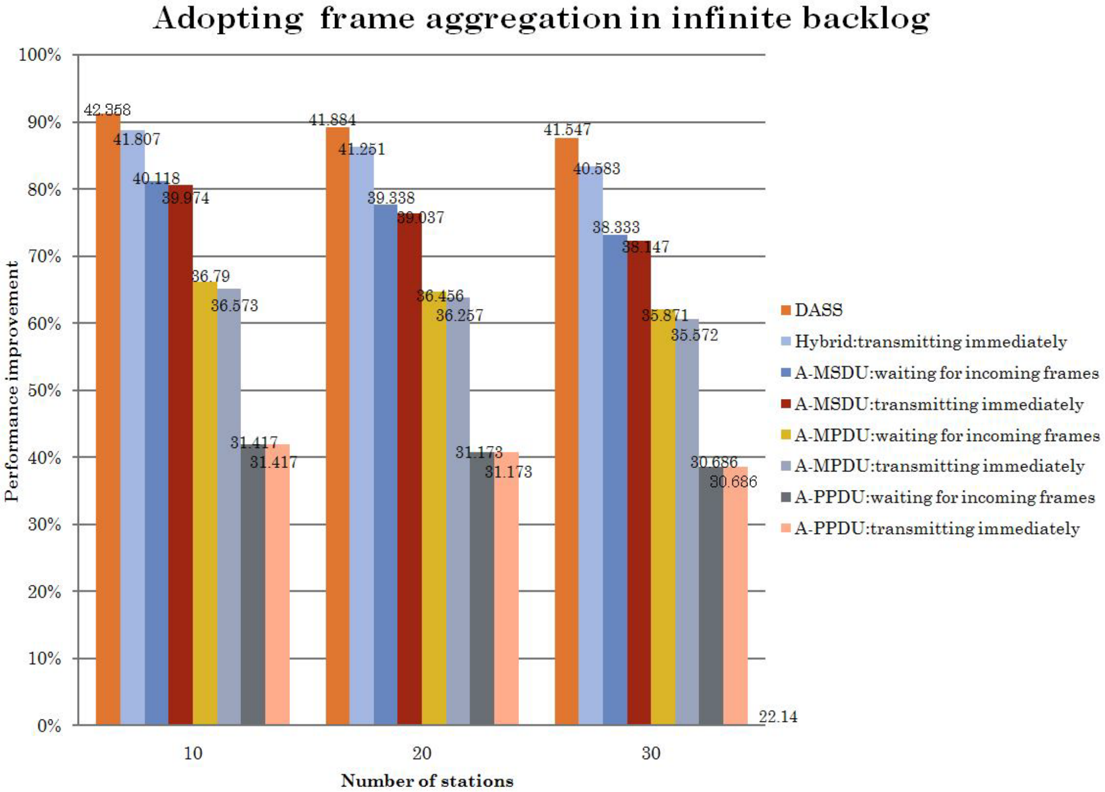


Fig. 7: Frame aggregation in infinite backlog

The other topic is to discuss whether the waiting mechanism for courting better throughput performance is necessary or not. Thus, for this topic, the simulation was carried out with the unsaturated traffic and the increase of the number of STAs step by step. Figure 8 shows the unsaturated throughput for frame aggregation. Comparisons with the simulation results show that the degree of throughput improvement with the consideration for the waiting mechanism is apparently much better than without waiting mechanism. To contrast with no frame aggregation, DASS could almost promote the overall throughput for 95%. Another phenomenon we observed is that the degree of throughput improvement increases with the increase of number of STAs. The reason should be that the total transmitted data is raised up since the channel is fully utilized and the throughput increases.

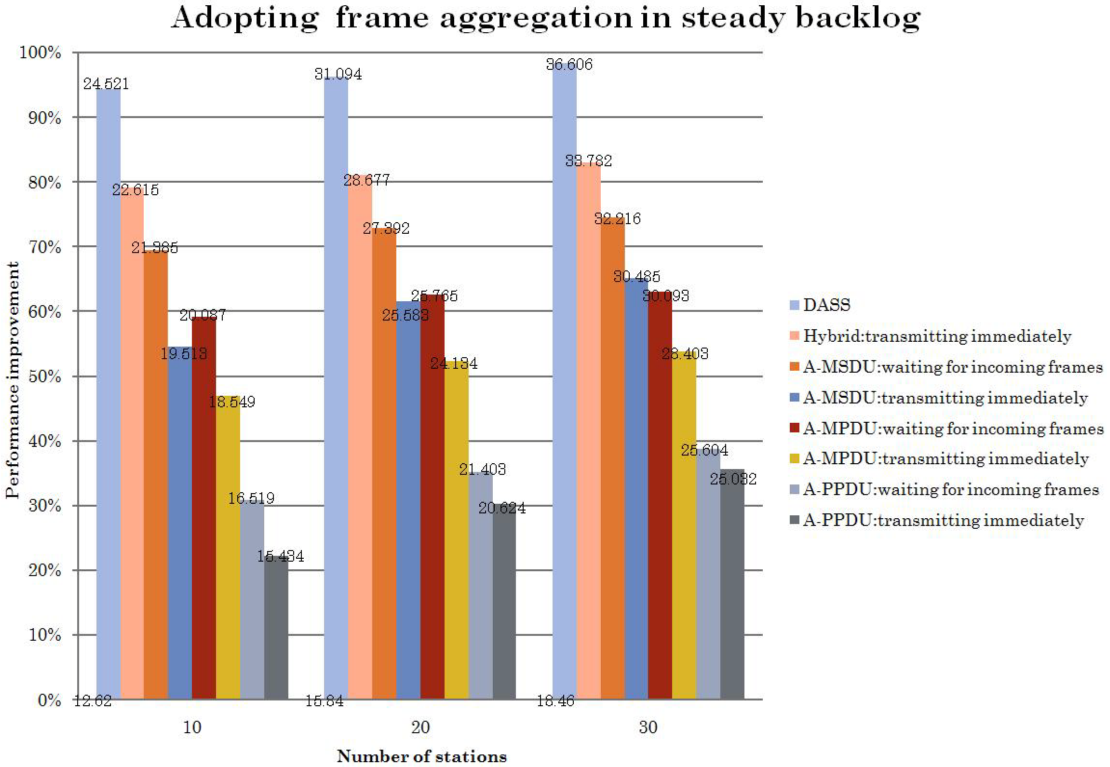


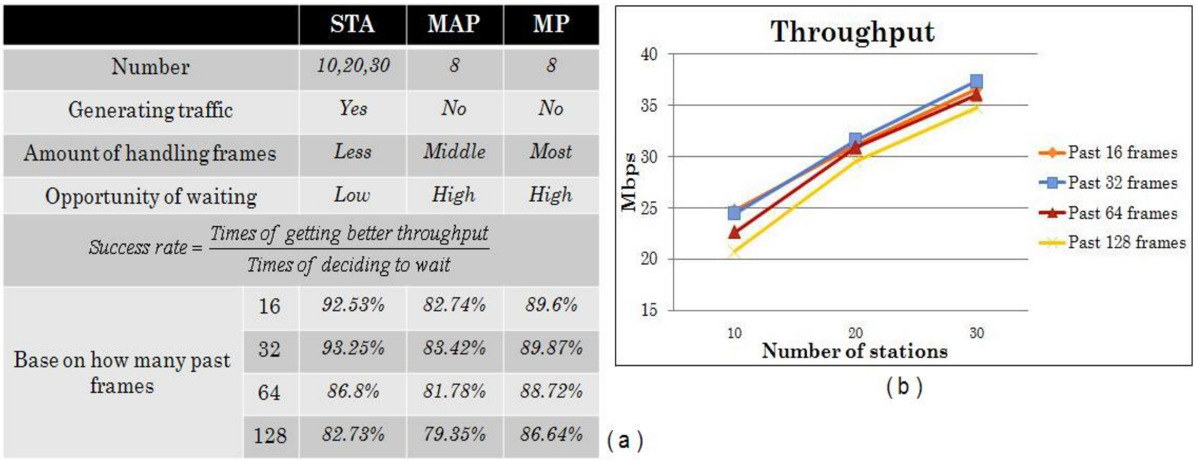
Fig. 8: Frame aggregation in steady backlog

4.2.2 Accuracy of Prediction of Frame Arrival Rate

In the DASS algorithm, through prediction of frame arrival rate, we can analyze and then decide whether to wait for the follow-up frames to aggregate to court better throughput. From the numerical results discussed above, for some traffic patterns under hybrid adoption in the frame aggregation mechanisms, the degree of throughput improvement with the additional waiting mechanism is further enhanced than the one without waiting mechanism. However, do the formulas in DASS for prediction of frame arrival rate determine the right time accurately? Therefore, an experiment was designed to observe the success rate, which is defined as the ratio of the times really gains better throughput to the times decides to wait, according to the playing roles in mesh. And the analysis of the success rate depends on variable number of past frames is also shown below. Figure 9 is the simulation results. From figure 9(a), the times of deciding to wait adopted by the MAPs and the MPs are much more than by STAs.

This is because the CBR traffic is generated by the STAs, the frame arrival rate of the STAs is much steady than others. Since the effect of stability, the success rate in the STAs is relatively high and approaches to 92.53%.

Except the discussion above, we also observed and analyzed the influence of changing the number of past frames used to predict frame arrival rate with exponential increase. Figure 9(a) illustrates the success rate of each aggregation point commonly drops off when the number of the referred frames increase to 128, and the degree of degradation is especially severe and evident for the STAs. We found this unusual phenomenon is caused by the CBR sources, which are off and on without stabilizing the traffic flow. If the packets generated from the STAs are transmitted continuously, with the increase of the number of the referred frames the success rate will converge and approach to a fixed value gradually. Figure 9(b) illustrates the throughput is relatively better while prediction of frame arrival rate is more precise. Obviously, the extra waiting time caused by the failure of prediction will make the throughput abate.



		STA	MAP	MP
Number		*10,20,30*	*8*	*8*
Generating traffic		*Yes*	*No*	*No*
Amount of handling frames		*Less*	*Middle*	*Most*
Opportunity of waiting		*Low*	*High*	*High*
$\text{Success rate} = \dfrac{\text{Times of getting better throughput}}{\text{Times of deciding to wait}}$				
Base on how many past frames	16	*92.53%*	*82.74%*	*89.6%*
	32	*93.25%*	*83.42%*	*89.87%*
	64	*86.8%*	*81.78%*	*88.72%*
	128	*82.73%*	*79.35%*	*86.64%*

(a)

(b)

Fig. 9: Accurate rate of predicting frame arrival rate

4.2.3 Comparisons between Different Selection Strategies

Other important issues are the frame-selection and queue-selection problems, which come up when there are many frames could be aggregated inside the queue or many queues have sufficient frames to aggregate to reach the maximum throughput at the same time. In the DASS algorithm, queue selection is to take turns between those candidates, and frame selection is to depend on the hop counts from the source to the aggregation point. A frame with more hop counts has a higher priority to be aggregated so that the deviation of access delays from their mean would be gradual. Figure 10 shows the average latency and the throughput performance compared for the five selection strategies. The former four strategies are for frame selection, and the last one is for queue selection. The strategies for different purposes can be mixed to seek for better performance, for example, the combination of the second and the fifth.

From figure 10(a), based on the channel quality between the senders and the receivers to select the aggregated queue will decrease the average latency so that improves the throughput performance further. In order to reach the goal above, the system implemented with the multi-path scenarios is prerequisite. There is one thing worthy to be discussed is that the channel quality here is exactly the BER value measured in the second stage of DASS algorithm. Besides, the average latency we observed for different frame selection strategies varies not too much. If we analyze the variation in the average latency, it is found that the standard deviation of using FCFS is highest and the standard deviation of considering the propagation delay is lowest. This work does not discuss painstakingly limits of transmission timeout from upper layers. Users can take account of the second strategy to reduce the opportunity for timeout in reality.

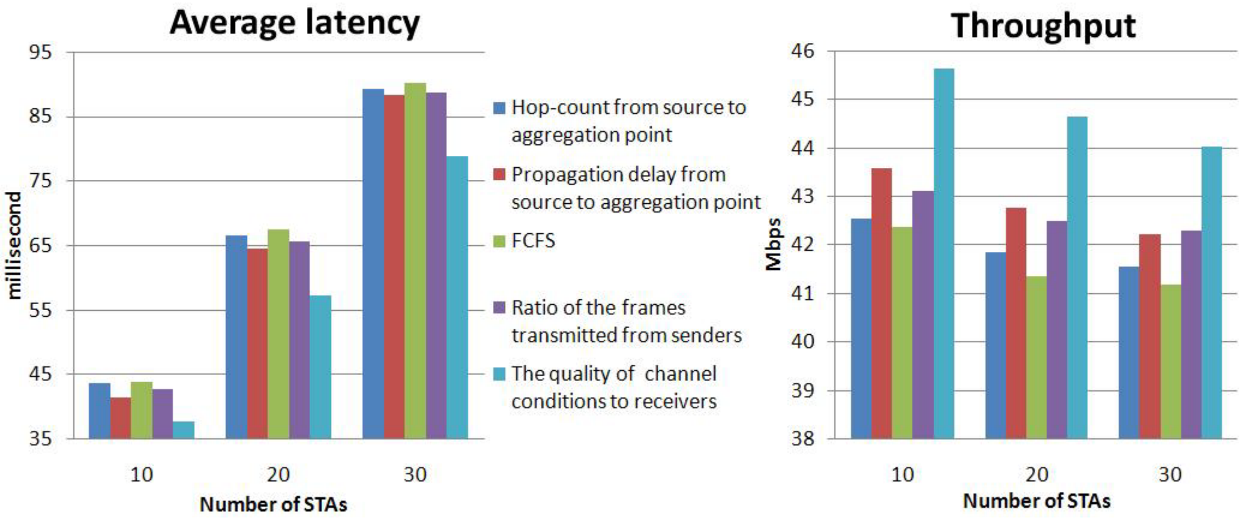


Fig. 10: Comparisons between different Selection Strategies

Chapter 5 Conclusions and Future Works

This work aims at designing a dynamic aggregation adoption algorithm for IEEE 802.11s mesh networks in order to promote poor bandwidth utilization caused by the overhead of CSMA/CA and slow down throughput degradation caused by multi-hop transmissions.

The *Dynamic Aggregation Selection and Scheduling* (DASS) is proposed to achieve a high-throughput and high-efficiency mesh network. It could dynamically adopt an appropriate aggregation mechanism according to the bit error rate (BER), the communication pair, the transmission type, and the quantity and the distribution of frames in the transmission queue to maximize the bandwidth utilization as high as possible. And through the considerations above and the analysis of past traffic, we predict how many incoming frames to be aggregated, and then determine an appropriate time to send the aggregated frame.

Simulation results demonstrated that DASS algorithm actually increases the channel efficiency of the 802.11 MAC and further improves the overall throughput 95% compared with no aggregation. We have also showed that increasing PHY layer transmission rate alone does not offer higher throughputs as PHY and MAC overhead degrades the overall performance.

All types of aggregation schemes are highly recommended as they resolve the fundamental problem of existing overhead. However, the IEEE 802.11n draft only identifies the basic concepts and the data frame structures. In a flawless environment it could deliver attractive results but in terms of its functionality in a realistic environment there are still some issues that need further investigation. For example, the processing time needed to compute these mechanisms can increase the overall delays. Actually, as the efficiency of aggregation increases, its operation becomes more complex (e.g., two-level aggregation).

Future work includes taking two-level aggregation into account and the co-existence of IEEE 802.11s draft 2.0, which is released recently and defines aggregation schemes additionally. Furthermore, mathematical modeling should be investigated to analyze the throughput performance.

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